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HCIP-Video Conference V2.0

Huawei H11-861_V2.0

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QUESTION NO: 1

The standby GK completes the following procedures ( ) in response to a synchronization request.

A. Check whether the system data status is valid for the primary use

B. Start the synchronization task process (lowest priority)

C. A timed 1 second checks whether the synchronization is over

D. Task pause 1ms

ANSWER: A B C

Explanation:

When a standby gatekeeper (GK) receives a synchronization request from the active/primary GK, it first checks whether the system-data state is valid for primary use. This validation ensures that the standby system has a usable, consistent baseline before synchronization data is processed. It then starts the synchronization task process at the lowest priority. Assigning this background replication work the lowest priority protects call-control and other real-time conferencing functions from being delayed by state synchronization. During the synchronization process, the standby GK also uses a one-second timer to periodically check whether synchronization has completed. This polling mechanism lets the device recognize task completion and proceed with the appropriate synchronization-state handling without consuming excessive processing resources. Together, these operations support high availability: the standby GK keeps registration, conference, and other relevant service data aligned with the active GK, so it can take over with the most current feasible system state if a failover is required. Gatekeeper operation is part of the H.323 architecture described by the ITU-T in Recommendation H.323; Huawei enterprise support resources are available through the Huawei Enterprise Support portal.

QUESTION NO: 2

In the Early state, dialog can be terminated by (). (3 answers).

A. The caller sends a BYE

B. The caller sends CANCLE

C. Called Failure Response

D. Called to send BYE

ANSWER: A B C

Explanation:

An early SIP dialog exists after a provisional response establishes dialog-identifying information, but before a successful final response confirms the dialog. The caller can terminate its attempted session by sending a BYE on the early dialog. SIP explicitly permits a user agent client to send BYE for an early dialog before it receives a final response to its INVITE. The caller can also send CANCEL to cancel the outstanding INVITE transaction; this ends the attempted call setup and causes the early dialog to be terminated when the INVITE is completed. Finally, a failure final response from the called side—meaning a non-2xx final response to the INVITE—terminates the early dialog because no confirmed dialog is established. These behaviors allow an originating endpoint to stop call setup proactively, while also ensuring that an unsuccessful destination-side decision cleanly removes provisional dialog state. The SIP rules for ending an attempted session, including use of BYE on an early dialog and the restriction on a user agent server sending such a BYE, are specified in RFC 3261, Section 15. CANCEL processing and its relationship to the pending INVITE are defined in RFC 3261, Section 9.

QUESTION NO: 3

Under static NAT networking, the signaling description of the public network terminal called the private network terminal is correctly (). (3 answers).

A. The Caller CS Addr address is the NAT address of the private network terminal

B. Callee CS addr is the NAT address

C. The DST address is the NAT address of the private endpoint

D. The DST address actually processed by the firewall is a private network address

ANSWER: B C D

Explanation:

For an incoming call from a public-network endpoint to an endpoint deployed behind static NAT, the public endpoint must address the called endpoint by its configured public, translated address. Therefore, *Callee CS addr is the NAT address* is correct: the called-side call-signaling address advertised and used across the public network is the NAT-mapped address, not the endpoint's RFC 1918 address. Likewise, *The DST address is the NAT address of the private endpoint* is correct for the packet as it arrives at the NAT/firewall's external interface. Static destination NAT uses that public destination address as the match for the translation rule. After the firewall applies the static NAT/DNAT mapping, the packet's destination is rewritten to the endpoint's inside address, so *The DST address actually processed by the firewall is a private network address* correctly describes the translated destination used for delivery on the internal network. This behavior lets an internal conferencing terminal remain on private addressing while still being reachable through a stable public address. Huawei documentation describes NAT as translating public and private addresses and identifies destination NAT as translating a packet's destination address for access to internal servers; see Huawei NAT configuration documentation and Huawei firewall NAT documentation.

QUESTION NO: 4

On top of UDP, the H.323 protocol is divided into (). (3 answers).

A. Terminal Control

B. Media Control

C. Data Application

D. Bandwidth Control

ANSWER: A B C

Explanation:

Within the H.323 multimedia architecture, UDP serves as the transport foundation for the real-time media plane, with functions commonly organized as terminal control, media control, and data application. **Terminal Control** covers the endpoint-oriented control capabilities needed for an H.323 terminal to participate in a multimedia session, including coordination of its supported communication functions. **Media Control** represents the real-time audio and video transport/control role, normally using RTP for media delivery and RTCP for media monitoring and feedback over UDP. RTP is specifically designed to carry real-time media across packet networks, while RTCP provides associated control information such as reception-quality feedback and stream synchronization. **Data Application** represents collaborative data-conferencing functionality, such as shared data and related application services, which is part of the broader H.323 conferencing environment. Together, these three functional areas express the terminal, media, and data components required for multimedia conferencing rather than merely a bandwidth-management function. The ITU's H.323 recommendation defines the framework for packet-based multimedia communication, and the RTP specification describes RTP/RTCP's real-time transport role: ITU-T Recommendation H.323 and RFC 3550: RTP.

QUESTION NO: 5

Network recording can record the audio and video output of any venue in the video conference (voice, SD/HD/Zhizhen) and play it back on demand via web.

A. True

B. False

ANSWER: A

Explanation:

True is correct. In Huawei video-conferencing solutions, network recording is a centralized recording capability that can capture conference media from participating sites, including audio-only venues, standard-definition and high-definition video venues, and immersive telepresence venues such as Zhizhen. The recording system stores the conference audio and video as a recording resource rather than requiring a user to make a local recording at an endpoint. This supports consistent retention and later access to the content produced during a conference.

After a recording is created and published or made available according to the organization's access policy, authorized users can use the web-based recording portal to search for and play the recording on demand. Web playback is therefore a core part of the network-recording workflow: conference media is captured centrally, stored by the recording service, and retrieved through a browser for subsequent viewing. This is especially useful for training, meeting review, compliance retention, and sharing information with people who could not attend the live conference. Huawei's enterprise communications portfolio describes centralized video-conferencing and collaboration capabilities at Huawei Video Conferencing; Huawei's support portal provides product documentation and configuration guidance for enterprise conferencing and recording components at Huawei Enterprise Support.

QUESTION NO: 6

A short message is one that can send ( ) messages to a designated venue or to all venues for inter-venue communication and information prompting. (Fill in the blanks)

A. Text

ANSWER: A

Explanation:

The correct completion is **Text**. In Huawei video-conferencing and collaboration systems, the short-message function is used to distribute written notifications to one selected venue or broadcast them to all venues participating in an inter-venue environment. This lets a conference administrator provide immediate operational prompts, meeting reminders, agenda updates, or other brief information without interrupting the active audio and video discussion. The message is rendered as textual content on the receiving conference endpoint or its associated conference interface, which is why "Text" accurately identifies the type of message sent by this feature. This capability supports orderly meeting management because important notices can be delivered simultaneously and consistently to remote participants while the conference remains in progress. It is therefore distinct from the conference's media streams: the short-message mechanism is intended for concise written communication and notification rather than for spoken media or video content. Huawei positions its enterprise collaboration portfolio around integrated conferencing and communication capabilities, including tools that improve information sharing across distributed meeting participants. See Huawei's Enterprise Collaboration product information and the Huawei Enterprise Support documentation portal for product documentation and configuration guidance.

QUESTION NO: 7

Under the video service, the public network call private network is not accessible, and the public and private network traversal technology used is generally the same as the private network call public network

A. NAT

B. SNP

C. ALG

D. H.460

ANSWER: B D

Explanation:

SNP and H.460 are applicable public/private-network traversal mechanisms for video-conferencing services. In a typical NAT deployment, an endpoint on a private network can establish an outbound registration or signaling relationship with a publicly reachable service. That relationship enables the public-side service to locate the endpoint and coordinate an inbound call without requiring the private endpoint to be directly routable from the Internet. SNP is used in Huawei video-conferencing deployments to support this kind of public/private-network service interaction and traversal arrangement.

H.460 is the standards-based H.323 traversal framework. In particular, the H.460 NAT/firewall traversal extensions support signaling and media communication for H.323 endpoints located behind NAT devices and firewalls. This permits calls to be established in either direction through a traversal service, rather than depending on unrestricted direct public-to-private reachability. Therefore, both SNP and H.460 match the stated scenario: the traversal approach used to support a public-network endpoint calling a private-network endpoint is generally based on the same established traversal relationship and service mechanisms used for private-to-public calls. The H.460 recommendation family is published by the ITU at ITU-T H.460; NAT behavior and the resulting reachability considerations are also described in RFC 4787.

QUESTION NO: 8

Which parts must be included in the transaction. ( )

A. ACK

B. Zero or more provisional responses

C. A final response

D. One request message

ANSWER: B C D

Explanation:

In SIP, a transaction is the complete request-and-response exchange handled by the transaction layer. It begins with exactly one request message, which establishes the transaction context and is retransmitted or processed according to the relevant SIP transport and transaction rules. The transaction can then include zero or more provisional responses, also called temporary responses; these are the responses in the 100–199 range and may be sent while the request is still being processed. A transaction is completed by a final response, which is a response from 200 through 699 and communicates the definitive result of the request processing.

This model applies the core SIP transaction definition: a client transaction sends a request and receives all responses associated with that request, while a server transaction receives the request and sends those responses. Therefore, “One request message,” “Zero or more provisional responses,” and “A final response” describe the required transaction structure. For INVITE processing, ACK handling has special rules: an ACK associated with a non-success final response is handled in the INVITE transaction context, whereas an ACK for a successful final response is a separate end-to-end request. The normative SIP transaction definition is in RFC 3261, Section 17; INVITE-specific ACK behavior is detailed in RFC 3261, Section 17.1.1.3.

QUESTION NO: 9

Forward error correction FEC, when both parties establish a call and detect more than 0.1% of the media stream loss within 10s, the super error correction function is enabled , super loan correction

function, will be a part of the call bandwidth, used to transmit anti-packet loss of redundant code stream. ( )

A. True

B. False

ANSWER: A

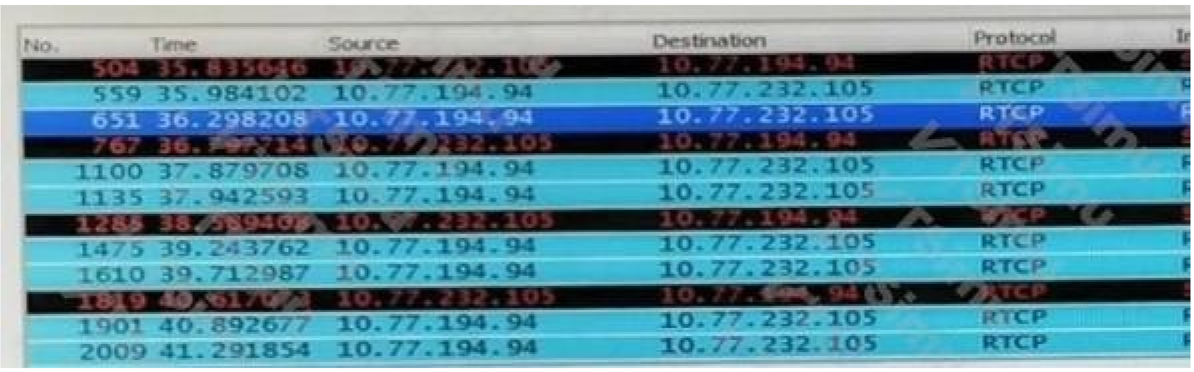
Explanation:

True is correct. In Huawei video-conferencing systems, Forward Error Correction is an adaptive media-resilience mechanism used during an established call. When the endpoints detect that media packet loss exceeds the configured trigger level of 0.1% over a 10-second observation period, Super FEC can be activated. The mechanism generates and sends redundant repair data along with the media stream. This redundancy allows the receiving endpoint to reconstruct certain lost packets without waiting for retransmission, which is particularly important for real-time audio and video because retransmissions can arrive too late to be useful.

Because the repair data must be carried across the same call path, Super FEC consumes a portion of the available conference bandwidth. This is an intentional tradeoff: allocating bandwidth to redundancy can improve perceived video continuity and audio quality when packet loss is present. The behavior described therefore accurately reflects adaptive FEC operation and its bandwidth cost in a Huawei video conference call. Huawei's enterprise communications portfolio is available at [Huawei CloudLink](). The underlying RTP FEC packet-repair approach is standardized in [RFC 5109]().

QUESTION NO: 10

As shown in the figure, the RTCP instance flow includes SR, RR, and ( ) messages.



No.	Time	Source	Destination	Protocol
504	35.835646	[illegible]	10.77.194.94	RTCP
559	35.984102	10.77.194.94	10.77.232.105	RTCP
651	36.298208	10.77.194.94	10.77.232.105	RTCP
767	36.[illegible]	[illegible]232.105	10.77.194.94	RTCP
1100	37.879708	10.77.194.94	10.77.232.105	RTCP
1135	37.942593	10.77.194.94	10.77.232.105	RTCP
1285	38.[illegible]	10.[illegible].232.105	10.77.194.94	[illegible]
1475	39.243762	10.77.194.94	10.77.232.105	RTCP
1610	39.712987	10.77.194.94	10.77.232.105	RTCP
1819	40.[illegible]	10.77.232.105	10.77.[illegible].94	[illegible]
1901	40.892677	10.77.194.94	10.77.232.105	RTCP
2009	41.291854	10.77.194.94	10.77.232.105	RTCP

A. SDES

ANSWER: A

Explanation:

SDES is the correct completion because it is a standard RTCP packet type used alongside Sender Report (SR) and Receiver Report (RR) packets in an RTCP reporting interval. RTCP provides control and monitoring information for an RTP media session rather than carrying the media itself. An SR is sent by an RTP participant that has transmitted media, while an RR supplies reception-quality feedback from a participant. SDES, meaning Source Description, identifies the RTP source and conveys descriptive items associated with that source; the canonical name (CNAME) is especially important for identifying participants and associating related RTP streams within a session.

The RTCP specification defines compound RTCP packets so that reports and source-description information are sent together according to the applicable packet-composition rules. Accordingly, a flow diagram that shows the normal RTCP exchange containing SR and RR messages is completed by SDES messages. This combination supports both quality monitoring and reliable source identification, which are core RTCP functions in video-conferencing deployments. The packet definitions and SDES requirements are specified in [RFC 3550, section 6.5]() and the compound-packet rules appear in [RFC 3550, section 6.1]().

QUESTION NO: 11

What elements are included in the SIP response message. ( )

A. Header

B. Status line

C. Message body

D. Request line

ANSWER: A B C

Explanation:

A SIP response has the standard SIP message structure defined by RFC 3261: a status line, followed by zero or more header fields, an empty line, and an optional message body. The *Status line* identifies the SIP version, a numeric response code, and a reason phrase, allowing the recipient to determine the result of the associated request. For example, a successful response begins with a status line such as `SIP/2.0 200 OK`.

The *Header* fields carry the signaling and routing information required to process the response. Typical response headers include `Via`, `From`, `To`, `Call-ID`, `CSeq`, and `Content-Length`. These fields correlate the response with its request and enable correct transaction handling.

The *Message body* is optional, but it is part of the SIP response message format and is commonly used to transport SDP session-description data, such as media IP addresses, ports, codecs, and capabilities. Its presence and type are indicated through headers such as `Content-Length` and `Content-Type`. RFC 3261 specifies this response grammar as `Status-Line *(message-header) CRLF [message-body]`. See RFC 3261, SIP message components and RFC 3261, response syntax.

QUESTION NO: 12

Terminal control mainly refers to the call signaling of the terminal, the negotiation of logic channels and AV capabilities, and the signaling of conference control.

A. True

B. False

ANSWER: A

Explanation:

True is correct. In a videoconferencing terminal, control functions are responsible for establishing, maintaining, modifying, and releasing the communication session rather than carrying the encoded audio and video media themselves. This includes call signaling, which manages call setup and teardown; audiovisual capability negotiation, which determines the mutually supported codecs, resolutions, bit rates, and media formats; and logical-channel negotiation, which opens and controls the channels used for the agreed media streams. Conference-control signaling is also a terminal-control responsibility because it enables a terminal to participate in conference operations and coordinate relevant control actions with conference resources or other endpoints.

This description aligns with the H.323 control architecture commonly used in enterprise videoconferencing. H.323 defines multimedia communication procedures, while H.245 specifies control signaling for capability exchange and logical-channel control. These mechanisms allow endpoints to agree on interoperable media parameters before media is transmitted and to adjust or close media channels during a conference. See the ITU-T H.323 recommendation and the ITU-T H.245 control protocol recommendation.

QUESTION NO: 13

In the MCU Web page, the format of the exported address book information is ( ).

A. vcf

B. xml

C. dat

D. cap

ANSWER: A

Explanation:

The exported address book information is in **vcf** format. A VCF file is a vCard file, which is a standardized electronic contact-card format used to store contact details such as endpoint names, numbers, addresses, and related directory information. Using VCF allows the MCU address book to be saved in a portable form that can be retained as a backup or transferred for later use in compatible video-conferencing and contact-management environments. This is particularly useful during system maintenance, device replacement, or migration, because directory data can be preserved separately from the MCU configuration itself. The vCard specification defines the general structure and exchange model for contact information, including the VCF file representation. Huawei enterprise product documentation and software resources are provided through the Huawei Enterprise Support portal, where the applicable MCU documentation can be selected by product and version. See the vCard standard in RFC 6350 and Huawei's Enterprise Support portal.

QUESTION NO: 14

Above the UDP layer, the H.323 protocol is divided into ( ).

A. Terminal Control

B. Media Control

C. Data Application

D. Bandwidth Control

ANSWER: A B C

Explanation:

Above the UDP transport layer, H.323 is commonly described through the functional areas of **Terminal Control**, **Media Control**, and **Data Application**. Terminal control covers endpoint registration, admission, call establishment, and related signaling functions. In the H.323 architecture, these functions are provided principally through H.225 signaling and RAS procedures, with H.245 used for capability exchange and logical-channel control. Media control coordinates the opening, management, and closing of media streams, while the actual real-time audio and video media is normally carried using RTP with RTCP providing monitoring and feedback. Data application refers to optional multipoint data-conferencing capabilities, historically associated with the T.120 family within the H.323 framework. Together, these areas describe how an H.323 endpoint controls a conference, negotiates and manages its media, and supports collaborative data services on top of IP transport. The ITU-T H.323 recommendation defines this packet-based multimedia communication system and its associated control and media components. See the ITU-T H.323 recommendation and the RTP transport specification in RFC 3550.

QUESTION NO: 15

Which of the following must the SIP request message contain?

A. Message body

B. Header Domain

C. Request line

D. Response line

E. Header fields

ANSWER: C E

Explanation:

A SIP request must begin with a **Request line**. RFC 3261 defines the request-line as the start-line of a request and specifies that it contains the SIP method, the Request-URI, and the SIP version. This line identifies both the requested operation and its target, so it is fundamental to parsing and routing a SIP request.

A SIP request also requires **Header fields**. SIP uses headers to carry the transaction and dialog information needed for protocol processing. In particular, SIP requests include required core fields such as Via, From, To, Call-ID, CSeq, and Max-Forwards, with additional mandatory fields depending on the request method and context. Header fields allow SIP entities to identify the transaction, route messages correctly, correlate requests with responses, and apply the requested service behavior.

The SIP message body is optional: it is used when a request needs to convey content such as SDP session-description data, but a valid SIP request can have no body. The normative message grammar and required request-header requirements are defined in RFC 3261, section 7 and RFC 3261, section 8.1.1.

QUESTION NO: 16

By default, the terminal reports the packet loss rate to the MCU once per ( ).

A. 20S

B. 15S

C. 6S

D. 1S

ANSWER: C

Explanation:

The correct interval is **6S**. In Huawei video-conferencing systems, endpoints periodically send packet-loss information to the MCU so that the MCU has current network-quality data for conference management. A six-second reporting cycle is the default cadence used for this packet-loss-rate reporting mechanism. It provides sufficiently timely visibility into changing media-network conditions while avoiding excessive control and management traffic. The MCU can use the reported packet-loss condition as part of its overall assessment of endpoint media quality and network status, supporting appropriate conference-level media handling and troubleshooting. Packet-loss monitoring is particularly important for real-time audio and video because loss directly affects received media quality and may require adaptive processing or operational attention. This behavior is part of Huawei's endpoint-and-MCU conference control design, which operates alongside the signaling and media-control concepts defined for videoconferencing systems. Huawei product documentation and technical resources are available through the Huawei Enterprise Support documentation portal. For background on videoconferencing control procedures, see the ITU-T H.245 recommendation.

QUESTION NO: 17

Stretch mode is stretched to a full screen proportional to the output, which captures part of the image. ( )

A. True

B. False

ANSWER: B

Explanation:

False is correct. In video-conferencing display configuration, Stretch mode enlarges the source picture independently in the horizontal and vertical directions until it fills the output display area. Its defining behavior is therefore geometric scaling: the full source image is retained, but its original aspect ratio may be altered, making people or objects appear wider or narrower than intended. Capturing, cutting off, or discarding portions of an image is associated with a crop-style display behavior, not with stretching. The statement incorrectly combines full-screen stretching with the characteristic result of cropping. This distinction matters when configuring an endpoint for displays whose aspect ratio differs from that of the incoming presentation or camera video. If preserving the whole image is important, Stretch can fill the screen but may distort proportions; it does not remove image edges. Huawei's enterprise support portal provides product documentation and endpoint configuration resources, including display and video-output guidance: Huawei Enterprise Support. Huawei's video-conferencing product information also describes the endpoint and display environment in which these image-adaptation settings are used: Huawei Video Conferencing.

QUESTION NO: 18

The wall riding scheme is a simple public and private network crossing method, which connects different network ports of the MCU in different networks to realize the terminals in different networks to access

the same conference.

A. True

B. False

ANSWER: A

Explanation:

True is correct. In Huawei video-conferencing deployment terminology, the wall-riding scheme is a straightforward method for interconnecting private-network and public-network video-conferencing participants. The MCU is provided with network-side connectivity in each relevant network domain—for example, one interface/addressing context for the enterprise intranet and another for an external or public-facing network. Because the MCU can receive and send conference media and signaling through the appropriate connected network, endpoints located on those separate networks can join and participate in the same multipoint conference.

This design is especially useful where direct endpoint-to-endpoint reachability across the network boundary is unavailable or undesirable. Rather than requiring every endpoint to use a complex traversal arrangement, the MCU effectively provides the conference resource that is reachable from both sides of the boundary. Correct routing, address planning, firewall policy configuration, and media-port permission are still required for the deployment to operate securely and reliably. Huawei's enterprise support portal provides the product documentation and deployment guidance for its video-conferencing systems: Huawei Enterprise Support. Huawei's video-conferencing product information is also available at Huawei Video Conferencing.

QUESTION NO: 19

Which of the following ways does HD terminal support to initiate peer-to-peer calls?

A. Venue name

B. IP

C. Number

ANSWER: A B C

Explanation:

Huawei HD video-conferencing terminals support several directly dialable addressing methods for peer-to-peer conferences. A remote site can be selected by its venue name when that site is available in the terminal's address book or directory, allowing users to place a call without entering a numeric address manually. The terminal can also initiate a call to an explicitly entered IP address, which is the standard direct-dial method when contacting an endpoint over an IP network. In addition, a remote terminal number can be dialed. This number is commonly used as the endpoint's registered E.164-style identifier in a video-conferencing deployment and can be resolved through the relevant call-control environment. These methods give users practical alternatives depending on whether the remote endpoint is known by directory name, network address, or assigned number. Huawei's enterprise collaboration documentation describes endpoint calling and address-book based dialing capabilities; see Huawei TE Series documentation and the Huawei Enterprise support knowledge base.

QUESTION NO: 20

Color temperature affects the reproducibility of the color of the video image, the higher the color temperature, the more color. ( )

A. Whitening

B. Bluedish

C. Reddish

D. Yellowing

ANSWER: B

Explanation:

Bluedish is correct. Color temperature describes the chromatic appearance of a light source or the white-balance setting used by a video camera, expressed in kelvins (K). As color temperature increases, the reference white point shifts toward blue. Daylight is commonly around 5,000–6,500 K, while much higher color-temperature values have a noticeably cooler, blue-tinted appearance. Conversely, lower color temperatures are associated with warm, red-to-yellow illumination. In a video-conferencing environment, accurate white-balance configuration is important because the endpoint camera uses its color-temperature reference to render neutral surfaces correctly and reproduce participant skin tones and room colors naturally. When a higher color-temperature setting is selected, the image's overall color rendition therefore becomes more bluish/cooler. This relationship follows the standard color-temperature concept based on the appearance of a black-body radiator: as its temperature rises, its emitted light progresses from warmer reddish tones toward cooler white and blue tones. See the International Commission on Illumination definition of colour temperature and the Encyclopaedia Britannica overview of color temperature.

QUESTION NO: 21

The following description of the cascading characteristics of the miniMCU is correct. () (3 answers).

A. MiniMCU terminals support up to three cascading conferences

B. Supports cascading conferencing in the primary miniMCU for conference control

C. Supports H.323 cascade port calls in cascading conferencing

D. Support for each terminal venue in a cascading conference to join the meeting as a SIP H323 call

ANSWER: A B C

Explanation:

MiniMCU cascading is designed to extend a conference beyond the capacity or geographic scope of a single endpoint-hosted conference. The supported topology allows MiniMCU terminals to form up to a three-level cascaded conference hierarchy, enabling a lower-level MiniMCU to aggregate local participants and connect upstream. Conference control is performed from the primary MiniMCU, which acts as the controlling conference node for the cascaded meeting and provides centralized management of the overall session. The cascade connection itself uses an H.323 cascade port call, so that the link between MiniMCU conference nodes is established as the dedicated H.323 cascade relationship rather than as an ordinary participant call. H.323 is an ITU-T multimedia conferencing recommendation and supplies the call-control framework used for this type of IP videoconferencing connection. Huawei documentation and software-version-specific endpoint documentation should always be consulted when deploying cascading, because endpoint models and releases can impose capacity or interoperability limits. See the Huawei Enterprise Support portal for product documentation and the ITU-T H.323 recommendation for the underlying protocol standard.

QUESTION NO: 22

In VR scenarios, the terminal cannot join the meeting ( ) in one of the following ways.

A. Call the conference access number

B. The MAC address

C. for the MCU Call the Unified Access Number

D. Ip Answer for Calling MCU

ANSWER: B

Explanation:

The terminal MAC address is not a meeting-dialing or conference-access identifier, so it cannot be used by a terminal to join a meeting in a VR scenario. A MAC address is a Layer 2 hardware identifier assigned to a network interface. Its purpose is local-network identification and frame delivery; it does not identify a conference, an MCU conference resource, or a routable call destination. In contrast, joining a video conference requires a service-recognized destination or access credential, such as a conference access number, a unified access number, or an IP-based call target. The conferencing platform uses those values to route the call to the appropriate MCU or meeting service and then associate the endpoint with the scheduled or active conference. Even where an endpoint's MAC address may be recorded for inventory, network access control, or device management, that administrative use does not make it a valid dial string or a meeting admission method. Huawei's enterprise video-conferencing portfolio is based on endpoint-to-conference call control and meeting-access services rather than Layer 2 addressing; see Huawei Video Conferencing. Huawei's meeting documentation likewise describes joining through meeting access and invitation mechanisms, not physical interface identifiers; see HUAWEI CLOUD Meeting User Guide.

QUESTION NO: 23

The function of the redirect server is to map the addresses in the request to zero or more new addresses, which are returned to the client.

A. True

B. False

ANSWER: A

Explanation:

True is correct. In SIP architecture, a redirect server receives a request and determines one or more alternative contact addresses for the target user or service. Rather than forwarding the request onward and remaining involved in signaling, it returns a redirection response to the requesting client. That response supplies zero or more new addresses, typically in Contact header fields, and the client can then use those returned addresses to initiate a new request toward an appropriate destination. This behavior supports flexible call routing, including locating a user who has moved, directing a request to a registered endpoint, or indicating that no current contact address is available. The defining behavior is therefore address mapping followed by returning the resulting address or addresses to the client, exactly as stated. RFC 3261 defines a SIP redirect server as a user agent server that generates responses to requests and redirects the client to contact alternative URI locations; its architecture section specifically describes mapping the request address into zero or more new addresses returned to the client. See RFC 3261, SIP terminology and RFC 3261, redirected requests.

QUESTION NO: 24

What are the following message interactions included in the process of a terminal calling by number using the H.323 protocol?

A. Connect

B. ACF

C. Setup

D. ARQ

ANSWER: D

Explanation:

ARQ is the Admission Request message used when an H.323 endpoint initiates a call through a gatekeeper-controlled zone. Before the terminal can establish the call, it requests admission from its gatekeeper and supplies relevant information such as the calling endpoint identity, the destination alias or number, bandwidth requirements, and call-related identifiers. This admission-control exchange lets the gatekeeper apply call-routing and bandwidth-management policies for a number-dialing request. When admission is granted, the gatekeeper returns the information needed for the endpoint to proceed with call establishment, including the signaling destination as applicable to the configured call model. Therefore, ARQ is the key terminal-to-gatekeeper interaction that represents the admission request in the numbered H.323 call process. The H.323 umbrella recommendation defines the system architecture and call-control framework, while H.225.0 specifies the call-signaling and registration, admission, and status procedures that include the ARQ mechanism. See the ITU-T H.323 recommendation and the related H.225.0 recommendation.

QUESTION NO: 25

Wireshark can support decoding of many protocols, such as ( ).

A. H.263

B. ip

C. RTP

D. TCP

ANSWER: A B C D

Explanation:

Wireshark is a packet analyzer with a large set of built-in dissectors that interpret captured packet data according to protocol specifications. H.263 is supported in the context of multimedia traffic analysis: Wireshark can identify and dissect H.263 video payload information carried in applicable streams, especially when RTP payload type and stream information are available. IP is a core network-layer protocol that Wireshark decodes to expose addressing, header fields, fragmentation information, and encapsulated upper-layer traffic. RTP is directly relevant to video-conferencing deployments because it commonly transports real-time audio and video media; Wireshark decodes RTP headers, sequence numbers, timestamps, synchronization source identifiers, and stream characteristics. TCP is also natively decoded, allowing analysis of connection setup, sequence and acknowledgment behavior, retransmissions, flow control, and application data carried over a reliable transport session. These protocol decoders can be used together in a single capture, since a packet may contain layered headers and media payloads. Wireshark's enabled-protocol documentation lists the available dissectors, while its RTP documentation describes RTP-stream analysis capabilities. See Wireshark enabled protocols and Wireshark RTP analysis.

QUESTION NO: 26

The large amount of data and the existence of data redundancy in the sound signal are the basis for the encoding of digital audio.

A. Compression

ANSWER: A

Explanation:

Compression is correct because digital audio streams contain substantial sample data, and adjacent samples commonly have predictable relationships. Audio compression encoding exploits this statistical and perceptual redundancy to reduce the bitrate required for storage or transmission while retaining the intended audio quality. In video-conferencing systems, this is essential: reducing the audio payload lowers network-bandwidth consumption, helps maintain media continuity under constrained network conditions, and enables efficient real-time transport alongside video and signaling traffic.

Audio codecs can use lossless techniques to represent repeated or predictable sample patterns more efficiently, or perceptual techniques to discard information that is less audible to human listeners. The latter approach is especially important for interactive communication because it achieves much lower bitrates while aiming to preserve speech intelligibility and natural sound. International codec specifications define such coding methods and their operational characteristics for telecommunications. For example, the ITU-T G.191 software tools support work involving speech and audio coding algorithms, while RFC 6716 specifies the Opus codec, which is designed for interactive audio communication and supports efficient speech and audio compression. See ITU-T Recommendation G.191 and RFC 6716: Definition of the Opus Audio Codec.

QUESTION NO: 27

Nlog v2.0 can export network packet drop data for all venues in a meeting

A. True

B. False

ANSWER: A

Explanation:

True is correct. Nlog V2.0 is intended to support post-meeting network-quality analysis by collecting and exporting packet-loss information for the venues participating in a conference. Packet loss is a key real-time media quality indicator because lost RTP media packets can cause visible video artifacts, freezes, audio gaps, or reduced overall conference quality. Exporting the packet-drop data across all venues allows an administrator to compare the network condition of each participant, identify whether impairment was isolated to one venue or widespread, and retain the results for troubleshooting and reporting. This capability is particularly useful when an issue is reported after a conference has ended, since the exported data provides an evidence-based view of the media-network experience rather than relying only on subjective user feedback. Huawei's enterprise support portal provides product documentation, software tools, and technical resources for video-conferencing maintenance and fault analysis: Huawei Enterprise Support. Huawei's video-conferencing product information also describes the operational and management focus of its enterprise collaboration solutions: Huawei Video Conferencing.

QUESTION NO: 28

In the figure, select an attribute of the packet, right-click and select the option within the ( ) group to specify this attribute as a filter condition, which must be clicked apply before it takes effect.

A. ApplyasFilter

ANSWER: A

Explanation:

"ApplyasFilter" identifies the *Apply as Filter* command used when examining packet details in a protocol-analysis view. After a packet field or attribute is selected, this command converts the selected field into a display-filter expression and places that expression in the filter toolbar. The filter is not active merely because the expression has been created; the user must select *Apply* to run the expression against the currently displayed capture and refresh the packet list. This workflow is useful because it turns a specific observed value, such as an address, protocol field, identifier, or message attribute, into a precise condition without requiring the administrator to manually construct the filter syntax. It therefore directly satisfies the blank's requirement for the right-click group command that specifies the selected attribute as a filter condition. Wireshark's User's Guide describes building display filters from packet-detail fields and applying the generated expression through the display-filter interface: Wireshark User's Guide: Building Display Filter Expressions. The command is also part of the standard packet-inspection workflow documented in the Wireshark manual page.

QUESTION NO: 29

During software installation, GK can only be installed on the C or D disk of the operating system .

A. True

B. False

ANSWER: A

Explanation:

True is correct. Huawei GK deployment guidance restricts the software installation location to the operating system's C or D disk. This requirement is intended to keep the installation on a local, fixed disk with the expected Windows file-system behavior, available capacity, and stable access permissions for GK program files, services, logs, and related operational data. Selecting one of these supported local volumes also makes the deployment conform to the installation and maintenance assumptions used by the product documentation and support procedures.

Before installing GK, the administrator should therefore verify that either the C or D disk has sufficient free space and that the account performing the installation has the necessary local administrator privileges. The selected disk should be a persistent local volume rather than removable or network-attached storage, so that the GK service remains available after restart and can reliably access its installed components. Following the documented disk-location restriction is part of a supported installation baseline and helps ensure predictable software operation and future maintenance. Huawei's enterprise support portal provides product documentation and installation resources at Huawei Enterprise Support; the documentation catalogue can be accessed through Huawei Enterprise Documentation.

QUESTION NO: 30

In multi-channel cascading, two MCUs in a conference support up to ( ) cascaded channels

A. 8

B. 16

C. 24

D. 32

ANSWER: D

Explanation:

32 is the supported maximum number of cascaded channels when two Huawei MCUs use multi-channel cascading in the same conference. Multi-channel cascading is designed to expand conference capacity and improve the distribution of media-processing resources between MCU nodes. Rather than relying on one inter-MCU media connection, the MCUs can establish multiple cascade channels so that participant video, audio, and presentation streams can be exchanged and processed at the required scale. For a two-MCU conference, Huawei's stated limit for this multi-channel cascade arrangement is 32 channels. This value is an inter-MCU cascading capability and should be understood separately from such platform specifications as the number of conference participants, ports, or simultaneous conferences. Correctly identifying the cascade-channel limit is important during solution design because the number of channels affects how media resources and bandwidth are planned between cascaded MCU sites. Huawei's enterprise support portal provides the product documentation and configuration guidance for videoconferencing systems, while Huawei's videoconferencing portfolio information describes the MCU-based architecture used to provide scalable conferencing services. See the Huawei Enterprise Support documentation portal and Huawei Video Conferencing solution overview.

QUESTION NO: 31

Packet loss retransmission ARQ, similar to TCP's retransmission mechanism, requires the peer to resend the UDP when the UDP packet loss of the video stream is detected in the network Stream, the

receiving side stores the received stream in a certain cache, and only opens the decoding after receiving the retransmitted stream.

A. True

B. False

ANSWER: A

Explanation:

True is correct. ARQ (Automatic Repeat reQuest) for a real-time video stream is an application-layer packet-loss recovery mechanism used over UDP/RTP. When the receiving endpoint detects that one or more media packets are missing, it sends feedback identifying the missing sequence numbers to the transmitting endpoint. The transmitter then retransmits the requested media packets, allowing the receiver to repair the sequence gap. To make this possible, the receiver maintains a jitter/reordering buffer: it temporarily holds subsequently received packets while awaiting the missing packet or retransmission result. After the expected packet sequence is restored, the media can be passed to the decoder in the proper order. This is conceptually similar to TCP retransmission in that detected loss triggers resend behavior, but it is designed for real-time media and uses timing, buffering, and feedback policies appropriate for video conferencing rather than TCP's reliable byte-stream delivery model. RTP retransmission is standardized through an RTP retransmission payload format, while packet-loss feedback is commonly conveyed through RTP/RTCP feedback mechanisms. See RFC 4588: RTP Retransmission Payload Format and RFC 4585: Extended RTP Profile for RTCP-Based Feedback.

QUESTION NO: 32

What features can the gatekeeper provide?

A. Conference Control

B. Bandwidth Control

C. Address Translation

D. Access Control

ANSWER: B C D

Explanation:

An H.323 gatekeeper is the central control entity for an H.323 zone and provides core endpoint-registration and call-admission services. **Bandwidth Control** is a gatekeeper function because it can evaluate bandwidth requests during admission and enforce zone bandwidth policies, preventing calls from being established when the available or permitted bandwidth is insufficient. **Address Translation** is also fundamental: the gatekeeper maps an endpoint's H.323 alias, such as an E.164 number or user name, to the endpoint's network transport address so that signaling can be routed to the appropriate registered endpoint. **Access Control** is provided through registration and admission procedures. The gatekeeper can determine whether an endpoint is permitted to register in its zone and whether it is authorized to place or receive a requested call according to configured policies. These functions allow centralized management of endpoint reachability, resource use, and calling permissions in an H.323 deployment. The ITU-T H.323 recommendation defines the architecture in which a gatekeeper performs these zone-management and admission-related roles; H.225.0 specifies the associated registration, admission, and status signaling procedures. See the ITU-T H.323 recommendation and the ITU-T H.225.0 recommendation.

QUESTION NO: 33

The advantages of static NAT include. ( )

A. The terminals in the four private networks are no different from the public network terminals, and the public network terminals can directly call the public network address after the NAT of the private network terminals.

B. The endpoint after port NAT is protected by a firewall.

C. Most endpoints support NAT functionality.

D. Back is very practical when there are a small number of private network terminals that need to access the public network.

ANSWER: A C D

Explanation:

Static NAT creates a fixed, one-to-one mapping between a private endpoint address and a public address. Therefore, "The terminals in the four private networks are no different from the public network terminals, and the public network terminals can directly call the public network address after the NAT of the private network terminals." describes its principal operational benefit: external videoconferencing terminals can consistently reach the mapped public address of an internal terminal. This predictable address mapping is especially useful for receiving calls, because the endpoint has a stable, externally routable identity.

"Most endpoints support NAT functionality." is also an advantage in a videoconferencing deployment context. Static NAT is a widely interoperable and straightforward NAT arrangement; endpoints and related signaling/media devices can use the configured public mapping without requiring a dynamic mapping to be discovered for each connection.

"Back is very practical when there are a small number of private network terminals that need to access the public network." correctly reflects the appropriate scale for static NAT. Each private endpoint normally requires its own dedicated public address, so it is practical when only a limited number of internal terminals need direct public-network reachability. RFC 3022 describes traditional NAT address mapping behavior, including static mappings: RFC 3022. Huawei's enterprise support portal also provides product documentation on NAT configuration and deployment: Huawei Enterprise Support.

QUESTION NO: 34

The ( ) function in Wireshark visualizes the conversational communication process graphically.

A. View

B. Flow Graph

ANSWER: B

Explanation:

The correct completion is **Flow Graph**. Wireshark's Flow Graph feature presents packets from a selected conversation as a time-ordered, graphical sequence of messages exchanged between endpoints. Each participant is shown as a separate vertical column, while arrows illustrate the direction and order of frames, making it much easier to follow signalling and media-related exchanges than reading packet rows alone. This is especially useful during video-conference troubleshooting, where an engineer may need to inspect the progression of call setup, negotiation, acknowledgements, or teardown messages within a captured trace. The graph can be generated from displayed packets, so capture and display filters can first narrow the analysis to the relevant call, protocol, IP addresses, or ports. Wireshark documents Flow Graph as a statistics feature and describes its ability to display packet flows between endpoints in a diagram. For operational guidance, see the Wireshark User's Guide: Flow Graph. Wireshark's official documentation index is also available through the Wireshark Documentation page.

QUESTION NO: 35

Under the static NAT networking, the packet of the private network terminal through the firewall will change, and the following description is wrong ( ).

A. The address of the terminal RAS addr after it becomes NAT

B. Endpoint CS addr becomes the address after NAT

C. NAT addr is the public egress address of the firewall

D. GK RAS addr becomes Address after NAT

ANSWER: D

Explanation:

“GK RAS addr becomes Address after NAT” is the incorrect description. In a static-NAT H.323 deployment, the firewall maps the private endpoint’s transport addresses to its configured public address. This allows external devices to send H.323 RAS and call-signaling traffic back to the endpoint through the fixed public-to-private mapping. The Gatekeeper RAS address, however, identifies the Gatekeeper that provides RAS service; it is not an endpoint address that should be substituted with the private endpoint’s NAT mapping. The endpoint uses the Gatekeeper address as the destination for registration, admission, and related RAS exchanges, while the NAT operation applies to the endpoint-side address information that must be reachable from outside the private network. Therefore, treating the GK RAS address itself as an address that “becomes” the NAT address reverses the relevant roles of the Gatekeeper address and the endpoint’s translated address. This behavior is consistent with H.323 NAT traversal principles, in which signaling must convey reachable translated endpoint transport addresses while preserving the logical addressing of the Gatekeeper service. See the ITU-T NAT traversal recommendation for H.323 at ITU-T H.460.18 and the NAT behavioral terminology in RFC 4787.

QUESTION NO: 36

The quality of digital audio depends on which factors. ( )

A. bandwidth

B. Sample frequency

C. Quantization of the number of digits

D. Channel count

ANSWER: B C D

Explanation:

Digital-audio quality is fundamentally determined by the sampling frequency, quantization precision, and channel count. Sample frequency sets the highest audio frequency that can be represented under the Nyquist sampling principle; a higher sampling rate can preserve a wider audible frequency range when the source and codec support it. Quantization of the number of digits describes the bit depth used for each sample. Greater bit depth provides more available amplitude values, reducing quantization noise and increasing the achievable dynamic range. Channel count determines how many discrete audio paths are carried, such as mono, stereo, or multichannel sound, and therefore affects spatial presentation and the ability to reproduce separate participant or program-audio signals in a conferencing system. These source-format parameters are distinct from network bandwidth, which is a transport resource; required network bitrate is influenced by the selected audio format and codec, but it is not itself a basic source-audio quality parameter in this context. The sampling and quantization foundations are specified in ITU-T’s audio-coding framework, while conferencing endpoints negotiate and transport audio streams according to multimedia conferencing standards. See ITU-T G.711 and ITU-T H.323.

QUESTION NO: 37

What addresses are typically included in the RRQ message for the GK registration record address when the node is registered? ( )

A. Caller Cs Addr

B. Poxy Addr

C. Caller Number

D. Caller RAS Addr

ANSWER: A C D

Explanation:

In H.323 gatekeeper registration, an endpoint sends a Registration Request (RRQ) over the Registration, Admission, and Status channel to make its reachable details known to the gatekeeper. **Caller Cs Addr** is correct because the RRQ provides the endpoint’s call-signaling address, allowing the gatekeeper to direct H.225 call-signaling traffic to the registered endpoint.

Caller RAS Addr is also correct because the RAS address identifies where the endpoint exchanges gatekeeper-control messages, including registration, admission, bandwidth, and status requests. **Caller Number** is correct in the sense of the endpoint's registered alias or aliases, such as an E.164 number; the gatekeeper uses these aliases to resolve a dialed number to the endpoint's registered addressing information. Together, the alias, call-signaling address, and RAS address form the core registration information needed for call routing and gatekeeper control. This is consistent with the RRQ structure defined by the ITU-T H.225.0 protocol and with Huawei's H.323 gatekeeper documentation. See ITU-T H.225.0 and Huawei enterprise product documentation.

QUESTION NO: 38

In the hierarchical relationship of SIP structure, what are the following main items of the request message?

A. Non-INVIT transaction

B. Request line/Response line

C. header field

D. Message Body

ANSWER: B C D

Explanation:

A SIP message is organized as a start line, followed by zero or more header fields, an empty line, and an optional message body. For a request, the start line is the request line, which identifies the SIP method, the request URI, and the SIP version. The wording "Request line/Response line" represents the start-line position in the common SIP message structure: requests use a request line, while responses use a status (response) line. Header fields carry the addressing, routing, dialog, transaction, capability, and content-related information needed to process the message. The Message Body is the optional payload after the header section; for example, an INVITE commonly carries an SDP offer describing media capabilities and transport parameters. These three items therefore describe the principal structural sections used when representing SIP request messages and the corresponding general SIP message format. The normative SIP syntax defines Request-Line, message-header, the required empty line separating headers from content, and an optional message-body. See the SIP protocol message definition in RFC 3261, Section 7 and the request-message syntax in RFC 3261, Section 7.1.

QUESTION NO: 39

What are the main management functions performed by RAS?

A. Number Registration

B. Status Inquiry

C. Access Control

D. Security Management

ANSWER: A B C D

Explanation:

RAS, meaning Registration, Admission, and Status in the H.323 architecture, provides the control-plane exchanges between endpoints and the gatekeeper. Its management scope includes number registration, through which an endpoint registers its identifiers and reachable addressing information with the gatekeeper. It also includes status inquiry, allowing the gatekeeper to obtain or receive endpoint status information needed for operational awareness and call-related control. Access control is a core RAS responsibility because admission procedures authorize endpoint participation and can apply call-routing, bandwidth, and policy decisions before a call is established. Security management is also supported through RAS-related authentication and authorization mechanisms, helping ensure that only validated endpoints and requests use conferencing resources. Together, these functions enable centralized endpoint management and controlled access to H.323 video-conferencing services. The ITU-T H.225.0 recommendation defines RAS signaling and its registration, admission, and status procedures; see ITU-T H.225.0. The IETF overview of H.323 also describes the gatekeeper's registration, admission, and status-control role: RFC 1639.

QUESTION NO: 40

The following description of nat mapping is correct () (3 answers).

A. Static NAT is the simplest and most implemented, where each host address in the internal network is permanently mapped to a legitimate address in the external network

B. Dynamic address NAT is a series of legitimate addresses defined in the external network that are mapped to the internal network using a dynamic allocation method

C. NAPT is a different port on which the internal address is mapped to an IP address of the external network on

D. The most common mapping method in video services is dynamic address NAT

ANSWER: A B C

Explanation:

Static NAT is correctly described as a persistent one-to-one mapping between a private/internal host address and a public/external address. Because the mapping is fixed, the internal endpoint has a predictable public identity, which is useful when inbound reachability is required. Dynamic address NAT is also correctly described: a public-address pool is configured and public addresses are allocated to internal hosts dynamically when translation sessions are needed. The allocation is not permanently bound to a particular internal host and is returned to the pool when no longer required. NAPT, also called port address translation, correctly extends address translation by distinguishing multiple internal sessions through transport-layer port numbers. It allows multiple private addresses to share one public IP address, with each translated flow identified by an appropriate public address and port combination. These three mechanisms are the standard NAT mapping models: fixed address translation, pooled dynamic address translation, and address-and-port translation. Their behavior and terminology are consistent with the traditional NAT model defined in RFC 3022 and the NAT terminology framework in RFC 2663.

QUESTION NO: 41

HD terminals support peer-to-peer () calling. (3 answers).

A. Number

B. Venue name

C. MAC address

D. IP Address

ANSWER: A B D

Explanation:

Huawei HD videoconferencing terminals support several identifiers for establishing a point-to-point call. A **Number** can be used as the endpoint's registered dialing number, such as an E.164-style number or other number assigned by the call-control environment. A **Venue name** can identify the remote conference site when that site name has been configured or resolved through the terminal's directory, address-book, or registration environment. An **IP Address** supports direct endpoint-to-endpoint calling: the caller enters the reachable address of the remote terminal and the terminal initiates the video call using the configured signaling protocol. These mechanisms allow calls to be placed either through a managed naming/numbering plan or directly to a known endpoint address, which is important for flexible deployment and inter-site conferencing. Huawei's videoconferencing portfolio documentation describes HD endpoints and their support for endpoint registration, directory-based dialing, and IP videoconferencing services. See the Huawei Videoconferencing product page and the Huawei Enterprise Support documentation portal for product documentation and configuration guidance.

QUESTION NO: 42

Illuminance, expressed in terms of the luminous flux accepted per unit area, is expressed in lux (Lux , lx). ( )

A. True

B. False

ANSWER: A

Explanation:

True is correct. Illuminance describes the amount of luminous flux falling on, or received by, a surface divided by that surface's area. Its SI derived unit is the lux, with symbol lx. One lux equals one lumen per square metre (1 lx = 1 lm/m²). In video-conference room design, illuminance is a practical lighting metric because it indicates how much useful light reaches participants, desks, walls, or presentation areas. Appropriate illuminance helps cameras capture faces clearly, supports natural colour reproduction, and avoids poor image quality caused by insufficient lighting. The wording "luminous flux accepted per unit area" reflects the standard concept of incident luminous flux per unit area, so the stated definition and unit are correct. The International System of Units recognizes lux as the special name for lumen per square metre, and the International Lighting Vocabulary defines illuminance in equivalent terms. See the BIPM SI Brochure and the CIE International Lighting Vocabulary definition of illuminance.

QUESTION NO: 43

What are the correct items in the following description of the RTCP protocol?

A. RTCP is primarily responsible for the real-time transmission of multimedia data streams

B. RTCP carries a permanent transport layer identifier for an RTP source, which is called the canonical name or CNAME

C. RTCP port number = RTP port number +1

D. RTCP provides monitoring of quality of service and delivers informational answers from participants in media sessions

ANSWER: B C D

Explanation:

RTCP is the RTP Control Protocol and operates alongside RTP to supply control and feedback functions for an RTP media session. The statement that RTCP carries a permanent transport layer identifier called the canonical name, or CNAME, is correct because RTCP Source Description (SDES) packets include the CNAME item. This identifier provides a persistent name for an RTP source and is used to associate streams from the same participant, including streams that may use different synchronization source identifiers.

The RTP/RTCP port-pairing convention is also correct in the standard deployment model: RTP uses an even-numbered UDP port and the associated RTCP traffic uses the next higher, odd-numbered UDP port. Thus, RTCP port number = RTP port number +1 is the expected convention in traditional RTP operation. RTCP also provides reception reports and sender reports containing packet-loss, jitter, sequence-number, and timing information. Participants use this feedback to monitor session quality, synchronize media, and obtain information about other participants. These functions are defined in RFC 3550, particularly the RTCP and SDES specifications. See also the RTP overview in IANA RTP Parameters.

QUESTION NO: 44

The following description of the advantages and disadvantages of packet loss retransmission ARQ is correct.

A. The latency requirements are high, and the existing network is difficult to use; Only for small packet loss scenarios; Aggravate image jitter

B. It basically does not occupy bandwidth resources and has little impact on image quality

C. Increase redundant data, occupy a certain amount of bandwidth, and reduce video efficiency; There is no way for sudden packet loss; Introduce a certain delay

D. Through the group combination check, it is possible to recover the packet loss data with a certain probability, which is more effective for the general packet loss scenario

ANSWER: A B

Explanation:

ARQ (Automatic Repeat reQuest) protects a media stream by detecting a missing packet and requesting that the sender retransmit that specific packet. This makes it most practical where packet loss is relatively limited and where the retransmitted packet can still arrive before its playback deadline. Consequently, it has stringent end-to-end delay requirements and can introduce additional delay variation: a receiver must wait for, receive, and reorder the retransmitted packet. In real-time video, that waiting can be perceived as increased image jitter when network delay is not sufficiently low or stable. These characteristics are captured by “The latency requirements are high, and the existing network is difficult to use; Only for small packet loss scenarios; Aggravate image jitter.”

Unlike proactive recovery methods that continuously send parity or duplicate data, ARQ does not add a standing redundancy stream during normal, loss-free transmission. Its bandwidth cost is therefore generally minimal when losses are rare, while successful recovery prevents missing media data from becoming visible video degradation. This is the intended benefit described by “It basically does not occupy bandwidth resources and has little impact on image quality.” RTP retransmission is specified as a mechanism for sending requested original RTP packets, and its usefulness depends on timely feedback and the packet still being relevant for playback. See RFC 4588: RTP Retransmission Payload Format and RFC 4585: RTP/AVPF Feedback Profile.

QUESTION NO: 45

During the work of the primary GK, the GK address registered by the terminal is the primary GK address.

A. True

B. False

ANSWER: B

Explanation:

False is correct. In a Huawei video-conferencing GK active/standby deployment, terminals should use the GK service address, commonly implemented as a floating or virtual IP address, rather than treating the physical address of the currently active primary GK as the registration address. This service address remains available to terminals regardless of which GK node is active. While the primary GK is operating, it provides the gatekeeper service through that shared service address; if a switchover occurs, the standby GK can assume the same service address and continue processing registrations and calls with minimal terminal-side reconfiguration. This design is fundamental to high availability because terminals do not need to detect and manually replace a primary node’s physical address after a fault or role change. H.323 defines gatekeeper discovery and Registration, Admission, and Status (RAS) procedures, under which terminals communicate with the gatekeeper service selected for the zone. Huawei’s enterprise support portal also provides product documentation for configuring video-conferencing GK redundancy and related service addresses. See the ITU-T H.323 recommendation and the Huawei Enterprise Support documentation portal.

QUESTION NO: 46

In multimedia services, RTP is used for real-time data packet transmission, and RTCP is used to detect the quality of service of an RTP session. ( )

A. True

B. False

ANSWER: A

Explanation:

True is correct. RTP (Real-time Transport Protocol) carries the actual real-time media stream, such as encoded audio and video, over an IP network. Its packet headers provide essential media-delivery information, including sequence numbers and timestamps, allowing receivers to reorder packets, identify packet loss, and maintain appropriate playback timing.

RTCP (RTP Control Protocol) operates alongside RTP and provides control and monitoring information for an RTP session. RTCP receiver reports and sender reports communicate metrics such as the fraction of packets lost, cumulative packet loss, extended highest sequence number received, interarrival jitter, and timing information. Endpoints can use these statistics to evaluate the effective quality of the media transmission and, where supported, adjust media behavior or report session quality. RTCP also supports participant identification and synchronization between media streams, particularly audio and video.

Therefore, the statement correctly distinguishes RTP as the protocol used to transport real-time media packets and RTCP as the companion protocol used to monitor and report RTP session delivery quality. This RTP/RTCP architecture is defined by the IETF in RFC 3550. Additional RTP protocol context is available from the IETF Datatracker.

QUESTION NO: 47

Mixing is actually mixing the loudest venues except for your own venue, not mixing all the venue sounds.

A. True

B. False

ANSWER: A

Explanation:

True is correct. In a multipoint video conference, the MCU’s audio-mixing behavior is designed to provide each participating venue with a useful combined audio stream while avoiding the venue’s own transmitted audio being returned to it. The system selects the active or loudest remote speaker streams according to its configured mixing capacity, combines those streams, and sends the resulting mixed audio to each endpoint. For a particular venue, its own audio is excluded from the audio stream delivered back to that same venue. This prevents an endpoint from hearing a delayed copy of its own microphone signal, which could otherwise create echo, distracting reverberation, or an unnatural monitoring experience. The wording “loudest venues” reflects the common voice-activated conference model: audio resources are prioritized for speakers with the highest current speech levels rather than indiscriminately combining every connected venue. This approach preserves intelligibility and keeps the mixed audio within the MCU’s supported audio-processing limits. Huawei’s enterprise video-conferencing portfolio and related technical documentation describe multipoint conferencing capabilities and MCU-based media processing: Huawei Video Conferencing and Huawei Enterprise Support Documentation.

QUESTION NO: 48

RAS performs management functions, mainly including :() (3 answers).

A. Registration

B. Admission control

C. Security

D. Status

ANSWER: A B D

Explanation:

RAS stands for Registration, Admission, and Status, and it is the H.323 control protocol used between an endpoint and a gatekeeper. Its central purpose is to let the gatekeeper manage endpoint participation and call-related resource decisions. **Registration** is a core RAS function because endpoints use RAS registration exchanges to announce themselves to the gatekeeper and provide addressing or capability-related information needed for management. **Admission control** is also fundamental: before a call is established, an endpoint uses RAS admission signaling to request permission, enabling the gatekeeper to apply policies such as bandwidth allocation, call authorization, and routing control. **Status** is the third principal function, allowing the gatekeeper to query or monitor endpoint state through RAS status-related exchanges. Together, these functions provide the gatekeeper with the information and control required to manage H.323 terminals and conferences. The ITU-T H.225.0 specification defines the call-signaling and RAS framework used by H.323 systems; see ITU-T Recommendation H.225.0. Huawei's enterprise video-conferencing portfolio uses standards-based conferencing technologies and related endpoint-management capabilities; see Huawei Video Conferencing.

QUESTION NO: 49

What are the correct items for the following description of the performance of the RSE6500 product?

A. Supports up to 30-channel meeting HD recording

B. Smooth scaling can be achieved by adding Licenses

C. Supports up to 20 HD live streams

D. Supports up to 2000 web-on-demand

ANSWER: A B C D

Explanation:

The RSE6500 is Huawei's recording and streaming media server for videoconferencing deployments. Its performance specification includes concurrent high-definition meeting recording, with capacity for up to 30 HD recording channels. It also supports up to 20 HD live streams, allowing recorded or live conference content to be distributed concurrently to viewers. For on-demand access, the platform supports up to 2,000 concurrent web video-on-demand users, which is intended for enterprise-scale viewing of archived conference recordings.

Capacity can be expanded smoothly through licenses. This licensing-based approach enables an organization to increase applicable service capacity without replacing the underlying RSE6500 hardware, supporting phased deployment as recording, streaming, and viewing demand grows. These capabilities must be considered together: the stated recording-channel, live-stream, and web-on-demand figures describe different service dimensions of the same product rather than mutually exclusive features. Huawei product documentation and downloadable materials can be located through the Huawei Enterprise Support search for RSE6500 and the Huawei Enterprise Communication product portfolio.

QUESTION NO: 50

Which of the following features does the gatekeeper provide (3 answers).

A. Address Translation

B. Admission Control

C. Bandwidth Control

D. Conference Control

ANSWER: A B C

Explanation:

In an H.323 videoconferencing network, a gatekeeper is the central control entity for a zone and provides essential registration, admission, and resource-management services. **Address Translation** is a core gatekeeper function: it resolves an endpoint's H.323 alias, such as an E.164 number or URI-style identifier, into the transport address needed to establish signaling with that endpoint. **Admission Control** is provided through the H.323 Registration, Admission, and Status (RAS) procedures. Endpoints request permission before placing or accepting calls, enabling the gatekeeper to apply zone policies and determine whether the call may proceed. **Bandwidth Control** is also performed during admission and can be adjusted during an active call; the gatekeeper authorizes requested bandwidth and helps ensure that aggregate media usage remains within the configured capacity of the zone. These functions allow Huawei video-conferencing deployments to use dialable aliases, control endpoint access to call resources, and manage available bandwidth centrally. For H.323 gatekeeper roles and RAS-based admission and bandwidth management, see the ITU-T H.323 recommendation and the ITU-T H.225.0 recommendation.

QUESTION NO: 51

The network diagnostic tool supports remote diagnostic mode, and the network diagnostic tool remotely controls another network diagnostic tool to initiate a network diagnosis and judgment against other

network diagnostic tools.

A. True

B. False

ANSWER: A

Explanation:

True is correct. Huawei's video-conferencing network diagnosis capability is designed for troubleshooting distributed endpoint networks, where a fault may be located on a different device or at a different site from the operator. Remote diagnosis allows one network diagnostic tool to act as the controlling side and instruct a diagnostic tool at another location to perform diagnostic operations. This makes it possible to collect connectivity and network-quality information from the remotely controlled side and use those results for diagnosis and fault judgment involving other diagnostic tools or sites.

This mechanism is useful because media-quality problems can be caused by path-specific conditions such as reachability failures, packet loss, delay, jitter, or routing issues. A centrally operated remote diagnostic workflow avoids requiring an engineer at every endpoint while still obtaining measurements from the relevant network locations. It therefore supports faster isolation of inter-site conferencing faults and aligns with Huawei's centralized management and O&M approach for enterprise video-conferencing solutions. Huawei product and documentation resources are available through the Huawei Enterprise Support portal and the Huawei video conferencing product page.

QUESTION NO: 52

RAS messages are a type of message specified in the H.225.0 protocol, which is a protocol used between nodes and gateways to perform management functions, mainly including registration, access

control, and status.

A. True

B. False

ANSWER: B

Explanation:

False is correct because H.225.0 RAS (Registration, Admission, and Status) signaling is specifically the control protocol used between an H.323 endpoint and a gatekeeper, rather than a protocol defined for communication "between nodes and gateways." The gatekeeper is the central control entity for an H.323 zone and uses RAS exchanges to manage endpoint participation and call authorization. Typical RAS procedures include endpoint registration and unregistration, admission requests before calls are established, bandwidth-change requests, disengagement, location resolution, and status or information reporting. These procedures enable the gatekeeper to maintain endpoint information and apply zone-level admission and bandwidth-control policies. Gateways can be H.323 endpoints and may therefore use RAS when they are registered in a gatekeeper's zone, but the defining RAS relationship remains endpoint-to-gatekeeper. The ITU-T H.225.0

recommendation identifies the registration, admission, and status signaling functions within the H.323 system architecture. See the [ITU-T H.225.0 recommendation] and the [ITU-T H.323 system recommendation] for the relevant protocol and entity definitions.

QUESTION NO: 53

After the endpoint enables the NAT setting, it can change the packet content and can be used for static NAT networking mode and DNAT/NAPT networking mode.

A. True

B. Error

ANSWER: B

Explanation:

Error is correct because the statement combines NAT behaviors in a technically inaccurate way. NAT is primarily a network-device function that translates addressing information in packet headers while forwarding traffic. Static NAT establishes a fixed one-to-one address mapping, DNAT translates a destination address according to a configured rule, and NAPT additionally multiplexes translations using transport-layer port numbers. These are distinct translation mechanisms, rather than a single endpoint setting that broadly "changes packet content."

For video conferencing, an endpoint NAT-related setting is intended to support the endpoint's signaling and media operation in a NAT environment, according to the supported deployment and traversal design. It should not be described as a general-purpose capability that modifies packet payload content for every static-NAT, DNAT, and NAPT scenario. Where application signaling contains embedded addressing information, successful traversal may require an appropriate NAT traversal mechanism, an application-layer gateway, or a properly designed conferencing-network deployment. This distinction is important because address translation and application-payload handling are separate functions.

The fundamental NAT terminology and behavior are defined in [RFC 3022: Traditional IP Network Address Translator]. Huawei's enterprise support portal provides the product documentation and configuration guidance applicable to the specific endpoint model and software version: [Huawei Enterprise Support].

QUESTION NO: 54

The system uses the H.245 protocol as the control protocol for controlling the communication channel. ( )

A. establish

B. Bandwidth limit

C. release

D. Maintenance

ANSWER: A C D

Explanation:

H.245 is the H.323-family control protocol used to negotiate and manage logical media channels between endpoints. Its logical-channel procedures cover opening a channel before media is sent, maintaining the channel while a call is active, and closing the channel when that media stream is no longer required or the conference ends. Therefore, *establish*, *release*, and *Maintenance* correctly describe communication-channel control functions performed through H.245.

In practical video conferencing, these procedures allow endpoints to agree on media capabilities and then create the appropriate unidirectional logical channels for audio, video, data, or control streams. Once established, H.245 messages can manage the state and operation of those channels during the session, including procedures associated with media flow and channel control. At call teardown or when a particular stream must stop, H.245 logical-channel closing procedures release the relevant channel cleanly. This role is fundamental to H.323 call control because signaling for call setup and registration is distinct from the detailed control of the individual media channels. The ITU-T H.245 recommendation defines these multimedia communication control procedures; see ITU-T Recommendation H.245. Additional H.323 protocol architecture context is available from ITU-T Recommendation H.323.

QUESTION NO: 55

In video conferencing, in order to improve the quality of speech, we can add some voice enhancement technology, such as ( ).

A. VAD

B. CNG

C. AAC-LD

D. EC

ANSWER: A B D

Explanation:

VAD, CNG, and EC are speech-processing technologies commonly used to make conversational audio clearer and more natural in video-conferencing deployments. VAD (voice activity detection) identifies whether a participant is currently speaking. This supports efficient handling of silent intervals and enables related functions such as discontinuous transmission, reducing unnecessary audio transmission without treating silence as active speech. CNG (comfort noise generation) supplies a low-level, background-like noise during periods in which speech transmission is suppressed. Maintaining this natural acoustic continuity prevents a far-end listener from perceiving abrupt, unnatural dead silence. RFC 3389 defines payload support for comfort noise in RTP-based real-time media: RFC 3389. EC (echo cancellation) removes acoustic echo caused when loudspeaker output is picked up by a microphone and returned to remote participants. It is particularly important for hands-free terminals and conference rooms, where echo can severely reduce intelligibility and disturb turn-taking. ITU-T's echo-canceller recommendation describes performance and operational requirements for this function: ITU-T G.168. Together, these functions improve perceived speech quality and the usability of full-duplex conferencing conversations.

QUESTION NO: 56

Which of the following video resolutions meet the 16:9 aspect ratio requirement?

A. 720*480

B. 1024*768

C. 1280*720

D. 1920*1080

ANSWER: C D

Explanation:

The resolutions **1280*720** and **1920*1080** meet the 16:9 aspect-ratio requirement because, in each case, the horizontal pixel count divided by the vertical pixel count equals 16 divided by 9. For 1280*720, simplifying both values by 80 gives 16:9. For 1920*1080, simplifying both values by 120 also gives 16:9. These are the widely used HD and Full HD progressive video formats commonly denoted as 720p and 1080p. In video-conferencing systems, selecting a standard 16:9 resolution ensures that the encoded image is displayed in the intended widescreen geometry when endpoints and displays use square pixels. The ITU-R HDTV production recommendation identifies 1280 × 720 and 1920 × 1080 as image formats with a 16:9 image aspect ratio; see ITU-R Recommendation BT.709. The broader HDTV system specification, including these formats, is also available from the ITU-R Recommendation BT.2020.

QUESTION NO: 57

The signal stream of the audio self-ring is. ( )

A. Microphone - > Audio In - > Audio Output -) Speaker

B. Audio Output - > Audio input. Speaker - Microphone

C. Microphone - > Audio Output - > Audio Output as -> Speaker

D. Speaker audio input - > audio output -) microphone

ANSWER: A

Explanation:

The correct audio self-loop signal path is **Microphone - > Audio In - > Audio Output -) Speaker**. An audio self-loop is a local verification procedure used to confirm that the endpoint's complete audio path is functioning. Sound captured by the microphone is first handled at the audio-input stage, where the endpoint receives and processes the microphone signal. The processed signal is then routed to the audio-output stage, which drives the connected loudspeaker so that the captured sound can be heard locally. This end-to-end path validates the microphone, input interface, internal audio processing, output interface, and speaker together. In video-conferencing deployment and maintenance, this type of loop test is useful for isolating local audio-path faults before investigating conference-side settings, network transport, or remote endpoints. Huawei's enterprise communications portfolio includes video-conferencing endpoints and related collaboration products whose installation, operation, and troubleshooting materials are available through the Huawei Enterprise support portal. See Huawei Enterprise Technical Support and Huawei Enterprise Communications.

QUESTION NO: 58

Which of the following signaling messages are capable of exchanging?

A. TCSR

B. TCS

C. TCSA

D. IRR

ANSWER: A B C

Explanation:

In an H.323 video-conferencing call, endpoint capabilities are exchanged through the H.245 terminal-capability-set procedure. **TCS** is the TerminalCapabilitySet message that an endpoint sends to advertise its supported receive capabilities, such as audio, video, data, and applicable codec-related parameters. The receiving endpoint processes that capability information and responds with either **TCSA**, the TerminalCapabilitySetAck message, when the received capability set is accepted, or **TCSR**, the TerminalCapabilitySetReject message, when the capability-set request cannot be accepted. These messages therefore belong to the same H.245 request/response exchange and are used to establish the mutual media capabilities required before logical channels can be opened for a conference. Capability exchange is fundamental to H.323 interoperability because each terminal must know what formats and media functions the peer can receive. The H.245 protocol definition identifies TerminalCapabilitySet together with its acknowledgement and rejection responses as part of this procedure. See the ITU-T H.245 recommendation and the ITU-T H.323 recommendation.