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Polycom Certified Videoconferencing Engineer (PCVE)

Polycom 1K0-001

Version Demo

Total Demo Questions: 32

Total Premium Questions: 326

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Topic Break Down

Topic	No. of Questions
Topic 1, Topic 0	173
Topic 2, Topic 1	133
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QUESTION NO: 1

PTZ Control may be performed by a camera automatically by using(select 2)

A. Music tracking.

B. Voice tracking.

C. Image tracking.

D. VTC microphone tracking.

E. Shadow tracking.

ANSWER: B C

Explanation:

PTZ cameras can automate pan, tilt, and zoom adjustments through voice tracking and image tracking. Voice tracking uses the active speaker's audio location—typically derived from microphone-array processing—to direct the camera toward the person speaking. This enables a remote participant to see the current speaker without requiring an operator to manually reposition the camera. Image tracking uses video-based analysis to identify and follow a person or subject in the camera image. The camera then adjusts its framing as that person moves, helping preserve an appropriate view during a presentation or meeting. These automatic tracking capabilities are intended to improve the remote viewing experience and reduce the need for manual PTZ control. Poly video solutions document camera-management and tracking-related configuration through their support resources; see the HP Poly Support portal for product documentation and software, and the Poly Support site for legacy Polycom documentation and support material.

QUESTION NO: 2

Which of the following are two examples of Fiber Optic interface problems? (Choose two of the following options.)

. Splice loss and connection loss

B. Noise

C. Gravity

D. Cable bends

E. EMI - Electro Magnetic Interference

F. RFI Radio Frequency Interference

nswer: ,D

Explanation:

A. Noise

B. Gravity

C. Cable bends

D. EMI - Electro Magnetic Interference

E. RFI Radio Frequency Interference

ANSWER: C

Explanation:

Cable bends is the correct available single-choice answer because bending an optical-fiber cable can impair the light path and cause attenuation. A severe bend, commonly called a macrobend, allows some guided light to escape from the fiber core rather than remaining confined by total internal reflection. This loss can reduce received optical power enough to cause unstable service, errors, or loss of link synchronization. Small-scale deformation or pressure on the fiber can also introduce bending-related loss. Proper fiber installation therefore requires adherence to the manufacturer's minimum bend-radius specification, both while pulling cable and after it is dressed in racks, trays, or patch panels. Fiber routes should avoid sharp corners, tight cable ties, pinching, and excessive compression. The Fiber Optic Association describes bend loss as an important installation and troubleshooting consideration, while Corning's installation guidance emphasizes maintaining bend radius to protect optical performance. See The Fiber Optic Association technical reference and Corning Optical Communications.

QUESTION NO: 3

Of the following, select three functions that the ISDN Functional Group NT-1 for Basic Rate

Interface provides.

A. May provide conversion between LEC and the PBX

B. May provide phantom power over the TX and RX loops of S/T reference

C. Provides conversion between the 2-wire U and 4-wire S/T references

D. Provides power to allow the U Reference to be longer from the Local Exchange

E. Allows the Local Exchange to Down speed the local loop

F. Allows the Local Exchange to access and perform maintenance to local loop

ANSWER: B C F

Explanation:

An NT-1 is the network-termination function used with Basic Rate ISDN to terminate the carrier-side two-wire U interface and present the customer-side four-wire S/T interface. Therefore, "Provides conversion between the 2-wire U and 4-wire S/T references" is a fundamental NT-1 function. This includes the line-interface and transmission functions necessary to bridge the distinct physical reference points defined for BRI service.

"May provide phantom power over the TX and RX loops of S/T reference" is also correct because an NT-1 can provide power toward compatible terminal equipment over the S/T interface. This arrangement supports connected ISDN devices without requiring separate local powering in every deployment, subject to the applicable equipment design and power budget.

"Allows the Local Exchange to access and perform maintenance to local loop" is correct because NT-1 equipment supports the physical-layer maintenance and testing relationship associated with the network-side local loop. The network can use this capability to supervise and maintain the U-interface connection between the local exchange and the NT-1. These roles align with the ISDN BRI layer-one characteristics and NT functions described in ITU-T Recommendation G.961 and Cisco's ISDN technology overview.

QUESTION NO: 4

Which of the following describe video dynamics as processed through a videoconference system?

A. Cameras react to light the same as ones eye and video monitors accurately render the video information so what one sees by eye is what the far-end sees.

B. Cameras have limits to the range of light levels which can be accommodated and tend to process the darker regions further toward black than what ones eye sees.

C. Codecs have the same limits as cameras and therefore will not change the perceived qualities of the video as rendered by the cameras.

D. Codecs have more restrictive limits than cameras, and the best way to accurately assess the video quality sent to the far-end is to view a video scene fully coded and then decoded.

ANSWER: B D

Explanation:

Cameras have limits to the range of light levels which can be accommodated and tend to process the darker regions further toward black than what ones eye sees. This reflects the fundamental difference between human vision and a camera imaging chain. The eye adapts dynamically to changing illumination and can perceive useful scene detail across a very broad brightness range. A camera sensor, lens exposure setting, and video-processing pipeline operate within finite exposure and signal ranges. Consequently, low-light areas can lose shadow detail or be represented closer to black, even where a person standing in the room can still distinguish objects.

Codecs have more restrictive limits than cameras, and the best way to accurately assess the video quality sent to the far-end is to view a video scene fully coded and then decoded. In a videoconference, the source image is not the final transmitted experience: it is compressed, packetized, received, decoded, and displayed. Compression can affect fine detail, motion rendition, gradients, and dark-area detail, particularly when bandwidth is constrained or the scene contains motion and complex textures. Assessing the decoded far-end image therefore evaluates the complete practical video path rather than only the camera output. Poly's video-system documentation and ITU video-coding standards provide useful background: Poly Support and ITU-T H.264 Recommendation.

QUESTION NO: 5

Defective Local Loop may introduce what two problems? (Pick two of the following options.)

A. Cyclic Redundancy Errors

B. Network Channel Clocking Delay

C. Over-voltage to codec power supply

D. Over-voltage to the codec camera

E. Intermittent codec network connections

ANSWER: E

Explanation:

Intermittent codec network connections is the best single-choice answer because a defective local loop is a physical-access-network fault that can cause the Ethernet or WAN path serving the video endpoint to become unstable. Local-loop defects may be caused by damaged cabling, poor terminations, water ingress, electrical interference, or an unreliable carrier circuit. These conditions can produce brief loss of signal, repeated link renegotiation, or temporary loss of packet connectivity. From the codec's perspective, the practical symptom is that its network connection drops and then returns, which can interrupt registration, calls, management access, or media transport.

Poly video systems depend on a continuously available IP connection for signaling and media. When the access circuit repeatedly loses connectivity, the endpoint can appear to disconnect from the network even if its local configuration is correct. Physical-layer and access-link conditions should therefore be verified during troubleshooting, including cabling, switch-port status, carrier handoff status, and link-event logs. This aligns with Ethernet's requirement that the physical interface provide reliable transmission for network-layer communication; see RFC 1122. For Poly support documentation and endpoint troubleshooting resources, see the Poly Support portal.

QUESTION NO: 6

Under normal operating conditions, Microphone and Audio amplification system should have sufficient ________ to operate without clipping.

A. amplitude bias space

B. 100db padding

C. no padding

D. headroom

E. videoroom

F. voltage feedback

ANSWER: D

Explanation:

headroom is the correct term for the level margin deliberately retained between the normal operating signal level and the maximum level a microphone preamplifier, mixer, amplifier, digital signal processor, or loudspeaker path can handle before clipping. Speech and other program material have short-duration peaks that can be substantially higher than their average level. A properly designed conferencing-audio gain structure therefore leaves sufficient headroom so those peaks pass cleanly while routine speech remains at an appropriate nominal level.

Clipping occurs when a stage is driven beyond its available voltage range or digital full-scale limit. It truncates waveform peaks and creates audible distortion, which can reduce speech intelligibility and degrade far-end conference participants' experience. In Polycom room-audio deployments, setting input gains, output gains, and processing levels so that normal speech and expected peak levels remain below overload thresholds is a fundamental commissioning practice. The needed margin is called headroom; it allows level variation without requiring the system to operate at or beyond its limit. For further audio-system terminology and gain-structure context, see the Audio Engineering Society standard on digital-audio engineering guidelines and Shure's audio gain structure guidance.

QUESTION NO: 7

Which of the following are the two most correct definitions of ISDN reference points?

A. U Reference point is between the ISDN terminal and the network termination

B. U Reference point is between the Network Termination and the Network Exchange Equipment

C. S/T Reference point is between ISDN Terminal Equipment and the Network Termination

D. S/T Reference point is between the Network Termination and the Network Exchange Equipment

ANSWER: B C

Explanation:

The correct definitions are "U Reference point is between the Network Termination and the Network Exchange Equipment" and "S/T Reference point is between ISDN Terminal Equipment and the Network Termination." In the ISDN reference configuration, the U reference point identifies the access transmission interface between the customer-side network termination, normally the NT1, and the carrier's local exchange or network-side line equipment. It is therefore the point at which the ISDN access loop connects to the public network.

The S and T reference points describe the customer-premises interface used by terminal equipment. More precisely, S is the interface between terminal equipment and an NT2, while T is between an NT2 and an NT1. In many practical Basic Rate Interface implementations, the NT1 and NT2 functions are combined or the distinction is not exposed to the endpoint; this is commonly referred to as the S/T interface. Consequently, describing S/T as being between ISDN terminal equipment and the network termination is the appropriate general definition for this context. These standardized demarcation points identify the functional boundaries that enable compatible ISDN terminal and network interconnection. See the ITU-T ISDN reference configuration in ITU-T Recommendation I.411 and the overview of ISDN reference points in Cisco ISDN documentation.

QUESTION NO: 8

When connecting to a Virtualization Manager (DMA) for the first time, which is the correct username and password?

A. admin / admin

B. admin / password

C. POLYCOM / POLYCOM

D. SUPPORT / SUPPORT

ANSWER: A

Explanation:

The correct initial credentials for a Polycom RealPresence DMA (formerly Polycom DMA) system are **admin / admin**. The DMA appliance uses the built-in `admin` account for first-time access to its management interface and initial configuration workflow. This initial administrative sign-in allows the installer to complete foundational deployment tasks, including network configuration, licensing, integration with the enterprise directory where applicable, and creation of additional administrator accounts. Because this is a default credential, it should be changed promptly as part of commissioning the system and protected according to the organization's access-control policy. Using the documented initial account is important because it provides the privileges required to complete the first-run setup process; it is not merely a user-level sign-in. Poly documentation and support resources for RealPresence DMA products are available through the Poly Support portal. The currently available product documentation library, including legacy Polycom documentation access, is provided at Poly Documentation.

QUESTION NO: 9

The Physical layer takes care of:

A. how the 0s and 1s are sent across the physical network

B. how any encryption is applied to the payload

C. what kind of data is being sent

D. whether MAC addresses or IP addresses are used

ANSWER: A

Explanation:

"how the 0s and 1s are sent across the physical network" is correct because the Physical layer defines the transmission of an unstructured bit stream over the actual communication medium. Its responsibilities include representing binary values as electrical, optical, or radio signals; signal timing and synchronization; connector and pin characteristics; cable or wireless-media requirements; and physical data rates. In a video-conferencing deployment, this layer includes the practical network path that carries endpoint traffic, such as Ethernet cabling, switch ports, fiber links, Wi-Fi radio transmission, and the signaling characteristics needed for those links to operate reliably. The Physical layer does not interpret the meaning of the transmitted data; it ensures that the bits can be placed onto, propagated through, and received from the medium. This basic role is important when troubleshooting endpoint connectivity: link state, cable quality, port speed, duplex behavior, optical levels, and wireless signal conditions are physical-network concerns that can prevent communication before higher-layer protocols can function. Cisco's OSI model overview describes the Physical layer as transmitting raw bits across the network medium, while IEEE 802.3 defines widely used Ethernet physical-media specifications. See Cisco's Ethernet and OSI model documentation and IEEE 802.3 Ethernet standards information.

QUESTION NO: 10

Which H.263 Annex is used to implement 60 / 50 Fields per Second Intra Video -Sync

Operation?

A. Annex N

B. Annex F

C. Annex P

D. Annex W

E. Annex Q

ANSWER: D

Explanation:

Annex W is used for 60/50 fields-per-second Intra Video-Sync operation. This H.263 optional annex defines the intra-video coding support needed for interoperability in video applications that must synchronize intra content to the 60 Hz or 50 Hz field rates associated with television-based video sources. In a videoconferencing environment, this capability is relevant when encoding or receiving interlaced-origin video and maintaining the intended temporal relationship of the fields. Identifying Annex W correctly matters because H.263 capabilities are negotiated as distinct optional annexes; endpoints must advertise and support the applicable annex to use its associated coding behavior reliably.

The authoritative H.263 specification is maintained by the ITU-T and includes the base video-coding syntax together with its optional annexes. Consult the ITU-T H.263 recommendation at ITU-T Recommendation H.263. The ITU-T recommendations catalogue also provides the current publication context and available editions at ITU-T H-Series Recommendations. Therefore, the applicable selection is Annex W.

QUESTION NO: 11

A signal which is made up of a sequence of discrete values over time is:

A. A beacon

B. Analog

C. Television

D. Digital

ANSWER: D

Explanation:

Digital is correct because a digital signal represents information using distinct, separate values rather than a continuously variable quantity. In communications systems, those values are commonly binary states, such as 0 and 1, represented electrically or optically by defined voltage, power, or symbol levels. The signal can therefore be sampled, encoded, transmitted, regenerated, and decoded while retaining its intended discrete representation. This is fundamental to videoconferencing and unified communications: voice and video are converted into digital samples, compressed by codecs, packetized, transported across IP networks, and reconstructed at the endpoint. The word “over time” does not change the classification; it simply recognizes that the discrete symbols occur in an ordered sequence. A signal is digital when its possible amplitudes or symbols come from a finite set of defined levels, even though those symbols are sent at particular times. For a general technical definition of digital signals and their use in representing data, see Encyclopaedia Britannica: digital signal. Background on digitizing audio through sampling and quantization, a typical source for media signals in conferencing systems, is available from the FCC’s digital radio overview.

QUESTION NO: 12

What does Automatic Gain Control (AGC) do?

A. It turns up the volume to the far-end microphone.

B. It selects among multiple microphones.

C. It turns down the volume to the far-end microphone.

D. It builds the volume to the near-end microphone.

ANSWER: D

Explanation:

Automatic Gain Control (AGC) automatically adjusts the gain applied to the local, or near-end, microphone signal so that people in the room are transmitted at a more consistent speech level. When a nearby participant speaks quietly or is positioned farther from the microphone, AGC can raise the microphone signal level—described here as building the volume to the near-end microphone—so their speech is more intelligible to remote participants. This processing occurs on the captured local audio before it is sent to the far site. AGC is dynamic rather than a fixed volume setting: it continually responds to changes in incoming speech level to maintain a usable level while supporting natural conferencing audio. In Polycom conferencing deployments, microphone placement, room acoustics, and appropriate audio configuration remain important, but AGC helps compensate for ordinary differences in how loudly near-end participants speak. Polycom documentation and support resources are available through the Poly Support site; additional conferencing-product documentation can be located through the HP Poly support portal.

QUESTION NO: 13

Which of the following correctly describes the purpose of an Acoustic Echo Canceller

(AEC)?

A. An AEC provides a means to reduce room noise sent to the far-end.

B. An AEC provides a means to reduce room reverberation as heard by the far-end.

C. An AEC provides compensation for room acoustics to improve microphone sensitivity and gain.

D. An AEC reduces the room loudspeaker contribution to the local microphone sent to the far-end.

E. An AEC provides compensation for room acoustics to better allow the reduction of echo returned to the far-end.

ANSWER: D E

Explanation:

An Acoustic Echo Canceller operates by modeling the acoustic path between a room's loudspeaker and its microphone. During a conference, audio received from the far end is reproduced through the local loudspeaker. Some of that reproduced signal can be picked up by the local microphone and transmitted back to the far end as a delayed echo. "An AEC reduces the room loudspeaker contribution to the local microphone sent to the far-end" correctly states this core function: the canceller identifies and removes the loudspeaker-originated component from the microphone transmit signal while preserving the local talker as effectively as possible.

"An AEC provides compensation for room acoustics to better allow the reduction of echo returned to the far-end" is also correct. Echo cancellation adapts to the characteristics of the room, including the loudspeaker-to-microphone acoustic path and associated delays, so that the far-end reference signal can be subtracted accurately before transmission. This is particularly important in conferencing rooms where loudspeaker volume, microphone placement, and changing room conditions affect the echo path. The relevant performance principles for voice echo cancellers are specified in ITU-T Recommendation G.168. Poly conferencing support resources are available through HP Poly Support.

QUESTION NO: 14

What is the best way to power down an HDX endpoint?

A. Using the on/off button on the rear panel

B. Unplug the power

C. Using the browser interface

D. Using the power button

ANSWER: D

Explanation:

Using the power button is the correct method because it initiates the HDX system's controlled shutdown process. A controlled shutdown allows the endpoint to stop active services, close system processes, and write configuration or operational data before electrical power is removed. This protects the integrity of the operating software and stored configuration, and it leaves the endpoint in a known safe state for its next startup. On HDX systems, the power control is intended for normal local power-down operation; depending on the endpoint model and installed interface, this may be the system power control or the corresponding power function available through the user interface. The operator should allow the shutdown sequence to finish completely before removing AC power or performing maintenance. Polycom's HDX documentation distinguishes routine power-down procedures from abrupt loss of power and directs administrators to use the normal power control for a graceful shutdown. Refer to the Polycom HDX Administrator Guide resources on HP Support and the Poly documentation and support portal for legacy HDX product documentation.

QUESTION NO: 15

Which four digital interfaces are capable of operating at rates up to 2 Million Bits per second?

A. RS-232

B. V.35

C. V.25

D. RS530

E. X.21

F. RS449

G. X.25

H. Q.931

ANSWER: B D E F

Explanation:

The four applicable interfaces are **V.35**, **RS530**, **X.21**, and **RS449**. These are synchronous serial physical-interface standards used for connecting data terminal equipment to data circuit-terminating equipment, including leased-line and packet-network access devices commonly encountered in legacy videoconferencing deployments. Their electrical and mechanical characteristics were designed for higher-speed serial WAN operation, with 2 Mbps being a commonly cited upper operating rate for these interface families in this context.

V.35 is the traditional high-speed serial interface associated with synchronous data transmission and is widely used at rates approaching 2 Mbps. **RS530** provides an EIA/TIA synchronous serial interface intended for high-speed operation, while **RS449** similarly defines balanced and unbalanced serial-interface arrangements suitable for these data rates. **X.21** is an ITU-T synchronous DTE/DCE interface recommendation used with digital public data networks and supports the required rate range. These interfaces therefore satisfy the question's requirement for four digital interfaces capable of operation up to 2 million bits per second. See the ITU-T V.35 recommendation and the ITU-T X.21 recommendation.

QUESTION NO: 16

Two examples of companding algorithms are:

A. B-law

B. A-law

C. u-law

D. O-law

ANSWER: B C

Explanation:

A-law and **u-law** are the two standardized logarithmic pulse-code modulation companding characteristics used with G.711 digital telephony audio. Companding combines compression at the encoder with complementary expansion at the decoder. This logarithmic mapping gives greater effective resolution to lower-level speech signals, where quantization noise would otherwise be more noticeable, while accommodating the broad dynamic range of human voice using 8-bit PCM samples. A-law is the companding characteristic traditionally used in many regions outside North America and Japan. The character commonly written as “u” is the Greek letter mu (μ); μ-law is traditionally used in North America and Japan. Both methods are explicitly defined by the ITU-T G.711 recommendation, which specifies the 64 kbit/s PCM coding of voice frequencies and its two companding laws. In Polycom and other interoperable VoIP or video-conferencing environments, recognizing A-law and μ-law is important when configuring or troubleshooting G.711 audio payloads, gateways, and PSTN-facing interfaces. See the ITU-T G.711 recommendation and RFC 3551 G.711 payload guidance.

QUESTION NO: 17

Which three options can be unlocked on an HDX endpoint using option keys?

A. 1080P

B. RTV

C. SIP

D. TIP

ANSWER: A B D

Explanation:

1080P, RTV, and TIP are features that can be enabled on applicable Polycom HDX systems through installed option keys. The 1080P option key enables the system’s 1080p video capability where the HDX model and connected components support it. RTV is an optional interoperability capability that enables use of Microsoft’s Real-Time Video codec in supported Microsoft integration scenarios. TIP is the Cisco TelePresence Interoperability Protocol capability, which allows a supported HDX endpoint to participate in compatible Cisco TelePresence environments after the appropriate license key is installed. Polycom documents these as licensed system capabilities managed from the endpoint’s administration interface, where the installed key determines which optional functions are available. This licensing approach lets an organization purchase and activate advanced video-resolution and interoperability functions without replacing the endpoint hardware. The applicable model, software release, and installed hardware must support the option before its key can be used. See the Polycom HDX administrator documentation available through the Poly support portal and the product documentation section at HP Poly Support.

QUESTION NO: 18

Which of the following apply to a Video Conferencing System? (Select three of the following options.)

. Coder / Decoder

B. udio Translator

C. udio, Video, Content, and Data Translator

D. nalog to Digital and Digital to nalog Converter

E. Video, udio, Content, and Data Multiplexer

nswer: ,D,E

Explanation:

A. Audio Translator

B. Audio, Video, Content, and Data Translator

C. Analog to Digital and Digital to Analog Converter

D. Video, Audio, Content, and Data Multiplexer

E. Coder / Decoder

ANSWER: C D E

Explanation:

A video conferencing endpoint performs several core media-processing functions. A **Coder / Decoder**, commonly called a codec, compresses outgoing media into a format suitable for transmission and decompresses received media for presentation. This encoding and decoding function is fundamental to standards-based conferencing because real-time audio and video must be represented efficiently while preserving usable quality and synchronization.

An **Analog to Digital and Digital to Analog Converter** is also a fundamental endpoint function. Cameras and microphones produce or handle physical signals, while displays and speakers ultimately require signals suitable for human presentation. The conferencing system therefore interfaces between analogue input/output devices and the digital media-processing path.

A **Video, Audio, Content, and Data Multiplexer** combines the relevant media and control/data streams into a coordinated transmitted stream or channel structure, with corresponding separation on reception. Multiplexing supports the concurrent exchange of video, audio, presentation content, and conferencing data over the available communications connection. These functions align with the audiovisual communication architecture described by the ITU-T H.320 framework and its multiplexing specification, ITU-T Recommendation H.320 and ITU-T Recommendation H.221.

QUESTION NO: 19

Which of the following is the cabling required for 2 pair (4 wire) 100BaseFast Ethernet?

A. CAT6

B. CAT5e

C. CAT3

D. CAT5

ANSWER: D

Explanation:

CAT5 is the applicable cabling category for the traditional 100BASE-TX Fast Ethernet requirement. 100BASE-TX transmits over two balanced twisted pairs, meaning four individual conductors: one pair carries transmitted data and one pair carries received data. In standard eight-position modular Ethernet wiring, these are the pairs associated with pins 1–2 and 3–6. Category 5 cabling was specified to provide the electrical performance needed for this 100 Mb/s copper Ethernet implementation, including the required bandwidth and crosstalk characteristics. This is why CAT5 is the expected answer when the question asks for the base cable requirement for two-pair Fast Ethernet. Category 5e is an enhanced, backward-compatible cabling category commonly deployed in newer installations, but CAT5 is the legacy category directly associated with the 100BASE-TX requirement stated here. The IEEE 802.3 Ethernet standard defines the family of Ethernet physical-layer specifications, including Fast Ethernet operation; see the [IEEE 802.3 standard page]. Cisco's Fast Ethernet overview also describes 100BASE-TX as a twisted-pair Fast Ethernet technology: [Cisco Fast Ethernet].

QUESTION NO: 20

When VSW mode is enabled in an RMX conference profile, how many resources does each participant use at 1080P resolution?

A. 1

B. 0.5

C. 3

D. 6

ANSWER: A

Explanation:

1 is correct. In Polycom RealPresence Collaboration Server (RMX) Video Switching (VSW) conferences, the system uses a switching-oriented resource model rather than continuous-presence composition for each endpoint. At 1080p, each connected participant consumes one video resource. This allocation rule is important for conference planning because the available RMX video-resource capacity determines the maximum number of 1080p VSW participants that can be admitted to a conference. VSW is intended for conferences where the RMX switches the active video stream among participants, instead of creating a composite layout that combines multiple sites. The conference profile's VSW setting therefore affects both the conferencing experience and the capacity calculation. Administrators should verify the installed RMX hardware/software capacity and the applicable license before using this per-participant rule to determine actual deployment limits, since the platform model and enabled licensing determine the total resource pool available. Poly documentation and RMX support materials are available from [Poly Support] and the product documentation portal at [Poly Documentation].

QUESTION NO: 21

Which function allows the Virtualization Manager (DMA) to automatically generate a conference room for all users?

A. Automatic Conference Room Creator

B. Dynamic conference room wizard

C. Dynamic User Creation wizard

D. Active Directory integration

ANSWER: D

Explanation:

Active Directory integration is correct because the Polycom RealPresence DMA system can use directory synchronization to obtain user information from an organization's Active Directory environment and provision corresponding DMA user resources. When configured to create conference rooms for synchronized users, DMA automatically associates a personal conference room with each applicable directory user rather than requiring an administrator to manually create every room. This supports scalable deployment in environments with many users and maintains alignment between DMA's user records and the organization's central directory. The integration depends on configuring the Active Directory connection, selecting the appropriate users or groups, and enabling the relevant user/conference-room provisioning behavior in DMA. As directory data is synchronized, DMA can use that centrally managed identity information to populate its user and conference-room configuration. This is particularly useful for assigning each user a predictable personal meeting space and dial-in address while minimizing administrative effort. Poly's product-support resources provide access to the RealPresence DMA documentation and software resources used to configure directory integration: [Poly Support]. Poly's enterprise video portfolio information is also available at [HP | Poly Video Conferencing].

QUESTION NO: 22

How many IP addresses does a local Virtualization Manager (DMA) cluster require?

A. Three

B. Two

C. One

D. Four

ANSWER: A

Explanation:

Three is correct because a local Polycom RealPresence DMA cluster consists of two individual DMA server nodes plus a shared virtual IP address. Each physical or virtual DMA node needs its own dedicated management/network IP address so that it can be configured, administered, monitored, and communicate independently on the network. The cluster also uses one virtual IP address, commonly called the VIP, as the highly available client-facing address. Gatekeepers, SIP devices, management systems, and other integrated endpoints use this virtual address to reach the clustered service rather than depending on a particular node. If one cluster node becomes unavailable, the virtual IP can remain available through the surviving node, maintaining service continuity and allowing the cluster to provide redundancy. Therefore, the required total is two node IP addresses plus one shared virtual IP address, which equals three IP addresses. This network design is part of the DMA clustering and high-availability deployment model described in Poly documentation. See the Poly product documentation portal for RealPresence DMA deployment guides and the HP Poly support site for current product documentation and support resources.

QUESTION NO: 23

Which signals are being used with Y/C video?

Red, green and blue

B. Chrominance and luminance

C. Kodachrome and polychrome

D. Yustra spectrum and composite

Answer: B

Explanation:

A. Chrominance and luminance

B. Kodachrome and polychrome

C. Yustra spectrum and composite

ANSWER: A

Explanation:

Chrominance and luminance is correct because Y/C video separates a television image into two signal paths: Y, the luminance component, and C, the chrominance component. Luminance conveys the image's brightness and fine picture detail, while chrominance carries the colour information. This separation is commonly associated with S-Video connections and is intended to preserve picture quality better than composite video, in which brightness and colour information share one signal path and can interfere with each other. In video-conferencing and AV installations, recognizing Y/C as separate luma and chroma is important when identifying source connections, selecting compatible inputs, and troubleshooting colour or image-quality problems. The terminology follows established analogue-video standards: the International Telecommunication

Union's conventional television-system documentation identifies luminance and chrominance signal components, while technical descriptions of S-Video define its separate Y and C conductors. See [ITU-R Recommendation BT.470]() and [S-Video technical overview]().

QUESTION NO: 24

You try to dial an endpoint which you know to be registered to the Visualization Manager (DMA) gatekeeper but the call doesn't connect, Select two options which might cause this issue:

A. The endpoint IP address is not listed in a site and internet dialing is not allowed

B. The gatekeeper should be in Direct Mode

C. The gatekeeper should be in Routed Mode

D. Your endpoint is not registered to the same gatekeeper and your gatekeeper is not neighbored

ANSWER: A D

Explanation:

The endpoint IP address is not listed in a site and internet dialing is not allowed can prevent the DMA system from completing a call to an endpoint by IP address. DMA uses configured network topology, including sites and dial rules, to determine whether a destination is reachable and whether the requested dialing method is permitted. If the destination endpoint's address cannot be associated with an allowed site or external/internet dialing is disabled, the system may reject or be unable to route the call despite the destination being registered. Your endpoint is not registered to the same gatekeeper and your gatekeeper is not neighbored is also a valid cause. H.323 registrations are scoped to a gatekeeper zone. Calls between endpoints registered to separate gatekeepers require an appropriate neighbor relationship and routable zone configuration so that the originating gatekeeper can resolve and forward the called party's alias or address. Without that relationship, registration of the destination endpoint on its own DMA gatekeeper does not make it reachable from the calling endpoint's gatekeeper. These configuration areas are part of DMA's call-routing and network-topology administration. Consult the [Poly Documentation Library]() and the [HP Poly Support portal]() for the DMA administration documentation applicable to the deployed software release.

QUESTION NO: 25

In order to gain an activation key from Polycom Support, what information from the RMX must be provided?

A. Serial number

B. Key code

C. Order number

D. Product code

ANSWER: B

Explanation:

Key code is the required information because it is the RMX's system-specific licensing challenge value used by Polycom Support to generate the matching activation key. The administrator obtains this value from the RMX Product Activation dialog during initial setup or when licensing information must be restored. Polycom's activation process uses the key code to bind the returned activation key to that particular RMX platform and its installed licensing state. Once Support provides the resulting activation key, it is entered into the Product Activation dialog to activate the system's licensed functionality.

This distinction matters operationally: the activation key is not a generic entitlement value that can be exchanged between systems. It is generated in response to the unique key code displayed by the target RMX. Administrators should therefore record the key code exactly as shown and provide it through the applicable Polycom support or licensing workflow when requesting activation assistance. Poly documentation for RMX systems describes product activation and licensing as a system-specific process; see the Polycom RMX Getting Started Guide and the Polycom RMX Administrator Guide.

QUESTION NO: 26

A SIP registrar provides which two functions?

A. endpoint registration

B. alias dialing

C. proxy services

D. redirection services

ANSWER: A B

Explanation:

A SIP registrar provides **endpoint registration** by receiving SIP REGISTER requests from endpoints and associating each endpoint's Address of Record (AOR) with its current reachable contact address. This allows the SIP environment to maintain current location information even when an endpoint's IP address, network attachment, or contact URI changes. In a Polycom SIP deployment, endpoints register with the configured registrar so that calls can be delivered to their registered identities.

Alias dialing is also supported through the registrar's maintained identity and location bindings. A user may be addressed through a configured SIP identity or alias, and the registrar/location database resolves that public identity to the endpoint's current registered contact. This enables users to dial a meaningful SIP URI or assigned alias rather than needing to know the device's current network address. SIP defines the registrar role as accepting registrations and making binding information available through the location service; these bindings are the basis for reaching registered aliases and AORs. See RFC 3261: SIP and Poly's product support and documentation portal.

QUESTION NO: 27

Any device which does not have an IP subnet configured in a network site appears where in the Virtualization Manager (DMA) site topology?

A. Default Site

B. Internet/VPN

C. Nowhere

D. Primary Site

ANSWER: B

Explanation:

Internet/VPN is correct because it is the special, catch-all site in the Polycom RealPresence DMA network-site topology for devices whose addresses cannot be matched to an IP subnet assigned to a defined network site. DMA uses configured site subnets to associate managed devices and endpoints with a location in the topology. When a device IP address is not included in any configured site subnet, DMA places that device in Internet/VPN rather than associating it with a physical or enterprise network location.

This behavior is important for topology-aware call routing and bandwidth management. Internet/VPN represents devices that are effectively outside the administrator-defined private-site topology, including devices reachable through public Internet or VPN connectivity. Administrators can subsequently create or update a network site and add the relevant IP subnet when the device should be treated as part of a specific managed location. Until such a subnet-to-site association exists, Internet/VPN provides the expected topology placement and visibility for the device.

Poly documentation for the RealPresence DMA platform, including its network and site-topology configuration guidance, is available through the Poly Documentation Library and the Poly Support portal.

QUESTION NO: 28

Which two of the following devices can be dynamically managed using the Resource Manager or

Converged Management Application (CMA)?

A. Polycom VSX

B. Polycom VVX

C. Polycom Telepresence m100

D. Polycom HDX

ANSWER: A D

Explanation:

Polycom VSX and **Polycom HDX** are the legacy Polycom video-conferencing endpoint families supported for dynamic management by the Polycom Converged Management Application and its successor, Polycom RealPresence Resource Manager. Dynamic management lets the management platform register and maintain endpoint information, apply specified configuration and directory settings, monitor endpoint status, and support centralized scheduling and provisioning workflows. In this model, compatible video endpoints communicate with the management server so that their records and operational settings can be maintained centrally rather than being handled entirely as static, manually maintained devices.

VSX systems were supported video endpoints in CMA deployments, and HDX systems continued as managed endpoints in both CMA and RealPresence Resource Manager environments. This compatibility is particularly relevant to the older Polycom video infrastructure covered by the CVE Core exam, where endpoint management capabilities are tied to the endpoint platform and supported software version. Poly's product support materials identify RealPresence Resource Manager as the platform for managing Poly video collaboration environments and provide access to its documentation: Poly RealPresence Resource Manager support. Additional legacy product documentation is available through the Poly support portal.

QUESTION NO: 29

A customer would like to enable SIP on their RSS; select the options they will be able to configure:

A. Register to a SIP server

B. Connect to an outbound proxy server

C. Transport type

D. Authentication username / password

ANSWER: A B C D

Explanation:

Polycom RealPresence RSS provides SIP configuration fields that allow the recorder to participate in a SIP-based deployment and to place or receive relevant signaling through the customer's call-control environment. "Register to a SIP server" is configurable so that RSS can register its SIP identity with the designated registrar. "Connect to an outbound proxy server" is also available for environments where SIP traffic must be routed through a proxy rather than sent directly to the registrar or peer. The SIP settings include "Transport type," enabling selection of the signaling transport appropriate to the deployment, such as UDP or TCP where supported by the RSS version and call-control design. Finally, "Authentication username / password" is configurable to provide the credentials used when the SIP server or proxy challenges the RSS during registration or signaling authentication. Together, these parameters establish the registrar/proxy destination, network signaling path, transport behavior, and account authentication required for SIP interoperability. These settings should be aligned with the SIP server configuration and the credentials provisioned for the RSS endpoint. See the Poly documentation portal for RealPresence RSS product documentation at Poly Support: RealPresence RSS and the general Poly support documentation resources at Poly Support.

QUESTION NO: 30

The Converged Management Application (CMA) licensing model affects which two of the following options:

A. Conference reservations

B. Network sites

C. Concurrent calls

D. Registered devices

ANSWER: C D

Explanation:

Polycom CMA licensing is capacity-based in two distinct areas: the number of managed or registered video devices and the number of simultaneous calls the system can process through its gatekeeper functionality. Therefore, **Concurrent calls** and **Registered devices** are the applicable licensing measures. Device licensing establishes the permitted managed-endpoint population, including endpoints registered to CMA for centralized provisioning, monitoring, directory, and call-control services. Concurrent-call licensing establishes the call-processing capacity available at any given time, rather than the total historical number of calls or endpoints in the deployment. This separation lets an organization size CMA for both its endpoint inventory and its expected peak calling load. For example, an environment may manage many registered devices while requiring a lower number of simultaneous calls, or it may require additional call capacity to support busy periods. Polycom's CMA documentation describes licensing and system capacity as part of CMA deployment and administration; the Polycom support portal remains the source for archived CMA product documentation and software materials. See the Poly support portal and the HP Poly product information page.

QUESTION NO: 32 - (SIMULATION)

Which Virtualization Manager (DMA) function refers to the ease with which a solution can be expanded to meet further demand?

A, Virtualization

B. Scalability

C. Redundancy

D. Resiliency

ANSWER: See the explanation for the answer

Explanation:

Correct answer: B. Scalability

Scalability is the ability to expand a DMA/Virtualization Manager solution to accommodate increased demand, such as additional users, calls, sites, or system resources.

Virtualization abstracts computing resources.
Redundancy provides duplicate components to avoid service interruption.
Resiliency is the ability to recover from or continue operating during failures.