Implementing Cisco Collaboration Core Technologies (CLCOR 350-801) Practice Test (Sample)

Study Guide

BY EXAMZIFY

Everything you need from our exam experts!

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Introduction

Preparing for a certification exam can feel overwhelming, but with the right tools, it becomes an opportunity to build confidence, sharpen your skills, and move one step closer to your goals. At Examzify, we believe that effective exam preparation isn't just about memorization, it's about understanding the material, identifying knowledge gaps, and building the test-taking strategies that lead to success.

This guide was designed to help you do exactly that.

Whether you're preparing for a licensing exam, professional certification, or entry-level qualification, this book offers structured practice to reinforce key concepts. You'll find a wide range of multiple-choice questions, each followed by clear explanations to help you understand not just the right answer, but why it's correct.

The content in this guide is based on real-world exam objectives and aligned with the types of questions and topics commonly found on official tests. It's ideal for learners who want to:

- Practice answering questions under realistic conditions,
- Improve accuracy and speed,
- Review explanations to strengthen weak areas, and
- Approach the exam with greater confidence.

We recommend using this book not as a stand-alone study tool, but alongside other resources like flashcards, textbooks, or hands-on training. For best results, we recommend working through each question, reflecting on the explanation provided, and revisiting the topics that challenge you most.

Remember: successful test preparation isn't about getting every question right the first time, it's about learning from your mistakes and improving over time. Stay focused, trust the process, and know that every page you turn brings you closer to success.

Let's begin.

How to Use This Guide

This guide is designed to help you study more effectively and approach your exam with confidence. Whether you're reviewing for the first time or doing a final refresh, here's how to get the most out of your Examzify study guide:

1. Start with a Diagnostic Review

Skim through the questions to get a sense of what you know and what you need to focus on. Your goal is to identify knowledge gaps early.

2. Study in Short, Focused Sessions

Break your study time into manageable blocks (e.g. 30 – 45 minutes). Review a handful of questions, reflect on the explanations.

3. Learn from the Explanations

After answering a question, always read the explanation, even if you got it right. It reinforces key points, corrects misunderstandings, and teaches subtle distinctions between similar answers.

4. Track Your Progress

Use bookmarks or notes (if reading digitally) to mark difficult questions. Revisit these regularly and track improvements over time.

5. Simulate the Real Exam

Once you're comfortable, try taking a full set of questions without pausing. Set a timer and simulate test-day conditions to build confidence and time management skills.

6. Repeat and Review

Don't just study once, repetition builds retention. Re-attempt questions after a few days and revisit explanations to reinforce learning. Pair this guide with other Examzify tools like flashcards, and digital practice tests to strengthen your preparation across formats.

There's no single right way to study, but consistent, thoughtful effort always wins. Use this guide flexibly, adapt the tips above to fit your pace and learning style. You've got this!

Questions

1. **Which statement about SIP normalization is true?**
    - A. To compress audio streams.
    - B. To convert SIP to H.323.
    - C. To apply TLS encryption to SIP signaling.
    - D. To adjust SIP headers/SDP behavior for interoperability.

2. **Which command helps you quickly view current active voice sessions on a gateway?**
    - A. show version
    - B. show call active voice brief
    - C. show ip route
    - D. show running-config

3. **Which component provides SRST functionality at a remote site?**
    - A. Cisco ISR Gateway
    - B. Cisco Unified Border Element
    - C. Cisco Expressway Edge
    - D. Cisco ASA firewall

4. **IM&P stands for Cisco Unified IM and Presence and provides what service?**
    - A. Cloud-based calling (Webex Calling)
    - B. Instant messaging
    - C. Voicemail transcription
    - D. Video conferencing

5. **Which of the following is NOT considered a common source of voice latency?**
    - A. Propagation delay
    - B. Queuing delay
    - C. Processing delays
    - D. Encryption overhead

6. Which protocol secures signaling between Cisco Unified Communications Manager (CUCM) and endpoints?

A. UDP

B. RTP

C. TLS (SIP over TLS)

D. TCP

7. What does MTP do?

A. Media Termination Point—controls media routing.

B. Media Transmission Protocol—relays voice data.

C. Media Timeslot Processor—schedules media streams.

D. Media Termination Point—anchors media for certain signaling/media features.

8. What is the function of NTP in a collaboration deployment?

A. Certificate validation

B. Time synchronization

C. Network latency measurement

D. Firmware updates

9. Which media resource supports DTMF interworking?

A. SIP Normalization Proxy

B. Media Gateway

C. MTP

D. DTMF Relay

10. Which protocol is used to reserve network bandwidth for calls?

A. DNS

B. SMTP

C. NTP

D. RSVP

Answers

1. D
2. B
3. A
4. B
5. D
6. C
7. D
8. A
9. C
10. D

Explanations

1. Which statement about SIP normalization is true?

A. To compress audio streams.

B. To convert SIP to H.323.

C. To apply TLS encryption to SIP signaling.

D. To adjust SIP headers/SDP behavior for interoperability.

SIP normalization focuses on harmonizing the signaling and media descriptions so different SIP implementations can interoperate smoothly. It works by inspecting SIP headers and SDP content and adjusting them to a consistent, supported form. This includes canonicalizing URIs, standardizing header fields (such as Contact, From, To, and Via), and removing or rewriting nonstandard or vendor-specific extensions. It also ensures SDP attributes like codecs, payload types, and media negotiations align across endpoints. The goal is to prevent interoperability problems, call setup failures, or media issues when different devices or networks interact. The other options aren't about normalization: compressing audio streams relates to codecs or transcoding, converting SIP to H.323 is protocol interworking via a gateway, and applying TLS encryption is about security, not message normalization.

2. Which command helps you quickly view current active voice sessions on a gateway?

A. show version

B. show call active voice brief

C. show ip route

D. show running-config

Viewing current active voice sessions on a gateway is done with a command that directly queries the gateway's voice subsystem and returns a concise list of what's happening in real time. The command show call active voice brief gives you a quick snapshot of all ongoing voice calls, including details like the call legs, endpoints, and basic status. This makes it ideal for immediate troubleshooting or monitoring, because you can see at a glance which calls are in progress and whether any are not progressing as expected. The other options don't focus on live calls. Show version reveals software and hardware information, not active sessions. Show ip route displays the routing table, which has nothing to do with current calls. Show running-config shows the current configuration, which again isn't about active calls.

3. Which component provides SRST functionality at a remote site?

A. Cisco ISR Gateway

B. Cisco Unified Border Element

C. Cisco Expressway Edge

D. Cisco ASA firewall

Survivable Remote Site Telephony is provided by the router at the remote site, specifically the Cisco Integrated Services Router (ISR) acting as the local call-processing gateway. When the link to the central call manager (CUCM) is up, phones register and are processed by the central system. If the WAN link goes down, SRST on the ISR takes over so basic call processing can continue locally—allowing internal calls, hold, transfer, and other essential features to function until connectivity is restored. This behavior is why the ISR gateway is the correct component for SRST at a remote site. The other devices serve different roles: a border element (CUBE) handles voice connectivity to external trunks, not remote-site survivability; Expressway Edge focuses on collaboration-edge access and federation; and the ASA firewall provides security—not call processing.

4. IM&P stands for Cisco Unified IM and Presence and provides what service?

A. Cloud-based calling (Webex Calling)

B. Instant messaging

C. Voicemail transcription

D. Video conferencing

Cisco Unified IM and Presence provides real-time text messaging between users, along with presence information that shows who is online and available. This enables instant messaging across Jabber clients and other Cisco apps, delivering the core chat capability teams rely on. Presence data enhances readiness by indicating user status, helping teams decide when to message or call. The other options point to different Cisco services—cloud-based calling is Webex Calling, voicemail transcription comes from Unity-based voicemail solutions, and video conferencing comes from Webex Meetings—so the service particular to IM&P is instant messaging (with presence).

5. Which of the following is NOT considered a common source of voice latency?

A. Propagation delay

B. Queuing delay

C. Processing delays

D. Encryption overhead

Voice latency mainly comes from three kinds of delays: propagation, queuing, and processing. Propagation delay is the time it takes for the voice signal to traverse the physical path, increasing with distance. Queuing delay happens when packets wait in buffers at routers during congestion. Processing delays cover the time devices spend encoding, decoding, packetizing, and making forwarding decisions. Encryption overhead, while it can add some processing time, is not typically treated as a separate common latency source in standard voice-latency models; it's absorbed into the overall processing delay. So encryption overhead isn't considered a separate common source of voice latency.

6. Which protocol secures signaling between Cisco Unified Communications Manager (CUCM) and endpoints?

A. UDP

B. RTP

C. TLS (SIP over TLS)

D. TCP

Securing signaling between CUCM and endpoints means protecting the SIP control messages that set up and manage calls. Using TLS wraps SIP signaling in encryption, providing confidentiality, integrity, and server/client authentication through certificates. This is typically implemented as SIP over TLS (often called SIP over TLS or SIPS), running with certificates issued by a trusted CA and configured on both CUCM and endpoints. This ensures signaling cannot be eavesdropped or tampered with during call setup. UDP and TCP are transport methods, not encryption by themselves—UDP offers no security and unreliability; TCP provides reliable transport but not encryption unless paired with TLS. RTP carries the media stream, not the signaling. So the secure signaling choice is TLS.

7. What does MTP do?

A. Media Termination Point—controls media routing.

B. Media Transmission Protocol—relays voice data.

C. Media Timeslot Processor—schedules media streams.

D. Media Termination Point—anchors media for certain signaling/media features.

MTP stands for Media Termination Point. Its job is to terminate the media path (RTP) and provide a stable point in the media plane where signaling and media features can be applied. This is why it's described as anchoring media for certain signaling/media features—having a fixed media anchor lets features like hold, transfer, and conferencing work reliably, even if endpoints negotiate different codecs or need transcoding. It isn't simply about routing media or relaying voice data, and it isn't about scheduling media streams. Those functions describe other parts of the system, while the MTP specifically terminates and stabilizes the media path to support advanced signaling and media features.

8. What is the function of NTP in a collaboration deployment?

A. Certificate validation

B. Time synchronization

C. Network latency measurement

D. Firmware updates

Time synchronization is what NTP provides. In a collaboration deployment, devices like call managers, endpoints, meeting servers, and recording systems rely on all clocks being aligned. When clocks stay in sync, certificates stay valid within their time windows, logs and call detail records line up across components for accurate troubleshooting, and time-based features and schedules behave predictably. NTP itself doesn't validate certificates—that's done by the PKI/trust setup—and it doesn't measure network latency or perform firmware updates.

9. Which media resource supports DTMF interworking?

A. SIP Normalization Proxy

B. Media Gateway

C. MTP

D. DTMF Relay

DTMF interworking happens when two network paths use different ways of carrying DTMF tones, so the media path itself must translate and relay those digits correctly between sides. The Media Termination Point (MTP) is designed for advanced media processing in Cisco deployments, including interworking DTMF across different transport methods (for example, converting between RFC2833 DTMF in RTP and in-band tones or other DTMF representations). By sitting in the media path, the MTP ensures digits pressed by users are understood on the other end, regardless of which DTMF method each side uses. SIP Normalization Proxy handles signaling normalization, not the actual media DTMF translation. A Media Gateway bridges media between networks and may carry DTMF, but interworking specifically hinges on the media-processing capability of an MTP. DTMF Relay focuses on relaying DTMF signaling, not performing the full media interworking function. So the MTP is the correct choice for DTMF interworking.

10. Which protocol is used to reserve network bandwidth for calls?

A. DNS

B. SMTP

C. NTP

D. RSVP

Guaranteeing quality for real-time traffic relies on signaling that reserves resources along the path. The protocol that does this is RSVP, the Resource Reservation Protocol. RSVP works within the Integrated Services framework by letting a sender describe a data flow and the required QoS, then having routers along the route reserve the needed bandwidth for that flow. Path messages carry the flow’s characteristics, and the routers install reservation state so the required bandwidth and QoS are available for the duration of the session. This makes VoIP and other real-time calls more reliable by preventing congestion from degrading voice quality. In contrast, DNS translates domain names to IP addresses, SMTP handles email transfer, and NTP synchronizes clocks, none of which reserve bandwidth for calls.

Next Steps

Congratulations on reaching the final section of this guide. You've taken a meaningful step toward passing your certification exam and advancing your career.

As you continue preparing, remember that consistent practice, review, and self-reflection are key to success. Make time to revisit difficult topics, simulate exam conditions, and track your progress along the way.

If you need help, have suggestions, or want to share feedback, we'd love to hear from you. Reach out to our team at hello@examzify.com.

Or visit your dedicated course page for more study tools and resources:

https://ciscoclcor350801.examzify.com

We wish you the very best on your exam journey. You've got this!