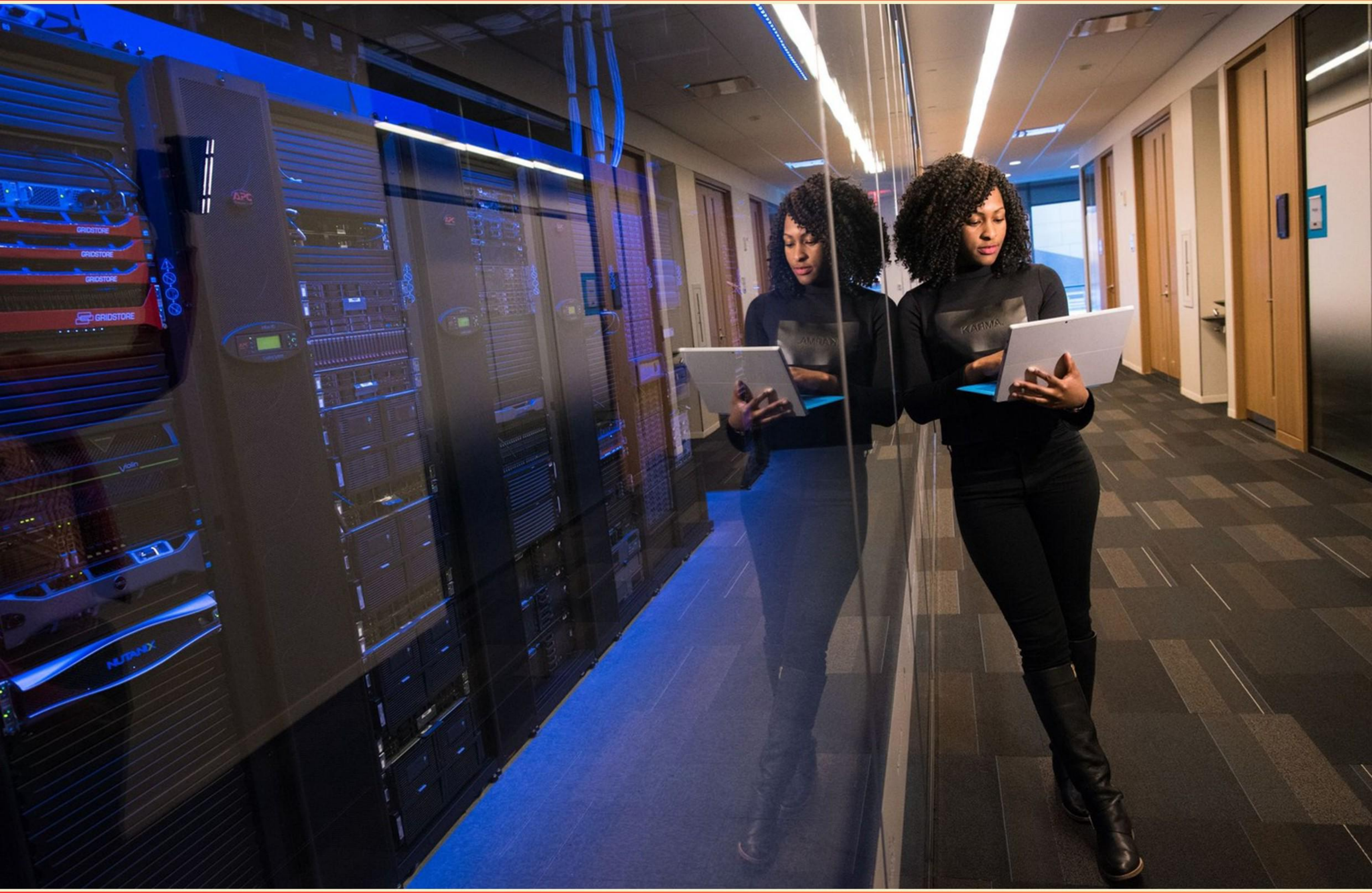
IMPLEMENTING CISCO COLLABORATION CORE TECHNOLOGIES (CLCOR 350-801) PRACTICE TEST (SAMPLE)

STUDY GUIDE



SAMPLE STUDY GUIDE WITH LIMITED QUESTIONS

Notice: Examzify makes every reasonable effort to obtain accurate, complete, and timely information about this product from reliable sources.

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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. **In a CUCM cluster, which services must be functioning together to support cluster operation?**
   - A. Only DNS/NTP.
   - B. Database replication plus core CUCM services and DNS/NTP.
   - C. Core CUCM services only.
   - D. Licensing server connectivity.

2. **In Cisco IOS gateways, DSP is required when?**
   - A. When hardware-based media services are needed.
   - B. When software-based voice processing is sufficient.
   - C. Only for video endpoints.
   - D. When SIP trunking is not available.

3. **Which component replaces on-prem TFTP for Webex devices?**
   - A. Webex Device Server
   - B. SFTP Server
   - C. Cisco Cloud
   - D. Local USB provisioning

4. **What are the typical effects of packet loss on VoIP voice quality?**
   - A. Choppiness, missing syllables, and concealment artifacts
   - B. Jitter only
   - C. Higher bandwidth usage
   - D. No perceptible effect

5. **During codec negotiation, what is the fundamental process?**
   - A. The first codec in the list is always selected
   - B. The network selects the codec independently of offers
   - C. The process chooses a mutually supported codec from the exchanged offers
   - D. The media is always transcoded to a fixed format

6. **Which command shows DSP usage?**
   - A. show voice dsp
   - B. show voice dsp summary
   - C. show dsp resources
   - D. display voice processor status

7. What is the packet loss threshold for voice when sustained losses become noticeable?

A. 0%

B. 0.5%

C. 1%

D. 2%

8. What is TLS handshake in SIP?

A. The negotiation that establishes an encrypted TLS session and validates certificates

B. The process of starting a SIP session

C. The exchange of codecs

D. The exchange of user IDs

9. Which pattern type is used to normalize dialed numbers before routing?

A. Line Group

B. Route List

C. Translation Pattern

D. Route Pattern

10. Hybrid calling refers to integrating on-prem calling with cloud services.

A. Involves only cloud-based calling

B. Uses on-premises hardware exclusively

C. Requires a VPN connection to the cloud

D. Integrates on-prem calling with cloud services

Answers

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

Explanations

1. In a CUCM cluster, which services must be functioning together to support cluster operation?

A. Only DNS/NTP.

B. Database replication plus core CUCM services and DNS/NTP.

C. Core CUCM services only.

D. Licensing server connectivity.

In a CUCM cluster, keeping data consistent across all nodes, ensuring the essential processing services are available, and having proper time and name resolution are all needed for the cluster to work together. Database replication ensures every node stores the same configuration and directory data. Without it, nodes would diverge, leading to registration, routing, and feature inconsistencies as phones and devices rely on shared data. Core CUCM services are the actual processing components that handle call setup, signaling, and related functions. If these services aren't running, the cluster cannot process calls or provide core functionality, even if other pieces are up. DNS and NTP provide critical infrastructure support. DNS allows nodes to resolve names and discover cluster components, while NTP keeps all devices on the same time base. Time mismatches or resolution issues can break authentication, logging, certificates, and synchronization of events across the cluster. Licensing server connectivity, while important for activating features and ensuring compliant operation, is not strictly required for basic cluster operation. The combination of database replication, core services, and DNS/NTP is what makes the cluster function reliably, so that is the best-fit requirement.

2. In Cisco IOS gateways, DSP is required when?

A. When hardware-based media services are needed.

B. When software-based voice processing is sufficient.

C. Only for video endpoints.

D. When SIP trunking is not available.

DSPs in Cisco IOS gateways provide hardware-based media processing, handling real-time tasks such as transcoding between codecs, conferencing, and advanced audio features like echo cancellation. They're required whenever you need hardware-accelerated media services because relying on the main CPU for these tasks can become a throughput or latency bottleneck. If the system can meet quality and performance requirements with software-based voice processing on the CPU, DSPs aren't strictly required. This isn't determined by SIP trunking availability, and while some video scenarios can benefit from DSP processing, the deciding factor is whether you need hardware-assisted media handling rather than the type of endpoint.

3. Which component replaces on-prem TFTP for Webex devices?

A. Webex Device Server

B. SFTP Server

C. Cisco Cloud

D. Local USB provisioning

Cloud-based provisioning replaces on-prem TFTP for Webex devices. Webex devices enroll and receive their configuration and firmware from Cisco Cloud, enabling centralized, scalable deployment without an on-prem TFTP server. This approach supports zero-touch provisioning and centralized management via Webex Control Hub, making it easier to roll out and update devices across multiple locations. Manual or on-device methods like local USB provisioning are not scalable for large deployments, and SFTP or similar on-prem alternatives aren't the provisioning mechanism used for Webex cloud-based devices.

4. What are the typical effects of packet loss on VoIP voice quality?

A. Choppiness, missing syllables, and concealment artifacts

B. Jitter only

C. Higher bandwidth usage

D. No perceptible effect

Packet loss directly creates gaps in the audio stream. When voice packets are dropped, parts of speech disappear, so the call sounds choppy and syllables can be missing. To hide these gaps, receivers apply packet loss concealment, which fills in the gaps with interpolations or repeats, producing concealment artifacts such as muffled or stuttering sounds. Jitter is about timing variation, not actual loss, and higher bandwidth usage isn't a direct result of packet loss in real-time VoIP, while no perceptible effect is unlikely once loss occurs. So the typical effects you hear are choppiness, missing syllables, and concealment artifacts.

5. During codec negotiation, what is the fundamental process?

A. The first codec in the list is always selected

B. The network selects the codec independently of offers

C. The process chooses a mutually supported codec from the exchanged offers

D. The media is always transcoded to a fixed format

Endpoints advertise the codecs they can handle and then agree on a mutual one. The process centers on exchanging offers and answers that list supported codecs, and selecting a codec that both sides support. This mutual agreement lets both ends encode and decode the media with the chosen codec. It isn't simply picking the first option, and the network doesn't choose independently. If there's no overlap, negotiation may fail or require bridging/transcoding, but the normal path is to settle on a mutually supported codec.

6. Which command shows DSP usage?

A. show voice dsp

B. show voice dsp summary

C. show dsp resources

D. display voice processor status

Understanding DSP usage means knowing how hardware DSP resources are allocated to voice processing tasks like transcoding, conferencing, and other DSP-intensive functions, and whether enough resources are available for current and future calls. The command show voice dsp provides a real-time, per-DSP view: it shows each DSP's status (Idle or Busy), what process is using it (for example transcoding or mixing), and how many channels are currently active on that DSP. With this detail, you can quickly see overall capacity, identify bottlenecks, and plan for capacity or troubleshoot issues where calls fail or are degraded because DSPs are fully utilized. This is the best choice because it delivers the actual usage data for each DSP, not just a high-level summary. The alternative that offers a summary only doesn't reveal per-DSP occupancy or which processes are consuming DSP resources. The other commands either don't pertain to DSP usage in this context or provide status information that isn't focused on DSP resource utilization.

7. What is the packet loss threshold for voice when sustained losses become noticeable?

A. 0%

B. 0.5%

C. 1%

D. 2%

Voice traffic is highly sensitive to packet loss because each lost packet can create a gap in the audio stream. While codecs can conceal some occasional losses, when losses are sustained, the gaps become audible and disrupt the rhythm of speech. Around 1% sustained loss is the point where listeners typically begin to notice degradation in voice quality. Below this, many listeners won't perceive a problem; above it, quality worsens noticeably. So the best choice is 1%. In practice, engineers aim to keep loss well under 1% and supplement with QoS, buffering, and reliable paths to protect voice.

8. What is TLS handshake in SIP?

A. The negotiation that establishes an encrypted TLS session and validates certificates

B. The process of starting a SIP session

C. The exchange of codecs

D. The exchange of user IDs

The TLS handshake in SIP is the negotiation that establishes a secure TLS session and validates certificates. Before SIP signaling messages travel over TLS, the client and server negotiate the TLS protocol version and a cipher suite, exchange and verify certificates (and optionally perform mutual authentication with a client certificate), and perform the key exchange to derive the symmetric keys that will encrypt and authenticate the SIP signaling. This process ensures confidentiality and integrity of all SIP messages on the signaling path. This isn't about starting a SIP session, which is about initiating calls; it doesn't cover media codec negotiation, which happens later for the RTP stream; and it doesn't involve exchanging user IDs, which occur within SIP messages themselves rather than during the TLS handshake.

9. Which pattern type is used to normalize dialed numbers before routing?

A. Line Group

B. Route List

C. Translation Pattern

D. Route Pattern

Dialed-number normalization is done with translation patterns. These patterns examine the digits a user dials and rewrite them into a form that the routing logic can understand and use. By matching a dialed string and replacing it with another sequence (for example, stripping a prefix, adding an area code, or converting an extension to a full E.164 number), translation patterns ensure the number fits the format expected by the Route Patterns that actually determine where the call goes. Once the number has been normalized, the system can reliably match it to a Route Pattern and route it over the correct trunk or to the proper destination. Line Group, Route Pattern, and Route List don't perform digit normalization themselves; they are about grouping lines, defining destinations, or organizing routing paths once a properly formatted number is determined.

10. Hybrid calling refers to integrating on-prem calling with cloud services.

A. Involves only cloud-based calling

B. Uses on-premises hardware exclusively

C. Requires a VPN connection to the cloud

D. Integrates on-prem calling with cloud services

Hybrid calling means blending on-premises calling infrastructure with cloud services, so users can place and receive calls across both environments and take advantage of features from the cloud while keeping on-prem systems. The choice that describes integrating on-prem calling with cloud services matches this idea exactly. It captures the essence of hybrid by pointing to a combined setup rather than a cloud-only or an on-prem-only approach. The other statements describe scenarios that are exclusively cloud-based, exclusively on-premises, or insist on a VPN, which is not what defines hybrid calling.

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!