# 3CX Academy Intermediate Certification Practice Test (Sample)

**Study Guide** 



Everything you need from our exam experts!

Copyright © 2025 by Examzify - A Kaluba Technologies Inc. product.

### ALL RIGHTS RESERVED.

No part of this book may be reproduced or transferred in any form or by any means, graphic, electronic, or mechanical, including photocopying, recording, web distribution, taping, or by any information storage retrieval system, without the written permission of the author.

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



# **Questions**



- 1. While using a "Pro Edition" key, do Ring Groups have more than two ringing strategies available?
  - A. Yes
  - B. No
  - C. Only if upgraded
  - D. Yes, but with restrictions
- 2. What format must custom prompts be in for use during a Digital Receptionist announcement?
  - A. MP3, encoded in PCM, 8 kHz, 16 bit, mono.
  - B. WAV, encoded in Stereo, 44 kHz, 16 bit.
  - C. OGG, encoded in PCM, 16 kHz.
  - D. MP3, encoded in stereo, 8 kHz, 16 bit.
- 3. How can latency be monitored in a 3CX installation?
  - A. Only by restarting the system every hour
  - B. Using network analysis tools or third-party solutions
  - C. By measuring the volume of calls
  - D. With traditional logging methods
- 4. How can users improve call quality in a 3CX system?
  - A. By using outdated hardware and ignoring network stability
  - B. By ensuring a stable network connection, using quality hardware, and proper configuration of the codec settings
  - C. By limiting the number of calls made simultaneously
  - D. By using lower quality audio settings to reduce data usage
- 5. What controls the selection of the Digital Receptionist destination in relation to DTMF tones?
  - A. The configuration settings of the phone system
  - **B.** DTMF tones received
  - C. Caller ID information
  - D. A set list of commands

- 6. What method can be employed for setting up remote extensions in 3CX?
  - A. Using a static IP address only
  - B. Through STUN or VPN configuration
  - C. Configuring local devices without any network adjustments
  - D. Only through manual extension entry
- 7. What timezone is call routing based upon if the host uses UTC?
  - A. UTC only
  - B. EST or GMT+2
  - C. PST or GMT-8
  - **D. GMT+0**
- 8. How do users verify their extensions on a remote connection?
  - A. By using a dedicated phone application
  - B. By entering credentials in the 3CX app or web client
  - C. By sending a verification request to IT support
  - D. By using a VPN connection to the local network
- 9. Can Inbound Rules route calls to destinations based on the time of day?
  - A. Yes, but only during business hours
  - B. Yes, always
  - C. No, they cannot route based on time
  - D. Yes, but requires additional configuration
- 10. What should users do if they encounter network configuration errors?
  - A. Ignore them and hope they resolve
  - B. Consult the official documentation or support resources
  - C. Change their internet provider
  - D. Contact all users individually

# **Answers**



- 1. B 2. A 3. B

- 3. B 4. B 5. B 6. B 7. B 8. B 9. B 10. B



# **Explanations**



- 1. While using a "Pro Edition" key, do Ring Groups have more than two ringing strategies available?
  - A. Yes
  - B. No
  - C. Only if upgraded
  - D. Yes, but with restrictions

When utilizing a "Pro Edition" key in the 3CX system, Ring Groups indeed offer more than just the basic ringing strategy options. However, the "Pro Edition" version specifically does not expand the number of ringing strategies available for Ring Groups beyond the standard capabilities that are already present in the system. For this edition, users have specific options available such as ringing all members simultaneously, sequential ringing, or distributing calls in various manners, but the core functionality does not change significantly to include additional strategies. The limitations in this scenario clarify that while users are accessing a feature-rich environment, the ring groups retain a standard level of options as defined by the licensing terms of the "Pro Edition." This aligns with the operational expectations set forth within the context of how the 3CX system is intended to function under this particular licensing structure.

- 2. What format must custom prompts be in for use during a Digital Receptionist announcement?
  - A. MP3, encoded in PCM, 8 kHz, 16 bit, mono.
  - B. WAV, encoded in Stereo, 44 kHz, 16 bit.
  - C. OGG, encoded in PCM, 16 kHz.
  - D. MP3, encoded in stereo, 8 kHz, 16 bit.

Custom prompts for use during a Digital Receptionist announcement must be in a specific audio format to ensure compatibility and optimal performance within the system. The correct format consists of MP3 files that are encoded in PCM (Pulse Code Modulation), at a frequency of 8 kHz, with a 16-bit depth, and in mono sound. This particular configuration is ideal because it balances the quality of audio output with the requirements of a Digital Receptionist. The lower sampling rate and mono format keep file sizes manageable while still producing clear and comprehensible announcements. Such specifications are common in telephony applications, where clarity over telephone lines is paramount. In contrast, the other formats and specifications mentioned do not meet the requirements for Digital Receptionist prompts. WAV files in stereo with a higher sampling rate may provide higher quality sound, but they exceed the necessary requirements, which can lead to larger file sizes and potential compatibility issues. OGG files encoded at a different frequency would not be supported, and the MP3 configurations listed in the other choices either have stereo output or incorrect sampling rates that would not be suitable for the use case.

### 3. How can latency be monitored in a 3CX installation?

- A. Only by restarting the system every hour
- B. Using network analysis tools or third-party solutions
- C. By measuring the volume of calls
- D. With traditional logging methods

Monitoring latency in a 3CX installation is crucial for ensuring call quality and overall system performance. The correct approach involves utilizing network analysis tools or third-party solutions. These tools are designed to provide real-time insights into network performance metrics, including latency, jitter, and packet loss, which are essential in a VoIP environment. Network analysis tools can perform detailed examinations of the data packets being transmitted, allowing administrators to quickly identify and diagnose any latency issues that may arise. This method is proactive and offers in-depth monitoring capabilities, enabling the optimization of the network configuration for better performance. While other methods like restarting the system periodically or measuring call volumes may affect performance temporarily or provide indirect indicators of system state, they do not provide the specific and detailed insights necessary for effective latency monitoring. Traditional logging methods primarily track event occurrence rather than real-time network performance metrics, making them less effective for this particular task. Thus, using dedicated analysis tools remains the most reliable strategy for monitoring latency in a 3CX installation.

## 4. How can users improve call quality in a 3CX system?

- A. By using outdated hardware and ignoring network stability
- B. By ensuring a stable network connection, using quality hardware, and proper configuration of the codec settings
- C. By limiting the number of calls made simultaneously
- D. By using lower quality audio settings to reduce data usage

To enhance call quality in a 3CX system, ensuring a stable network connection, utilizing quality hardware, and properly configuring codec settings are all essential factors. A stable network connection minimizes packet loss and latency, which are crucial elements for maintaining clear and uninterrupted voice communication. If the network is unstable, even the best hardware won't be able to deliver satisfactory call quality. Using high-quality hardware, including reliable SIP phones or headsets, contributes significantly to the overall audio experience. Poor hardware may introduce distortions or complications that can affect call clarity. Codec settings also play a pivotal role in call quality. Proper configuration helps optimize bandwidth usage while maintaining audio fidelity. Selecting an appropriate codec can directly affect how voice data is compressed and transmitted, impacting both call quality and the ability to handle network fluctuations. This comprehensive approach ensures that all aspects of the calling environment are addressed, leading to a superior calling experience for users. Other choices may not support the same level of enhancement in call quality, focusing instead on limiting resources or utilizing inadequate technology.

# 5. What controls the selection of the Digital Receptionist destination in relation to DTMF tones?

- A. The configuration settings of the phone system
- **B. DTMF tones received**
- C. Caller ID information
- D. A set list of commands

The selection of the Digital Receptionist destination is primarily controlled by the DTMF tones received during a call. DTMF, or Dual-Tone Multi-Frequency, refers to the signals generated when a caller presses buttons on their telephone keypad. Each button corresponds to a unique DTMF tone, and these tones serve as input commands for the phone system. When a caller interacts with a Digital Receptionist, they can navigate through options and make selections by pressing specific keys. Each key corresponds to a defined action or destination within the system's configuration. For instance, pressing '1' may direct the call to sales, while '2' might connect the caller to support. The Digital Receptionist interprets these DTMF tones to determine the appropriate destination based on what the caller has selected. In contrast, the other options do not specifically dictate the real-time input mechanisms involved in controlling the call flow. Configuration settings of the phone system outline how the Digital Receptionist operates but do not actively influence the current user interaction. Caller ID information provides details about the incoming caller but does not guide the selection process during an interactive session. A set list of commands outlines expected inputs but relies on the DTMF tones themselves to determine which command the caller intends to

# 6. What method can be employed for setting up remote extensions in 3CX?

- A. Using a static IP address only
- **B.** Through STUN or VPN configuration
- C. Configuring local devices without any network adjustments
- D. Only through manual extension entry

Using STUN (Session Traversal Utilities for NAT) or a VPN (Virtual Private Network) configuration is the most effective method for setting up remote extensions in 3CX. STUN allows remote extensions to communicate through NAT (Network Address Translation) devices by discovering their public IP address and the type of NAT they are behind. This enables proper routing of VoIP traffic and seamlessly connects remote users with the 3CX server. On the other hand, VPNs create a secure, encrypted connection between a remote device and the office network. This approach not only addresses potential security concerns but also ensures that remote extensions can access internal resources as if they were connected locally. Using STUN or VPN provides flexibility and security, making it easier to deploy and manage remote extensions. Other methods, such as relying solely on a static IP address, do not provide the same level of flexibility in dynamic environments. Configuring local devices without any network adjustments ignores the complexities associated with NAT and network configurations for VoIP. Finally, only manual extension entry lacks the automation and ease of use that STUN or VPN methods provide for remote connectivity. Thus, utilizing STUN or VPN is the recommended approach for setting up remote extensions effectively in 3CX.

# 7. What timezone is call routing based upon if the host uses UTC?

- A. UTC only
- B. EST or GMT+2
- C. PST or GMT-8
- D. GMT+0

In the context of call routing and timezones, if the host is using UTC (Coordinated Universal Time), it is important to note that UTC itself serves as a reference point for various local timezones around the world. The correct reasoning for the appropriate answer hinges on understanding that UTC does not change and is a fixed reference. Therefore, any timezone adjustments will either add or subtract hours from UTC. The answer indicates specific timezones that may be considered in relation to UTC. Using options like EST (Eastern Standard Time), which typically corresponds to UTC-5 during standard time or UTC-4 during daylight savings time, or GMT+2 (which is two hours ahead of UTC), indicates that call routing might consider these variations depending on the location of the users. This means that for accurate call routing, understanding local time adjustments to the UTC is crucial. Other options reflect specific offsets from UTC that would not be applicable as primary bases for call routing decisions, as they could be too narrow or irrelevant depending on the scenario. Therefore, considering a broader spectrum like EST or GMT+2 makes sense, as it encompasses more global user time specifications in call routing scenarios.

# 8. How do users verify their extensions on a remote connection?

- A. By using a dedicated phone application
- B. By entering credentials in the 3CX app or web client
- C. By sending a verification request to IT support
- D. By using a VPN connection to the local network

Users verify their extensions on a remote connection by entering their credentials in the 3CX app or web client. This process is straightforward and allows users to authenticate themselves securely from anywhere, as long as they have internet access. When users input their credentials—typically their extension number and password—they are granted access to their account and can utilize the full range of features offered by the 3CX system, making it a seamless experience. Using a dedicated phone application might be useful for accessing the 3CX system, but it does not inherently provide the verification process that is facilitated through the app or web client. The option of sending a verification request to IT support may involve additional delays and is not a direct method for immediate verification. A VPN connection to the local network is not necessary for users to verify their extensions, as the 3CX app is designed to operate over the internet without requiring a VPN for authentication purposes. Therefore, entering credentials in the 3CX app or web client is the most effective method for verification on a remote connection.

- 9. Can Inbound Rules route calls to destinations based on the time of day?
  - A. Yes, but only during business hours
  - B. Yes, always
  - C. No, they cannot route based on time
  - D. Yes, but requires additional configuration

Inbound Rules in 3CX can indeed route calls based on the time of day, making the correct choice affirmatively solid. This feature allows businesses to define specific conditions under which inbound calls are directed to various destinations, facilitating better call management and customer service. By utilizing this capability, organizations can optimize their operations by setting different call handling strategies for different times—for instance, routing calls to different departments or voicemail outside of regular business hours. This flexibility is essential for businesses that want to ensure they are responsive while also managing call traffic efficiently. It is important to note that while some routing features may require additional settings or configurations, the core ability to route based on time is inherent to the functionality of the inbound rules set within 3CX.

- 10. What should users do if they encounter network configuration errors?
  - A. Ignore them and hope they resolve
  - B. Consult the official documentation or support resources
  - C. Change their internet provider
  - D. Contact all users individually

When users encounter network configuration errors, consulting the official documentation or support resources is the most effective course of action. This option provides access to comprehensive guides, troubleshooting steps, and best practices that are specifically designed to address a wide range of issues. Official resources often include insights from experienced professionals or community forums, which can help users understand the root cause of the error and how to resolve it efficiently. Additionally, official documentation is usually updated with the latest information and fixes that reflect current knowledge and common challenges faced by users. This structured approach avoids unnecessary delays and increases the likelihood of resolving the issue quickly.