MOC 50571 A Partner Lync Support Training - Pre-Read Material

Course Summary

Description

This course provides students with the knowledge and skills in Voice technology. This course focuses on an introduction to voice technologies, dial plans, SIP architecture, routing, SIP message syntax, SDP payload used in voice, SIP trunking, RTP and RTCP, DTMF handling, voice compression, queuing, and quality of service.

Objectives
At the end of this course, students will be able to:

- Dial Plans
- SIP Architecture
- Trunking
- Routing
- SDP Payloads
- RTP and RTCP
- Voice compression
- Queuing
- Voice Quality of Service

Topics

- VoIP Technologies from Microsoft OCS/UM and Beyond
- Dial Plans
- SIP Architecture
- Call Routing
- SIP Message Headers used in VOIP
- SDP payload used in VOIP
- SIP Trunking
- RTP/RTCP
- DTMF Handling
- Voice Compression
- Queuing
- QoS

Audience

This course is intended for Premier Lync Server partners and is perquisite reading material for partners wishing to attend the Partner Lync Support Training - Tier One or Tier Two courses.

Prerequisites

There are no prerequisites for this course.

Duration

One day

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Course Outline

I. VoIP Technologies from Microsoft OCS/UM and Beyond

This module provides an essential understanding of the SIP control protocol and RTP media negotiation/delivery. It intends to provide Enterprise Voice support professionals the knowledge required to deploy, maintain, and troubleshoot problems that are common in voice applications that – share the IP network with other applications, are more time latent and frequently compete with the bandwidth, and offer timely delivery of voice packets across a packet switched network.

A. VoIP Introduction
B. VoIP Protocols

After completing this module, students will be able to:

- Describe SIP and RTP with respect to how it is used in Voice applications
- Define common components in VOIP deployments extending beyond the IP network.
- Usewire tracing and stack tracing to troubleshoot VOIP issues.
- Identify issues and recommend solutions to common problems in VOIP deployment scenarios.

II. Dial Plans

This module introduces the concept of a "dial plan", and explains how telephony systems handle digit manipulation for call routing.

A. Dial Plan Routing

After completing this module, students will be able to:

- Describe the hierarchy of calls routed from caller to called party.
- Understand and troubleshoot problems of normalization and routing.
- Use a normalization rule to modify a dialed number.

III. SIP Architecture

This module explains the function of SIP in relation to VOIP.

A. The Role of the UAC and UAS

After completing this module, students will be able to:

- Understand the purpose of a UA and UAS.
- Define how SIP controls media flow.
- Analyze and troubleshoot a VOIP call setup/teardown using SIP.

IV. Call Routing

This module introduces the concept of routing with the SIP signaling. Since the path of the signaling can travel over any IP network and is likely to be processed along the path (UAS), tracking the path of the signal is critical so that the UAC and UAS do not miss critical call processing data.

A. SIP Call Routing

After completing this module, students will be able to:

- Track the way SIP call processing data is routed from caller to called party and between UASs and SIP proxies.
- Understand and troubleshoot problems with SIP signaling failures.

V. SIP Message Headers used in VOIP

This module will explain the SIP message headers that are used in VoIP

A. SDP

After completing this module, students will be familiar with:

- All SIP message headers

VI. SDP payload used in VOIP

This module explains SDP payload and how it is used in VOIP.

A. SDP Payload used in VOIP

After completing this module, students will become familiar with:

- Explain SDP Payloads used in VOIP

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Course Outline (cont'd)

VII. SIP Trunking
This module discusses the next logical evolution into Total VoIP across the telecom industry. There are significant barriers into this next generation of voice. Microsoft and other VoIP solution providers have a large stake in the evolutionary state of the services and software, as a solution to the otherwise non-standardized solutions currently and historically used.

A. Lines vs. Trunks

After completing this module, students will be able to:

- Understand the challenges currently before the industry and where VoIP plays a role.
- Position Microsoft Voice software solution in the total solution for Total VoIP.

VIII.RTP/RTCP
This module discusses how voice is packetized and transported across the IP network. Breaking a single stream of audio up into tiny packets has its challenges in ensuring a reasonably accurate reproduction.

A. Real-Time Transport Protocol

After completing this module, students will be able to:

- Understand how VoIP streams Audio on the network.
- Understand how VoIP audio is achieved through encoding, packetization, packet re-assembly, decoding.

IX. DTMF Handling
This module discusses the concepts of DTMF (dual tone / multiple frequencies) in VoIP applications, including RFC 2833 / 4733 event based RTP. DTMF was originally designed to provide a simple audible signal over telephony equipment that would be less confusing and easy to use.

A. DTMF In-Band
B. DTMF Out-of-Band
C. DTMF using the SIP INFO Method

After completing this module, students will be able to:

- Understand how DTMF functions through VoIP.
- Distinguish between In-band and Out-of-band DTMF.

X. Voice Compression
This module discusses the concepts of Voice/Audio Compression and how it impacts the quality of the VoIP experience.

A. MOS
B. Codecs

After completing this module, students will be able to:

- Understand how compression impacts the quality of reproduction.
- Explain how compression can both compensate for transmission delay as well as improve transmission times.
- Present an argument *for* or *against* compression, specifically with respect to VOICE communications.

XI. Queuing
This module discusses the concepts of packet queuing across network boundaries. CoS (Class of Service) is a requirement for a successful QoS (Quality of Service) contract.

A. Slow Links Cause Queues

After completing this module, students will be able to:

- Understand packet queuing strategies.
- Define a traffic contract based on data classification.
- Describe how a given strategy can be used to solve the QoS problem.

XII. QoS
This module discusses the implementation of a QoS contract / Protocols.

A. Sources of delay
B. QoS and differentiated services