

## Voice over IP: Hands-On - 4 Days

### *Course 461 Overview*

- You Will Learn How To**
- Obtain the maximum benefit from Voice over IP (VoIP) and voice/video/data integration
  - Identify the costs, benefits and challenges of VoIP in multivendor networks
  - Interpret H.323, SIP and other industry standards for VoIP
  - Configure Session Initiation Protocol (SIP) IP phones, software phones and SIP IP PBX
  - Engineer high Quality-of-Service VoIP networks
  - Design networks to carry mixed voice, video and data traffic

**Course Benefits** Voice over IP reduces your telephony costs and provides unique opportunities for integrating voice, video and data. In this course, you identify challenges and benefits of VoIP integration. You gain the knowledge to use a data packet network to provide wide area voice communications. You learn how to ensure Quality of Service for VoIP traffic, take advantage of VoIP for new and innovative applications, and configure and operate SIP and H.323 for VoIP.

**Who Should Attend** Anyone interested in migrating voice and video conferencing traffic to new or existing data networks. A basic understanding of either TCP/IP internetworking or telephony concepts is assumed.

**Hands-On Training** Exercises provide you with practical experience in key aspects of Voice over IP technology and include:

- Configuring IP softphones with voice, video and messaging
- Conferencing with H.323 and SIP protocols
- Measuring the effect of codecs on network bandwidth consumption
- Configuring IP desk phones
- Calculating the degradation in voice quality due to packet loss, delay and jitter
- Configuring analogue phones with SIP analogue telephone adapters (gateways) and SIP IP PBX
- Engineering a network to carry VoIP using Erlang charts

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### Course 461 Outline

#### Introduction to VoIP

##### Traditional voice and data networks

- The Public Switched Telephone Network
- Packet-switched data networks
- OSI and IETF reference models

##### Realising the benefits of VoIP

- Using data networks to transmit voice and video traffic
- Taking advantage of new communications features available with SIP and H.323

#### Applying VoIP Fundamentals

##### Defining multimedia conferences

- Voice, video, data and mixed conversions
- Key VoIP hardware and software components
- Gateways and gatekeepers
- Signalling and data protocols

##### Deploying H.323 or SIP within your organisation

- SIP vs. H.323
- Configuring and testing SIP IP PBX and IP phones
- Setting up voice and video conferencing
- Creating ring groups for a call center
- Tuning network quality-of-service for user satisfaction

##### Analysing H.323 and SIP calls

- Capturing SIP and H.323 signalling exchanges
- Implementing and analysing silence suppression
- In-depth analysis of SIP signalling messages
- Registering with a gatekeeper or proxy
- RTCP messages

#### Leverage SIP for Seamless

##### Communication

##### Building the IP infrastructure

- Packetising voice for carriage through the IP network with RTP
- Employing reliable signalling with TCP
- Maintaining real-time voice performance with RTP

##### Analysing VoIP signalling

- Distinguishing between SIP, H.323, H.225 and H.245
- Applying SDP to SIP packets
- Call setup and teardown

##### Addressing within VoIP networks

- Mapping phone numbers to IP addresses

- Gatekeepers, proxies, locations and call servers

##### Measuring voice quality

- Mean opinion scores (MOS)
- Detecting flaws in transmitted voice
- Grade of service and capacity planning

#### Voice Quality Engineering

##### Classifying factors that affect voice quality

- Encoding voice effectively
- Comparing codecs
- Assessing the effects of delay and loss on quality

##### Selecting the best codec

- G.711, G.722, G.723, G.726, G.729
- Testing for differences in performance
- Adaptive techniques
- Voice companding:  $\mu$ -law, A-law

##### Open and multivendor systems

- Anatomy of IP phones
- Open source Asterisk IP PBX
- Gateways to the PSTN legacy phone network

##### Capacity planning for voice over packet

- Determining bandwidth needs
- Comparing LAN and WAN performance
- Assessing the impact on data networks

#### Designing Networks to Carry VoIP

##### Calculating bandwidth service requirements

- Traffic engineering
- Measuring sporadicity of signals
- Sizing service loading for multisite operation
- Designing VoIP service: case study

##### Minimising delay and delay variation

- Calculating queuing delays
- Sizing link capacity needs and required trunk speeds
- Calculating expected routing delays
- Controlling admission in commercial services
- Employing Random Early Discard (RED)
- Designing for high availability

#### Delivering Quality on Real Networks

##### Exploiting the Internet with VoIP services

- Making calls with IP phones over the Internet
- Providing international voice services
- Linking a call centre to the Internet
- Adding VoIP to an e-commerce website

##### Employing quality services on intranets

- Resource Reservation Protocol (RSVP)
- Configuring IP precedence
- Multiprotocol Label Switching (MPLS)
- Observing the effects of data traffic and fair queuing on voice services
- Mixing voice and data traffic effectively