Global Knowledge™

Voice over IP Foundations

Duration: 5 Days   Course Code: GK3277

Overview:

The aim of this course is for delegates to gain essential data networking and Voice over IP (VoIP) knowledge in a single, week-long class. In this course, you will learn how VoIP works, why VoIP works, and how to use VoIP

Target Audience:

This class is for people who need to understand VoIP technology. IT managers, technical sales/marketing personnel, consultants, network designers and engineers, product design engineers developing integrated-services products,telecom technicians and managers integrating PBX services within data networks, and systems administrators who will manage a converged network would benefit from this course.

Objectives:

- At the end of this course delegates will be able to;
- Core concepts of how Internet Protocol (IP) carries a VoIP packet
- Configure DHCP and DNS to support IP telephony
- Real-Time Transport Protocol (RTP)
- Session Initiation Protocol (SIP) - Call set up, Instant Messaging, Presence
- Session Description Protocol (SDP)
- The H.323 protocol suite, including H.225, RAS, and H.245
- The role of endpoints, gatekeepers, gateways, and MCU in an H.323 network
- SIP proxy, Session Border Controller (SBC), and SIP softswitch
- Media Gateway Control Protocol (MGCP) analysis
- MGCP architecture
- A technical comparison of H.323, SIP, and MGCP
- How to implement QoS to ensure the highest voice quality over your IP networks
- The impact of jitter, latency, and packet loss on VoIP networks
- How to use Wireshark to decode and troubleshoot RTP, SIP, MGCP, and H.323 call flows
- Configure the trixbox Softswitch and SIP proxy
- Configure SIP gateways and softphones
- Security issues to consider when setting up VoIP

Prerequisites:

The skills and knowledge required for a delegate to sit this course are as follows

- TCP/IP Networking
- Telecommunications Fundamentals

Testing and Certification

- There are no exams associated with this course.

Content:

Packetizing Voice

- Key architectural VoIP components
- End-to-end voice transmission
- Packetizing voice (encapsulation)
- Transmission time allocation
- QoS and capacity considerations
- Sources of delay
- Coder processing delay (Think Time)
- Algorithmic delay (Look Ahead)
- Packetization delay
- Serialization delay
- Queuing delay
- Jitter buffer function
- VoIP QoS requirements: Packet Loss, Latency, Jitter

VoIP in the LAN

- MAC address
- IP address and ARP
- Ethernet switching
- Logical and physical segmentation
- VLAN - 802.1Q/P
- 802.3af - Power over Ethernet (POE)

IP Networking

- IP addressing
- Static routing
- OSPF
- EIGRP

TCP/IP Review

- Transmission Control Protocol (TCP)
- VoIP protocols that use TCP
- User Datagram Protocol (UDP)
- VoIP protocols that use UDP

SIP-Related IP Services

- DNS
- How SIP uses DNS
- DHCP
- How IP telephony uses DHCP

Voice Compression

- G.711 u-law and a-law
- G.729
- G.723.1

Real-Time Transport Protocol (RTP)

- Dealing with packet Loss, latency, jitter
- How various protocols define the RTP session
- Session Description Protocol
- H.245 terminal capabilities
- The RTP profile
- The RTP payload type field
- RTP telephony events (RFC 2833)
- How RTP removes jitter
- How RTP handles packet loss
- How RTP identifies the talking party
- How RTP handles silence suppression
- How RTP is used to mix voice (conference calls)
- The RTP header
- RTP Control Protocol (RTCP)
- SDES
- Sender/receiver reports
- Bye reports

SIP Architecture

- SIP architecture
- Proxy: stateful, stateless, call stateful, Session Border Controller
- SIP methods: INVITE, ACK, BYE, CANCEL, REGISTER, INFO, PRACK, etc.
- SIP response codes: 1xx, 2xx, 3xx, 4xx, 5xx, 6xx
- SIP headers (To:, From:, Call-ID:, Allows: Required, Via)
- Session Description Protocol (SDP)
- SIP Addressing, Session Control, and Call Setup

SIP Uniform Resource Indicators (URIs)

- Understand the format of SIP URIs and how URIs interoperate with
- PSTN dialing plans, e-mail systems, and web pages
- Generic URI information (RFC 2396)
- Direct or Proxy
- PSTN number (RFC 2808)
- Instant messaging
- Presence
- In registrations
- Content-Type Header

SIP Call Flow Examples

- Review how SIP calls are set up for applications like PSTN, instant messaging,
- VoIP, and more in this technical, in-depth analysis of the protocol.
- Call attempt - unsuccessful
- Presence subscription
- Registration
- Presence notification
- Instant Message Exchange

SIP NAT Traversal

- One-Way Voice Results
- Full Cone NAT
- IP Address Restricted NAT
- Port Restricted NAT
- Symmetric NAT
- Simple Traversal of UDP through NATs
- Traversal Using Relay NAT
- NAT with Embedded SIP Proxy
- Public VoIP Example

Media Gateway Control Protocol (MGCP)

- Architecture
- Verbs: CRCX, MDCX
- Responses
- Packages (DTMF, Line, Trunk, Generic, etc.)
- Parameter lines\
- Sample call flow protocol analysis

H.323

- ASN.1 primer
- H.323 architecture
- Gatekeeper
- Gateway
- MCU
- Terminal
- H.323 versions
- H.323 gatekeeper-controlled call flow example

Queuing

- Priority queuing
- Weighted fair queuing
- Weighted precedence
- Traffic policing and traffic shaping
- Low latency queuing
- The effects of data traffic and fair queuing on VoIP
- Mixing voice and data traffic effectively
- Determining bandwidth needs for voice traffic
- Assessing the impact of voice on data networks
- Low speed links
- High speed links

QoS-Related Networking Protocols

- Differentiated Services (DiffServ)
- Call Admission Control

- Call setup - successful
- Cancel
- Vacant number
- 100rel
- www authenticate

SIP Syntax

- Request Message
- Response Message
- The Start Line
- Via Header
- SIP Dialog
- From Header
- To Header
- Call-ID Header
- Dialog State
- CSeq Header
- Max-Forwards Header
- Proxy-Authenticate Header
- Contact Header
- Expires Header
- User-Agent Header
- Content-Length Header
- Allow Header
- Supported Header

Session Description Protocol

- v= Header
- o= Header
- s= Header
- C= Header
- t= Header
- m= Header
- a= Header
- Offer/Answer Model
- SDP Offer/Answer Rules
- UPDATE Method
- RTP SEND and RECV Defined
- Media Direction and RTCP
- How RTCP Works
- Placing a Call on HOLD

Further Information:

For More information, or to book your course, please call us on Head Office 01189 123456 / Northern Office 0113 242 5931

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