Voice over IP Fundamentals

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.

Follow-on-Courses:

- Cipt1v6
- Citp2v6
- GK3285

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

Configure a DNS zone, NAPTR, SRV, and A records as needed to support VoIP services.

Lab 6: Implement DHCP

Configure DHCP services on your LAN to support VoIP gateways and phones.

Lab 7: Calling Without a SIP Proxy

Call without a SIP proxy.

Lab 8: UA Registration

Register a UA with a proxy.

Lab 9: VoIP Island Configuration

Configure VoIP islands.

Lab 10: SIP Ethernet Phone Configuration

Configure a SIP Ethernet phone.

Lab 11: Networking SIP Proxies

Network SIP Proxies.

Lab 12: Dial Plan Implementation

Implement the Dial Plan.

Lab 13: SIP Softphone Configuration

Configure a SIP softphone.

Lab 14: Capturing and Analyzing RTP using Wireshark

Use Wireshark and Port Spanning to capture and analyze RTP.

Lab 15: Codec MOS Testing

Lab 19: Codec Negotiation (Offer/Answer)

Configure automatic codec negotiation and observe how SIP negotiates codecs (OFFER/ANSWER).

Lab 20: DTMF RFC 2833 and SIP INFO

Configure two different techniques that support accurate and reliable DTMF transmission.

Lab 21: Using Wireshark for Capture and Analysis

Use Wireshark to capture and analyze RTCP (QoS) reports.

Lab 22: SIP REGISTER Authentication

Configure a SIP phone to authenticate prior to joining a SIP network.

Lab 23: SIP INVITE Authentication

Configure a SIP proxy to confirm the calling party prior to processing the call.

Lab 24: SIP Call Flow Analysis

Using Wireshark, analyze typical call processing such as a normal call, busy call, abandoned call, and call transfer. Learn how to use Wireshark to troubleshoot problems with call processing.

Lab 25: Wi-Fi Radio Configuration

Configure a Wi-Fi radio.

Lab 26: Wi-Fi SIP Phone Configuration

Configure a Wi-Fi SIP phone.

Lab 27: SIP Trunking

Configure SIP trunking between two SIP

- 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
- 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

Configure various codecs and make test calls to compare voice quality (G.711, G.729, and G.723.1).

Lab 16: Increasing Packet Intervals

Reduce bandwidth consumption by 50% or more by increasing packet intervals and witness the QoS tradeoff.

Lab 17: Codec Bandwidth Testing

Test the amount of bandwidth actually consumed by different types of voice compression.

Lab 18: Silence Suppression

Silence suppression and witness any QoS tradeoff. Activate and test silence suppression.

PBXs, and learn the process of connecting to the PSTN using ITSP rather than buying your own PSTN gateways and connecting using conventional TDM or analog methods.

Lab 28: trixbox Meet-Me Conferencing

Lab 29: trixbox Voice Mail

Lab 30: QoS Performance Testing

Install SolarWinds Engineer's Edition and use WAN Killer and SolarWinds SNMP tools to test QoS performance.

Lab 31: VoIP Gateway DiffServ Configuration

Configure DiffServ on your VoIP gateway.

Lab 32: Queuing Strategies and QoS Configuration

Configure various queuing strategies, apply service policies on your router, and witness the results. Perform file transfer and voice services on the same network and witness the results of proper and poor QoS configuration.

- 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

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

Labs

Lab 1: Network Hardware Installation

Install the network hardware.

Lab 2: Cisco IOS Command Line Interface Configuration

Configure Cisco IOS Command Line Interface via Telnet and console port access.

Lab 3: Configure VLAN

Configure VLAN for secure voice and data separation.

Lab 4: IP Network Configuration

Configure an IP network using static routing.

Lab 5: Implement DNS

Further Information:

For More information, or to book your course, please call us on 0800/84.009

info@globalknowledge.be

www.globalknowledge.be