Course Code – VOI12002

Voice over IP with SIP

This course focuses on theoretical and practical principles of Real-Time Media over Internet Protocol coupled with SIP (Session Initiation Protocol). Although the course is intended to be generic, examples of VoIP networks are highlighted utilising Cisco products and other vendor software and hardware.

Prerequisites
A basic understanding of Data Networking and Internet Protocol

Aim:
To provide delegates with a fundamental understanding of how Real-Time Media is delivered using packet switched IP Networks and SIP (Session Initiation Protocol).

Objectives:
By the end of the course you will be able to:

- Understand how VoIP and it's associated protocols fit in with the existing networking protocol models.
- Be able to explain the reasons for the use of VoIP.
- Describe the potential benefits of VoIP
- Understand voice quality issues associated with VoIP
- State the additional protocols that make VoIP possible
- Understand the differences between H.323 and SIP
- Understand Quality of Service (QoS) and what it means
- Appreciate how IP Multicasting plays a part in Real-time media delivery
- Understand the differences between Circuit Switches and Packet Switched Voice.
- Configure a simple VoIP application on a windows PC and make Phone Calls
- Set up a SIP Server for registration of Client Devices
- Configure a SIP Converter or SIP Phone
- Configure a SIP Server / IP PBX at multiple sites and make calls across a WAN.
- Configure and test basic QoS parameters and Test Voice Quality

Course Profile

Introduction to VoIP

- How it Works
- VoIP – Encapsulation
- VoIP – Associated Internet Protocols
- VoIP – Service Providers
- Networks
- Why VoIP?
- What does VoIP Offer?
- The Business Case for VoIP
- Internet Telephony Product Classes
- Today's Voice and Data Networks

- Integration
- VoIP Regulatory Bodies

Voice Encoding Schemes

- Waveform Encoding
- Digital Recording and Playback
- PAM – Pulse Amplitude Modulation
- Quantization
- Clipping
- Vocoders
- MOS – Mean Opinion Scores
- Voice Quality Measurement
- Voice Quality Issues
    - Latency
    - Jitter
    - Packet Loss
- Echo Problems in VoIP
    - Echo Suppression
    - Echo Cancellation

Protocols – Network and Transport Layers

- OSI 7 Layer Model
- Internet Protocol
- IP Addressing
- TCP – Transmission Control Protocol
- Ports and Sockets
- Windowing
- UDP – User Datagram Protocol

Protocols – Data Link and Physical Layers

- Data Link and Physical Layer in the LAN
- Routed Networks
- Switched Networks
- Networks with VLANs
    - What is a VLAN?
    - VLAN Membership
    - VLAN Tagging
    - IEEE 802.1p/q Ethernet Frame Format
- Data and Computer Networking
- Frame-Relay
- xDSL
- Broadband Access
- SONET and SDH
- PPP – Point to Point Protocol
- MTU Size
- Fragmentation and Reassembly

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VoIP Support Services

- Configuring DHCP to support VoIP
- Cisco DHCP Configuration Example
  - Option 150 and 66
- DHCP Relay Agents
- Cisco Router Relay Agent Configuration
- NAT – Network Address Translation
- NTP – Network Time Protocol
- DNS Support for VoIP
- STUN – Simple Traversal of UDP through NAT

Real Time Protocols

- RTP – Real Time Transport Protocol
- Encapsulation Overhead
- Header Compression
- RTP Translators
- Audio Conferencing (Mixers)
- RTCP – Real Time Transport Control Protocol
- RTCP Bandwidth Control
- RTCP Reports

VoIP Security

- Security Implications
- Eavesdropping
- Spoofing
- Interception
- Countermeasures
- Physical and Logical Security
- VoIP and Firewalls

VoIP Bandwidth

- Traditional Call Activity
- Trunk Activity
- Erlangs
- Blocking
- Voice Quality Delivery Options
- Configuring End-to-End QoS

H.323 Networks

- Brief Description for comparison with SIP

SIP – Session Initiation Protocol

- SIP Protocol Stack
- SIP Topology
- SIP Operation
- SIP URLs
- SIP Signalling Messages

Network Systems Training (UK) Ltd Tel: 0845 519 7752 Overseas: +44 1670 712585
Email: enquiries@nstuk.com Web: http://www.nstuk.com
Registered Address: 83 Beech Avenue, Cramlington, Northumberland NE23 6XS

- Typical SIP Transaction
- SIP Message Format
- SIP Cseq
- SDP – Session Description Protocol
- SIP Servers
    - SIP Registrar
    - SIP Proxy
    - SIP Redirect

SIP Trunking

- What is VoIP Trunking
- Traditional vs New
- Carrier Class SIP Trunking
- SIP and PSTN Internetworking
- SIP and ISUP
- SIP Telephony and ISUP Tunnelling
- Enhanced Telephony Services
- SIP Example Call Traces

Cisco VoIP Example (Brief)

- Cisco IP Telephony Components
- Cisco CallManager
- SRST – Survivable Remote Site Telephony
- Cisco Unified CallManager

QoS – Quality of Service

- Quality and Delivery Options
- Congestion Control
- Quality of Service Models
    - Best Effort
    - Integrated Services
        - RSVP
    - Differentiated Services
- Congestion Management
    - FIFO
    - Priority Queuing
    - Custom Queuing
    - Low Latency Queuing
    - Weighted Fair Queuing
- Delivering QoS in the core network
    - MPLS Overview (Brief)
- CAC – Call Admission Control
- WRED – Weighted Random Early Detection

VoIP Phone / Adapter and IP PBX Configuration