Hands-On

Voice Over IP - VoIP

Data and Voice

Course Description

This Hands-On course provides in-depth coverage on Voice over IP (VoIP), which integrates voice and data transmission, is quickly becoming an important factor in network communications.

It promises lower operational costs, greater flexibility, and a variety of enhanced applications. This course provides a thorough foundation to this new technology to help experts in both the data and telephone industries plan and support their VoIP networks.

This Hands-On course will allow participants to work with an in-class intranet infrastructure constructed with Ethernet switches and routers. Upon successfully constructing and testing the in-class Intranet, participants will observe the installation of voice and video hardware and software on PCs. Participants will test for voicevideo quality over the LAN and WAN. These labs are very useful methods to understand the A-Z of VoIP.

Students Will Learn

- **Explain the basics of telephony and TCP/IP**
- **Understand the engineering tools and procedures required for a voice network and the current technologies leading to the integration of voice and data networks**
- **Explain the basics of Voice over IP (VoIP)**
- **Understand existing and emerging standards for VoIP and network architectures to support VoIP**
- **Understanding Carrier Grade VoIP Technologies**
- **Describe the protocols that support VoIP calls and explain how IP works with the PSTN**
- **Identify some of the challenges VoIP faces in today's networks to demonstrate a good understanding of its capabilities**
- **Explore the latest enabling technologies**
- **Explain Softswicth/MGC, Media Gateways, SIP, Megaco, and MGCP**
- **Voice characteristics, compression standa Mean Opinion Scores (MOS)**
- **Learn about the functional components involved in using gateways to deploy VoIP networks**
- **Explain the concepts of quality of service enforcement techniques**
- **Explain performance and voice quality considerations**
- **Discuss VoIP OSS/BSS**
- **Review transitioning to the All-VoIP PSTN and VoIP Taxation**
- **Explore project planning process of VoIP**
- **Review successful VoIP deployments for wireline, wireless, and cable operators**
- **Discuss successful and unsuccessful VoIP deployments**
- **Step through a practical process for managing a VoIP deployment project**
- **Explore the current and future market trends**
- **Discuss Video Services Over IP**
- **And More...**

Target Audience

This course is designed for anyone who plans on using, evaluating or working with VoIP networks, applications and services.

Prerequisites

Basic telephony and understanding of Basic TCP/IP is recommended.

Course Outline

Module I: Introduction

- Telephony Terminology
- Public Switched Telephone Network (PSTN)

- Circuit Switched and Packet Switched
- Voice and Video Communications
- VoIP and Next- Generation Services
- Voice Compression
- Performance and voice quality considerations
- Voice & Data Convergence
- The Numbering Plan for Telephony - E.164
- Signaling
- Q.931 Messages
- Signaling System Number 7 (SS7)
- Examples of the SS7 Functions
- SS7 Signaling Architecture
- Challenges In Transitioning to an all VoIP
- Carrier Grade Services
- Regulations and Taxation

Module II: Basics of TCP/IP

- Transmission Control Protocol (TCP)
- User Datagram Protocol (UDP)
- Connection-Oriented Protocol vs. Connectionless Protocols
- Basic of IP Addresing
- DHCP (Dynamic Host Configuration Protocol)
- DNS
- Unicast, Broadcast, and Multicast Addresses
- Internet Control Message Protocol Technique

Module II: Basic of VoIP

- Network Architectures to Support VoIP
- Voice over IP Requirements
- Support of Real-Time Services
- Voice Data Transport
- Encapsulation of Voice Data
- Voice Encoding and Packetization
- Issues in Packet Voice Communication
- Real Time Transport Protocol (RTP)
- Real-Time Transport Protocol Architecture
- Real-time Control Protocol (RTCP)
- RTCP QoS
- Real-time Streaming Protocol (RTSP)
- VoIP Signaling Approaches
- H.323
- MGCP
- Megaco
- SIP
- Sigtran
- SCTP
- Evolution of VoIP Signaling Protocols

Module III: Voice Compression

- Voice Quality
- Voice Sampling
- Quantization
- Speech-Coding Techniques
- Components of a Conversation
- Voice Activity Detection
- Fixed Sampling
- Silence Suppression
- Mean Opinion Score (MOS)
- Code Excited Linear Prediction
- Local Delay Compression
- Analysis-by-Synthesis (AbS) Codecs
- G.71: Pulse Code Modulation (PCM)
- Linear Predictive Coders (LPCs)
- G.723.1 and G.729
- G.721
- G.723.1 ACELP
- G.726
- G.727
- G.728 LD-CELP
- G.729A
- Selecting Codecs
- Cascaded Codecs
- ADPCM
- LDCELP
- CS-ACELP
- MPMLQ
- FR
- EFR
- AMR

Module IV: Network Delay Considerations

- The IP Datagram
- TCP Congestion-Control
- Hypothetical Signaling Reference Connection (HSRC)
- End-to-end Voice Transmission
- Transmission time allocation
- Delay Problems
- Delay and Jitter
- Induced Delay
- Processing delay
- Network Congestion
- Call Setup Delays
- Post Dial Delay
- Propagation Delay
- Handling Delay
- Queuing Delay
- Lost Packet Compensation
- Jitter
- IP Precedence

- Priority Queuing
- Random Early Detection (RED)
- Weighted Precedence
- Weighted RED
- Weighted Fair Queuing

Module V: VoIP Signaling and Protocols

- VoIP Signaling Transport
- H.323 Architecture
- H.245 Control Signaling
- H.248/Megaco
- Session Description Protocol (SDP)
- Session Initiation Protocol (SIP)
- SIP Call Process
- Call Setup: SIP to ISDN
- Call Setup: SIP to SS7
- Gateway Protocols
- Media Gateway Control Protocol (MGCP)
- MGCP and Megaco Protocol Architecture
- MGCP and Megaco Protocol Evolution
- MGCP Architecture
- MGCP Stack
- Megaco Features
- Megaco and Megaco Enhancements
- Session Initiation Protocol vs. Softswitch (MGCP and MEGACO)
- MEGACO and H.248
- Protocol Comparison
- Simple Control Transmission Protocol (SCTP)
- Sigtran Architecture
- M2UA and M3UA
- Simple Conference Control Protocol (SCCP)
- Real Time Streaming Protocol (RSTP)
- Comparison of VoIP Call Control Protocols

Module VI: QoS Considerations in VoIP Network Design

- Voice over Packet (VoIP, VoATM)
  Quality of Service (QoS)
- QoS Building BlocksVoIP Quality of Service Provisioning
- VoIP QOS Requirements Delay, Loss, Blocking
- Issues Affecting Voice in an IP Network
- Delay in General
- Fixed and Variable Delay
- Variable Delay Factor: Jitter
- Variable Delay Factor: Packet Loss
- Variable Delay Factor: Sequence Errors
- Variable Delay and QoS Mechanisms
- Quality Measurements and Mean Opinion Scores (MOS)
- Echo and the Use of Echo Cancelers
- Delay Variation and Jitter Buffers
- Managing Delay Addressing Quality of Service

- Bandwidth Limitations
- Traffic Policing
- Traffic Shaping
- IP QOS Mechanisms
- Integrated Services
- Differentiated Services
- Resource Reservation Protocol (RSVP)
- Integrated Services (Int Serv) and QoS
- Int Serv Service ClassesIntegrated Services Architecture
- Differentiated Services (DiffServ) and QoS
- Common Open Policy Server (COPS)
- Multi-Protocol Label Switching (MPLS) and QoS
- MPLS Operation
- MPLS-TE

Module VII: VoIP OAM&P and deployment

- Overview of Management Layers
- VoIP SLA Management Architecture
- VoIP Management Data Sources
- Performance and Fualt Management Functions
- VoIP Provisioning
- Service Control and QoS
- Maintaining an SLA
- Billing and CRM
- Security and Reliability

Module VIII: VoIP Implementations

- VoIP Services
- VoIP over DSL/Cable Modems
- Large-Scale VoIP Service Deployment
- Business Operations Solutions
- VoIP BSS/OSS and NMS
- VoIP E-911 and CALEA
- High-Availability VoIP Network
- Standards and QoS
- Interworking with the Public Switched Telephone Network (PSTN)
- Implementing the PSTN Switch/VoIP Gateway Trunk
- IP Centrex
- Video Bundling With VoIP
- Softswitch
- Services and Implementation Issues
- Security Considerations for VoIP Networks
- Network Access Security
- Device Security
- Using IPSec
- VoIP Traffic Cases
- VoIP Implementations Lessons Learned
- Revenue Assurance for VoIP
- Potential hidden pitfalls
- Deployment issues and challenges

- A project framework for managing a VoIP deployment
- Pulling it All Together
- Case Studies

Notes

LAB 1 Using Lines and IP Bandwidth Calculation

It can be used to estimate the bandwidth required through an IP based network for a fixed number of voice paths. voip bandwidth calculator calculates the bandwidth requirements for voice over IP. This depends on the voice compression scheme used (e.g. G.711, G.723, G.728, G.729), the packet interval and the transport protocol (RTP, cRTP).

LAB 2 Erlangs to VoIP Bandwidth Calculation

The Erlangs and Bandwidth Calculator can be used to estimate the bandwidth which must be provided through an IP based to satisfactorily transport a given busy hour traffic level.

LAB 3 Build a shared Ethernet and share multimedia files.

LAB 4 Build a switched Ethernet and share multimedia files.

LAB 5 Engage in H.323 audio and video session using your workstation.

LAB 6 View a quality of service (QOS) policy built on a workstation.
Voice Over IP

LAB 7 Construct a router network and view its performance using various delay
measurements.

LAB 8 Configure your gateway for successful voice and fax transmission.

LAB 9 Using router utilities to view a Q.931/H.323 protocol connection establishment.

LAB 10 Using your protocol analyzer view multimedia packets in detail.

LAB 11 Configure various VOIP hardware and software products.

And More...

Delivery Method

Instructor-Led with numerous Hands-on Labs and exercises.

Equipment Requirements

(This apply's to our hands-on courses only)

BTS always provides equipment to have a very successful Hands-On course. BTS also encourages all attendees to bring their own equipment to the course. This will provide attendees the opportunity to incorporate their own gear into the labs and gain valuable training using their specific equipment.

Course Length

3 Days