

Course Description

Voice over IP is being widely implemented both within companies and across the Internet. The key problems with IP voice services are maintaining the quality of the voice service and ensuring reach-ability of all users.

Technicians are often required to troubleshoot services beyond just swapping out phones and switches and need to develop skills to identify the source of problems when interconnection between different vendor equipment is involved.

This course will teach you how to use protocol analyzers to capture traffic and analyze the source of problems. It will teach how VoIP services are carried between IP Phones and Analog Telephone Adaptor to IP PBX Servers. It will also teach how SIP protocols used to interconnect servers can be analyzed to test and verify services. Attendees will set up and examine calls inside the classroom and observe how an Internet simulator can be used to introduce the kinds of problems that may be observed when connecting over Internet services. They will then learn how to troubleshoot and fix typical faults.

Students will configure WireShark protocol analyzer, Soft Phones and Asterisk IP PBX software on their own Laptop computers. They will then use this software to learn how to build, test and troubleshoot VoIP services Hands-On.

Students Will Learn

- Compare Traditional Telecommunications Services With VoIP
- Analyse The Key VoIP Protocols Used To Deliver Quality Voice Over Packet Networks
- Recognize Signaling Protocols And Compare H.323 And SIP Approaches
- Size A VoIP Service For Both Enterprise And Carrier Environments To Deliver Voice Quality

- Identify Voip Codes Used From Wire Shark Captures
- Use Wire Shark To Monitor Voice Quality And Verify Voip Signaling
- Troubleshoot And Fault Find Voip Services Hands-On
- And More...

Target Audience

The course will be aimed at technicians and engineers.

Prerequisites

Assuming knowledge of Microsoft Windows and simple TCP/IP networking using PCs and basic telecommunications. It will not assume any prior knowledge of VoIP or packet voice technology.

Course Outline

Module I: Voice over IP Services

Evolution of VoIP Telecommunications

Circuit Switched voice

Packet Switching Data

Motivation: Why use VOIP

Comparison between current voice and data networks

One Integrated Network

Where VOIP can be deployed

Integration at the IP PBX

Integration at the PC

Integration at the desk with IP phones

Analog Telephone Adaptors

Which IP Network Signaling

Signaling using ITU-T H.323

Signaling using SIP

Media gateways and MGCP

Hands On Set up and use VoIP soft phone to place calls across the classroom

Hands-on Set up Wire Shark to capture VoIP traffic

Module II: Internet Protocol Suite Fundamentals for VoIP

Sources for Protocols: ITU and IETF

ITU and its standards used for VoIP

IETF RFCs

Protocol Structures

Layer 2 Frame level services

IP Datagrams

Routing

Routing Tables

ARP Tables

Hands-on Manipulating IP Addresses and Routing Tables

Module III: VoIP Media Streams

Carrying Voice over UDP

RTP and its functions

Recognizing VoIP CODECs

RTCP

Hands-on Taking apart VoIP Media Streams with Wire Shark

Module IV: Telephone Call Fundamentals

Principles of Circuit Switching

Digital voice circuits

CODECs

G.711 calls

Switching Capacity

Sizing a network or switch using Erlangs

Blocking and non-blocking services

Connecting a Call In ISDN

Call Map

Access signaling

Q.931 Signaling messages

Hands-on Calculating Blocking Using Spread Sheets

Module V: VoIP Architectures

Source of VoIP standards

ITU and H323

IETF SIP

H.323 Multimedia conference over packet network

How does a normal phone call get connected

Call Map

Conversion to digital

Dialing and Signaling

Alerting and Call Progress Tones

H323 Components

Map of H323 Components

Gateway (GW)

MCU

Gatekeepers (GK) and Call Managers

Capability Exchange

Negotiating codec

Making an H323 Call

Hands-on Session 3: H.323 Gatekeeper Managed Services

Setup a Gatekeeper managed service

Observe H.323 gatekeeper registrations and call connections

Observe Network Performance Using Netmeter

Setup and use an MCU and observe its performance

Module VI: VoIP using IETF Architecture SIP

SIP Components

SIP Addressing

Connection signaling

Capabilities exchange

SIP Message Format

Comparing SIP and H.323

Media Gateway Control Protocol (MGCP) and MEGACO H.248

Hands-on SIP Proxy Controlled Services

Setup a SIP Proxy controlled VoIP service

Configure a SIP application

Observe and capture SIP Registrar interactions and call connections

Observe Network Performance Using Netmeter

Module VII: Quality of the Voice

What Constitutes Quality

Delay

Availability

Understanding the speech

Recognizing the person speaking

Quality Measures

Mean end to end delay

Mean up time

Mean Opinion Scores

Codecs

Companded PCM

ADPCM

CELP

G.711,G.726, G.728, G.729, G.723.1

Hands-on Analyzing Voice Quality Using Different CODECS

Hands-on Using Wire Shark to Troubleshoot Quality over Internet Services

Hands-on Quality of service Planning

1. Size a VoIP service
2. Predict Delay and QOS performance
3. Mix voice and Data over a low speed Router-Router Link
4. Deliver QOS in practice

Module VIII: SIP Trunking

IP PBX

Interconnection IP PBXs

Configuration of Extensions and Trunks

Hands-on Setup of Asterisk IP PBX

Hands-on IP PBX SIP Trunk Configuration

Delivery Method

Instructor-Led with Hands-On labs and numerous 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

4 Days