

Training

A Training Title and Duration

VoIP Technologies

(Four days)

B Instructor(s)

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C Abstract

This training covers the key features introduced by the Voice over Internet Protocol (VoIP) technology and provides a deep dive into the technology and concepts of VoIP. This training provides a through foundation of VoIP technology, as well as the knowledge necessary to understand the complete VoIP protocol suite. VoIP is a foundational building block of next generation networks which will allow network operators to lower operations costs and take advantage of unified communications applications for Enterprise Networks. This training examines the major components of VoIP networks, the incorporation of VoIP into a data network, packaging and transmitting voice, maintaining voice quality, and addressing traffic issues. Other topics discussed include configuring networks for VoIP and designing complex VoIP solutions encountered in Wide and Local Area Networks (WAN and LAN).

Scope & Objective

The objective of this four days training is to provide a clear and concise technical introduction to VoIP. The participant will have description of the major concept of VoIP. The training is conveyed in eight mutually harmonised parts.

Voice over IP (VoIP) is forecasted to have explosive growth in the coming months and years. Many corporate environments have migrated, are actively migrating, or are researching the process of migrating to VoIP. Some long-distance providers are using VoIP to carry voice traffic, particularly on international calls. This module provides an overview of the basic telephony functions and devices, including PBXs, switching functions, call signaling, and multiplexing techniques. It also reviews the basic components of the Packet Telephony Network and identifies the different requirements in campus, enterprise, and service provider environments. Together, these concepts and techniques provide a solid introduction to the VoIP arena. This first chapter introduces the major components of VoIP and the considerations necessary for setting up and using a VoIP network.

The second chapter deals and discusses analog-to-digital voice signal conversion and signal coding. The delay effect on voice traffic and conversation quality will be studied. Particularly, how packet loss affects VoIP traffic, how to measure voice quality and how to calculate the bandwidth per call.

The chapter introduces also the Transport & Session Layer Technologies, the Open Systems Interconnection (OSI) and the IP Suite reference models, and specifically focuses on the protocols most important to VoIP in these reference models.

The third chapter provides clear ideas on signalling protocol suite for VoIP. First, it discusses the H323 protocol: the ITU standard that provides a foundation for audio, video, and data communications across IP-based networks. It identifies the components of the H.323 standard in a VoIP environment. Second, it introduces the Session Initiation Protocol (SIP): the IETF specification used for multimedia sessions, such as Internet telephone calls, multimedia conferences, and instant messaging. Then, it identifies how the protocol SIP enables support of multimedia sessions for IP telephony. Finally, a Comparison of SIP and H.323 within a VoIP environment will be presented to determine their respective benefits and capabilities.

The fourth chapter describes VoIP upper-layer protocols, such as Real-time Transport Protocol (RTP) and Real-time Transport Control Protocol (RTCP) and the methods used to implement QoS on a customer's data network. This part analyzes various methods available for implementing QoS prioritization of VoIP traffic to achieve the best voice quality, identifies queuing mechanisms available to handle network congestion and makes QoS recommendations to ensure optimal traffic flow.

Voice over Internet Protocol (VoIP) traffic sent across a public IP network is vulnerable to interception or disruption, because IP and the Internet were not originally designed with security in mind. This fifth chapter describes security and remote management considerations for the VoIP environment. Security methods covered in this chapter include Implementing firewalls and Network Address Translation (NAT), Using Virtual Private Networks (VPNs).

The sixth chapter deals with two basics considerations in VoIP network deployment: Dialling plan and network assessment. This lesson describes the attributes of a scalable numbering plan for voice networks, addresses the challenges of designing these networks, and identifies the methods of implementing numbering plans. To integrate VoIP networks into existing voice networks, network administrators must have the skills and knowledge to implement a comprehensive, scalable, and logical numbering plan. This chapter walks you also through the network assessment process for a customer's existing network, and the steps required for an assessment of a new installation

The seventh chapter deals with the migration of telephony systems, hardware and software market solutions in VoIP. It also explains some real world cases that present situations encountered in real life implementing the topics discussed and identify potential challenges which may arise during implementation.

The Labs and Demonstrations deal with the deployment of a small VoIP network based on IPBX BCM50a, IP Phone and Soft phone. Three Labs are proposed. The Using BCM50a Lab describes the features of the BCM50 Element Manager application and shows how to configure the basic parameters for a BCM50 system. The Programming trunks and target lines Lab discusses how lines are used in a BCM50 system and gives you an understanding about which lines you need to program, and how to program those lines. The IP telephony service programming Lab introduces you to Voice over IP (VoIP) sets and available services.

D Motivation

VoIP is a foundational building block of next generation networks and is a first step on emerged unified communications applications. This technical overview covers the major procedures and the main aspects of VoIP. This training is basic for who plans to pass Professional Certification Exam.

E Intended Audience

This training was designed to a wide range of audience who desire a foundation in voice, data, and VoIP technologies. Such individuals will acquire an introduction to the specific knowledge and skills they need to design and support complex VoIP solutions in Wide Area Network (WAN) and Local Area Network (LAN) environments.

Accordingly research and development engineers, telecommunication managers, academic researchers who are interested in the fundamentals and technology concepts of VoIP may find the wide coverage of this overview attractive.

F Prior History

It is recommended that individuals who plan to attend this training have a basic knowledge of Internet Protocol (IP) fundamentals and the concepts of networking. This knowledge is necessary to fully understand and appreciate the concepts presented in this training.

G Outline

- **Chapter I: Introduction to VoIP technologies**
 - ✓ Basic Components of a Telephony Network
 - ✓ Packet Telephony Networks
 - ✓ Identify VoIP Networks
 - ✓ Understand VoIP Network Components
 - ✓ Gauge the Viability of a Customer's Existing Network

- **Chapter II : Voice over IP fundamentals**
 - ✓ VoIP Network Considerations
 - ✓ CODECs
 - ✓ End-to-End Delay
 - ✓ Voice Quality
 - ✓ Bandwidth calculation
 - ✓ Bandwidth engineering
 - ✓ Transport layer technologies
 - ✓ Gateway and their role

- **Chapter III: VoIP Standardization and signaling Protocols**
 - ✓ Introduction to VoIP signalling Protocols
 - ✓ H323
 - Protocol stack
 - Major network components
 - ✓ SIP
 - Messages
 - Network elements
 - Scenarios
 - ✓ Similarities and differences between H.323 and SIP

- **Chapter IV: QoS in VoIP**
 - ✓ QoS Methods at Layer 2

- ✓ QoS Methods at Layer 3
- ✓ QoS Methods at Layer 4 and Beyond
- ✓ Network Policies

Chapter V: VoIP and Security Consideration

- ✓ Elements of a Security Policy
- ✓ Firewall and Address Translation
- ✓ Stateless filtering
- ✓ Stateful filtering
- ✓ Network Address Translation
- ✓ Virtual Private Networks

• Chapter VI: *Dial Plan and Network Assessment*

- ✓ Dial Plan
- ✓ Call routing
- ✓ VoIP Network Health Assessments
- ✓ Needs Assessment Process for Existing Network
- ✓ Pre-Sales Planning Phase
- ✓ Conduct a Readiness Audit

• Chapter VII: VoIP Solutions

- ✓ Migration
- ✓ *Software solutions*
- ✓ *Hardware solutions*
- ✓ *Real Word Cases in VoIP*

• Chapter VIII Labs and demonstrations

- Using BCM50a
 - ✓ Network connection
 - ✓ Load Element Manager
 - ✓ Describe the components of the Element Manager screen
 - ✓ .Using Element Manager, define the BCM50 as a new network element.
 - ✓ Describe the features of BCM50 Element Manager.
 - ✓ Describe how to use Element Manager to define the basic parameters for a BCM50 system.
- Programming trunks and target lines
 - ✓ Define target lines.
 - ✓ Assign target lines.
 - ✓ Perform lines programming.
 - ✓ Perform trunk/line data programming.
 - ✓ Assign line properties.
- IP telephony service programming
 - ✓ Program and register IP Phones
 - ✓ Use the IP softphone 2050.
 - ✓ Available services