Understanding Voice over IP/ IP Telephony

IP Telephony will be the packet technology of choice for most next-generation service providers and enterprise infrastructure managers. This course provides an overview of IP Telephony and explores how the voice traffic is carried over the IP transport. It summarizes the major motivating factors driving the industry and clearly describes the various ways in which Voice over IP can be implemented in the consumer, Enterprise, and carrier market segments. It describes the nodes in the Voice over IP architecture, introduces the key signaling protocols and their functional roles, and summarizes the issues associated with delivering real-time data over IP networks. Finally, it presents a description of the emerging standards for service creation and execution environments. This course also covers how Voice over IP calls are handled & VoIP Services. The benefits and challenges of supporting real-time applications such as VoIP over Wireless data networks are explored.

Course Objectives :

- Covers the basic PSTN, Signaling ,SS7 Architecture, IP stack, TCP/IP stacks,
- Multimedia Stacks,SS7 stacks etc.
- Understanding of the VoIP,SIP,MGCP,H323,Basic Protocol stacks.
- Discuss different ways of assessing call quality speech and MOS.
- Understating the integration of VoIP with the Wireless Technology.
- Understating SIP as a part of 3G Specification.
- Explore the perquisite for toll speech quality in VoIP.
- Discuss the benefits that IP Telephony offers to the communications industry and the obstacles it faces.
- Define how IP Telephony works: voice encoding, wireless vocoders, signaling, media
- Transport, internetworking, and network management.
- Examine how voice over the Internet works today.
- Explore the prevalent industry standards: H.323 and SIP, MGCP and Megaco/H.248.
- Explain the VoIP possibilities in cellular networks.
- Learn how IP Telephony applies to carrier, enterprise, and consumer market segments

Intended Audience:

This course seeks to address the needs of technical professionals who will be implementing and deploying IP Telephony products and services. This course meets the needs of technically oriented telecom professionals/Engineers who are working in VoIP.

Course Outline:

Module- 1 [Basic Telecom]

SS7 brief ,Architectural component ,Signaling, Types of Signalling,SS7 stack, IP stack, TCP/IP stack.

VoIP Introduction & Technical Overview of VoIP:

VoIP utilization TCP or UDP,Different Baseline technology supporting VoIP,What is VoIP, VoIP history,Why VoIP?,What is VoIP,Outlook of VoIP,How to Test VoIP Quality,Standard Bodies,VoIP Call Processing Layered Model,Speech Codecs,VoIP Advantages,Basic Topology,PC Configuration for VoIP,Toplogy of PC-to-phone,VoIP Applications,VoIP Bandwidth Requirement,VoIP Software Support ,VoIP Protocol Stack ,VoIP Interoperatibility,VOIP Networks,PC-to-phone ,Phone-to-phone,PC-to-PC VoIP Network,Internet Telephony,VoIP Standards, Functionality, VoIP Working,Bandwith requirement,Compression Algorithm-

PCM,Compression Algorithm-ADPCM, Working After Compression, RTP, RSVP, RTCP, RTSP,SCTP, Diff:VoIP vs PSTN,Planning for VoIP Interoperatibility, VoIP futures,VoIP Development Challenges, VoIP Future Challenges Gateway software,GateKeeper Software,VoIP Profitibality graph to Co.,Open Source VoIP Projects, VOIPVendors/Services,VoIP links,VoIP hardware,PSTN Gateways,Analog Telephone Adapter,VoIP Phones,Near future: VoIP over WiFi,VoIP Market.

Module –2 [VoIP Telephony Protocol]

H.323 Introduction,Scope of H.323,Why is H.323 Important?,H.323 Architecture: Components Terminal Gateway, Gatekeeper,Multipoint Control Unit (MCU),H.323 Architecture: Protocols and Procedures,H.323 Terminal Protocol Stack,Audio Codecs,Video Codecs,Data Conferencing Control and Signaling Mechanisms,H.225.0 Call Signalling,H.245 Media Control,H.225.0 RAS,H.323 Call Setup and Tear Down,Supplementary Services,Security issues,Challenges ,Interoperability and Implementation Issues,Lack of Value Added Services,Quality and Security Issues,Future Prospects,H.323 or SIP,H323 terminal stack.

Module-3 [VoIP Telephony Protocol]

MGCP Media Gateway Control Protocol Introduction,History of MGCP, MEGACO and H.248,H.248/MEGACO,H.248/mecago call flow,MGCP,Comparing SIP and MGCP/H.248,Conclusions MGCP, MEGACO/H.248,MGCP Basic network Architecture, MGCP Applications,MGCP,Advantages/Disadvantages ,MeGaCo/H.248,Model of VoIP (MGCP),MGCP Commands,MGCP Command Responses,MGCP Call Establisment,Media Gateway Control Interface,Endpoints

Module -4 [VOIP Network Operation]

QOS Fundamental limits,Testing of the VoIP quality,MOS,Trade off : Quality urface,Internet performance problem,Web performance problem,Packet losses,Sources of Fixed Delay,Sources of Variable delay,Queuing Packet Methods,DiffServ,Latency ,Accumulation Delay,Processing delay,Network Delay,,Polling Delay,Echo Jitter,Loss packets,Interpolation,Redundancy,Voice Coder,Congestion,Packet loss,Queuing system,Introduction to the Queuing models,Queuing behaviour.

Module -5 [VoIP Telephony Protocol]

Session Initiation Protocol` (SIP) Overview,Introduction,Protocol Architecture,Component Architecture,Addressing and locating SIP proxies,Programming SIP services,VoIP protocol architecture,SIP Application,SIP Addressing,SIP Addressing,SIP building Blocks,Back –to-back UA(B2BUA),SIP building block properties,SIP Architecture :Peer to Peer, outbound proxy,SIP Architecture :VoIP to PSTN, operation in proxy mode,SIP operation in the redirect mode,Locating SIP Users,Locating Registars and location servers,Basic User location mechanism,Basic SIP routing Mechanism,Outbound proxies, Locating Users DNS,SIP protocol operation,SIP request and responses,SIP Syntax, SIP methods, SIP Invitation and Media Negotiation,SIP responses,SIP Response routing,Forcing Request paths,Request Routing,SIP Request Forking,SIP sequential request forking,Parallel forking call flow,Invite Reliability,SIP Architecture,Additional call information,SIP based services Invitation modes,Call transfer,Security Issues,SIP authentication,Digest Authentication,Quality Of Services,SIP caller Preferences,Extended SIP contact Header,Contact Example,SIP Notify example,SIP home architecture,Programming SIP services,Examples,Call processing language,Abstract Structure,SIP Programming,Call redirect ,Example call forward busy/no answer.

Module-6 [VoIP services]

Implementation of the VoIP Management system,management Networks,Testing,Transastion to VoIP ,VoDSL,VoATM,Directory Services,VoCable,VoIP Security,Call Centres,Working with vendors,Universal messaging,GPS,Quality onsideration, Billing, Provisioning,CCMS,Fax Over IP ,Softphones /Softswitches ,implementation ,Universal Messaging,Voice Mails, VoIP Convergence, Triple play services, and IPTV.

Module –7 (Hot Coverage)

VoIP working with Wireless Technology
SIP Integration with 3G
MGCP Integration with 3G,
Introduction to EoIP,
UMTS,UTRAN, RNC.