VoIP and IP Telephony - Hands-on

Course No. 1223 **Type:** Hands-on **Duration: 5 Days**

Course Overview:
In this course you will discover the explosive dynamics of bringing voice and data together on a single network. You will learn How IP telephony has evolved; review the typology, architecture, protocols, and special issues.
The course participants will setup IP Phone and register to a Softswitch, define different services and applications which are widely used by SMB and large Enterprise.
Each one of the participants will access sophisticated Softswitch which will be on the same network for generating end-to-end IP telephony and Video calls.

Who should attend?
Professional people in the communication and IT, Engineers, software developer, Technical support, field engineers

Prerequisites:
Basic knowledge of:
- IP Networks
- Basic telephony

Lecturer: Eyal Tomer
Mr. Eyal Tomer is a senior lecturer at LOGTEL. Mr. Tomer has more than 20 years of experience in the telecommunication technology, particularly in the NGN technology.
Prior to founding ICVision, Mr. Tomer worked at Lucent as a project manager and was the leader of the Integration team for a Cable telephony project in Israel.Prior to Lucent, Mr. Tomer worked at Arelnet, which was later acquired by Airspan. At Arelnet, Mr. Tomer was responsible for the Product Management activities and for designing the new VoIP product line. During his work at Arelnet, Mr. Tomer cooperated with Intel, implementing a Carrier-Grade VoIP project including SS7 infrastructure. Before this, Mr. Tomer was the Product Department Manager of RADCOM Ltd., one of the leaders of advanced VoIP test-solutions. Mr. Tomer holds a B.Sc. degree in Electronics and Communications from Tel-Aviv University, Israel.

Course Content:

Day 1

1. Traditional versus IP Telephony
- Peer2Peer versus Master/Slave protocols
- VoIP typical deployment
- Class 4 and Class 5 services

2. Media GW
- MGW Profile and functions
- QoS Mechanisms
- PSTN Interfaces (FXS, FXO, PRI, etc.)
- Media over IP

3. RTP and RTCP
- Internet Multimedia Protocol Stack
- RTP Structure
- RTCP Profile and structure
- CRTP and Bandwidth consumption

4. QoS and QoE challenges for IP Telephony
- Definitions and terms
- The need for QoS
- Solutions to provide QoS
- QoS requirements for Triple-Play

5. Media Compression methods
- Coders types
- Bandwidth utilization
- Side effects due to compression

6. MGCP architecture and components
- MGCP protocol Stack
- MGCP Components (MGW, MGC, etc.)
- MGCP based network
- Call-flow Examples

Day 2

7. Media GW Controller
- MGC concept and Profile
- MGC functions
- ENUM concept- RFC2916
- MGC architecture

8. Megaco Architecture
- Master/Slave versus P2P
- Megaco Standards and components
- Megaco Architecture
- Megaco Commands
- Call-flow Examples

9. SIP Introduction
- What is Session Initiation Protocol
- The Incentive
- SIP Components
- SIP Servers (Proxy, Registrar, Redirect, Location)

10. SIP architecture
- Protocol Stack
- SIP Transactions and response codes
- Addressing format
- DTMF and VoIP (In-Band and Out of Band methods)

Continued...

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Course Content:

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11. Hands-on Session :Setup NGN Network (including WiFi Hotspot) and IP Phone/Softphone
- Setting and registration process
- CA List Configuration
- Network parameters
- Account setting
- Compression setting (Codecs)
- QoS setting

12. Hands-on Session : Real Time Registration capture analysis (using Wireshark)
- Real Time Call setup capture analysis (using Wireshark)
- Digest Authentication process analysis

Day 3

13. SIP Protocol structure
- SIP Timers for reliability
- Forking Methods
- Provisional Response Acknowledge method
- Call-flow Examples

14. Session Description Protocol
- SDP Main tasks
- SDP Messages
- Mandatory fields and optional fields

15. Softswitch and Class 5 services- Part 1
- Softswitch profile
- SIP new methods
- Softswitch as application platform
- Class 4 and Class 5 functionalities

16. Softswitch and Class 5 services- Part 2
- Advanced Voicemail
- Unified messaging
- Instant Message

17. Hands-on session: Softswitch administration
- Add new user
- User administration level
- System Administration level
- Define the Phones types (SIP, MGCP, etc.)

Day 4

18. H.323 Architecture
- H.323 Topology
- H.323 Architecture and Components
- H.323 Call-Flow
- H.323V6

19. Security Threats
- Security Criterias
- External and Internal threats
- Classic Threat Models

20. Security building blocks
- NAT and FW
- Network Access Control (NAC)
- IDS and IPS
- Session Border Controller

21. SBC and Lawful Interception
- Stateful and Stateless Session Controller
- Potential Threats and solutions
- NAT Traversal
- Lawful Interception

22. Hands-on Session- Real Live Capture Analysis using Wireshark
- Session Description Protocol Analysis
- RTP Capture analysis
- RTCP Capture analysis
- QoS and Media analysis

23. Hands-on session: VoIP Services
- Secured and Non secured conference
- DTMF over IP – Real Live capture analysis

Day 5

24. SIP Trunking Overview
- Introduction
- What is SIP Trunk
- SIP trunk Challenges
- References and relevant RFC

25. SIP and Telephony networks
- SCTP- Stream Control Transmission Protocol
- SCTP and adaptation Layers
- SIP-T and SIP-I (ISUP encapsulation and mapping methods)

26. VoIP Design and Implementation (Part 1)
- Network Impairments description
- The sources of network Impairments
- Correlation between the Impairments

27. VoIP Design and Implementation (Part 2)
- Objective and Subjective speech measurement
- MOS Definition
- PSQM, PAMS, PESQ, E-MODEL

28. Hands-on session: Advanced implementations
- Impairments Analysis (using Protocol Analyzer)
- QoS provisioning – Live demo

29. Summary