SIP - SESSION INITIATION PROTOCOL

Course Description

The Session Initiation Protocol - SIP was developed to create and handle multimedia sessions between different applications. Due to the inherit flexibility, combined with its simplicity and scalability, SIP was chosen by 3GPP as the protocol to be used for IMS.

The training Session Initiation Protocol – SIP, is a combines theory with exercises to cover both the fundamental and advanced features and functions of SIP.

The History of SIP, the protocol structure, the states and the more advanced functions and features such as messaging, SDP and security are explained.

Other topics covered are TCP/IP with related DNS, NAPTER, message forwarding and SIP session management.

Furthermore, SIP state handling, SIP extensions, SIP Security, the SIP services, and the use of SIP in the classic telephony and IMS are described.

Content

ORGANIZATIONS AND STANDARDIZATION

- TCP/IP and Internet History
- Internet standardization bodies, 3GPP & IETF
- IETF Workgroups



SIP INTRODUCTION

- · SIP History
- Voice over IP connection
- IP Network Overview
- SMTP & HTTP
- · Traffic case: Voice over IP
- SIP Components
- METHODs and Responses
- Basic Call Handling: REGISTRATION



- Basic Call Handling: INVITATION
- Syntax of a SIP Message
- SDP
- SIP CANCEL
- Forking Proxies
- SIP Addressing
- SIP Services
- IP Tunneling

SIP ADDRESSING AND REGISTRATION

- SIP Addresses and URI
- Registration
- Multiple Registrations
- Registration in- and output
- · Reusing addresses

TCP/UDP/SCTP, DNS AND NAPTER

- TCP/IP Protocol Stack
- TCP/IP Encapsulation Process
- IHL and Total length Example
- TCP
- SCTP
- · TCP vs. UDP
- UDP, TCP or SCTP
- Introduction Domain Name System
- Specific Domain Names and FQDNs
- · Records used in EPC
- DNS on IPX/GRX
- NAPTR

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- SRV Service Record
- · Address Record

SIP MESSAGE FORWARDING

- To the Destination
- Selecting Transport Protocol
- Selecting IP Address
- SIP Transport Issues
- The Via header Sent By
- The Via header received
- Simple Loop Detection
- Special Loops
- Forcing traffic a certain way
- · Recording PATHs

SIP SESSION MANAGEMENT

- SIP Methods
- SIP Extensions
- SIP Response Codes, Definitions & Examples
- Important rules
- Session setup SDP Usage
- An Offer/Answer Model with the SDP
- · Recording Routes
- Session Modifying
- Dialogs Creation, State

SIP STATE HANDLING

- · A scalable network architecture
- · Stateful or Stateless



- Transaction Lifetime
- Transaction Identification
- Responses and Transaction state
- 2xx and 3++ Responses to INVITE
- Forking Proxies
- · Stateful Cancel and 3++
- · Proxy core Response handling, forking
- Waiting and Selecting
- INVITE & Non-INVITE Transactions
- · SIP Timers
- Usefulness of 408 Responses
- · Recovering from packet loss
- · Rules and Exceptions

SIP PROTOCOL EXTENSIONS AND UPDATES

- Extending the Protocol
- Requiring Extensions
- Request-Disposition
- Callee Capabilities
- Callee Preferences
- · SIP, SDP and Quality of Service
- · Provisional responses and Reliability
- Reliable Provisional Responses
- Provisional Response ACK
- Early Media and Forking
- Session Timer

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SIP SECURITY

- SIP and Firewall
- N.A.T & P.A.T Basics
- Implementation issues
- Network Address Translation
- · STUN and TURN
- Interactive Connectivity Establishment
- Symmetrical (SIP) Responses
- Client Initiated Connections in SIP
- Securing SIP signalling
- Authentication Digest
- Privacy and Identification
- S/MIME
- Media Level Security

SIP SERVICES

- SIP Service Creation
- Call Processing Language
- Some useful headers
- SIP and call-transfer
- Globally Routable User-Agent URI
- 3rd Party Call Control
- Replacing sessions
- · Call transfer Attended
- Events
- Presence
- Instant Messaging
- · Back 2 Back User Agent



- Handling DTMF in SIP
- URI Lists

CLASSIC TELEPHONY USING SIP

- · SIP and PSTN
- Phone numbers vs. Sip-URI's
- E.164 Number Mapping (ENUM)
- SIP and PSTN
- IP Multimedia Subsystem
- SIP group (SiPiT)
- Open Source Developments
- Tools
- Simple SIP Checklist
- · SIP Swiss Army Knife
- SIP Scenario Generators

IP MULTIMEDIA SUBSYSTEM (IMS)

- IMS Introduction
- Why IMS?
- Mobile Network
- · Protocol stack
- · PSTN Breakout
- Public and Private IMS Identities
- IMS- Service Profile
- Traffic Case: IMS Invitation
- IMS Registration & De-registration
- WM-Map: IMS Registration & Invitation

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Target audience

The target group for this course are software developers, software testing engineers, multimedia operators, developers and other personnel working with either fixed or mobile networks and who need in-depth technical knowledge about SIP.

Pre-requisites

It is recommended that the participants have a basic understanding of data- and telecom, TCP/IP and Voice over IP.

Course length

3 days

Widermind communicates the knowledge you need to develop and implement new technologies for current and future network operations. Our clients are telecom operators, system integrators, system suppliers and consultancy firms.

Based in Stockholm, Sweden, we develop courses backed by a comprehensive network of associates. Our instructors employ technical and pedagogical skills that have made Widermind training well known and appreciated as one of the best services in the field.

You are warm welcome to contact our representatives at:

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