SIP Advanced (Session Initiation Protocol)
Underleverantör: IP-Solutions

Datum

- 4- 6 Apr, 2011
Stockholm
- 2- 4 May, 2011
Stockholm
- 30 May- 1 Jun, 2011
Madrid
- 15-17 Jun, 2011
Stockholm

Detta är en avancerad SIP-kurs med mycket praktiska övningar. Funktioner såsom transaktionshantering, dialoger, diverse felsituationer, timers och mycket annat gås igenom i detalj. Du kommer att lära dig hur SIP fungerar inom både IP-telefoni och multimedia-lösningar, såsom presence och Instant Messaging (IM). Utbildningen går igenom hur SIP fungerar i mobila och fasta nät och grundkunskaper inom IP-telefoni och SIP behövs.

Målgrupp

Målgruppen för denna kurs är utvecklare och testare samt personer som jobbar med implementationer av SIP lösningar. Kursen är riktad både mot fasta och trådlösa lösningar.

Förkunskaper

Grundläggande kännedom om datakommunikation motsvarande kursen Datakommunikation Grundkurs, grundläggande kännedom om TCP/IP motsvarande kursen TCP/IP samt goda kunskaper i grundläggande VoIP och SIP motsvarande kursen SIP Fundamentals.

Kunskapstest
Testa dina kunskaper inom SIP gratis redan idag. Våra rådgivare kontaktar dig med förslag på vilken av våra utbildningar som är mest optimal för dig.

SIP-test

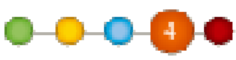
Övrigt

Denna kurs finns som schemalagd utbildning och presentationen ges på svenska eller engelska. Under kursen blandas teoretiska presentationer med praktiska övningar. Vi kan även hålla denna kurs företagsintern. Kontakta oss för att få reda på hur vi kan hjälpa er med anpassade kurser.

Längd

3 dagar

Svårighetsgrad



Agenda

VoIP Signalling Overview

- Functional components of a Voice over IP Network
- The need for VoIP signalling and the different alternatives
    - SIP, H.323, MEGACO, BICC
    - IETF Multimedia Architecture
    - Signalling and User plane separation
- Brief repetition of RTP and RTCP

SIP Refresher and SDP Update

- Background, history and Internet heritage
- Main components; servers and clients.
- Benefits and rules with different transport protocols
    - UDP, TCP, SCTP.
- Basic sessions and SIP mobility features
    - Proxy and Redirect mode
    - SIP Methods & Response codes
    - SIP Registration
- Session Description Protocol
    - SDP Offer/Answer Model
    - Quality of Service Extensions
    - Connection Oriented Transports in SDP
    - Media groupings in SDP

Exercise 1 – Basic Calls

- Basic SIP sessions with SIP proxy and Registrar
- Using Wireshark for traffic monitoring
- SIP Signalling and SDP Negotiation analysis
- Methods; INVITE, BYE, REGISTER

Protocol Foundation

- Message structure and format rules
    - Mandatory headers and parameters.
- Proxy and Client DNS Usage
    - NAPTR record type
    - SRV record type
    - A record type
- SIP message routing rules
    - Route headers
    - Record Routing
    - Via header response routing
- Detailed proxy behaviour
    - Location server lookup
    - Request forwarding
    - Response processing
- Statefullness in SIP Servers
    - Limitations of stateless servers
    - Transaction and dialog-stateful servers
    - Registration stateful servers
- Creating early dialogs for early media

Exercise 2 – Call Signalling Details

- Inter-domain call setup and routing
- DNS usage
- RTP details
- SIP Header analysis
- Usage of Request-URI
- Record-routing examples
- ReINVITE's or UPDATE for session re-negotiation

Step by step walkthrough of advanced call scenario Features and Functionality

- Extending the SIP protocol
    - Using OPTIONS
    - Negotiation extensions
    - Requiring extensions
    - Handling new SIP methods in old proxies
- Reliable provisional responses
    - PRACK
    - RSeq and RAck
- Forking and Cancelling requests
    - Cancel and stateful proxies
- Caller preferences and Callee Capabilities
    - Addressing and Registration extensions
- Using Early Media
    - Simplex or Full-duplex
    - Issues with forking.
- Quality of Service and SIP
    - Require Qos with SDP parameters
    - Using UPDATE in early dialogs

Exercise 3 – Forking and CANCEL

- Forking Calls When and Why?
- Response processing at forking
- Method: CANCEL
- Parallell and sequential forking

Security

- Firewalling SIP servers and clients.
- Encryption and Authentication – How?
- The Firewall and NAT problem
    - SIP-away firewalls
    - SIP signalling and NAT
    - Symmetric Responses
    - Managing Client Initiated Connections in SIP
    - Media NAT traversal: STUN, TURN, ICE
- SIP Privacy and Authenticity
    - S/MIME examples for end to end security
    - Privacy services
- Securing the media channel

Exercise 5 – Security and DNS

- User authentication and http digest
- SIP symmetric responses
- Understanding DNS queries

Services and Applications

- Service creation possibilities with SIP
  - o Overview of SIP-CGI, CPL, Java servlets, Jain, Parlay
  - o Service examples with CPL
- SIP Basic call-services and PBX-like features
  - o Call-forwarding, voicemail, CLIR/CLIP, etc
- Call-transfer and Call-Pickup
  - o REFER and Replaces:
- 3rd party call control
- SIP for events
- SIP and presence
- Instant messaging in SIP
  - o Stand-alone messages with MESSAGE
  - o Session based messaging with MSRP
- B2BUA (Back to back User Agent)
  - o Requirements and Possibilities

Exercise 4 – Services

- Presence
- Instant messaging
- Authorization and message encoding
- Methods; SUBSCRIBE, NOTIFY, PUBLISH, MESSAGE

Classic Telephony Using SIP

- Sending DTMF in VoIP
  - o DTMF and RTP, rfc2833
  - o DTMF and SIP
- Merging PSTN Networks and SIP
  - o SIP for telephones SIP-T
  - o Q.1912.5 – SIP-I
- Phonenumbers and SIP-addresses
  - o Tel: and SIP: URIs
  - o Address translation, interworking
  - o DNS and ENUM
- Mobile SIP Telephony
  - o IMS, IP Multimedia Subsystem

Summary and Future

- Summarizing the whole course
- What to read first – List of RFC's and Internet-Drafts
- Links and references