SIP, SDP and other NGN Protocols
Signaling & Protocol Analysis

Course Duration:

- 3 days.

Course Description:

- This course addresses the needs of engineers and technicians who need to analyze the signaling messages within and among complex Next Generation Networks.
- In that respect, some but not all emphasis is on IMS-based mobile networks as standardized by 3GPP for the enabling of IP-based multimedia services.
- The technical focus of the course lies on the understanding of SIP/SDP-based session control signaling among SIP-servers and between SIP-servers and SIP-terminals.
- Especially the perspective of the SIP-terminals (UA) is covered in detail. And intentionally the course does not focus only on standard voice calls but also covers other session types like instant messaging, gaming or VoD.
- The student is confronted with various SIP/SDP-related and practice-oriented questions and exercises to gain the maximum output from the course.
- After a comprehensive review of NGN's, SIP and SDP and other protocols like IPsec, MSRP or DIAMETER the course focuses on a detailed description of the various SIP- and SDPaspects (Network Elements, SIP/SDP-Terminology, SIP-Timers) and on advanced SIP/SDPoperation.
- The latter part deals among others with essential issues using SDP-preconditions and, very important, the handling of ungraceful session releases.
- The course continues with the presentation of typical SIP-scenarios for different applications like registration, video or voice call setup, application server access and interworking between SIP and ISUP.
- One chapter is dedicated to 3GPP-based and IMS-enabled mobile networks and the related SIP- and SDP-specifics.
- Note that this course represents the protocol view of the IMS. A more architecture and network layout related course about the IMS is also available. Please check at www.inacon.com.

Prerequisites:

- The student needs to have a solid background of the IP-protocol stack. Previous practical exposure to the configuration and operation of LAN/WAN is favorable.
- Fundamental knowledge of mobile networks (particularly GSM, GPRS and UMTS) is necessary.
- Practical experience with protocol testers and IP-sniffers (e.g. Ethereal) is necessary.

Course Target:

- After the course the student will have a clear understanding of all details of SIP and SDP operation.
- The student will understand how different session types are established through SIP over different types of IP-CAN's like DSL-lines, WIMAX or GERAN/UTRAN.
- The student is enabled to perform sophisticated network analysis in SIP-enabled networks and to nail down typical network failures.

Some of your Questions that will be answered:

- What is the difference between stateful and stateless SIP-proxies, B2BUA's and SBC's?
- How are SIP-messages identified that belong to the same transaction, to the same dialog and to the same call? And what are the differences between transactions, dialogs, sessions and calls with respect to SIP?
- How are SDP-items and attributes used to specify an audio, video, message or gaming session? How can I add, remove or change media during a session?
- How does the SIP-communication relate and interface to resource management within different types of IP-CAN's like GERAN/UTRAN, WIMAX or DSL/CableTV?
- What is a SIP-URI, a PRES-URI or an IM-URI and how do they differ from a TEL-URI?
- Which meaning do NAPTR, SRV and ENUM have?
- How is mutual authentication (IMS-AKA) achieved in the 3GPP-based IMS?
- How do 3GPP-specific procedures like GPRS-attachment and PDP-context activation relate to SIP-registrations and SIP-dialogs?

Who should attend this Course:

- Engineers, technicians and IP-professionals who are involved in the setup, configuration and maintenance of SIP-enabled networks.
- Network Operators and technical assistance center suppliers who need to perform error diagnosis and troubleshooting within NGN's.
- Everybody who requires detailed knowledge about the operation of SIP/SDP-based NGN's.

Table of Content:

Introducing the Playground of SIP / Reviewing SIP and SDP Basics

⇒ Response: 180 (Ringing)

⇒ Response: 200 (OK)

⇒ Request: ACK

⇒ Example of one of the 4300 Speech Frames

⇒ Request: BYE

⇒ Response: 200 (OK)

⇒ The Related Session Description Protocol (SDP) Contents

Detailed Consideration of Formal SIP-Protocol Aspects

- **SIP – Protocol Structure**
- **SIP-Network Architecture**

⇒ User Agents

⇒ SIP-Servers (generic)

⇒ Special SIP-Servers

⇒ Operation of Stateless SIP-Proxy Servers
Advantages of Stateless SIP-Proxies, Disadvantages of Stateless SIP-Proxies, Other Assets of Stateless SIP-Proxies

⇒ Operation of Stateful SIP-Proxy Servers

⇒ Operation of Registrars

⇒ Operation of Redirect Servers

⇒ Operation of Forking SIP-Proxy Servers (always stateful)

- **Operation of B2BUA and SBC**
  Example: VoD for a Mobile Client with limited Access Rates

⇒ Operation of Event Servers

⇒ Soft Switches and their Controllers

- **Summary**
- **Important SIP-Terminology / Step 1: Two UA's …**
  Transaction, Dialog / Call / Early Dialog (Definition)
- **Session (Definition)**

⇒ Dialog Identification (two Users / with or w/o Proxies)
Session Identification and Distinction

⇒ Transaction Identification (two UA's / no Proxies)
The Cseq Parameter , The Branch Parameter, Magic Cookie "z9hG4bK", Example: Transaction Identification, Sequence Numbering (CSeq)

- **(1) Transaction Identification (two UA's / with Proxies)**
  Transaction Identification through "branch" is done Hop-by-Hop
- **(2) Transaction Identification (two UA's / with Proxies)**
  Transaction Numbering through "CSeq:" applies end-to-end
- **Practical Exercise:**
- **Transaction-specific Messaging**
  Option 1: Request = INVITE / Transaction = successful

  ⇒ Option 2: Request = INVITE / Transaction = unsuccessful

  ⇒ Option 3: Request = INVITE / Transaction = cancelled

  ⇒ Option 4: Request = REGISTER

  ⇒ Option 5: All Other Requests
- **Practical Exercise**
- **Amendments in case of more than two Peers**
  Introducing Different Contact Addresses per User, Behavior of Forking Proxies, The Terms Call, Dialog, Session and Transaction in case of Forking
- **(1) Message and Parameter Details**
- **(2) Message and Parameter Details**
- **Summary**
- **SIP-Timers**
- **INVITE Transaction (UAC-Side - Response: 200-OK)**
- **INVITE Transaction (UAC-Side - Response: 3XX – 6XX)**
- **INVITE Transaction (UAC-Side – no Response received)**
- **INVITE Transaction (UAS-Side sent 3XX – 6XX – wait for ACK)**
- **INVITE Transaction (UAS-Side sent 200-OK – wait for ACK)**
- **"None"-INVITE Transaction (UAC-Side - Response: 2XX – 6XX)**
- **"None"-INVITE Transaction (UAC-Side - no Response)**
- **"None"-INVITE Transaction (UAS-Side - final Response sent)**

  ⇒ SIP-Message Format
  General Information, Request Messages, Response Messages

  ⇒ Selected SIP Header Parameters
  Example: SIP Logfile with Header and Routing Info
- **SIP-Message Contents**
- **(1) The Different Method-Types**
  REGISTER, INVITE, ACK, CANCEL, BYE, OPTIONS, INFO
- **(2) The Different Method-Types**
  MESSAGE, SUBSCRIBE, NOTIFY, PUBLISH, PRACK, REFER, UPDATE

- **SIP-related Identities**
  Public User Identities, Private User Identities, Service Identities

  ⇒ Address Specification / Request-URI

  ⇒ The SIP(S)-URI
- **The Status Line**
- **The “From:” and the “To:” Header Fields**
- **The “Call-ID:” and “Max-Forwards:” Header Fields**
- **The “CSeq:” Header Field**
- **The “Via:” Header Field**
- **The “Contact:” Header Field**

Detailed Consideration of Formal SDP-Protocol Aspects

- **Session Description Protocol**
- **Logfile Example: Session and Media Descriptors through SDP**
- **Session Description Items**
- **The “o=”-Line (Origin)**
- **The “c=”-Line (Connection Info)**
- **Time Description Items**
- **Media Description Items**

  ⇒ The “m”-Line (Media Announcement)
  “m”-line Media Type Attribute (MIME) / some Examples, Media Type = message / Subtype = CPIM, (1) “m”-line / Details of the Transport Protocol Types, (2) “m”-line / Details of the Transport Protocol Types, (3) “m”-line / Details of the Transport Protocol Types, (4) “m”-line / Details of the Transport Protocol Types

  ⇒ Use Case Example: Floor Control (BFCP) during Push-to-Talk
- **BFCP-Operation during a Conference Session**
- **The “b”-Line (Bandwidth Information)**
- **Details of the Bandwidth Modifiers “RR” and RS”**
- **“a”-Lines (Attributes)**
- **Example 2: AMR-Codec Definition and Parameterization**
- **Example 3: TCP-Connection Definition**
- **The Offer / Answer Model**
- **Session Identification Parameters at both Peers**

Advanced Use of SIP and SDP

- **Reviewing the SIP-Scenarios: Critical Issues**
- **Question 1: How to find the called party across networks and how to route SIP-messages?**
- **Answers …**
  SIP-clients need to register to “their” SIP-registrar to bind their current IP-address to their SIP-URI.
- **But how does a User Agent find “its” Registrar?**
- **DNS-Queries with NAPTR- and SRV-Records**
- **Question 2: What happens if Provisional Responses get lost?**
- **Solution: Provide for the Option to acknowledge provisional Responses**
- **Indicating Lacking Support for a Required Feature**
- **Using PRACK to acknowledge Provisional Responses**
- **Transaction Abort in Case of Lacking PRACK**
- **Summary**
- **Question 3: How to assure appropriate Resource Allocation in both Ways before alerting the Called Party?**
- **Answer: We define an additional Handshaking Procedure**
- **Overview: Resource Management using SIP and SDP**
  Positive Outcome – Resource Reservation successful
- **(1) Option 2: Related SDP was contained in a SIP-Response**
- **(2) Option 2: Related SDP was contained in a SIP-Response**
- **One Media Stream is rejected altogether**
- **Handling the Precondition Attributes “a = curr:” and “a = des:”**
  - ⇒ The new Option Tag “precondition”
  - ⇒ The “m = …” Line / Port Number and Payload Type
  - ⇒ Interpretation of “local” and “remote” Direction-Tags
  - ⇒ Interpretation of the “current-status” Attribute (“a = curr:”)
  - ⇒ The “desired-status” and “confirm-status” Attributes
  - ⇒ Preconditions fulfilled: the final Status
    Example 1: Resource Reservation if IP-CAN = GERAN/UTRAN, Example 2: Resource Reservation if IP-CAN = WIMAX, Example 3: Resource Reservation if IP-CAN = IntServ-aware, Example 4: Unsuccessful Outcome, Example: LTE Resource Reservation / Policing
  - ⇒ Network Initiated (IMS triggered during Call Establishment)
  - ⇒ Example for VoIP-IMS in LTE – S1-MME/UserPlane
    Three EPS Bearers for a Voice Call?

SIP, SDP and DBP in 3GPP-Networks

- **The IMS-AKA Authentication Process**
- **Application of IPsec between MS/UE and P-CSCF**
- **(1) Registration to the IMS in 3GPP (Detailed Scenario)**
- **(2) Registration to the IMS in 3GPP (Detailed Scenario)**
- **(3) Registration to the IMS in 3GPP (Detailed Scenario)**
- **(4) Registration to the IMS in 3GPP (Detailed Scenario)**
- **Mobile Originating Calls**
- **Call towards the PSTN**
- **Call from the PSTN**
- **VoLTE – The Alternatives**
  Architecture in case of IMS-based Voice Services
- **Single Radio Voice Call Continuity (SRVCC)**

  ⇒ SRVCC in Release 8

  ⇒ Circuit Switched Fallback (CSFB)
  The SGs Interface
- **IMS based VoLTE – A Brief Comparison**