SISTEMAS DE SEÑALIZACION SIP I & II
(@-SIP1&2)

Contenido

1. Why SIP? Gain an understanding of why SIP is a valuable protocol despite competing technologies like ISDN, SS7, H.323, MEGACO, SGCP, MGCP, and IPDC.
Circuit Switching
VoIP Protocols
SIP and the Softswitch
A short history of SIP

2. IP Routing and Switching Overview
IP Routing
Ethernet Essentials

3. TCP and UDP Essentials
How VoIP uses TCP
How VoIP uses UDP

4. SIP Architecture – Understand the components and Core SIP Protocol.
The SIP architecture
UA, Proxy, Redirect, Forking, B2BUA
Session Border Controller
Multimedia Architecture
RTP/RTCP
SDP
Methods
REGISTER
INVITE and ACK
UDATE
OPTIONS
REFER
CANCEL
BYE
SIP responses
1xx Informational
2xx Final
3xx Redirection
4xx Client Error
5xx Server Error
6xx Global Failure

HAND-ON LABs

1 Configure PCs and Build an IP network
2 Learn how to use Wireshark (Ethereal)
3 Configure the SIPURA ATA
4 Configure X-Lite (CounterPath) SIP soft client
5 Configure the ONDO SIP Proxy

6 Configure the Asterisk based trixbox
7 Perform Call trace analysis
SIP REGISTER without authentication
SIP REGISTER with authentication
Simple SIP Call without INVITE authentication
SIP call with INVITE authentication
Busy call
Vacant Number (Call a number that does not exist)
Abandoned Call (Hang up on an unanswered call)

5. SIP Uniform Resource Indicators (URIs) - Understand the format of SIP URIs, and how URIs interoperate with PSTN dialing plans, email systems, and web pages.
Generic URI information (RFC 2396)
Direct or Proxy
PSTN number (RFC 2808)
Instant messaging
Presence
In registrations

6. SIP Headers - Learn how the straightforward format of SIP messages organizes the information needed to process a call. Understand the information contained in these messages and how it is formatted.
Via:, Branch, Max-Forwards:, SIP Dialog (To, From, tag= fields, Call-ID:)
CSeq, Proxy-Authenticate:, Proxy-Authorize:, Contact: Expires:
User-Agent:, Content-Length:, Allow:, Supported:, P-Access-Network-Info
P-Charging-Vector:, P-Preferred-Identity:, P-Asserted-Identity:, Authorization:
Security-Client:, Security-Server:, Content-Type:

7. Session Description Protocol (SDP) - Learn how SIP uses SDP to define technical parameters that support a voice over IP media channel.
Session parameters
SDP format
Extending SDP
SDPng
Media negotiation
Changing session parameters

8. SIP Utilization of the DNS – How to use DNS to find the caller party.
DNS basics
A-record
SOA
NS record
MX record
SRV record – RfC 2782
Locating SIP servers – RFC 3263
NAPTR records – RFC 2915
ENUM – RFC 3761

9. SIP and DHCP
DHCP service
DHCP RFC 2131
RFC 3361 DHCP option for SIP servers

10. Call Flow Examples - Review how SIP calls are set up for applications like PSTN, instant messaging, VOIP, and more in this technical, in-depth analysis of the protocol.
Call attempt – unsuccessful
Presence subscription
Registration
Presence notification
Instant Message Exchange
Call setup – successful
Call hold
Call transfer
RFC 3515 REFER method
RFC 3725 3rd party call control

11. Call Routing - Discover the power and flexibility of SIP in intelligently routing calls over virtually any network.
Direct call
Proxy routed call
Forking
Loops and spirals
Response path
Creation of via-path
Response merging
Record route
Heterogeneous Error Response Forking Problem (HERFP)
Control models
Third party
Multi-party

12. RTP and RTCP (Real-Time Control Protocol) - Explore how the universal protocol RTP carries voice across an IP network once SIP has established a session. We will also cover RTCP's function of QoS reporting and management.
Dealing Packet Loss, Latency, Jitter
How RTP defines the session
Session Description Protocol
H.245 Terminal Capabilities
The RTP profile
The RTP payload type field
RTP telephony events (RFC 2833)
How RTP removes jitter
How RTP handles packet loss
How RTP identifies the talking party
How RTP handles silence suppression
How RTP handles fixed length packets (padding)
How RTP is used to mix voice (conference calls)
The RTP header
RFC 2833
RTP Control Protocol (RTCP)
SDES
Sender/receiver reports
Bye reports

13. DTMF handling
DTMF Inband
DTMF using SIP INFO
RFC 2833

14.. Presence
• SIMPLE - SIP for Instant Messaging and Presence Leveraging Extensions
• Terminology
• Framework
• Resource List Manipulation Requirements
• Authorization Policy Manipulation
• Acceptance Policy Requirements
• Notification Requirements
• Content Requirements
• General Requirements

15. SIP Timers
T1 , T2, T4
Timer A – K

16. SIP Security - Learn how SIP interoperates with firewalls and other security systems to prevent theft of service while still allowing phone calls.
Authentication
Security for call setup
Digest Authentication
S/MIME
TLS
Privacy and identity

17. NAT Traversal
Firewall traversal
SIP NAT traversal
RTP NAT traversal
SIP Application Level (layer) Gateway (ALG)
Network Address Translation function
Full and restricted cone NATs
Symmetric cone NATs
Simple Traversal of UDP through NATs (STUN) RFC 3489
Traversal Using Relay NAT (TURN)

18. QoS-Related Networking Protocols
Differentiated Services (DiffServ) RFC 2427
Queuing methods FQ, WFQ, LLQ, RR, and FIFO
Service policy

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Call Routing (Multiple Proxies)
Via: routing
Record-Route: and Route: “routing”
100rel (PRACK)
Call routing and registration using the DNS SRV record
Call routing and registration using the DNS NAPTR record
RTP packet interval impact on QoS and Bandwidth
RTP jitter buffer analysis and impact on Quality, (small, large, dynamic, etc.)
RTP relay and the session border controller
Bye message with RTCP call information
SDP - CODEC Negotiation
SDP - RTP session establishment
DTMF - SIP INFO
DTMF - RFC 2833
DTMF – in band
Response 405 (example: X-lite phone, DTMF not supported)
SIP NOTIFY (voice mail indication example)
SIP SUBSCRIBE and NOTIFY (presence)
SIP MESSAGE (Instant Messaging)
Call Forward Immediate
Call Forward No Answer
Call Transfer (REFER)
NAT Traversal (RTP Relay)
NAT Traversal (STUN)
SIP Timers effect on call processing
Call park and retrieve
Conference