The SIP School
Learn and Qualify

The SIP School

Course Objectives

This course will take delegates through the basics of SIP into some very technical areas and is suited to people who will be installing and supporting SIP solutions of all kinds. It is also of value for people who need to have a good understanding to help them sell SIP Solutions and Services along with planning their implementations.

Being a modular training course, delegates can work through all modules or simply choose the module they really need. If delegates are new to SIP, they can start at the center of the circle and work their way outwards

Each module has it's own 'mini' quiz at the end to help delegates 'gauge' how well they are doing and the 'Final' SSCA™ accreditation test is available from the SIP School Login Page

The Modules are as follows with detailed descriptions further in this document

- Core SIP
- SIP Trunking
- SIP-T and the PSTN
- Firewalls, NAT and Session Border Controllers
- SIP Security
- SIP and Voice over IP
- SIP and Unified Communications
- ENUM and DNS
- Testing and Troubleshooting

Total Running time for this course (including time taken to work on all the labs) is approximately 14 ½ hours from the start to finish. This does not include study time for the SSCA™ or the taking of the SSCA™ final test itself.

Core SIP

Module times

- Running time = 63 minutes
- Quizzes = 7 minutes
- Total = 70 minutes

SIP (The Session Initiation Protocol) is described in this module along with the many other Components and Services that will be encountered on a SIP based network

Topics:

SIP - Who benefits?
SIP – The Session Initiation Protocol
SIP 'Official Summary'
Based on HTML
Where does SIP fit in?
SIP Clients and Servers
SIP User Agents
Simple Call Session Setup
SIP System Architecture
The URI - Unique Resource Identifier
SIP Addressing
SIP Addressing Examples
SIP Servers and Operation
Registration
Re-Registration
SIP Proxy servers and why we need them
SIP Server – Proxy Mode
SIP Server – Re-Direct Mode
Proxy Server 'State' types
Location Services
Registration
Re-Registration
DHCP and SIP
SIP Proxy – Trapezoid Model
SIP Server in Proxy Mode
SIP Server in Proxy Redirect Mode
Stateful and Stateless Proxies
Location Server
Location Server – Components
Location Server – Information
Sources
Location Server – Example
SIP Messaging
Request Methods
Response Codes
SIP Headers
INVITE – Example
RESPONSE – Example
SIP Request Methods
SIP Response Codes
SIP Headers
SIP HEADER - INVITE
SIP HEADER - 200 Response
SDP – The Session Description Protocol
SDP in a SIP Message
An SDP Example
Extending SDP
Changing Session Parameters
Call Hold example
Multiple 'm' lines
SDP – The Session Description Protocol
SDP Component in a SIP Message
SDP Example
Extending SDP
Changing Session Parameters
SDP Example - Put a call on Hold
SDP Example - Call Hold Trace
INVITE and reINVITE
SIP Mobility
SIP Call Forking - Parallel
SIP Call Forking - Sequential
Call Forward - No Answer
Call Forward to Voicemail
More Proxy Server details
Headers
Record-Route Defined
Record Route Example
How do we keep track?
Call leg and Call ID
Tag and Branch ID
More on Proxies and SIP Routing
VIA Headers
Record-Route and Route Session Policies
MIME
Multiple MIME parts
SIP and the PSTN
SIP to PSTN Call Flow
SIP to PSTN Detail
SIP Codes and the PSTN
SIP and B2BUA
B2BUA - Back to Back User Agent
B2BUA Example
B2BUA Benefits and Features
Request for Comments
New RFCs
SIPIT
The Call Process

SIP Trunks

Module times

- Running time = 36 minutes
- Quizzes = 7 minutes
- Lab – 'Setting up SIP Trunks' ~ approx 120 minutes
- Total = 163 minutes

This module teaches the theory of connecting a SIP based PBX to the PSTN and it is the foundation of vendor specific **Trunking** modules.

Topics:

A Basic Overview
Benefits of SIP Trunking
SIP Trunking – more depth
SIP Trunking in the Network
SIP Trunk Capabilities
SIP Trunking Network Examples
SIP Peering
Peering problems?
Least Cost routing (LCR)
Disaster Recovery
SIP PBX Requirements
Enterprise PSTN Identities
P-Preferred and P-Asserted
Call Progress Tones
Trunking 'Variations'
Single Site, TDM PBX
Single Site, No 'Forklift'
Single Site, Converged
Converged – SIP/IP PBX
Multiple Site, 'Converged'
Media Gateways
SIP PBX to Non-SIP PBX
SIP PBX to Non-SIP PBX, Call Flow
SIP Trunks Performance
The ADSL issue
Codecs, Voice and Data
Symmetric DSL (SDSL)
Bandwidth Calculator
Testing your link
Configuration
Security and SIP Trunks
SIP Trunk Security - Overview
Session Border Controllers
Setting up a SIP Trunk
Add a VoIP Provider
Provider SIP Servers
Authentication
Stun and the Firewall test
Add a Dialling Rule
Trunk setup complete
Registration Trace
Call out Trace
Next Generation Networks
An Example – British Telecom
Troubleshooting and Interops
SIP Trunks and Common Problems
The SIP Forum
SIPits
SIPit Results
SIP Connect Document.
Choosing an ITSP
ITSP Offerings

SIP-T and the PSTN

Module times

- Running time = 25 minutes
- Quizzes = 7 minutes
- Total = 32 minutes

SIP Networks will of course have to allow connections to and from the PSTN. This module works through SIP and PSTN connectivity

Topics:

SIP to PSTN Overview
SIP to PSTN Call Flow
SIP to PSTN Detail
PSTN to SIP Call Flow
SIP to PSTN Call Failure
SIP to PSTN Call trace
Early Media
Early Media - SIP to PSTN Call
Early Offer / Delayed Offer
Gateways
Default Gateway?
Gateway Location and Routing with TRIP
TRIP Example
SIP-T and PSTN Bridging
SIP-T
SS7, ISDN and SIP
ISUP and SIP Messages
ISDN User Part (ISUP) to SIP Codes
PSTN to PSTN via SIP
ISUP Encapsulation
ISUP Encapsulation / SDP
Addressing Notes
SIP and DTMF
DTMF - Quick Re-Cap
What is DTMF?
DTMF Transport methods
DTMF 'Inband'
RFC 2833 'Trace' example

Firewalls, NAT and Session Border Controllers

Module times

- Running time = 24 minutes
- Quizzes = 7 minutes
- Total = 31 minutes

Inevitably, all IP traffic comes across a Firewall / NAT device and in the case of SIP they can stop the flow of SIP message. This module looks at the problems and the solutions including Session border controllers.

Topics:

Firewalls
What does a Firewall do?
Are Firewalls effective?
What is NAT?
NAT Request
NAT Response
Multiple NATs
The NAT Problem
Types of NAT
NAT – Full Cone
NAT – Restricted Cone
NAT – Port Restricted Cone
NAT – Symmetric
The NAPT or (PAT) Problem
Problems with NAT, Firewalls and SIP
The Solutions
STUN (Simple Traversal of UDP)
STUN (Simple Traversal of UDP)
STUN and rport
Problems with STUN
TURN (Traversal Using Relay NAT)
Interactive Connectivity Establishment (ICE)
How ICE works – Simplified!
More on ICE
Universal Plug and Play (UPnP)
The RTP Problem
The Firewall Problem
Solving the RTP Problem
Symmetric RTP
Media Proxy
Application Level Gateway
SIP Aware Firewalls - Incoming
SIP Aware Firewalls - Outgoing
Session Border Controllers
SBC for the Enterprise
SBC for the ITSP
Enterprise SBC – in Action!

SIP Security

Module times

- Running time = 35 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx 120 minutes
- Total = 162 minutes

SIP **Security** is a complex issue and this modules covers many SIP Security problems along with possible solutions

Topics:

Authentication and Authorization
SIP Proxy Authentication
401 and 407 Authorization
SIP Authorization
PROXY Authentication
SSL with MD5 Cracked !
MD5 v SHA
Encryption
Why Encrypt SIP?
Certificates and HTTPS
Certificate Authorities
Certificate Example
Self-Signed Certificates
Format type
Securing SIP and VoIP
SSL and TLS
SIP and TLS
TLS Thoughts
TLS and SIP in Action
SIPS and SIP Addressing
Secure RTP (SRTP)
Setting SRTP on SIP Devices
Secure RTP (SRTP) - Example
SRTP and SRTCP
Caller Identity
DTLS/SRTP
S/MIME and SIP
MIME and ISUP
SIP Trunking and Security
Enhancing SIP Trunk Security
Alternatives - IPSec, ZRTP
Attacks and Responses
Phishing and SIP exploit
RFC 4475
Try for Yourself
Types of Attack on a VoIP/SIP Network
Responses and Protection
TLS v SSL
Response Identity – A Problem!
Rogue SIP Proxy
More Examples
Try for yourself!
Cain
nmap
NIST Recommendations

SIP and VoIP

Module times

- Running time = 34 minutes
- Quizzes = 7 minutes
- Total = 41 minutes

This module is a refresher module on the basics of **Voice over IP** and also focuses on components that are important to a SIP based Network

Topics:

What is VoIP?
What is Voice over IP?
VoIP – 'A Basic Call'
VoIP and TCP / UDP
VoIP over the Internet
Branch to Branch VoIP
IP PBX
Voice Sampling and Codecs
Encoding
Codecs for Voice
MOS – Mean Opinion scores
The Real Time Protocol (RTP)
Payload Type Identification
Sequence Numbering
Timestamps
Delivery Information
RTP Encapsulation
RTP Header Trace
Real Time Control Protocol
RTCP-XR (Extended Reports)
RTP / RTCP and UDP Ports
Quality of Service
QoS Issues
Measuring Delay
Jitter and Packet Loss
General VoIP Acceptance Criteria
QoS on the Network
802.1Q – VLANs
802.1Q/P Tagging
802.1P - L2 Classification
TOS and DiffServe
Layer 3 Classification
Codecs and Bandwidth
Symmetric DSL (SDSL)
Testing your link
SIP, SDP and VoIP
SIP in the TCP/IP Model
SIP and SDP Messages
SIP and SDP Codec mapping
Where does SIP fit in?
SIP, SDP and VoIP INVITE
Audio and Video in the SDP body

Testing and Troubleshooting

Module times

- Running time = 29 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx 240 minutes
- Total = 276 minutes

Learn how to Monitor and Test SIP devices and services using Wireshark. This tool enables delegates to analyze call control messages to establish where a fault may lie in your SIP infrastructure. Full examples are provided and delegates are encouraged to follow the exercises to try for themselves.

Topics:

Setting up a Test Environment
SIP Phones
Choosing a 'Trial/Test' ITSP
Download a Free Soft Phone
Free ITSP Accounts
Configuring the Softphone
Even more SIP Softphones
The SIP Phones @ The SIP School
Wireshark
Load Wireshark
Network interface setup
Wireshark - Basic Layout
Wireshark Icons
Using Wireshark - Capturing
Using Wireshark – Simple Filters
Using Wireshark – SIP Statistics
Using Wireshark – SIP ladders
Using Wireshark – RTP Statistics
Saving Captures
Where to Capture?
Common Sip Problems
Will it ever work?
What else can you do?
Common SIP/VoIP Problems
Troubleshooting SIP Trunks
4xx — Client Failure Responses
5xx — Server Failure Responses
6xx — Global Failure Responses
More SIP Testing Tools
SIP Scenario
SIP Scan
TestYourVoIP.com
HoverIP
NSLookup
Using the NET to find answers
The SIP Wiki

SIP and Unified Communications

Module times

- Running time = 45 minutes
- Quizzes = 7 minutes
- Total = 52 minutes

SIP and Unified Communications shows you how SIP underpins all the elements of Unified Communications to realize efficiencies that a successful implementation promises to business.

Topics Include

Communication Breakdown
IM Clients
IM Client Features
Enterprise Clients
More in IM Clients
IM and Mobile devices
The Background Stuff
The IMPP working group
IMPP and CPP
More IMPP work
SIMPLE
How it all works
Presentity
A Basic SIP subscription
Multiple Presence States
Presence and P2P
A Presence Network
Getting inside the SIP packets
2 places at one time
Presentity and more!
A Basic SIP Subscription
Multiple Presence States
Presence and P2P
A Presence Network
Get inside the SIP packets
The Packet Structure
PIDF Message Body

XML
Tuples
Example Presence doc with
Tuples (using a Mobile Phone)
Rich Presence
The METHODS in Action
PUBLISH STATE
PUBLISH and PIDF/XML body
SUBSCRIBE METHOD
202 OK Response
NOTIFY
MESSAGE
Add A Buddy/Subscribe
is-composing
Alternative 'Presence States'
2 Places at the same time
Conferencing
What SIP does in Conferencing
INITIATE a conference
JOIN a conference
LEAVE / EXIT a conference
INVITE other participants
REFER conference server to invite
or others to join
EXPEL participants
CONFIGURE the media stream
CONTROL a conference

Why SIP?
Centralized conferencing
Centralized Signalling
Centralized Mixing (optional)
Centralized Authentication
B2BUA (Discussed in core
module)
Conference Components
The Focus
More than one Focus
Conference Setup
iscomposing in Conference
MESSAGE in conference
BYE in conference
Alternative INVITE
SDP BODY OF INVITE
IETF work and Conferencing
XMPP v SIP/SIMPLE
What is XMPP?
SIMPLE and/or XMPP
Gateways
Federations
What is Federation?
Multiple Presence sources
Super-Aggregation
Inter-Domain Federation
RFCs Galore

ENUM and DNS

Module times

- Running time = 20 minutes
- Quizzes = 7 minutes
- Lab – 'Registering / Testing ENUM' ~ approx 20 minutes
- Total = 47 minutes

ENUM (along with **DNS**) is developing into an essential protocol on SIP networks and its purpose is to assist in finding destination SIP devices from a single SIP address.

Topics:

What is E.164?
What is ENUM?
Why ENUM?
Call Routing and ENUM - Example
Why are we using DNS?
DNS and the Web
The e164.arpa Domain
Approved ENUM Delegations
TIERS 0, 1, 2 and 3
TIERS and Registrars
DNS and AOR
e164.arpa Domain in action
Example - ENUM in the UK
Address of Record
Reseaux IP Europeens
PSTN to SIP UA - Example
The ENUM Query
NAPTR Records
DNS Response to an ENUM query
Calls Flows
PSTN to SIP UA – Example (2)
IP to PSTN (Simplified)
Different 'Types' of ENUM
The Problems with 'Public' ENUM
Example – 'Private' ENUM
Example – 'Operator' ENUM
A few providers
SIP User agent and ENUM
Register your number
Testing ENUM
How is ENUM moving forward?
Useful Links