The SIP School Certified Associate – SSCA™

Exam Objectives

The SSCA exam is designed to test your skills and knowledge on the protocol SIP (Session Initiation Protocol). Everything that you need to cover in order to pass this test is covered in the SIP School training modules but if you decide to learn about SIP elsewhere then these are the topics that you should learn about in order to be prepared for the test.

Please note that if you go along an alternate training path it is possible that you may get a question that may not have been covered in that path. It's up to you!

Please view the following pages for the complete topic list….

Core SIP

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 Trunks

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

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

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

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

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

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

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

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