The SIP School

Learn and Qualify

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 'official' training program** 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

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 – 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 Mobility
- SIP Call Forking - Parallel
- SIP Call Forking - Sequential
- How do we keep track?
- Call leg and Call ID
- Tag and Branch ID
- Call Forward - No Answer
- Call Forward to Voicemail

More on Proxies and SIP Routing

- Stateless Proxy
- Stateful Proxy
- More Proxy information
- VIA and Record Route
- VIA Details
- Record-Route Defined
- Record Route Example
- Session Policies

MIME

- MIME
- Multiple MIME parts

SIP and the PSTN

- 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

SIP Summary

- Request for Comments
- New RFCs
- SIPIT
- The Call Process

SIP Trunks

Topics:

SIP Trunks
- What is a SIP Trunk
- Alternative to TDM
- Separate Data and Voice connections
- Converging the network
- SIP Trunks and Codec
- SIP Trunk Benefits

SIP Trunking – In More Depth
- SIP Trunk Capabilities
- SIP Trunking Network Examples
- SIP Peering
- Peering problems?
- Least Cost routing (LCR)
- Disaster Recovery

Some 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 Trunk Performance
- The ADSL issue
- Codecs, Voice and Data
- Symmetric DSL (SDSL)
- Bandwidth Calculator
- Testing your link

Security and SIP Trunking
- 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
- What are NGNs?
- An Example – British Telecom

Troubleshooting and Interops
- SIP Trunks and Common Problems
- The SIP Forum

- SIPits
- SIPit Results
- SIP Connect Document.

Choosing an ITSP

- Understanding ITSP Offerings

SIP-T and the PSTN

Topics:

SIP-T and the PSTN

- 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 and Delayed Offer

- Early Offer / Delayed Offer

Gateways

- Default Gateway?
- Gateway Location and Routing with TRIP
- TRIP Examples

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
- SIP INFO ‘Trace’ example

Firewalls, NAT and Session Border Controllers

Topics:

Overview
- Issues to address

Firewalls
- What does a Firewall do?
- Are Firewalls effective?

NAT or Network Address Translation
- What is NAT?
- NAT Request
- NAT Response
- Multiple NATs
- The NAT Problem

Types of NAT
- 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

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

- Types of Attack on a VoIP/SIP Network
- Responses and Protection
- TLS v SSL
- Response Identity – A Problem!
- Rogue SIP Proxy
- Phishing and SIP exploit
- More Examples RFC 4475
- Try for yourself with recommended software tools

NIST Recommendations

- NIST Recommendations on securing VoIP

SIP and VoIP

Topics:

What is VoIP or Voice over IP?

- 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 Codec

- Encoding
- Codecs for Voice
- MOS – Mean Opinion scores

The Real time Protocol or RTP

- 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

Topics:

Setting up your test environment

- Using SIP IP Phones
- Using SIP Softphones
- SIP Communicator
- Choosing a 'Trial/Test' ITSP
- Get a SIP URI of your own
- Using 'Test Numbers'
- Multiple Setup options for you to try
- Getting Free ITSP Accounts
- Configuring your Softphone
- Even more SIP Softphones
- Example - The SIP Phones @ The SIP School™

Wireshark

- Loading Wireshark
- Network interface setup for capture
- Wireshark - Basic Layout
- Understanding 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?

Interoperability Testing

- Interop Testing
- Why Interop can be tough
- Different interpretations in the RFC 3261
- Interop Test Scenario
- Interop Test operations
- Sample Interop Traces
- Wireshark example videos to help understand interop issues
- SIPIT events

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

Topics:

Communication Breakdown
- Playing Voicemail tag
- Can't find people
- Available but not Available..!
- More Examples of communication problems

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
- 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 Signaling
- 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 SIMPLE

- What is XMPP?
- SIMPLE and/or XMPP
- Gateways

Federations

- What is Federation?
- Multiple Presence sources
- Super-Aggregation
- Inter-Domain Federation

Unified Communications

- What's all the fuss?
- Unified Confusion
- Components involved
- What should UC do?
- 21st Centrury Dial tone
- The Unified inbox
- Unified aware applications
- Find me – Follow me
- Device awareness
- Unified Comms for Business
- Do your Homework
- Humans and UC
- UC in a SIP network

Relevant RFCs

- RFCs Galore

ENUM and DNS

Topics:

ENUM Explained

- What is E.164?
- What is ENUM?
- Why ENUM?
- Call Routing and ENUM - Example

Enum, DNS and Domains

- 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)

Types of ENUM

- Different ‘Types’ of ENUM
- The Problems with ‘Public’ ENUM
- Example – ‘Private’ ENUM
- Example – ‘Operator’ ENUM
- A few providers

Try for yourself

- Register your number
- Testing ENUM

ENUM and the future

- How is ENUM moving forward?
- Useful Links