The SIP School
Learn and Qualify

SSCA® Certification

Become a 'SIP School Certified Associate' endorsed by the Telecommunications Industry Association (TIA)

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 **SSCA® SIP 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.

This list is the same as the 'course topics' list also found under the 'outline' button next to the course name in the Catalog.

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

Module times
- Running time = 74 minutes
- Quizzes = 7 minutes
- Total = 81 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'
- What is an RFC?
- IETF Working groups
- 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 Client Configuration
- Configuration scenarios
- Some basic elements

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
- Call Hold – Old and New Methods
- Music on Hold example
- INVITE and reINVITE

SIP Mobility
- SIP Mobility
- SIP Call Forking - Parallel
- SIP Call Forking - Sequential
- Call legs, dialogs and Call IDs
- Dialog trace example
- Dialogs and Transactions
- Branch Ids
- 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
- Loose and Strict Routing
- 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

Wireshark

Module times
- Running time = 15 minutes
- Quizzes = 1 minutes
- Lab – 'Various' ~ approx 80 minutes
- Total = 96 minutes

- Note: If the student wishes to take more time over the module – this could run to a few hours of learning. It's completely dependent on the students desire to learn Wireshark

This module on Wireshark is an introduction and is intended to get students setup quickly so that they can capture traffic to analyze during the Core module and the rest of the course. More advanced Wireshark training can be found in the Troubleshooting, Testing and Interoperability module of this course.

Topics:

Wireshark
- What is Wireshark?
- Initial Setup
- Free SIP Account options
- Free @thesipschool.com account from OnSIP
- Jitsi client for testing
- Blink client for testing
- X-lite client for testing
- PhonerLite client for testing
- Social Study directory
- SIP test numbers
- Download Wireshark
- Wireshark – Basic Layout
- Wireshark icons
- Using Wireshark – Capturing
- Using Wireshark – Simple Filters
- Using Wireshark – More SIP statistics
- Using Wireshark – RTP Statistics
- Saving Captures
- Over to you!
- What are the codes?
- Link to Troubleshooting module for Advanced Wireshark

SIP, the PSTN and SIP-T

Module times
- Running time = 27 minutes
- Quizzes = 7 minutes
- Total = 34 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-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 and SIP-I
- 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
- RFC 4733 replaces 2833
- RFC 4734
- SIP INFO 'Trace' example

SIP, VoIP and QoS

Module times
- Running time = 42 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx 10 minutes
- Total = 59 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 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
- Signaling paths
- Speech paths
- IP PBX

Voice Sampling and Codec
- Encoding
- Codecs for Voice
- Try the Codec Test
- High Definition (HD) Voice
- Sound tests
- Wideband (HD) codecs
- 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 described
- 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

SIP Security

Module times
- Running time = 40 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx 120 minutes
- Total = 167 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
- sdes and the Crypto attribute
- Crypto attribute example
- Crypto multiple streams
- RFC 4474 for Caller Identity
- 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
- FBI network examples
- 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

Firewalls, NAT and Session Border Controllers

Module times
- Running time = 33 minutes
- Quizzes = 7 minutes
- Total = 40 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:

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
- UDP Hole punching
- Hairpinning
- 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 (Session Traversal Utilities for NAT)
- STUN and rport
- Problems with 'Classic' STUN
- STUN RFC 5389
- TURN (Traversal Using Relays around NAT)
- Interactive Connectivity Establishment (ICE)
- ICE 'In Theory'
- Candidate information and other 'ICE stuff'.
- ICE 'In practice'
- ICE tags
- More on ICE
- ICE Client settings
- More on ICE
- Universal Plug and Play (UPnP)
- Near end NAT
- Far end NAT

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
- Recommended Session Border Controller features
- SBCs in Action!

SIP Trunking

Module times
- Running time = 59 minutes
- Quizzes = 7 minutes
- Lab – 'Setting up SIP Trunks' ~ approx 120 minutes
    - *(If student has access to a SIP server)*
- Total = 186 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:

SIP Trunks
- What is a SIP Trunk
- Alternative to TDM
- Separate Data and Voice connections
- Converging the network
- SIP Trunks and Codecs
- 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
- Disaster Recovery 'Expanded detail'
- Disaster Recovery – Last resort?
- Number Consolidation
- Virtual Presences

Trunking Variations
- Single Site, TDM PBX
- Single Site, No 'Forklift'
- Single Site, Converged
- Converged – SIP/IP PBX
- Multiple Site, 'Converged'
- Multiple Site, 'Converged' + central SBC

Media Gateways
- SIP PBX to Non-SIP PBX
- SIP PBX to Non-SIP PBX, Call Flow

SIP Trunk Performance
- Connection types
- The ADSL issue
- Codecs, Voice and Data
- Symmetric DSL (SDSL)
- Bandwidth Calculator
- Testing your link
- ADSL Developments
- Fibre Options

SIP Trunking and MPLS
- MPLS, basic explanation
- MPLS Label format
- MPLS in a MAC frame
- MPLS example network
- MPLS benefits

Testing, Troubleshooting and Interoperability

Module times
- Running time = 73 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx 240 minutes
- Total = 320 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.

Setting up your test environment
- Using SIP IP Phones
- Using SIP Softphones
- Jitsi, Blink, X-Lite and PhonerLite setup – revisited.
- Even more SIP Softphones
- Jitsi
- Choosing a 'Trial/Test' ITSP
- Getting Free ITSP Accounts
- Configuring your Softphone
- Get a SIP URI of your own
- SIP2SIP accounts
- Configuring Jitsi with a SIP2SIP account
- Using 'Test Numbers'
- Multiple Setup options for you to try
- Configure Blink and Jitsi on the same PC for testing
- 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
- Saving Captures
- Wireshark in more depth!
- SIP Statistics
- RTP / VoIP Capture and Playback
- More 'SIP ladder' analysis
- Coloring rules
- More 'filter expressions'
- More Help on Wireshark if you need it
- You try
- Where to Capture?

Interoperability Testing
- Interop Testing
- Why Interop can be tough
- Different interpretations in the RFC 3261
- BLISS – Basic Interoperability for SIP Services
- Interop Test Scenario
- Interop Test operations
- Sample Interop Traces
- Wireshark example videos to help understand interop issues
- Video call testing
- Video tests with Wireshark trace analysis
- 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 Workbench
- SIP Scan
- TestYourVoIP.com
- HoverIP
- NSLookup
- SIP Center and Voip-info for more tools!
- Using the NET to find answers
- The SIP Wiki

ENUM, DNS and VoIP Peering

Module times
- Running time = 40 minutes
- Quizzes = 7 minutes
- Lab – 'Registering / Testing ENUM' ~ approx 20 minutes
- Total = 67 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:

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
- DNS 'Mirrors'
- 'Finding' Domain name servers using NSLookup
- 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
- 'Finding SIP servers using the tool - DIG
- Calls Flows
- PSTN to SIP UA – Example (2)
- IP to PSTN (Simplified)
- RFC 6140

Types of ENUM
- Different 'Types' of ENUM
- The Problems with 'Public' ENUM
- Example – 'Private' ENUM
- Example – 'Operator' ENUM

VoIP Peering
- Stay 'On-Net
- From ITSP to PSTN and Back…!
- Loss of features with the PSTN
- Peering Profiles and Agreements
- Bi-lateral Peering
- Multi-lateral Peering
- A few providers
- ENUM in North America
- A complete 'infrastructure'

ViPR
- Verification Involving PSTN Reachability (ViPR).
- What is ViPR
- ViPR and P2P
- Initial PSTN Call
- ViPR Call Record
- Query the DHT
- DHT query and Validation
- The Next call is a SIP Call
- ViPR Summary

Try for yourself
- Register your number
- Testing ENUM
- DIG and NAPTR records

ENUM and the future
- How is ENUM moving forward?
- Useful Links

SIP and Fax over IP

Module times
- Running time = 33 minutes
- Quizzes = 7 minutes
- Total = 40 minutes

A lot of companies are now trying to run Fax services across SIP trunks and finding it's not an easy service to get working successfully. This module intends to describe the various flavors of Fax over IP along what should be focused on in order to troubleshoot any issues.

Topics Include

Faxing Basics
- Faxing background
- T.30 Fax signaling
- Associated tones and protocols
- The ITU and TIA standards

Fax over IP
- Fax over IP benefits
- From the old to the new
- Intro to FoIP
- FoIP and SIP trunks
- Protocol conversions

Fax Protocols
- G.711 Pass-through
- T.37 Store and Forward
- T.38 Relay
- Where does SIP fit in?
- UDPTL
- Protocol options for the future

FoIP in action
- SIP in FoIP – Call Flow
- SIP INVITE
- INVITE for T.38
- The INVITE SDP body
- Wireshark FoIP example
- SIP T.38 Call flows – IETF draft document

Bandwidth
- T.38 and G.711 network traffic

Troubleshooting
- The basics
- More complex issues to watch out for

Ongoing Efforts
- RFC 6913 and sip.fax tag
- Use DTMF events instead?

SIP and Unified Communications

Module times
- Running time = 49 minutes
- Quizzes = 7 minutes
- Total = 56 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
- Playing Voicemail tag
- Can't find people
- Available but not Available..!
- More Examples of communication problems

IM Clients
- IM Client Features
- Enterprise Clients
- Enterprise UC client comparisons
- 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 Century 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

UCI Forum
- The UCI Forum - Challenges
- UCI Forum goals
- UCIF website

Relevant RFCs
- RFCs Galore