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

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

- 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
- 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

Topics:

Wireshark

SIP-T and the PSTN

Topics:

SIP, VoIP and QoS

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
- 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 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

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
- 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

Topics:

SIP Trunking

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
- Disaster Recovery 'Expanded detail'
- Disaster Recovery – Last resort?

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

SIP Trunking and MPLS

- MPLS, basic explanation
- Your own VPLS
- but 'Not the only client'
- Separate MPLS networks

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

Some PBX Requirements

- Enterprise PSTN Identities
- P-Preferred and P-Asserted
- Call Progress Tones

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
- SIP Connect 'Missing pieces'
- SIP Connect 1.1

Choosing an ITSP

- Understanding ITSP Offerings

Resource Websites

- TMCnet – Sip trunking
- Siptrunk.org
- No Jitter - Hotzone

Testing, Troubleshooting and Interoperability

Setting up your test environment

- Using SIP IP Phones
- Using SIP Softphones
- Even more SIP Softphones
- SIP Communicator
- Choosing a 'Trial/Test' ITSP
- Getting Free ITSP Accounts
- Configuring your Softphone
- Get a SIP URI of your own
- SIP2SIP accounts
- Configuring SIP Communicator with a SIP2SIP account
- Using 'Test Numbers'
- Multiple Setup options for you to try
- Configure X-Lite and SIP Communicator 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
- 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 Workbench
- SIP Monitoring example app
- SIP Scan
- TestYourVoIP.com
- HoverIP

- NSLookup
- SIP Center and Voip-info for more tools!
- Using the NET to find answers
- The SIP Wiki

ENUM and DNS

Topics:

SIP and Unified Communications

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
- 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