The SIP School™

Overview
The SIP School™ is 'the' place to learn all about the Session Initiation Protocol also known as SIP. There is so much information on the internet about SIP that is both hard to read and poorly presented making it difficult for people to learn about this most important protocol. So The SIP School™ with its lively, clear and fully animated eLearning program has become the only place to enroll to learn about SIP.

Who would benefit?
Everyone…! This training is designed to suit anyone working with SIP such as: Manufacturers of IP PBX and IP Phone equipment, SIP Security equipment manufacturers, SIP Trunk service providers and Carriers, Network Design specialists, Sales and Marketing personnel working with VoIP equipment and services; all of these will benefit from this program.

What's in The SIP School™?
Once you've enrolled with The SIP School™ you'll see nine modules. You can work through the modules in order or simply choose the ones you are most interested in. The modules are listed here but for more detail, please look further into this document.

How long will it take to work through?
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.

Become a SIP School Certified Associate or SSCA®
The SIP School™ training program is accompanied by the SSCA® Certification test. This certification is recognized in the Telecommunications world as the only certification on SIP to strive for. It is endorsed and supported by the TIA (Telecoms Industry Association) along with Bicsi and a rapidly growing number of Manufacturers, Service providers and Carriers.

To prepare for the certification test, each module has it's own 'mini' quiz at the end to help delegates 'gauge' how well they are doing.

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 – 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

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:

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.
- SIP Connect 1.1

Choosing an ITSP

- Understanding 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:

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:

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

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
- Sdes and the Crypto attribute
- Crypto attribute example
- Crypto and Multiple streams
- RFC 4474 for Caller ID
- Caller Identity explained
- 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

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 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'
- HD Voice
- Let's add some Music
- Wideband (HD) Codec example
- 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

Module times

- Running time = 40 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx 240 minutes
- Total = 287 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
- 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

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

- 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

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: