

# The SIP School

Learn and Qualify

## The SSCA® SIP training program

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### 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 from the SSCA® SIP training program?

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 SSCA® SIP training program?

Once you've enrolled with 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.

1. [Core SIP](#)
2. [Wireshark](#)
3. [SIP-T and the PSTN](#)
4. [SIP, VoIP and QoS](#)
5. [SIP Security](#)
6. [Firewalls, NAT and Session Border Controllers](#)
7. [SIP Trunking](#)
8. [Testing, Troubleshooting and Interoperability](#)
9. [ENUM and DNS](#)
10. [SIP and Unified Communications](#)

### How long will it take to work through?

Total Running time for this program (including time taken to work on all the labs) is approximately 17¾ hours from the start to finish though the time will vary based on the student's own experience and of course, how much time they want to spend on the material and if they want to replay some modules.

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®

You can gain access to the test *separately* or with a 'bundle' license – check license 'purchase' options carefully.

The SSCA® certification is recognized in the Telecommunications world as the only certification on SIP to strive for 'Globally'. 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 SIP training module has its own 'mini' quiz at the end to help delegates 'gauge' how well they are doing.

# Core SIP

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## Module times

- Running time = 66 minutes
- Quizzes = 7 minutes
- Total = 73 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
- 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

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## Module times

- Running time = 10 minutes
  - Quizzes = 1 minutes
  - Lab – ‘Various’ ~ approx 80 minutes
  - Total = 91 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?
- Your Initial Setup
- SIP account with Voipuser.org
- X-lite client for testing
- Configure X-Lite
- 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-T and the PSTN

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## 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:

### **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
- RFC 4733 replaces 2833
- SIP INFO 'Trace' example

# SIP, VoIP and QoS

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## Module times

- Running time = 37 minutes
- Quizzes = 7 minutes
- Lab – ‘Various’ ~ approx 10 minutes
- Total = 54 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
- 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

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## Module times

- Running time = 38 minutes
- Quizzes = 7 minutes
- Lab – ‘Various’ ~ approx 120 minutes
- Total = 165 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
- 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

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## Module times

- Running time = 32 minutes
- Quizzes = 7 minutes
- Total = 39 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 NAT 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
- Recommended Session Border Controller features
- SBCs in Action!

# SIP Trunking

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## Module times

- Running time = 40 minutes
- Quizzes = 7 minutes
- Lab – ‘Setting up SIP Trunks’ ~ approx 120 minutes
  - *(If student has access to a SIP server)*
- Total = 167 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
- 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

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## Module times

- Running time = 65 minutes
- Quizzes = 7 minutes
- Lab – ‘Various’ ~ approx 240 minutes
- Total = 312 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
- 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

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## Module times

- Running time = 32 minutes
- Quizzes = 7 minutes
- Lab – ‘Registering / Testing ENUM’ ~ approx 20 minutes
- Total = 59 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
- 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)
- MARTINI

### Types of ENUM

- Different ‘Types’ of ENUM
- The Problems with ‘Public’ ENUM
- Example – ‘Private’ ENUM
- Example – ‘Operator’ ENUM
- Stay ‘On-Net’
- From ITSP to PSTN and Back...!
- Peering Profiles and Agreements
- A few providers

### 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

### ENUM and the future

- How is ENUM moving forward?
- Useful Links

# SIP and Unified Communications

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## Module times

- Running time = 48 minutes
- Quizzes = 7 minutes
- Total = 55 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?
- 21<sup>st</sup> 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