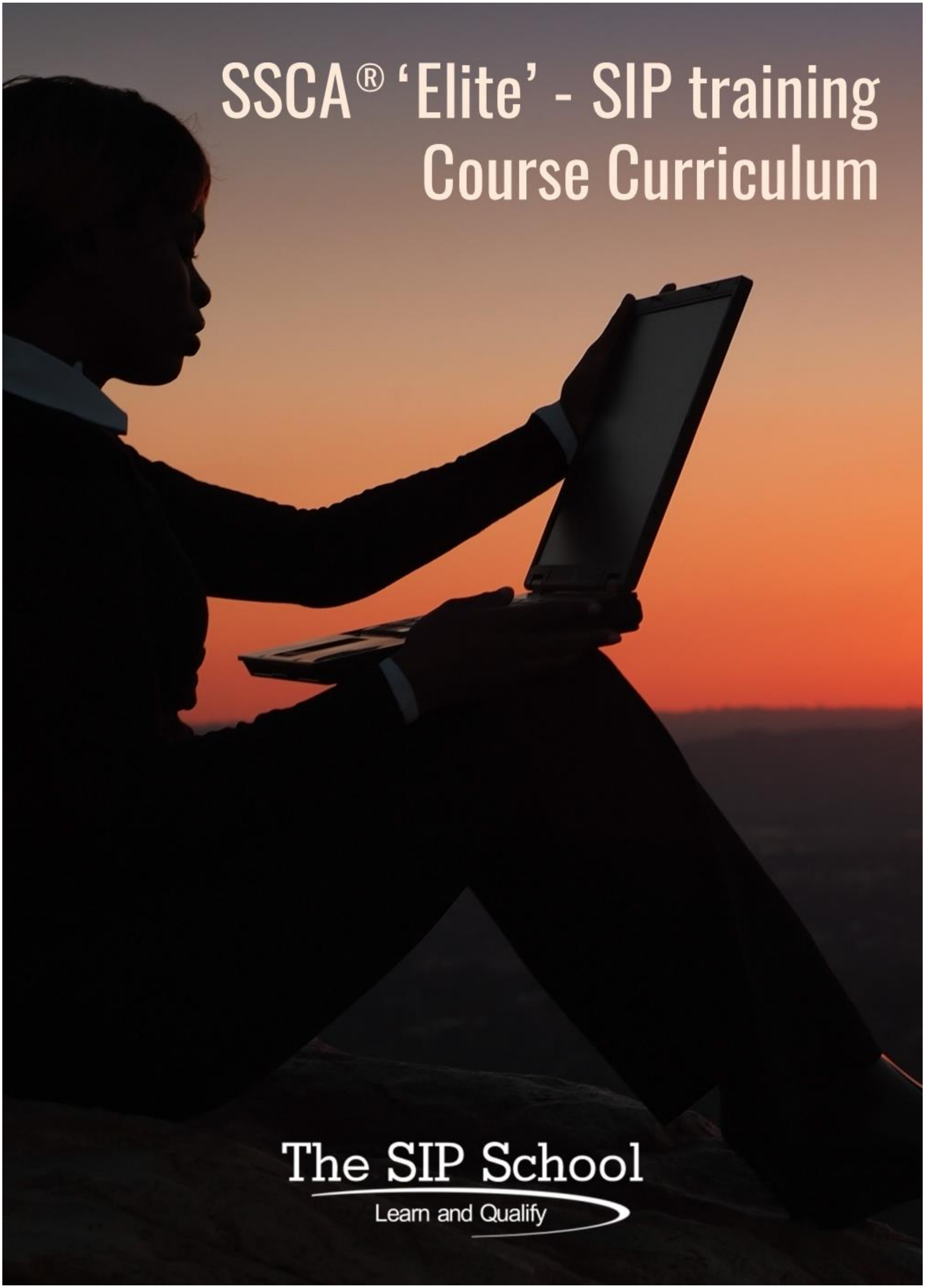
SSCA® 'Elite' - SIP training
Course Curriculum
The SIP School
Learn and Qualify

The SSCA® 'Elite' - SIP training course

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 you need to learn about SIP.

Who would benefit from the SSCA® 'Elite' - 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, Hosted/Cloud service providers, Carriers, Mobile Network Operators, Network Design specialists, Sales and Marketing personnel working with Voice and Video over IP equipment and services; all of these will benefit from this program.

What is in the SSCA® 'Elite' - SIP training program?

Once you have enrolled, you will see 14 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 by clicking on the module names in the list below.

How long will it take to work through?

Running times for this program are **approximate** as 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.

NOTE:

- The time it takes to 'Play' all the slides and Videos (also known as the 'running' time) plus complete all of the quizzes is = **16hours 19mins.**
- The TOTAL time ***will*** be more than this and dependent on factors such as slides being replayed, note taking, working on Labs (some of which can take a few hours), also doing some 'extra' work with the Software tools provided for the labs which we believe is a great idea as it increases student skills.
- Further study time for the SSCA® 'Elite' and the taking of the SSCA® 'Elite' final test itself should also be accounted for.

Is there a Pre-requisite to this program?

This program assumes the student has a 'good' understanding of Data networking technologies along with the 'basics' of Voice and Video over IP. This could be gained through long-term working experience, other certifications such as Cisco's CCENT/CCNA/CCNP, even The SIP School's own 'Networking for VVoIP program' also available via the website. Please check carefully as having the skills required will make the SIP learning experience a more productive one.

Become a (SIP School Certified Associate) SSCA® 'Elite'

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

The **SSCA® 'Elite'** certification is recognized in the Telecommunications world as the only certification on SIP to strive for 'Globally'. It is endorsed and supported by USTelecom, Incompas, the Comms Council UK along with BICSI and an extensive number of Manufacturers, Service providers, Cloud providers, Carriers and Mobile Network Operators

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.

NOTE: An access license for any training course and certification test is for 12 months from the date of purchase.

SSCA® 'Elite'
Core SIP

Description

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

Module times

Running time
86 minutes

Quizzes
10 minutes

Total
96 minutes

1 Core SIP topics

SIP

- **Why SIP?**
- What is SIP?
- SIP 'from the RFC'
- What are 'Requests for Comments' – RFCs?
- More than just 3261
- New RFCs
- IETF Working groups
- Based on HTTP
- Where does SIP fit in?
- SIP Clients and Servers
- SIP User Agents
- SIP Dialog - INVITE
- 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
- Proxy Server 'State' types
- DHCP and SIP
- SIP Proxy – Trapezoid Model
- SIP Server – Proxy Mode
- SIP Server – Re-Direct Mode
- Location Services
- SIP Server in Proxy Mode
- SIP Server in Proxy Redirect Mode
- Stateful and Stateless Proxies
- Location Server
    - Components
    - Information Sources
    - Example

SIP Client Configuration

- Configuration scenarios
- Some basic elements needed to configure a client

SIP Messaging

- Request Methods
- Response Codes
- SIP Headers
- INVITE – Example
- RESPONSE (200 OK) – Example
- More on Headers
- Support and Require Headers
    - Timer (Session Times)
    - 100rel (PRACK)
- Short form 'compact' Headers

SDP - the Session Description Protocol

- SDP – The Session Description Protocol
- SDP in a SIP Message
- An SDP Example
- Extending SDP
- Multiple 'm' lines
- 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 to Voicemail
- Call Forward - No Answer
- Replaces header
- Diversion headers
- History-info

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

SIP and B2BUA

- B2BUA - Back to Back User Agent
- B2BUA Example
- B2BUA Benefits and Features

SIP 'Call Process' Summary

- The Call Process

Description

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

Module times

Running time
44 minutes

Quizzes
1 minute

Labs equate to exercises suggested within the module

Approx. lab time
80 minutes

Total
125 minutes

2 Wireshark topics

Wireshark

- What is Wireshark?
- Initial Setup
- Free SIP Account options
- Free @thesipschool.com SIP account / address
- **Test Numbers**
- Desktop clients
  - Jitsi client for testing
  - Blink client for testing
  - Bria Solo client for testing
  - PhonerLite client for testing
- Mobile clients
  - **Bria Solo for testing**
  - MizuPhone for testing
  - Linphone for testing
  - WeePhone SIP for testing
- **SIP phone in a Browser**
- **SIP Browser clients**
- Free DID and Credit
- Security and SIP in Wireshark
- Social Study directory
- Security and SIP in Wireshark
- Download Wireshark
- Wireshark
  - Introduction
  - Menus, Screens and Views
  - Capturing traffic
  - Profiles
  - Display Filters
  - Capture Filters
  - SIP Packet Analysis
  - SIP ladders and Audio Playback
  - Other Menu options
  - SIP INVITE Analysis
  - Follow a UDP Stream
  - Frame Relationships
  - Colouring Rules
  - RTP Streams
- **Use the Cloud**
- **PCAPs from 'other' places**
- LAB Exercises
- What are the codes?

Description

SIP Networks will of course have to allow connections to and from the PSTN. This module works through SIP and PSTN connectivity

Module times

Running time
25 minutes

Quizzes
7 minutes

Total
32 minutes

3 SIP and the PSTN topics

SIP 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 Codes and the PSTN

Early Media

- Early Media explained
- Early Media - SIP to PSTN Call

Early Offer and Delayed Offer

- Early Offer / Delayed Offer

Gateways

- Default Gateway?
- Gateways and expectations

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?
- Inband vs Out-of-band
- RFC 2833 'Trace' example
- RFC 4733 replaces 2833
- RFC 4734
- SIP INFO 6086
- RFC 2833 'Trace' example
- SIP INFO 'Trace' example

4 SIP, VVoIP and QoS topics

Description

This module starts as a 'refresher' module on the basics of Voice over IP before digging deeper. It then moves on to cover Video over IP and throughout the module there is a big focus on the components and (good) QoS practices that are important to a SIP based Network.

Module times

Running time
82 minutes

Quizzes
7 minutes

Labs equate to exercises suggested within the module

Approx. lab time
10 minutes

Total
99 minutes

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 Codecs

- Encoding
- Codecs for Voice
- Dynamic [RTP Payload type]
- **The 'Codec Test'**
- MOS, R-Factor and High Definition (HD) Voice
- Sound tests
- Codecs and Bandwidth
- Packet Rate / Packets per second
- Variable bit rate / Constant bit rate codecs
- Wideband (HD) codecs
- Opus codec
- Opus audio examples

The Real Time Protocol or RTP

- RTP Intro
- RTP Encapsulation
- RTP Header Trace
- Real Time Control Protocol (RTCP)
- 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 across all Networks
- 802.1Q – VLANs
- 802.1Q/P Tagging
- 802.1P - L2 Classification
- TOS and DiffServ
- Layer 3 Classification
- DSCP with Assured forwarding (AF)
- Bandwidth decisions
- Link options – Symmetric DSL (SDSL)
- Bandwidth (kbps) vs. Packet per Second (pps)
- Network Behavior Analysis
- Issues that can affect QoS
- QoS Summary
- Testing your link

SIP, SDP and VoIP

- SIP in the TCP/IP Model
- SIP and SDP Messages (e.g. Invite and 200OK)
- SIP and SDP Codec mapping

Video over IP

- What is Video over IP?
- Streaming Voice and Video – 1 Way Transmission
- Two-way Conferencing with RTP
- **Codec and Bandwidth Considerations**
- Video bitrate Calculator
- Setting Video Codecs on Devices
- Audio and Video in the SDP body

Assured SIP Services

- Assured SIP intro
- Service Provider Architecture
- Proxy and Access Router functions
- Resource-Priority
- Video 'example'
- Reason Header for Pre-emption Events
- More Proxy details
- Multi-Level Pre-emption and Precedence (MLPP)
- Summary

SSCA® 'Elite'
SIP and Media Security

Description

SIP and Media Security is a complex subject to address yet is imperative for safe communications today. This module covers many Security problems and challenges along with possible solutions.

Module times

Running time
55 minutes

Quizzes
5 minutes

Labs equate to various exercises suggested within the module

Approx. lab time 60 minutes

Total
120 minutes

5 SIP and Media Security

topics

Authentication and Authorization

- SIP Proxy Authentication – in detail
- 401 and 407 Authorization
- SIP Authorization
- PROXY Authentication
- Hashing Algorithms [MD5, SHA etc.]

Encryption

- Why Encrypt SIP?
- Encryption types (Symmetric / Asymmetric)
- Keying and Hashing
- **Certificate Authorities**
- **Certificate Example**
- **The Certificate application process**
- **Installing your new Certificate**
- **Backup your Private key**
- **Self-Signed Certificates**
- **Public Key Infrastructure - PKI**

TLS - Transport Layer Security

- **TLS in Action**
- **TLS 1.2 Capture example**
- **TLS 1.3**
- **SSL/TLS checking**

Securing SIP signaling

- **Securing SIP Signalling and then the media**
- 'SIPS' addressing
- **TLS and SIP in action**
- **Combinations of what you may see…**

Securing the Media Stream

- Secure RTP (SRTP)
- Setting SRTP on SIP Devices
- Secure RTP (SRTP) - Example
- SRTP and SRTCP
- sdes and the Crypto attribute
- Crypto attribute example
- SRTP Call example 'showing' Crypto
- Crypto – multiple streams
- **DTLS/SRTP**
- SRTP with ZRTP
- Encryption summary

SIP trunks and Security

- SIP trunks and Security
- Enhancing SIP Trunk Security

Attacks and Responses

- **Types of Attack on a VoIP/SIP Network**
- FBI network examples
- **Responses and Protection**
- Response Identity – A Problem!
- Rogue SIP Proxy
- Phishing and SIP exploit
- More Examples RFC 4475
- Try for yourself with 'example' software tools

NIST Recommendations

- NIST Recommendations on securing VoIP

3rd party training to extend your knowledge

- **The SANS institute**

SSCA® 'Elite'
STIR/SHAKEN and the 'identity' problem

Description

There is an existing (and growing) issue with illegal robocalls, and spam calls and this module aims to teach the student what is being done initially in North America with an eye on the rest of the world. STIR/SHAKEN is the FCC mandated solution for proving identity on a call and it is happening now.

Module times

Running time
119

Quizzes
10 minutes

Total
129 minutes

6 STIR/SHAKEN and the 'identity problem' topics

STIR/SHAKEN

- Introduction and topics

Who's calling?

- The PSTN Caller ID Spoofing problem
- The 'scale' of the problem (USA)

Caller Identity

- Caller Identity
- Enterprise Identities
- P-Preferred and P-Asserted
- CNAM/eCNAM

Spoofing

- Spoofing a number - Video

STIR/SHAKEN

- Simple Definitions
- Robocalling and more
- Why this is a problem
- A First Step: STIR/SHAKEN
- STIR/SHAKEN in a Nutshell
- What is a PASSporT?
- Haven't I Heard of SIP Identity Already?
- STIR/SHAKEN Architecture
- Signed INVITE Example
- PASSporT Token from Example
- PASSporT Token in JSON
- PASSporT Token Protected Header
- PASSporT Token Payload
- The 'digital signature'
- Fetching Certificate
- Success Call Flow
- Failure Call Flow – Missing Identity Header
- Failure Call Flow – Bad Identity Header
- Certificate management for STIR/SHAKEN
- What's happening in Canada?
- Partner system
- STI Certificate for Authentication
- Attestation
- The SIP School 'test system'
- Verstat
- STIR/SHAKEN in action
- Video - Authentication to Verification
- Service providers with SHAKEN

Enterprises and the 'A'

- The 'Attestation gap'
- How to 'fix' the gap – some options
- Delegate Certificates
- Delegate Certificates base PASSporT
- Delegate Certificates for OTT providers
- Enterprise Certificates
- TN Databases
- Registered Caller

- Distributed Ledger
- Trust
- Getting 'Creative'
- Which option is best?

Rich Call Data

- What is Rich Call Data?
- Rich Call data location
- Adding Rich Call Data
- Rich Call Data in the token
- RCD jCard / rcdi
- RCD and Delegated certs
- RCD PASSporTs
- Will RCD 'reach' all handsets?

International STIR/SHAKEN

- +1 Numbers and Scenarios
- International Attestation
- ATIS and International calls – Bilateral
- ATIS and International calls – Central Registry

Out of Band STIR/SHAKEN

- Why is this a problem?
- Out of Band (OOB) STIR with TDM
- Another OOB example
- How will OOB progress?

Call Diversion

- Diverted call flow
- "div" in a SIP INVITE
- "div-o"

Call Analytics

- An overview

Call Blocking

- Call Blocking

What's happening now

- The Traced Act
- Where are we now?
- 'Other Services and Techniques
- Bringing it all together
- Possible extensions
- FCC mandate
- Robocall mitigation
- The FCC Robocall Mitigation Database
- Find the call originator
- Industry Traceback Group (ITG)

Resources

- 'Some' other companies offering STIR/SHAKEN
- ATIS testbed
- STIR and SHAKEN references
- STIR/SHAKEN conference
- Best practices.

SSCA® 'Elite'
Firewalls, NAT and Session Border Controllers

Description

Inevitably, all IP traffic traverses a Firewall / NAT device and in the case of SIP these devices can stop the flow of SIP messages. This module looks at the problems and the solutions including a focus on Session Border Controllers.

Module times

Running time
64 minutes

Quizzes
10 minutes

Total
74 minutes

7 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
- UDP Hole punching
- NAT Hairpinning
- Media Hairpinning/Tromboning
- Multiple NATs

NAT in more detail

- Types of NAT
- NAT – Full Cone
- NAT – Restricted Cone
- NAT – Port Restricted Cone
- NAT – Symmetric
- New Terminologies
    - Mapping and Filtering
- Endpoint Independent Mapping
- Address Dependent Mapping
- Address and Port Dependent Mapping
- NAT Filtering Rules

The NAT & Firewall 'problem'

- The NAT problem
- The NAPT or (PAT) Problem
- The Firewall Problem

The Solutions

- Interactive Connectivity Establishment (ICE)
- 'Classic STUN' (Session Traversal Utilities for NAT)
- VIA received parameter
- VIA rport parameter
- Problems with 'Classic' STUN
- Symmetric RTP
- **STUN RFC 8489**
- **Request and Response example**
- **TURN (Traversal Using Relays around NAT)**
- ICE 'In Theory'
- Candidate information and other 'ICE stuff'.
- ICE 'In action'
- **ICE tags**
- ICE-Lite and Trickle-ICE
- **ICE Client settings**
- **More on ICE**
- Media Proxy

The Solutions (continued)

- Application Level Gateway
- SIP Aware Firewalls - Incoming
- SIP Aware Firewalls - Outgoing
- Universal Plug and Play (UPnP)
- 'Near end' NAT
- 'Far end' NAT
- GRUU (Globally Routable User Agent)

Session Border Controllers

- SBC for the Enterprise and SBC for the ITSP
- Recommended Session Border Controller features
- SBCs in Action!
- SBCs and message manipulation / normalization
- SIP 'Refer' problems
- SBC 'Interop' example
- SBC Manufacturers – examples
- **SBCs in the Cloud / as a Service**

Description

This module teaches the theory of connecting a SIP based PBX into an ITSPs own network and also focuses on Network technologies, Security, Troubleshooting as well as offering advice on how to select an ITSP for your company or clients.

Module times

Running time
79 minutes

Quizzes
7 minutes

Labs equate to multiple exercises 'suggested' within the module such as SIP PBX and trunk configuration.

Approx. lab time 120 minutes

Total
206 minutes

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

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
- Trunk 'bursting'
- Elastic SIP

SIP Trunks, MPLS and SD-WAN

- MPLS, basic explanation
- MPLS Label format
- MPLS in a MAC frame
- MPLS example network
- MPLS benefits
- Your own private WAN
- but 'Not the only client'
- Separate MPLS networks
- VPLS explained
- WAN Optimization, Hybrids and SD-WAN
- Software Defined WANs explained
    - Orchestrator
    - Policies
    - SD-WAN device capabilities

Setting up a SIP Trunk

- SIP trunk configuration on 'sample' PBX
- Outbound 'Dialling' Rule
- Calling across the trunk
- Call analysis with Wireshark
    - Call Flow
    - SIP ladder

Modes of Operation

- Registration Mode
- Static Mode

Security and SIP Trunks

- **SIP Trunk Security - Overview**

Microsoft (a little)

- Microsoft Teams and Calling plans
- Microsoft Teams and Direct Routing

Troubleshooting and Interops

- SIP Trunks and Common Problems
- The SIP Forum
- SIP Connect
- SIP Connect 1.1 onto 2.0
- Interoperability testing

Choosing an ITSP

- Understanding ITSP Offerings
- 'Sticking points'?
- What you may need in the future
- SIP trunk 'connectivity'
    - Things to watch out for when connecting to your ITSP
- **'Finding' an ITSP**
- SIP trunking Checklist for ITSP evaluation

SSCA® 'Elite'
Testing, troubleshooting and Interop.

Description

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 a SIP infrastructure. Full examples are provided, and delegates are encouraged to follow the exercises to try for themselves.

Module times

Running time
57 minutes

Quizzes
7 minutes

Labs equate to various exercises suggested within the module

Approx. lab time 240 minutes

Total
304 minutes

9 Testing, Troubleshooting and Interoperability — topics

Setting up your test environment

- Your Setup
- Using SIP IP Phones and Softphones
- **Jitsi, Blink, Bria Solo and PhonerLite setup – reminder.**
- Choosing a 'Trial/Test' ITSP
- **Get 'another' SIP account**
- SIP2SIP account
- Configure Blink and Jitsi on the same PC for testing
- **Using 'Test Numbers'**

Wireshark

- Where to 'capture'
- **More options for Packet Capturing**
- Wireshark 'Revisited'
- Colours and the Intelligent Scrollbar
- Packet 'Marking' and 'Comments'
- New Packet Window
- Exporting 'Specified' Frames
- RTP Streams
- TShark (Terminal-based Wireshark)
- PCAP-ng and PCAP formats
- **Alternatives to Wireshark**
- You try!

Interoperability Testing

- Interop Testing and why Interop can be tough
- Different interpretations in the RFC 3261
- Interop Test Scenario
- Interop Test Operations
- Sample Interop Traces with Wireshark
- Wireshark example videos to help understand interop issues
- More Sample captures
- Video call testing
- Video tests with Wireshark trace analysis
- 'Basic' Interop Test List
- SIPIT events

Common SIP problems

- Will it ever work?
- Where can you start checking?
- 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
- Visualware for testing
- HoverIP
- NSLookup
- **Pointers to more tools and SIP Servers**
- **Using the NET to find answers**
- **Using AI to help**
- Other SIP Resources

SSCA® 'Elite'
ENUM, Peering and Interconnect

Description

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. Peering is also discussed as more and more services providers are 'connecting' together to allow a full IP to IP experience. Inclusion of the IP-NNI recommendation builds on 'Peering' to enable ITSPs to 'Peer' in a more effective manner.

Module times

Running time
62 minutes

Quizzes
7 minutes

Total
69 minutes

10 ENUM, Peering and Interconnect — 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 Operation
- DNS Root Server 'Mirrors'
- 'Finding' Domain name servers using NSLookup
- The e164.arpa Domain
- Approved ENUM Delegations (RIPE)
- TIERS 0, 1, 2 and 3
- e164.arpa Domain 'in action'
- ENUM Delegations
- Address of Record
- PSTN to SIP UA – Example
- The ENUM Query
- DNS Response to an ENUM query
- NAPTR and DNS records
- Finding SIP servers using the tool - DIG
- IP to PSTN (Simplified)
- RFC 6140

Types of ENUM

- Different 'Types' of ENUM
- The Problems with 'Public' ENUM
- Example – 'Private' ENUM
- 'Carrier' ENUM and e164enum.net

Peering and Interconnect (for VoIP and Video)

- 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
- Back to ENUM
- A complete 'infrastructure'
- Who's involved?

IP-NNI

- Network-to-Network interface [NNI]
- ATIS and the SIP Forum for NNI
- Benefits of SIP NNI
- History of IP NNI Effort
- Layers of Interconnection
    - IP Interconnection Profile
    - IP Interconnection Routing
- IP NNI Profile
- IP NNI Trust Model

IP-NNI (continued)

- Identities
- Codecs
- DTMF and Fax
- Fault Isolation and Troubleshooting
- QoS
- SIP-Specific Details of IP NNI
- IP Interconnection Routing
- Aggregate Approach
- Per-Telephone Number (TN) Approach
- What's Next for NNI

SSCA® 'Elite'
SIP in the Cloud

Description

SIP is critical to phones and servers involved in a hosted VoIP and Video service. This module aims to show the student what 'the cloud' is all about along with its many different deployment types. Videos will take the student through 'cloud' deployment of an SBC and PBX along with SIP trunk connectivity for a full 'cloud based' VoIP service.

Module times

Running time
64 minutes

Quizzes
7 minutes

Labs in this module are optional and do not count towards course running time. Labs could take from ½ day upwards - all depending on what the student would like to attempt.

Total
71 minutes

11 SIP in the Cloud topics

'Types' of 'Cloud'

- **Public, Private and Hybrid**

Hosted SIP

- What Hosted SIP service is
- Hosted functions and features
- Example Network including 'failover'
- 'Hosted' clients in action
- Why Hosted – Benefits and things to consider
- Why on-site PBX – Benefits and things to consider

The Cloud and 'Anything as a Service'

- **Pizza as a Service**
- **IaaS / PaaS / SaaS**
- **SaaS in 'reality'**
    - **What is Virtualization?**
    - **Virtual Machines**
    - **Emulation**
    - **Virtual Machines (contd.)**
    - **Network Functions Virtualization (and VNF)**
    - **SBCs in the Cloud / as a Service**
- **Virtualization of the PBX**
- **Our own Network examples**
- **Moving to the Cloud**
    - **Example with - AWS / Azure and Twilio**
    - **Call flow in the example 'Cloud based' system**

Video demos of 'Cloud systems'

- **Visualising the migration to the cloud**
- **Cloud marketplaces**
- **Azure – Anyone VM SBC**
- **RDP connection to the SBC**
- **Anynode configuration**
- **SBC and Twilio**
- **AWS instances**
- **AWS and 3CX (PBX)**
- **SBC, PBX and Twilio**
- **Capturing the 'Cloud call'**

Auto Provisioning

- Auto Provisioning Example
- Boot Server
- Client Config
- Client boot sequence
- Client config download
- RFC **6011**
- **Zero-Touch Provisioning**
- **Zero-touch example**
- Benefits of Hosted SIP Service
- Benefits of Onsite PBX and SIP trunks

Troubleshooting

- **Troubleshooting a cloud service**
- **What to look for – Dashboards**
- **What to look for – PCAP files**
- **Monitoring across cloud-based services**

SSCA® 'Elite'
SIP in Cellular networks

Description

SIP is a critical part of VoLTE and VoNR calling across Cellular networks. This module aims to make students aware of SIPs role in all these environments.

Module times

Running time
51 minutes

Quizzes
5 minutes

Total
56 minutes

12 SIP in Cellular networks

topics

SIP in Cellular networks

- Network Overview
- RAN, eNodeB, EPC, IP Core and 3GPP
- **4G, LTE, LTE Advanced LTA-Pro, WiMAX2**
- The RAN and EPC
- Default Bearer Setup
- Introduction to the Servers and Functions in the IMS
    - CSCF
    - S-CSCF
    - P-CSCF
    - I-CSCF
    - Home Subscriber Server HSS
    - Application Server
    - TAS
    - PSCF
    - DNS and ENUM
- Device Registration (with SIP)
- SIP Registration packet example
- SIP in the IMS – Call Flow explained
- Introduction to VoLTE and the threat of OTT services
- Making VoLTE work
    - SIP Preconditions in Action
    - With Codec examples within SDP
- SIP Call flow for VoLTE
- Quality settings 'recap'
- VoLTE media flow
- More on VoLTE
- The IMS
- Layers architecture
    - Application
    - IMS / Session Control
    - Access and Transport
    - 3GPP
- Multiple access devices
    - RCS and OTT
- Who provides IMS solutions?
- IPX and Peering for Security, QoS and SLAs
- GSMA and IR.92
- HD Voice News
- VoLTE media flow
- More on VoLTE
- The IMS
- Layers architecture
    - Application
    - IMS / Session Control
    - Access and Transport
    - 3GPP
- Multiple access devices
    - RCS and OTT
- Who provides IMS solutions?
- IPX and Peering for Security, QoS and SLAs
- GSMA and IR.92

5G

- Benefits of 5G
- 5G service examples
- Voice over 5G
- 5G – NSA Option 3x (and more)
- Mandatory Codecs
- SIP in 5G
- Summarizing the state of 5G
- Resources
- Coverage Checker

Description

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.

Module times

Running time
38 minutes

Quizzes
7 minutes

Total
45 minutes

13 SIP and Fax over IP topics

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?

SSCA® 'Elite'
SIP in UC, UCaaS and CPaaS

Description

SIP in UC, UCaaS and CPaaS shows you how SIP underpins all the elements of Unified Communications services and CPaaS applications to realize efficiencies that a successful implementation promises to business.

Module times

Running time
55 minutes

Quizzes
7 minutes

Total
62 minutes

12 SIP in UC, UCaaS and CPaaS topics

Communication Breakdown

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

IM Clients

- **IM Client Examples and Features**
- **Clients and UC providers**
- **More IM Clients**

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)
- The METHODS in Action
- PUBLISH
- SUBSCRIBE
- NOTIFY
- MESSAGE
- is-composing
- Rich Presence
- 2 Places at the same time

SSCA® 'Elite'
SIP in UC, UCaaS and CPaaS

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
- Creating a Conference
- Creating a Conference: Details
- Adding a participant
- Adding a participant: Details
- Alternative INVITE with REFER
- IETF work and Conferencing

Unified Communications

- What's all the fuss?
- Unified Confusion
- What is Unified Communications?
- From UC to UCaaS
- 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
- **Migrating to UCaaS**
- **UCasS, SIP and the WAN**

CPaaS and APIs

- **Introduction to CPaaS and APIs**
- **What is an API?**
- **Communications API examples**
- **REST APIs and more...**
- **More examples of API use with your PBX**
- **Creating CPaaS applications**
- **Creating an IVR**
- **UCaaS v CPaaS**