The SSCA® SIP training program

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® 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 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's in the SSCA® SIP training program?

Once you've enrolled, you'll see 12 modules (with one optional one). 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, the PSTN and SIP-T
4. SIP, VVoIP and QoS
5. SIP Security
6. Firewalls, NAT and Session Border Controllers
7. SIP Trunking
8. Testing, Troubleshooting and Interoperability
9. ENUM, DNS and Peering
10. SIP in the Cloud, LTE, the IMS and VoLTE
11. SIP and Fax over IP
12. SIP and Unified Communications

    'Recommended', optional module

13. WAN Broadband Assessment for SIP trunking

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.

Times, not including the optional 13th module.

- Running time for the content = **11hours 44mins**
- Running time for the content, WITH 'all' Lab exercises = **21hours 34mins**

This does not include study time for the SSCA® or the taking of the SSCA® final test itself.

Continues on next page….

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 fruitful one.

Become a 'SIP School Certified Associate' or SSCA®

You can gain access to the certification 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), USTelecom, Comptel, the ITSPA along with BICSI and a rapidly growing number of Manufacturers, Service 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.

Core SIP

Module times
- Running time = 84 minutes
- Quizzes = 7 minutes
- Total = 91 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
- 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
- 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
- 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
- Location Server – Components
- Location Server – Information Sources
- Location Server – 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

Wireshark

Module times
- Running time = 20 minutes
- Quizzes = 1 minutes
- Lab – 'Various' ~ approx. 80 minutes
  - *Labs equate to exercises suggested within the module*
- Total = 101 minutes

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.

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

Topics:

Wireshark
- What is Wireshark?
- Initial Setup
- Free SIP Account options
- Free @thesipschool.com SIP account / address
- Desktop clients
  - Jitsi client for testing
  - Blink client for testing
  - X-lite client for testing
  - PhonerLite client for testing
- **Mobile clients**
  - **Media5 for testing**
  - **Linphone for testing**
  - **WeePhone SIP 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
- **Viewing online with Cloudshark**
- Over to you!
- What are the codes?
- Link to Troubleshooting module for Advanced Wireshark

SIP, the PSTN and SIP-T

Module times
- Running time = 26 minutes
- Quizzes = 7 minutes
- Total = 33 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 6086**
- RFC 2833 'Trace' example
- SIP INFO 'Trace' example

SIP, VVoIP and QoS + access to the 'Optional' Wan assessment module

Module times
- Running time = 81 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx. 10 minutes
  - *Labs equate to exercises suggested within the module*
- Total = 98 minutes

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.

Access to a 'LIVE' WAN Broadband assessment module is available also here.

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
- **Opus codec**
- **Opus audio examples**
- **Codec choices and MOS – Mean Opinion scores**
- **Packet Rate / Packets per second**

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 DiffServe
- 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

'OPTIONAL' Module – Refer to WAN Broadband Assessment for SIP trunking

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

SIP Security

Module times
- Running time = 41 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx. 120 minutes
  - *Labs equate to exercises suggested within the module*
- Total = 168 minutes

SIP **Security** and 'privacy' are complex issues and this module 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
- **SRTP Call example 'showing' Crypto**
- **SRTP with ZRTP**
- RFC 4474 for Caller Identity
- Caller Identity
- DTLS/SRTP
- **Ongoing developments for Identity**
- S/MIME and SIP
- MIME and ISUP
- SIP Trunking 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

Firewalls, NAT and Session Border Controllers

Module times
- Running time = 62 minutes
- Quizzes = 7 minutes
- Total = 69 minutes

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.

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
- Hairpinning
- 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
- **BEHAVE**

The Solutions
- STUN (Session Traversal Utilities for NAT)
- STUN and rport
- Problems with 'Classic' STUN
- TURN (Traversal Using Relays around NAT)
- STUN RFC 5389
- Interactive Connectivity Establishment (ICE)
- ICE 'In Theory'
- Candidate information and other 'ICE stuff'.
- ICE 'In practice'
- ICE tags
- **ICE-Lite and Trickle-ICE**
- ICE Client settings
- More on ICE
- Universal Plug and Play (UPnP)
- 'Near end' NAT
- 'Far end' NAT
- **GRUU (Globally Routable User Agent)**

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 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
- **From SIP to WebRTC (and back)**

SIP Trunking

Module times
- Running time = 65 minutes
- Quizzes = 7 minutes
- Lab – 'Setting up SIP Trunks' ~ approx. 120 minutes
  - *Labs equate to exercises suggested within the module and also if student has access to a SIP server.*
- Total = 192 minutes

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.

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

SIP Trunking and MPLS
- 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

Security and SIP Trunking

Testing, Troubleshooting and Interoperability

Module times
- Running time = 75 minutes
- Quizzes = 7 minutes
- Lab – 'Various' ~ approx. 240 minutes
  - *Labs equate to exercises suggested within the module*
- Total = 322 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 a 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.
- Linphone, Media 5 and WeePhone SIP setup - revisited
- Even more SIP Softphones
- Choosing a 'Trial/Test' ITSP
- Get a SIP URI of your own
- SIP2SIP account
- Configure Blink and Jitsi on the same PC for testing
- Using 'Test Numbers'
- 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 – RTP 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?
- **Alternatives to Wireshark**

Interoperability Testing
- Interop Testing
- 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
- SIP Scan
- Visualware for testing
- HoverIP
- NSLookup
- Voip-info for more tools!
- Using the NET to find answers
- Other SIP Resources

ENUM, DNS and Peering

Module times
- Running time = 43 minutes
- Quizzes = 7 minutes
- Lab – 'Registering / Testing ENUM' approx. 20 minutes
  - *Labs equate to exercises suggested within the module*
- Total = 70 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. Peering is also talked about as more and more services providers are 'connecting' to allow a full IP to IP experience.

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
- PSTN to SIP UA – Example (2)
- NAPTR Query – a different view
- 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 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'**

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

SIP in the Cloud, LTE, the IMS and VoLTE

Module times
- Running time = 53 minutes
- Quizzes = 7 minutes
- Total = 60 minutes

SIP is becoming critical to phones and servers involved in a hosted setup. SIP is also critical when used by VoLTE calls in order to make voice sound great on a mobile device. This module aims to make students aware of SIPs role in all of these environments.

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**

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

SIP, LTE, the IMS and VoLTE
- **Network Overview**
- **RAN, eNodeB, EPC, IP Core and 3GPP**
- **4G, LTE, LTE Advanced, 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**

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 = 43 minutes
- Quizzes = 7 minutes
- Total = 50 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

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
- MESSAGE in conference
- BYE in conference
- iscomposing in Conference
- Alternative INVITE with REFER
- IETF work and Conferencing

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

Relevant RFCs
- RFCs Galore

WAN Broadband Assessment for SIP Trunking 'Optional'

Module times
- Running time = 54 minutes
- Quizzes = 7 minutes
- Total = 61 minutes

This is a free 'optional' module that has been provided to allow students of The SIP School to get some 'real' experience of working with a SIP trunk configuration in order to test if it can cope with the demands that is required of it by a business. Although it is 'optional' it really does compliment and re-enforce what has been learned in other modules of this course. It is NOT part of the SSCA® certification test, hence it's 'optional' status.

Topics:

Planning Steps
- Servers and Operation
- Do you have BYOB or OTT?
- Your ISP and ITSP – do they support QoS?
- What is a good time to run assessment tests
- Assess the components and configuration of your WAN
- Set up the WAN assessment network

The Assessment System
- Logging onto the Assessment system
- VoIP Assessment items
  - SIP Call flows
  - Call Metrics
  - Jitter
  - Lost Packets
  - MOS estimates
  - Traffic Flows
  - Reporting
- Starting a test
  - Test types
  - Codec selection
  - Number of trunks
  - Number of calls
  - Assessment period
  - Traceroute test?
  - ToS and QoS requirements
- Analyzing a test
  - Call Metrics
    - Lost Packets
    - Jitter
    - Types of traffic flow i.e. UDP / TCP, Real Time etc.
    - MOS results
- Generating Reports
- Analyzing Reports
  - Analyzing MOS
  - Analyzing Jitter / Delay / Packet Loss
  - Router RTT responses
  - How to spot problems in the network from the report
- Advice on how to configure a WAN network