

A silhouette of a person in profile, sitting and using a laptop. The person is wearing a dark suit jacket and a light-colored shirt. The background is a gradient of orange and red, suggesting a sunset or sunrise over a body of water. The overall mood is professional and focused.

SSCA® SIP training Course curriculum

The SIP School
Learn and Qualify

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/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's in the SSCA® SIP training program?

Once you've enrolled, you'll see 12 modules (with one optional extra). 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.

1. [Core SIP](#)
2. [Wireshark](#)
3. [SIP and the PSTN](#)
4. [SIP, VVoIP and QoS](#)
5. [SIP Security and Identity](#)
6. [Firewalls, NAT and Session Border Controllers](#)
7. [SIP trunking](#)
8. [Testing, Troubleshooting and Interoperability](#)
9. [ENUM, Peering and Interconnect](#)
10. [SIP in the Cloud, LTE, the IMS and VoLTE](#)
11. [SIP and Fax over IP](#)
12. [SIP with Unified Communications](#)

NOTE:

This program was last updated on March 11th 2019. All new / edited sections are shown in a **bold, blue font**.

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: Timings do not include the additional Skype for Business material and labs.

- The time it takes to 'Play' all of the slides and Videos (also known as the 'running' time) plus complete all of the quizzes is = **13hours 31mins.**
- 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® and the taking of the SSCA® 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' 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 USTelecom, Incompas, the ITSPA along with BICSI and an extensive 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.

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
85 minutes

Quizzes
10 minutes

Total
95 minutes

1 Core SIP

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

MIME

- MIME
- Multiple MIME parts

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
41 minutes

Quizzes
1 minute

Labs equate to exercises suggested within the module

Approx. lab time
80 minutes

Total
122 minutes

2 Wireshark

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**
 - **MizuPhone for testing**
 - Linphone for testing
 - WeePhone SIP for testing
- Social Study directory
- **Free DID and Credit**
- SIP test numbers
- 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
- 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
26 minutes

Quizzes
7 minutes

Total
33 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
- Telephony routing over IP (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?
- 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

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
81 minutes

Quizzes
7 minutes

Labs equate to exercises suggested within the module

Approx. lab time
10 minutes

Total
98 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]
- Try 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 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

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

Description

SIP Security and Identity are complex issues and this module covers many Security and Identity problems along with possible solutions.

Module times

Running time
75 minutes

Quizzes
10 minutes

Labs equate to various exercises suggested within the module

Approx. lab time
60 minutes

Total
145 minutes

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SIP Security and Identity

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

SSL and TLS

- Certificate Authorities
- Certificate Example
- Self-Signed Certificates
- TLS in Action

Securing SIP signaling

- Securing SIP Signaling and then the media
- TLS and SIP in Action - recommended
- More on TLS and SIP
- 'SIPS' addressing

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
- SRTP with ZRTP
- Encryption summary

Caller Identity

- RFC 4474 for Caller Identity
- Caller Identity
- DTLS/SRTP
- Ongoing developments for Identity
- Enterprise PSTN Identities
- P-Preferred and P-Asserted
- CNAM
- STIR/SHAKEN
- The PSTN Caller ID Spoofing Problem
- Types of Fraudulent calls
- 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

Caller Identity (continued)

- RFC 4PASSporT Token from Example
- PASSporT Token in JSON
- PASSporT Token Protected Header
- PASSporT Token Payload
- Fetching Certificate
- Success Call Flow
- Failure Call Flow – Missing Identity Header
- Failure Call Flow – Bad Identity Header
- Next steps and references

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**

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
67 minutes

Quizzes
10 minutes

Total
77 minutes

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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 5389
- 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
- Application Level Gateway
- SIP Aware Firewalls - Incoming
- SIP Aware Firewalls - Outgoing
- Universal Plug and Play (UPnP)

The Solutions (continued)

- '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

Virtualization of the SBC

- What is Virtualization?
- Virtual Machines
- Emulation
- Virtual Machines (contd.)
- Network Functions Virtualization (and VNF)
- 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
87 minutes

(not including the 'optional' Skype material)

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
214 minutes

NOTE: Total including the 'optional' Skype material is approx. 274 minutes

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SIP trunking

+ 'optional' Skype for Business access

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 Trunking, 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

SIP Trunking, MPLS and SD-WAN (continued)

- Software Defined WANs explained
 - Orchestrator
 - Policies
 - SD-WAN device capabilities

Modes of Operation

- Registration Mode
- Static Mode

Security and SIP Trunking

- SIP Trunk Security - Overview
- Session Border Controllers

More on SBCs

- The 'corporate' SBC
- SIP REFER issues

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
- Skype for Business and SIP trunks
 - **'Optional' Lab exercises**
 - *Skype for Business 'Network Environment'*
 - *Topology Builder*
 - *Control Panel*
 - *Management Shell and basic commands*
 - *Installing Skype for Business Client*
 - *Making Calls*
 - *Using Wireshark to monitor calls from a Skype network environment to the PSTN across a SIP trunk*
 - **Microsoft Teams**
 - **Microsoft Teams and SIP connectivity**

Some PBX Requirements

- Call Progress Tones

Troubleshooting and Interops

- SIP Trunks and Common Problems
- The SIP Forum
- SIPits
- SIPit Results
- SIP Connect
- SIP Connect 1.1 onto 2.0

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

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
56 minutes

Quizzes
7 minutes

Labs equate to various exercises suggested within the module

Approx. lab time
240 minutes

Total
303 minutes

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Testing, Troubleshooting and Interoperability

topics

Setting up your test environment

- Your Setup
- Using SIP IP Phones and Softphones
- Jitsi, Blink, X-Lite and PhonerLite setup – reminder.
- Linphone, MizuPhone and WeePhone SIP 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
- SIP Scan
- Visualware for testing
- HoverIP
- NSLookup
- Voip-info for more tools!
- Using the NET to find answers
- Other SIP Resources

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
63 minutes

Quizzes
7 minutes

Total
70 minutes

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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

Try for yourself

- Testing ENUM
- DIG and NAPTR records

Description

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.

Module times

Running time
68 minutes

Quizzes
7 minutes

Total
75 minutes

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SIP in the Cloud, LTE, the IMS and VoLTE

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

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

PBX in the Cloud with SIP Trunks

- Cloud and SIP trunk Config overview
- Configuring a SIP trunk on the 'Hosted' PBX (in the Cloud)
- E.164 Outbound routing example
- Calling from Softphone via Cloud PBX
- PCAP for analysis

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'

SIP, LTE, the IMS and VoLTE (continued)

- 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

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
33 minutes

Quizzes
7 minutes

Total
40 minutes

11 SIP and Fax over IP

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?

Description

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.

Module times

Running time
42 minutes

Quizzes
7 minutes

Total
49 minutes

12

SIP and Unified Communications

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

'Presence' Federations

- What is Federation?
- Multiple Presence sources
- Super-Aggregation
- Inter-Domain Federation

Conferencing

- What SIP does in Conferencing
- INITIATE a conference
- JOIN a conference
- LEAVE / EXIT a conference

Conferencing (continued)

- 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
- UCaaS, SIP and the WAN