The SIP School
Learn and Qualify

The SIP School- 'Mitel Style'

Course Objectives

This course will take delegates through the basics of SIP into some very technical areas and is suited to people who will be installing and supporting SIP solutions of all kinds. It is also of value for people who need to have a good understanding to help them sell SIP Solutions and Services along with planning their implementations.

Being a modular training course, delegates can work through all modules or simply choose the module they really need. If delegates are new to SIP, they can start at the center of the circle and work their way outwards

Each module has it's own 'mini' quiz at the end to help delegates 'gauge' how well they are doing and the 'Final' test is available on Mitel Online

The Modules are as follows with detailed descriptions further in this document

- Core SIP
- SIP Trunks
- SIP-T and the PSTN
- Firewalls, NAT and Session Border Controllers
- SIP Security
- SIP and Voice over IP
- SIP and Unified Communications
- ENUM and DNS
- Testing and Troubleshooting
- The Mitel 3300 and SIP
- The Mitel 3300 and SIP Trunking

Core SIP

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 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
INVITE and reINVITE
SIP Mobility
SIP Call Forking - Parallel
SIP Call Forking - Sequential
Call Forward - No Answer
Call Forward to Voicemail
More Proxy Server details
Headers
Record-Route Defined
Record Route Example
How do we keep track?
Call leg and Call ID
Tag and Branch ID
More on Proxies and SIP Routing
VIA Headers
Record-Route and Route Session Policies
MIME
Multiple MIME parts
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
Request for Comments
New RFCs
SIPIT
The Call Process

SIP Trunks

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:

- A Basic Overview
- Benefits of SIP Trunking
- SIP Trunking – more depth
- SIP Trunking in the Network
- SIP Trunk Capabilities
- SIP Trunking Network Examples
- SIP Peering
- Peering problems?
- Least Cost routing (LCR)
- Disaster Recovery
- SIP PBX Requirements
- Enterprise PSTN Identities
- P-Preferred and P-Asserted
- Call Progress Tones
- Trunking 'Variations'
- Single Site, TDM PBX
- Single Site, No 'Forklift'
- Single Site, Converged
- Converged – SIP/IP PBX
- Multiple Site, 'Converged'
- Media Gateways
- SIP PBX to Non-SIP PBX
- SIP PBX to Non-SIP PBX, Call Flow
- SIP Trunks Performance
- The ADSL issue
- Codecs, Voice and Data
- Symmetric DSL (SDSL)
- Bandwidth Calculator
- Testing your link
- Configuration
- Security and SIP Trunks
- SIP Trunk Security - Overview
- Session Border Controllers
- Setting up a SIP Trunk
- Add a VoIP Provider
- Provider SIP Servers
- Authentication
- Stun and the Firewall test
- Add a Dialling Rule
- Trunk setup complete
- Registration Trace
- Call out Trace
- Next Generation Networks
- An Example – British Telecom
- Troubleshooting and Interops
- SIP Trunks and Common Problems
- The SIP Forum
- SIPits
- SIPit Results
- SIP Connect Document.
- Choosing an ITSP
- ITSP Offerings

SIP-T and the PSTN

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

Topics:

- 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 / Delayed Offer
- Gateways
- Default Gateway?
- Gateway Location and Routing with TRIP
- TRIP Example
- 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

Firewalls, NAT and Session Border Controllers

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:

- Firewalls
- What does a Firewall do?
- Are Firewalls effective?
- What is NAT?
- NAT Request
- NAT Response
- Multiple NATs
- The NAT Problem
- Types of NAT
- NAT – Full Cone
- NAT – Restricted Cone
- NAT – Port Restricted Cone
- NAT – Symmetric
- The NAPT or (PAT) Problem
- Problems with NAT, Firewalls and SIP
- The Solutions
- STUN (Simple Traversal of UDP)
- STUN (Simple Traversal of UDP)
- STUN and rport
- Problems with STUN
- TURN (Traversal Using Relay NAT)
- Interactive Connectivity Establishment (ICE)
- How ICE works – Simplified!
- More on ICE
- Universal Plug and Play (UPnP)
- The RTP Problem
- The Firewall Problem
- Solving the RTP Problem
- Symmetric RTP
- Media Proxy
- Application Level Gateway
- SIP Aware Firewalls - Incoming
- SIP Aware Firewalls - Outgoing
- Session Border Controllers
- SBC for the Enterprise
- SBC for the ITSP
- Enterprise SBC – in Action!

SIP Security

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
- 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
- Phishing and SIP exploit
- RFC 4475
- Try for Yourself
- Types of Attack on a VoIP/SIP Network
- Responses and Protection
- TLS v SSL
- Response Identity – A Problem!
- Rogue SIP Proxy
- More Examples
- Try for yourself!
- Cain
- nmap
- NIST Recommendations

SIP and VoIP

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?
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 Codecs
Encoding
Codecs for Voice
MOS – Mean Opinion scores
The Real Time Protocol (RTP)
Payload Type Identification
Sequence Numbering
Timestamps
Delivery Information
RTP Encapsulation
RTP Header Trace
Real Time Control Protocol
RTCP-XR (Extended Reports)
RTP / RTCP and UDP Ports
Quality of Service
QoS Issues
Measuring Delay
Jitter and Packet Loss
General VoIP Acceptance Criteria
QoS on the Network
802.1Q – VLANs
802.1Q/P Tagging
802.1P - L2 Classification
TOS and DiffServe
Layer 3 Classification
Codecs and Bandwidth
Symmetric DSL (SDSL)
Testing your link
SIP, SDP and VoIP
SIP in the TCP/IP Model
SIP and SDP Messages
SIP and SDP Codec mapping
Where does SIP fit in?
SIP, SDP and VoIP INVITE
Audio and Video in the SDP body

Testing and Troubleshooting

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.

Topics:

Setting up a Test Environment
SIP Phones
Choosing a 'Trial/Test' ITSP
Download a Free Soft Phone
Free ITSP Accounts
Configuring the Softphone
Even more SIP Softphones
The SIP Phones @ The SIP School
Wireshark
Load Wireshark
Network interface setup
Wireshark - Basic Layout
Wireshark Icons
Using Wireshark - Capturing
Using Wireshark – Simple Filters
Using Wireshark – SIP Statistics
Using Wireshark – SIP ladders
Using Wireshark – RTP Statistics
Saving Captures
Where to Capture?
Common Sip Problems
Will it ever work?
What else can you do?
Common SIP/VoIP Problems
Troubleshooting SIP Trunks
4xx — Client Failure Responses
5xx — Server Failure Responses
6xx — Global Failure Responses
More SIP Testing Tools
SIP Scenario
SIP Scan
TestYourVoIP.com
HoverIP
NSLookup
Using the NET to find answers
The SIP Wiki

SIP and Unified Communications

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
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
2 places at one time
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 Signalling
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 SIP/SIMPLE
What is XMPP?
SIMPLE and/or XMPP
Gateways
Federations
What is Federation?
Multiple Presence sources
Super-Aggregation
Inter-Domain Federation
RFCs Galore

ENUM and DNS

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:

What is E.164?
What is ENUM?
Why ENUM?
Call Routing and ENUM - Example
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)
Different 'Types' of ENUM
The Problems with 'Public' ENUM
Example – 'Private' ENUM
Example – 'Operator' ENUM
A few providers
SIP User agent and ENUM
Register your number
Testing ENUM
How is ENUM moving forward?
Useful Links

The Mitel 3300 and SIP

In this module delegates will find out about the Mitel 3300's capabilities with regards to SIP. SIP phone connectivity is the main area covered here.

Topics include:

Mitel and SIP
The Mitel 3300 and SIP
Licensing and SIP Devices
SIP Phone Support
SIP Device – Supported Features
The Mitel 5224 as SIP Device
Change to SIP Mode
The 5224 – Booting
Review Network Settings
5224 Web Management
Mitel IP Phone Web Management
Complete the Programming
Feature Configuration
Dial by URL
Authenticate Methods
Network Configuration
Protocols
Media Configuration
Setting SRTP on Mitel IP Phones
SIP Device capabilities form
The Snom 300 as a SIP Device
The Snom and the 3300
Making Calls
The X-Lite Soft Phone as a SIP Device
Configuring X-Lite for the 3300
An X-Lite account on the Mitel 3300
The Mitel 5302
SIP Devices and the 3300
The Vocale Test Network
X-Lite Registering with the Mitel 3300
X-Lite to Snom via the Mitel 3300
3300 as a B2BUA Example
Sample Traces
Wireshark
Troubleshooting SIP Devices
Firmware for Mitel IP Devices
3300 Maintenance commands
Common Problems
Sources of Information
3300 Technical information
5215/5220 SIP Phone Guides
Phone user Guides
For information on SIP Firmware levels for Mitel IP Devices
Mitel Knowledgebase
Firmware for Mitel IP Devices
SIP Maintenance Commands on the Mitel 3300
Lab exercise
Access to a 3300
Setup a 5224
Setup X-Lite [Free]
Setup a Snom [Budget]
Make Calls and examine the traces in Wireshark
Try Call Hold, Transfer and other Phone Features – again looking at the traffic in Wireshark
The Mitel 3300 and SIP

The Mitel 3300 and SIP Trunking

SIP Trunking is a technology that allows for low cost lines to be configured from the 3300 controller to a SIP Trunk provider and out onto the PSTN. Low cost is the driver for this technology and this module delegates are shown how to set up SIP trunks on a Mitel 3300 in easy steps

Topics include:

What is SIP Trunking?
Benefits of SIP Trunking
What are the components?
Request for Comments
The Mitel 3300 and SIP Trunking - The Main components.
Don't Forget Bandwidth!
1 x SIP Trunk per Call
SIP Trunk Programming
Trunking Examples
Trunking to a Test ITSP
The Programming Forms
Licensing
Class of Service
Network Element Assignment
System IP Port Assignment
Trunk Service Assignment
SIP Peer Profile
Digit Modification Assignment
Route Assignment
ARS Digits Dialled
DID for CPN Substitution
Outgoing DID Ranges
Incoming DID and SIP Peer Profile Assignment
What's happening on the Wire
Mitel 3300 to Mitel 3300
3300 to 3300 Peering
3300 to an Asterisk Soft switch
SIP Peer Profile – More Options
SIP Peer Profile - Extra Information
Satellite Offices
Satellite – Incoming Calls
Satellite – Redundancy
Satellite – Fax Support
Manufacturers and Compatibility
SIP Firewalls
What does a Firewall do? - Refresher
Firewalls and SIP Trunking
3rd Party Manufacturers
Outbound Proxies
Outbound Proxy Options
Configuring an Outbound Proxy
Troubleshooting
Sample Wireshark Traces
Wireshark Traces
SIP Trunk Initialization
Alarms
SIP Trunk Maintenance
Commands on the Mitel 3300
Common Trunk Problems
Maintenance Logs
3300 Technical information
Mitel Knowledgebase
3rd Part Manufacturers own Web Sites and Forums