



WebRTC Deployment

- 5 Day Course
- Lecture and Labs (with Instructor led Training)

Learn how to build, deploy and troubleshoot web-rtc services using all open source components. Finish the course with a working system available by cloning the Alta3 “sipgate” github repository which provides a working framework, ready to provide the foundation for production web-rtc applications.

1. NAT Essentials - Understand the technical details of traversing Full cone, ip restricted, port restricted, and symmetrical NATs
 2. STUN - A detailed analysis of the STUN protocol
 3. TURN - A detailed analysis of the TURN protocol
 4. ICE - Learn how ICE manages NAT traversal.
 5. STUN/TURN/ICE labs - configure and deploy coturn.
 6. NGINX - Learn how to configure NGINX as a reverse proxy, SSL edge, and preread forwarder.
 7. websocket essentials - Deploy, manage and troubleshoot websocket based services, especially as deployed in web-rtc.
 8. Kamailio ws (web socket) - Deploy a secure proxy to act as carrier grade websocket to SIP gateway.
 9. RTP Engine - Learn how to manage, troubleshoot and deploy the portion of NAT traversal logic that performs the work of RTP relay.
 10. js.sip - Learn the essentials of js.sip, arguably the most popular open javascript client in common use today.
 - A valid domain: sip.alta3.com
 - NTP config update
 - A SSL certificate: lets-encrypt
 - SIP client: js.sip
 - The web server: NGINX
 - Kamailio Secure Web socket to SIP gateway: WSS to SIP
 - siremix: kamailio DB manager
 - NAT Traversal RTP proxy: rtpengine
 - A turn server for NAT traversal: coturn
 - A SIP target to call: asterisk server
- None required
 - Any company or individual who wants to advance their comprehension of web based real time communication
 - Session Initiation Protocol
 - VoLTE and the IMS
 - 5G Essentials

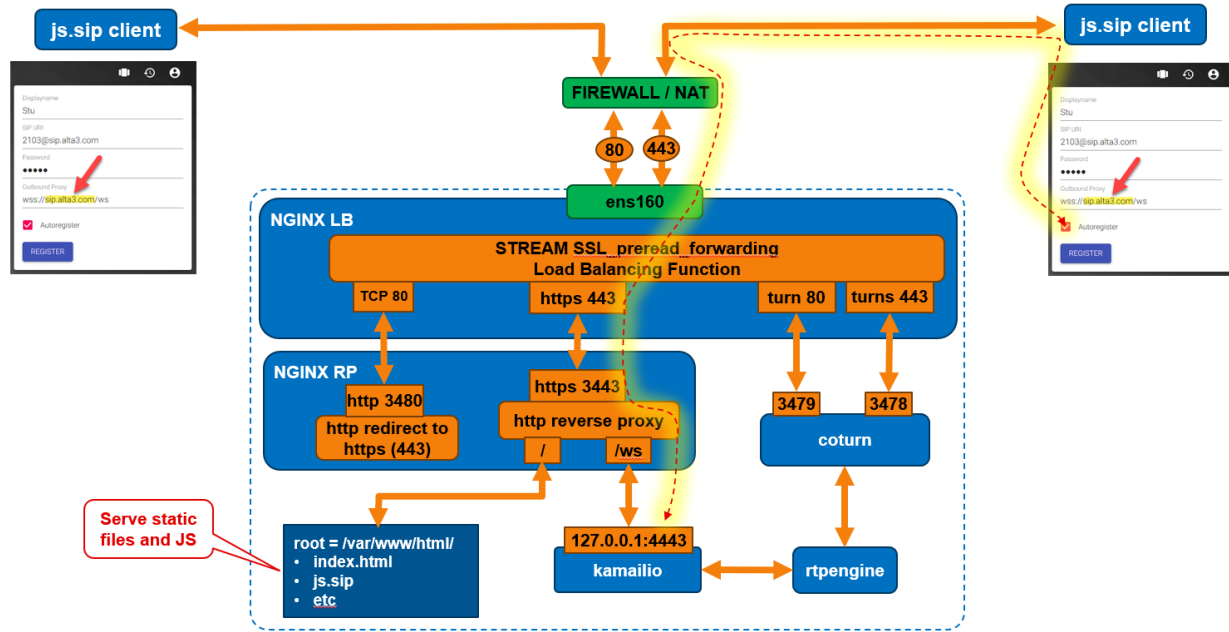


Figure 1: labs diagram

Course Overview

Outline

AI LLM Toolkit

- 📖 Lecture + Lab: Large Language Model toolkit for AI Solution Assistance


Software Control Management

- 📖 Lecture + Lab: SCM Option #1 - GitHub







Installing WebRTC Playground

- 📖 Lecture + Lab: Introducing the WebRTC Playground
- 🗣️ Lecture: RTPEngine
- 📖 Lecture + Lab: Install RTPEngine
- 🗣️ Lecture: Kamailio
- 📖 Lecture + Lab: Install Kamailio
- 🗣️ Lecture: SIP-JS
- 📖 Lecture + Lab: Install SIP-JS
- 📖 Lecture + Lab: Install nginx
- 🗣️ Lecture: STUN/TURN
- 📖 Lecture + Lab: DEMO-Install coturn
- 📖 Lecture + Lab: SIP REGISTER
- 📖 Lecture + Lab: SIP Domains
- 📖 Lecture + Lab: Dial Plan-PDT
- 📖 Lecture + Lab: DialPlan module
- 📖 Lecture + Lab: IP Tables
- 📖 Lecture + Lab: IP Table testing
- 📖 Lecture + Lab: Analyzing websockets





Installing a SIP B2B-UA

-  Lecture + Lab: Install Asterisk



SIP Fundamentals

-  Lecture + Lab: Introduction to VoIP
-  Lecture + Lab: Termshark ##### SIP Registrars
-  Lecture + Lab: SIP Architecture
-  Lecture + Lab: Successful REGISTER by a User Agent
-  Lecture + Lab: REGISTER Fails Auth
-  Lecture + Lab: deREGISTER Log Out



SIP INVITE

-  Lecture + Lab: Regular Expression
-  Lecture + Lab: Routing the INVITE
-  Lecture + Lab: The SIP INVITE
-  Lecture + Lab: SIP INVITE Packet Analysis with Wireshark







Establishing Calls

-  Lecture + Lab: SIP Dialog
-  Lecture: SIP Entities


Call Flows

-  Lecture + Lab: Basic SIP Call Flows
-  Lecture + Lab: SIP 3xx Redirection


SIP Proxies

-  Lecture + Lab: Call Routing
-  Lecture + Lab: INVITE Relay by SIP Proxies
-  Lecture + Lab: No Record Routes
-  Lecture + Lab: SIP URIs
-  Lecture + Lab: CANCELED SIP call
-  Lecture + Lab: Global Failures or 6xx responses



Supporting Systems

-  Lecture + Lab: SIP and the DNS



SIP Headers

-  Lecture + Lab: Common SIP Headers



Session Description Protocol

-  Lecture + Lab: Session Description Protocol
-  Lecture + Lab: Session Description Protocol
-  Lecture + Lab: SDP Video Call Setup
-  Lecture + Lab: SDP Video Call Setup Fails


Real-Time Transport Protocol

-  Lecture + Lab: Real-time Transport Protocol
-  Lecture + Lab: One-Way Media


Dual Tone Multi Frequency

-  Lecture + Lab: Transmitting DTMF
-  Lecture + Lab: Methods for Transport of DTMF

SIP Timers

-  Lecture + Lab: SIP Timers

SIP Security

-  Lecture + Lab: SIP Security

NAT Issues

59108ec54 2024-01-08