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 symetitrical NATs
2. STUN - A detailed analsysis 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 deply 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

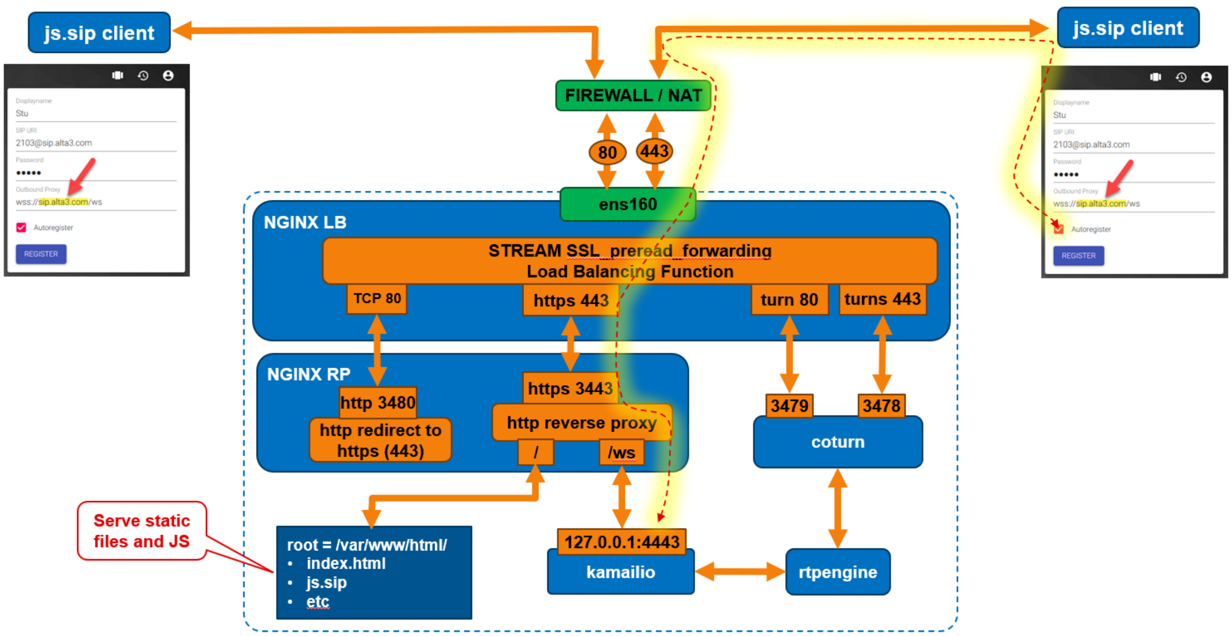

Figure 1: labs diagram

Course Overview

Review this course online at https://www.alta3.com/courses/web-rtc

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