



Nguyen Vu Minh Tuan

VoIP Engineer | Telecom Infrastructure | AI Automation

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About Me

Hi, I'm Tuan — a VoIP engineer, telecom infrastructure builder, and AI automation enthusiast. Over 6+ years I've gone from building PBX systems and SBC solutions to operating one of the largest FreeSWITCH-based voice platforms in the United States — from SIP protocol debugging and C/C++ systems programming to CI/CD pipelines and production fleet management.

At Telnyx, I run the FreeSWITCH-based voice platform that powers millions of calls a day across 18 datacenters. I own the full lifecycle: shipping 200+ releases through phased rollouts, building the CI/CD pipeline from code merge all the way to fleet deployment, setting up 800+ alert rules, and serving as on-call for a fleet handling 100,000+ concurrent channels.

What sets me apart is that I don't just fix problems — I automate them away. I built an AI operations platform purpose-built for FreeSWITCH that handles crash triage, anomaly detection, and deduplication around the clock. It's changed how I work — the routine is handled by AI, so I can focus on the hard problems that need a human.

Every incident becomes a pattern, every pattern becomes automation. That's the mindset I bring to any team — leave the system better than I found it.

I once heard: "Musicians don't retire — they stop when there's no more music in them." I still have a lot of music to offer.

Work Experience

Present — May 2023

TELNYX LLC

FreeSWITCH Operations Engineer

Initiative

- Designed and built Telnyx Internal AI workflow for telephony
- Led Jenkins → GitHub Actions migration across the core telephony platform
- Led Graylog to Grafana/Loki migration for improved log correlation and unified observability
- Created a Python-based out-of-band tester for WebRTC service
- Designed black-box and load test scenarios for call flows including 3PCC, UPDATE handling, codec renegotiation, and SRTP crypto changes
- Evaluated and built audio quality analysis tooling for automated voice quality assessment in customer escalation workflows
- Implemented FreeSWITCH unit testing and regression test suites in CI/CD pipeline with custom dialplan entries to trigger specific call behaviors
- Designed and built Council of Claws — an autonomous AI operations platform
- Migrated the FreeSWITCH-based B2BUA service from containers to Kubernetes pods, serving production customer traffic and enabling faster reaction to traffic changes
- Deployed and trained an on-call bot, a help bot, and a deployment bot to improve my squad's efficiency by assisting with on-call duties, answering help questions, and speeding up deployments

Responsibilities

- Own end-to-end operations of the FreeSWITCH-based B2BUA service — spanning QA, deployment, production operations, troubleshooting, maintenance and decommission
- Support B2BUA integration with upstream platform services, interfacing with routing, media, billing, and QA teams
- Diagnose and resolve latency, scalability, and performance issues across the distributed fleet
- Handle cross-team and customer escalations for B2BUA-related issues
- Build and maintain monitoring and alerting infrastructure to ensure production system health
- Deploy releases and hot-fixes via zero-downtime phased deployment strategy
- Automate deployment pipelines and operational processes to accelerate release cycles and capacity scaling
- Serve in on-call rotation and perform ongoing maintenance of the B2BUA service

Incident Response

- Investigated crash cases across 14+ distinct failure families using GDB backtraces, core dumps, and production traffic pattern reproduction
- Resolved customer escalations with detailed RCA reports, PCAP analysis, and resolution timelines
- Authored incident response procedures and post-mortems with regression test scenarios covering core protocol failure modes

May 2023 — Nov 2021

TEL4VN ICT Training Center

System Team Leader

Responsible:

- Fail-over solution for high availability for FreeSWITCH [▶ Video](#)
- Load-balance solution for scaling concurrent calls on FreeSWITCH [▶ Video](#)
- Use ISO Debian live image to create a custom ISO image containing VoIP System which is documented to install completely offline without a connection to the internet
- Write 2-month plan and personally train interns fresh out of college, or university on SIP, RTP, VoIP Systems, and Linux so they can support all current customers and handle any running project, 3 people have become full-time employees with my guidance
- R&D new VoIP solution
- Successfully using FreeSWITCH to detect both voice and keypress in the same calls with mod_umimrcp on the FreeSWITCH server to detect voice and keypress to detect medical records for a foreign medical company
- Successfully using RTPengine to enable recordings on the SBC side instead of the PBX
- Write lua code to detect events from FreeSWITCH server and setup that lua code in mod_lua to detect specific header in SIP message and process it with custom dialplan, for example, when receiving SIP message 480 with Reason header cause is 600 and text is voicemail this is a voicemail calls
- Research, apply, document, and train DevOps tools for VoIP Engineer, System Admin, 2 people have become proficient in coding Jenkins pipeline and Ansible playbook to handle my tasks
- Code Jenkins for CI and Ansible for deployment, migrate data, backup, restore, load test, and stress test, some of them can be found on my GitHub repositories and some of them even have a video explain how it works [▶ Video](#) [▶ Video](#) [▶ Video](#)

[▶ Video](#)[▶ Video](#)

- All VoIP Engineer and System Admin responsibilities

Nov 2021 — Jun 2019

TEL4VN ICT Training Center

VoIP Engineer, System Admin

Responsible:

- Proactively manage service performance. Provide in-depth analysis of usage trends and make recommendations for modifications to ensure services have sufficient capacity and increase/decrease capacity as needed.
- Operate Zabbix-server as a monitoring server
- Operate Homer as a SIP capture server
- Operate Kamailio as a class 4 Softswitch server with carrier_route module for routing SIP numbers for PBX server
- Operate OpenSIPs as a SBC (Session Border Control) Server with mid_registrar module and RTPengine for re-write SDP for WebRTC SIP
- Write a bash shell script for keeping a recording directory of a tenant on the FreeSWITCH server always below the given size or date but with the most recent date (tenant A keeping 5GB of the most recent recordings, tenant B keeping 10GB of the most recent recordings, tenant C keeping last 3 months of the most recent recordings, ...)
- Optimize and tuning database PostgreSQL and MariaDB or MySQL server to maximize concurrent calls and extensions can be run on a PBX server and Autocall server
- Write SIPP code, using SIPP to load test, and stress test with many scenarios to determine how many concurrent calls and extensions can be run on different specs, for example, 2 cores - 2 GB RAM - can run 800 extensions and 200 concurrent calls, 4 cores - 4 GB RAM can run 800 extensions and 200 concurrent calls, ...
- Planning backup and restore solutions when disaster happens
- Build and deploy VoIP Systems such as FusionPBX, Autocall Server based on Asterisk, and OpenSIPs on the cloud and on-premise server using a list of step-by-step commands
- Understanding how VoIP Systems running behind NAT should handle NAT
- Understanding SIP and RTP
- Works with PSTN provider
- Operate Asterisk as an Autocall server
- Operate FreeSWITCH as a PBX server
- Operate VoIP peripheral

Skills

VoIP & Protocols

- SIP/RTP/SRTP/WebRTC
- STUN/TURN
- Asterisk - FreePBX
- FreeSWITCH - FusionPBX
- GOautodial
- Homer
- Kamailio
- OpenSIPs
- RTPengine

System & Observability

- Consul
- Grafana
- Graylog
- Loki
- OpsGenie
- Prometheus/PromQL
- Vault
- Apache2
- Corosync/Pacemaker
- HAProxy
- Linux
- MariaDB
- Nginx
- PostgreSQL
- Zabbix

DevOps & CI/CD

- GitHub Actions
- Jinja2
- Ansible
- Docker
- Git/GitHub
- Jenkins
- Kubernetes
- Terraform
- Vagrant

Programming

- C/C++
- Bash
- Lua
- Python

Debugging & Testing

- Audio Quality Tools
- GDB/Core Dumps
- PCAP Analysis
- SRTP Decryption
- sngrep/tcpdump
- SIPp Load Testing
- Wireshark/tshark

AI & Automation

- Context Engineering
- OpenClaw
- Telnyx Internal AI

Education

Engineer of Telecommunications

Sai Gon University, Viet Nam
2016 — 2020

Courses & Certificates

Google Cybersecurity

Feb 2026
GOOGLE

IBM Data Engineering

Jan 2026
IBM

Computer Systems and Architecture

Nov 2025
COURSERA

All of AI: ChatGPT, Midjourney, Stable Diffusion & App Dev

May 2025
UDEMY

ASL | First 500+ Basic Signs | American Sign Language

Dec 2024
UDEMY

The C++20 Masterclass: From Fundamentals to Advanced

Apr 2024
UDEMY

AWS Certified Solutions Architect – Associate

Jan 2023
AWS

Google IT Automation with Python

Sep 2022
GOOGLE

Linux Engineer (LPIC-2)

Aug 2022
LPI

Terraform Foundation

Jun 2022
TEL4VN

Kubernetes Administrator

Sep 2021
TEL4VN

Asterisk Certified Essentials

Aug 2021
SANGOMA

**Certified Linux
Administrator (LPIC-1)**

Mar 2021

LPI

Ansible Automation

Jan 2021

TEL4VN

Jenkins Fundamentals

Sep 2020

TEL4VN

LPIC-2: Linux Engineer

May 2020

TEL4VN

**Docker Certified
Associate**

Aug 2019

TEL4VN

**LPIC-1: System
Administrator**

Jun 2019

TEL4VN

**895 TOEIC Listening and
Reading**

Nov 2018

ETS

**VoIP Open-source Carrier
Certificate**

Jul 2018

TEL4VN

**VoIP Open-source
Administrator Certificate**

May 2018

TEL4VN

Languages & Hobbies

Languages

Vietnamese

Native

English

Fluent

Hobbies

- Movies & series
- Music
- Languages

Get in Touch

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