

## PERSONAL INFORMATION

## David Villasmil Govea

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Sex Male | Date of birth 21/09/1971

## WORK EXPERIENCE

2017 - Present

Business or sector Telecommunications

### Senior VoIP Engineer

SingleComm, LLC./Richmond, VA, United States (Remote/Subcontracted).

- Designed and implemented the complete VoIP infrastructure upgrade from legacy architecture to up-to-date redundant and fault-tolerant architecture based on AWS cloud utilizing Kamailio, freeSWITCH and OpenSIPS.
- Designed and implemented the monitoring system for the VoIP infrastructure based on AWS Cloud.
- Designed and implemented an automated testing suite, as well as Stree-teting suite.
- Deployed Homer (<http://sipcapture.org/>) tracing and monitoring.
- Designed and implemented VoIP-related QoS and KPIs.
- Automated UI testing based on Selenium.
- VoIP Server/Client troubleshooting.
- Implementation and deployment based on AWS CloudFront.

2015 – 2017

Business or sector Telecommunications

### Senior VoIP Engineer

Orange France/Libon, Paris, France (Remote/Subcontracted).

- Designed and implemented a Test Platform to validate automatically all voice components, as well as stress-testing.
- General voice platform maintenance based on Kamailio and freeSWITCH
- Voice platform troubleshooting
- Development and Implementation of alarm system for VoIP errors received from PSTN providers using Homer sipcapture
- Developed a freeSWITCH module for mass-calling campaigns for stress-test and/or marketing
- Agile methodology

2008 - 2015

Business or sector Telecommunications

### Implementations Manager / QA Monitoring

ELEPHANT TALK COMMUNICATIONS, Spain. (Now Pareteum, <https://www.pareteum.com/>)

- General network operations
- MVNO Implementation
- Project management
- Incidence management
- New product design, development and implementation
- Integration with Mobile Network Operators
- Development and implementation of GSM/GRPS/UMTS services such as:
  - Voice services
  - Value added services
  - Data connectivity
  - Video streaming integration
  - Network elements management, i.e.
  - HLR/SMSC/MMSC/MNP
- Designed and implemented a C7/SS7 passive probe and early alerting system. This is based on Wireshark/tshark, sniffing signalling links, dissecting and soring all network signaling. The dissecting is forward to a proprietary application developed by me in C++ which in turn associates all signaling packets into on transaction and stores it in MySQL and MongoDB. This is implemented in 4 countries.
- Design, development and implementation of monitoring and QA system that raises early-warnings and alarms based on threshold.
- Design and implementation of KPI Reporting system for the MVNE voice, SMS and Data services.

- 2006 - 2008 **Business or sector** Telecommunications  
**General Manager/CEO**  
ICETECH, S.L. Madrid, Spain.
- Telecommunications consulting services. Founder.
  - Creating VoIP network for residential customers.
  - Routing services.
  - VoIP planning services.
  - Provisioning services.
- 2002 - 2006 **Business or sector** Telecommunications  
**General Manager/CEO**  
Interactive Communications Europe, S.L., Madrid, Spain
- Co-founder
  - Company with TELES and EXCEL (Lucent/AT&T) as its infrastructure providing services for call shops and prepaid card services.
  - Design and development of real-time proprietary software for monitoring switching equipment ASR, statistics and billing.
  - Management of technical team.
  - Trouble Ticket system's management.
  - Business strategy planning.
  - Personnel under my supervision: 3.
- 2001- 2002 **Business or sector** Telecommunications  
**Implementation Manager / VoIP**  
YIP Telecom, Brazil
- Design and implementation of VoIP services for residential and business subscribers using OpenSIPs and RTPProxy as core and Asterisk as the backend for billing, value-added services.
  - Design and implantation of the billing system.
  - Design and implementation of monitoring infrastructure.
  - 250.000 Subscribers base
  - VoIP Infrastructure implementation
  - Job systems automation
- 2001 - 2002 **Business or sector** Telecommunications  
**Operations Manager**  
Creative Communications Engineering, S.L. Madrid, Spain
- Co-founder
  - Company with TELES and EXCEL (Lucent/AT&T) as its infrastructure providing services for call shops and prepaid card services.
  - Design and development of real-time proprietary software for monitoring switching equipment ASR, statistics and billing.
  - VoIP Infrastructure implementation
  - Job systems automation
  - Management of technical team.
  - Trouble Ticket system's management.
  - Personnel under my supervision: 2.

## EDUCATION AND TRAINING

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- 2016 **Kamailio Advanced Training**
- 2003 - 2004 **Executive Management of Telecommunications Companies.**  
Instituto de Empresa, IE Business School
- Executive Education, MBA. Strategy, management y planning.
- 1989 - 1993 **Computer Science Degree**  
Universidad Privada Rafael Bellosillo Chacín, Maracaibo, Venezuela

## PERSONAL SKILLS

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Mother tongue(s) Spanish

Other language(s)	UNDERSTANDING		SPEAKING		WRITING
	Listening	Reading	Spoken interaction	Spoken production	
English	Bilingual	Bilingual	Bilingual	Bilingual	Bilingual

**Communication skills** ▪ Good communication skills gained through my experience as Project Manager and in creating the companies I have founded in the past.

**Organisational / managerial skills** ▪ Leadership (I have had up to 4 people under my management)

**Job-related skills**

**Operating Systems:**

- Linux/Unix, Mac, Microsoft Windows

**Databases:**

- Microsoft SQL Server, MySQL, PostgreSQL, DBF, MongoDB

**Scripting:**

- Perl, Python, Bash, Lua, AWK, VBScript, Javascript; Python

**Programming Languages:**

- C/C++, Java, HTML, XML Microsoft Visual Basic

**Protocols and Switching Equipment:**

- Excel Switching (Lucent Excel/AT&T/Cantata) Programmable switch
- TELES AG, AS50/AS500/iXESS
- OpenSER/Kamailio/OpenSIPs
- Asterisk
- FreeSWITCH
- GSM Gateways (Topex/Teles)
- VoIP/SIP/H323/IAX/VoFR
- SS7/C7, EuroSDN, SCCP, TCAP, GSM, CAMEL, SIGTRAN and others

**AWS SysOps capabilities.**

**Other skills** ▪ Very fast learner, love challenges.

## Publications, Contributions

▪ **FreeSWITCH**

FreeSWITCH is a free and open source application server for real-time communication, WebRTC, telecommunications, video and Voice over Internet Protocol. Multiplatform, it runs on Linux, Windows, MacOS and FreeBSD.

- Console filter, rework to allow for SIP Messages to be filtered by IP Address.  
(<https://freeswitch.org/stash/users/davidcsi/repos/freeswitch/compare/commits?sourceBranch=refs%2Fheads%2Ffeature%2Fsofia-filter&targetBranch=refs%2Fheads%2Fmaster>)
- Dialer Module, New module to add auto-dialer feature to the project.  
([https://freeswitch.org/stash/users/davidcsi/repos/freeswitch/compare/commits?sourceBranch=refs%2Fheads%2Ffeature%2Fmod\\_dialer&targetBranch=refs%2Fheads%2Fmaster](https://freeswitch.org/stash/users/davidcsi/repos/freeswitch/compare/commits?sourceBranch=refs%2Fheads%2Ffeature%2Fmod_dialer&targetBranch=refs%2Fheads%2Fmaster))
- Billing Platform (<https://github.com/davidcsi/FreeSWITCH-Billing>)

▪ **Homer Sipcapture**

HOMER is part of the SIPCAPTURE stack: A robust, carrier-grade and modular VoIP and RTC Capture Framework for Analysis and Monitoring with native support for all major OSS Voice platforms and vendor-agnostic Capture agents. HOMER counts thousands of deployments worldwide including notorious industry vendors, voice network operators and fortune 500 enterprises, providing advanced search, end-to-end analysis and packet drill-down capabilities for ITSPs, VoIP Providers and Trunk Suppliers using and relying on VoIP services and RTC technologies - All 100% Open-Source

- Add ASR/ACD calculation per-component (<https://github.com/sipcapture/homer-docker/pull/44/commits/867436c0cc09bba873d68c6386f82f5e529b642b>)

▪ **Kamailio**

Kamailio (successor of former OpenSER and SER) is an Open Source SIP Server released under GPL, able to handle thousands of call setups per second. Kamailio can be used to build large platforms for VoIP and realtime communications – presence, WebRTC, Instant messaging and other applications. Moreover, it can be easily used for scaling up SIP-to-PSTN gateways, PBX systems or media servers like Asterisk™, FreeSWITCH™ or SEMS.

- Added pseudo-variables to sipcapture module (<https://github.com/kamailio/kamailio/pull/995>)