

Building Telephony Systems with OpenSIPS 1.6

Build scalable and robust telephony systems using SIP

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About the Author

Flavio E. Goncalves was born in 1966 in Brazil. Having always had a strong interest in computers, he got his first personal computer in 1983 and since then it has been almost an addiction. He received his degree in Engineering in 1989 with a focus on computer-aided design and computer-aided manufacturing.

He is also the CEO of V.Office Networks in Brazil – a consulting company dedicated to the areas of Networks, Security, and Telecommunications and a training center since its foundation in 1996. Since 1993, he has participated in a series of certification programs and been certificated as Novell MCNE/MCNI, Microsoft MCSE/MCT, Cisco CCSP/CCNP/CCDP, Asterisk dCAP, and some others.

He started writing about open source software because he thinks that the way certification programs were organized in the past was very good for helping learners. Some books today are written by strictly technical people who, sometimes, do not have a clear idea of how people learn. He tried to use his 15 years of experience as an instructor to help people learn about the open source telephony software. His experience with networks, protocol analyzers, and IP telephony combined with his teaching experience give him an edge to write this book. This is the third book written by him; the first one was "*Configuration Guide for Asterisk PBX*", *BookSurge Publishing*.

As the CEO of V.Office, Flavio E. Goncalves balances his time between family, work, and fun. He is a father of two children and lives in Florianopolis, Brazil, one of the most beautiful places in the world. He dedicates his free time to water sports such as surfing and sailing.

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Writing this book has been a process that involved many people. I would like to thank the staff at Packt Publishing who worked in all the processes of reviewing and editing the book. I would like to thank Bogdan Andrei Iancu for the countless tips on OpenSIPS and the book itself and Adrian Georgescu for his contribution for CDRTool and Media Proxy. I would also like to thank several students, who took courses in the OpenSIPS Bootcamp for their feedback. Finally, I would like to thank my family for all the support they gave me during all these years.

About the Reviewers

Bogdan-Andrei Iancu entered the SIP world in 2001, right after graduating in Computer Science from the "Politechnica" University of Bucharest, Romania. He started in the early days of SIP as a researcher at the Fokus Fraunhofer Institute, Berlin, Germany. For almost four years, Bogdan-Andrei Iancu accumulated a quick understanding and experience of VoIP/SIP, being involved in research and industry project and following tight the evolution of the VoIP world.

In 2005, Bogdan-Andrei Iancu started his own company Voice System. The company entered the open source software market by launching the OpenSER/OpenSIPS project – a free GPL-SIP proxy implementation. As CEO of Voice System, Bogdan-Andrei Iancu pushes the company in two directions: developing and supporting the OpenSIPS public project (Voice System being the major contributor and sponsor of the project), and creating professional solutions and platforms (OpenSIPS based) for the industry. In other words, Bogdan's interest was to create knowledge (by the work with the project) and to provide the knowledge where needed (embedded in commercial products or in raw format as consultancy service).

In the effort of sharing the knowledge of the SIP/OpenSIPS project, together with Flavio E. Goncalves, the author of this book, he started to run OpenSIPS Bootcamp since 2008, an intensive training dedicated to people who want to learn and get hands-on experience on OpenSIPS from the most experienced people.

Bogdan-Andrei Iancu's main concern is to research and develop new technologies or new software for SIP-based VoIP (actually, this is the reason for his strong involvement with the OpenSIPS project), and to pack all these cutting-edge technologies as professional solutions to the industry.

SIP and OpenSIPS became a key factor in the VoIP world along the year – telephony providers, telcos, carrier grades started to adopt and use OpenSIPS as the core component of their VoIP network, because of its stability, performance, and security, but most importantly, because of its reliability as a project.

Justin Thomas Zimmer has worked in the contact-center technology field for twelve years. During that time, he has performed extensive software and computer telephony integrations using both PSTN and IP telephony. His current projects include system designs utilizing open source soft switches over more traditional proprietary hardware-based telephony and the integration of these technologies into market-specific CRM products.

As the Technical Partner of Unicore Technologies out of Phoenix, Arizona, Justin is developing custom business solutions utilizing open source software. Unicore's solutions present businesses with low startup costs in a turbulent economy.

He has worked on *The Hopewell Blogs* – a science fiction adventure novel that will be released online chapter-by-chapter, and available in print once the final chapter has been released.

I'd like to thank the countless community contributors who have provided enough online documentation to make this book as accurate and helpful as possible. And I'd like to thank my wife Nicole for putting up with the extra hours spent reviewing this book, as well as my boys Micah, Caden, and daughter Keira for giving up some of their daddy-time for this project.

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