VoIP and SIP Trunking Architecture and Maintenance Training Course



20 - 24 Dec 2026

Manama

4400 € (Per Person)

Ref: #NO9854_231048

Course Introduction / Overview:

This course provides a comprehensive exploration of Voice over IP (VoIP) and Session Initiation Protocol (SIP) Trunking, from fundamental principles to advanced architectural design and maintenance strategies. In today's digitally-driven world, migrating from traditional telephony to IP-based systems is essential for organizational agility, scalability, and cost-efficiency. This program is meticulously designed to equip participants with the practical skills needed to design, implement, and manage robust VoIP and SIP trunking solutions. We will delve into the core protocols, including SIP, RTP, and RTCP, and explore the critical role of Session Border Controllers (SBCs) in modern networks. Drawing on foundational concepts established by pioneers like Henning Schulzrinne, a key contributor to the development of SIP, the course bridges theory with real-world application. Participants will gain insights aligned with industry-leading literature, such as "Internet Telephony: The Complete Guide to VoIP" by David Collins, ensuring their knowledge is both current and deeply rooted in established principles. BIG BEN Training Center offers this immersive experience to empower professionals to master the complexities of unified communications and confidently navigate the evolving landscape of enterprise voice technology.

Target Audience / This training course is suitable for:

- Network Engineers and Administrators.
- Telecommunications Specialists.
- IT Managers and Directors.
- Voice and Unified Communications Engineers.
- Systems Integrators and Consultants.
- Solutions Architects.
- Technical Support Professionals involved with voice networks.

Target Sectors and Industries:

- Telecommunications and Internet Service Providers.
- Information Technology and Services.
- Financial Services and Banking.
- Healthcare and Hospital Administration.
- Education Sector, including universities and schools.
- Retail and E-commerce.
- Government and Public Sector Agencies.
- Hospitality and Customer Service Centers.

Target Organizations Departments:

- Information Technology (IT) Department.
- Network Operations Center (NOC).
- Telecommunications Department.
- Unified Communications (UC) Team.
- Infrastructure and Operations.
- Information Security (InfoSec).
- Technical Support and Help Desk.

Course Offerings:

By the end of this course, the participants will have able to:

- Design scalable and resilient VoIP network architectures.
- Configure and deploy SIP trunks to connect with service providers.
- Implement Quality of Service (QoS) policies to ensure high-quality voice communication.
- Secure VoIP infrastructure against common threats like toll fraud and eavesdropping.
- Troubleshoot complex call flow, registration, and audio quality issues.
- Analyze SIP and RTP traffic using network protocol analyzers.
- Manage and maintain Session Border Controllers (SBCs) effectively.
- Develop comprehensive dial plans and call routing strategies.
- Understand the interoperability challenges between different vendor systems.
- Evaluate and select appropriate codecs for different network conditions.

Course Methodology:

The training methodology at BIG BEN Training Center is designed to be highly interactive, practical, and engaging, ensuring that participants not only learn the theory but can also apply it effectively in their professional roles. Our approach moves beyond traditional lectures to create an immersive learning environment. The course combines expert-led instruction with hands-on lab sessions where participants will configure routers, switches, and IP PBX systems in a simulated enterprise environment. Real-world case studies will be analyzed to understand common challenges and successful implementation strategies in VoIP and SIP trunking projects. Collaborative group discussions and problem-solving activities encourage participants to share experiences and learn from one another. Instructors facilitate these sessions, providing personalized feedback and guidance. A significant portion of the course is dedicated to practical troubleshooting exercises, allowing participants to diagnose and resolve realistic network and voice quality issues. This blend of theoretical knowledge, practical application, and peer-to-peer interaction ensures a comprehensive and lasting learning outcome.

Course Agenda (Course Units):

Unit One: Fundamentals of VoIP and SIP Protocols

- Introduction to Voice over IP (VoIP) technology.
- Comparing legacy PSTN with modern IP telephony.
- Deep dive into the Session Initiation Protocol (SIP) architecture.
- Understanding SIP messages, methods, and response codes.
- The role of Real-time Transport Protocol (RTP) for media delivery.
- Exploring RTP Control Protocol (RTCP) for quality monitoring.
- Overview of common voice codecs like G.711, G.729, and Opus.

Unit Two: Designing Resilient VoIP Architectures

- Conducting a network readiness assessment for VoIP.
- Core components of a VoIP ecosystem (IP-PBX, Gateways, IP Phones).
- Principles of high availability and redundancy in voice networks.
- Architectural considerations for on-premise vs. cloud-hosted PBX.
- Introduction to Session Border Controllers (SBCs) and their functions.
- Capacity planning for bandwidth and concurrent calls.
- Designing logical network segmentation for voice and data traffic.

Unit Three: SIP Trunking Implementation and Management

- Understanding the concept and benefits of SIP trunking.
- Process of selecting and connecting to an Internet Telephony Service Provider (ITSP).
- Key configuration parameters for a SIP trunk.
- Developing and implementing effective dial plans and call routing logic.
- Managing Direct Inward Dialing (DID) and number portability.
- Addressing E911 regulations and configuration requirements.
- Troubleshooting common SIP trunk registration and interoperability issues.

Unit Four: Ensuring Voice Quality and Network Security

- Identifying the causes of poor voice quality like latency, jitter, and packet loss.
- Implementing Quality of Service (QoS) mechanisms (CoS, DSCP).
- Prioritizing voice traffic across the network infrastructure.
- Common VoIP security threats and vulnerabilities.
- Securing SIP signaling with TLS (Transport Layer Security).
- Encrypting media streams with SRTP (Secure Real-time Transport Protocol).
- Best practices for securing endpoints, servers, and SBCs.

Unit Five: Advanced Troubleshooting and Maintenance

- Using network protocol analyzers like Wireshark for call flow analysis.
- Diagnosing and resolving one-way audio and no-audio issues.
- Troubleshooting failed calls and analyzing SIP error codes.
- Proactive monitoring strategies for VoIP network health.
- Best practices for routine maintenance and system updates.
- Backup and disaster recovery planning for voice systems.
- Final project simulating a real-world troubleshooting scenario.

FAQ:

Qualifications required for registering to this course?

There are no requirements.

How long is each daily session, and what is the total number of training hours for the course?

This training course spans five days, with daily sessions ranging between 4 to 5 hours, including breaks and interactive activities, bringing the total duration to 20 - 25 training hours.

Something to think about:

How will the rise of WebRTC and cloud-native communication platforms influence the traditional architecture and security models of SIP trunking in the next decade?

What unique qualities does this course offer compared to other courses?

This course distinguishes itself by focusing intensely on the practical, real-world application of VoIP and SIP trunking principles, moving far beyond theoretical lectures. While other programs may focus on a single vendor's product, our curriculum is vendor-agnostic, providing a holistic understanding of the underlying technologies that empowers participants to work in any environment. The emphasis on hands-on troubleshooting, using tools like Wireshark to dissect real call flows, is a core component that builds tangible, job-ready skills. We dedicate significant time to architectural design and security, two areas critical for building robust and resilient communication systems but often overlooked in introductory courses. The content is continuously updated to reflect the latest industry challenges, including interoperability issues and emerging security threats. The expertise of the instructors at BIG BEN Training Center, who bring years of field experience, ensures that every module is enriched with practical insights, cautionary tales from actual deployments, and best practices that are not found in textbooks. This approach transforms participants from passive learners into confident practitioners ready to manage complex voice networks.