

VOIP & IP TELECOM

MASTER TECHNICAL TRAINING

Comprehensive, hands-on training to design, deploy, manage and troubleshoot enterprise VoIP & IP telecom solutions.



DURATION
03 MONTHS
(72 HOURS)



COURSE FEE
Rs. 24,000.00



CERTIFICATE
Upon Successful
Completion



ONLINE VIA
ZOOM




LMS ACCESS
Full Course
Access



LIFETIME
RECORDING
ACCESS



WHAT YOU WILL LEARN

- ✓ VoIP Fundamentals & Protocols
- ✓ IP PBX Installation & Configuration
- ✓ SIP Trunks & Call Routing
- ✓ IVR, Voicemail & Auto Attendant
- ✓ Intercom & Paging Solutions
- ✓ Security, Monitoring & Troubleshooting
- ✓ Real-world Projects & Industry Best Practices



100% PRACTICAL & INDUSTRY FOCUSED
Hands-on labs and real-world scenarios to build job-ready skills.



EXPERT INSTRUCTOR SUPPORT
Learn from experienced VoIP professionals with industry expertise.



CAREER OPPORTUNITIES
Be prepared for roles in VoIP engineering, networking, telecom and IT support.



WEB :
www.academy.oneaccess.lk



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academy@oneaccess.lk



TEL :
+94 702122214 | +94 763008308



DURATION

03 MONTHS
(12 DAYS,
24 HOURS)



FEES

RS. 24,000.00
(INSTALMENT -
RS 8000 X 3)



CERTIFICATE

Upon Successful
Completion



HYBRID MODE

**ONLINE &
PHYSICAL**



LMS ACCESS

Lifetime LMS
Access



LIFE TIME RECORDING ACCESS

Lifetime
Recording Access

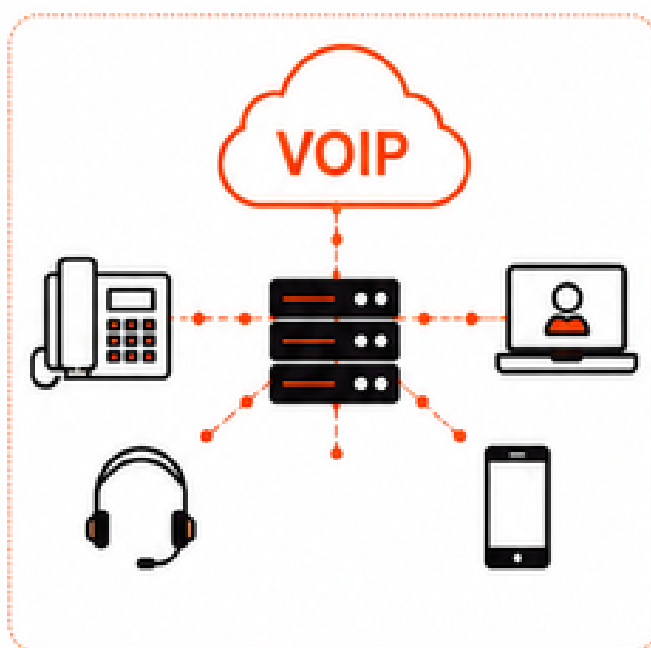


COURSE OVERVIEW

The VoIP & IP Telecom Master Technical Training Program is designed to provide a strong foundation in modern VoIP and IP Telephony technologies through a combination of structured theoretical learning and hands-on practical experience.

Participants will learn VoIP fundamentals, SIP protocols, IP PBX systems, VoIP networking, security, troubleshooting, and enterprise communication solutions while gaining practical experience with Asterisk, FreePBX, and Grandstream UCM platforms.

By the end of the program, students will have the knowledge and skills required to plan, deploy, configure, manage, and troubleshoot modern VoIP and IP Telecom solutions in real-world environments.



PREREQUISITES

No prior VoIP experience is required to attend this training program. However, a basic understanding of the following areas will be beneficial:

- ✓ Basic Computer Operations
- ✓ Fundamental Networking Concepts (IP Addressing, Subnetting, TCP/IP)
- ✓ Basic Knowledge of Windows or Linux Operating Systems
- ✓ Interest in VoIP, Networking, Telecommunications, or IT Infrastructure



This course is suitable for beginners, IT students, network engineers, system administrators, telecom professionals, and anyone looking to build a career in VoIP and IP Telephony.



TARGET AUDIENCE

This training program is ideal for:

- ✓ Network Engineers and Network Administrators
- ✓ System Administrators and IT Support Engineers
- ✓ Telecom and VoIP Engineers
- ✓ Cyber Security Professionals
- ✓ Technical Support and Helpdesk Engineers
- ✓ University Students and Graduates
- ✓ Diploma and Degree Students in IT, Networking, Telecommunications, or Computer Science
- ✓ Professionals looking to transition into the VoIP and IP Telecom industry
- ✓ Anyone interested in learning VoIP, IP Telephony, and Enterprise Communication Technologies



Whether you are a beginner seeking to enter the industry or an experienced IT professional looking to expand your technical expertise, this program provides the knowledge and practical skills required to succeed in modern VoIP environments.





COURSE CONTENT

MODULE 1



6 LESSONS

1 INTRODUCTION TO VoIP

- Overview of VoIP and its advantages
- VoIP vs traditional telephony (PSTN)
- Overview of VoIP Brands and Models
- H.323 Protocol
- Analogue and IP Phones
- Key components: IP phones, gateways, softphones, PBX

MODULE 2



4 LESSONS

2 VoIP PROTOCOLS

- Session Initiation Protocol (SIP)
- Real-Time Transport Protocol (RTP)
- Media Gateway Control Protocol (MGCP)
- WebRTC basics for browser-based voice communication

MODULE 3



4 LESSONS

3 VoIP NETWORK FUNDAMENTALS

- IP addressing and subnetting for VoIP
- VLANs and QoS (Quality of Service)
- NAT traversal and firewall considerations
- Bandwidth requirements and call quality

MODULE 4



8 LESSONS

4 CONFIGURE ASTERISK VoIP PBX ON LINUX

- Understanding Asterisk Architecture
- Install Asterisk on Ubuntu
- Configure SIP Endpoints
- Configure the Dialplan with advanced Features
- Configure Voicemail
- Enable External Calling (PSTN / Trunk)
- Create IVR Menus
- Set Up Call Queues Create



Whether you are a beginner seeking to enter the industry or an experienced IT professional looking to expand your technical expertise, this program provides the knowledge and practical skills required to succeed in modern VoIP environments.



COURSE CONTENT

MODULE 5



7 LESSONS

5 FREEPBX / GRANDSTREAM SETUP

- Setting up FreePBX / Grandstream
- Configuring extensions, trunks, and dial plans
- Softphones and IP phone setup
- Call routing, voicemail, and auto-attendant
- IVR Setup, Call Queue
- Grandstream Zero Config
- High Availability (HA)

MODULE 6



4 LESSONS

6 VoIP SECURITY

- VoIP threats: eavesdropping, toll fraud, DoS attacks
- Securing SIP endpoints and servers
- Encryption with SRTP and TLS
- Security considerations for cloud VoIP deployments

MODULE 7



4 LESSONS

7 VoIP TROUBLESHOOTING & MONITORING

- Call flow analysis and logging
- Common issues and resolutions
- Monitoring tools for VoIP networks
- AI-powered call analytics for performance monitoring

MODULE 8



4 LESSONS

8 HANDS-ON LABS & PROJECTS ~ ON-SITE

- Configuring a complete VoIP system (on-premise and cloud)
- Simulating calls between IP phones and softphones
- QoS and bandwidth testing
- Security testing and troubleshooting

MODULE 9



5 LESSONS

9 FINAL PRACTICAL EXAM

- Basic Network Setup - Configure IP addressing of PBX server, Softphone, IP phones Ensure connectivity (ping test)
- PBX Installation & Setup - Install or access PBX (FreePBX / Grandstream / 3CX) 2-3 extensions (e.g., 1001, 1002), Register softphones
- SIP Configuration - Configure SIP accounts, Verify registration status, Make internal calls between extensions
- Call Routing (Dial Plan) - Configure Internal calling, Outbound dial rule (e.g., 9 + number), inbound route & Outbound Route
- VoIP Testing - Make test calls: Extension ↔ Extension, Extension → Outside



Whether you are a beginner seeking to enter the industry or an experienced IT professional looking to expand your technical expertise, this program provides the knowledge and practical skills required to succeed in modern VoIP environments.

CERTIFICATION OF AFTER COMPLETION

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CONNECTING THE WORLD



GSCS-2025-0401028

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UC Solution

Presented to

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EXCEL TECHNOLOGIES LIMITED

For successfully completing the requirements and
demonstrating the knowledge and skills in Grandstream UC Solutions.



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Tel: +1 (617) 546-9300
www.grandstream.com

Certificate No.: GSCS250401028

Valid Through: April 01, 2026

Abderrahim Ait Omar

Abderrahim Ait Omar
Chief Technology Officer
Grandstream Networks, Inc.



ONE ACCESS TRAINING INSTITUTE CERTIFICATE

ONE ACCESS
TRAINING INSTITUTE

CERTIFICATE OF COMPLETION

We are pleased to award this certificate to

Mohammed Sameer SanoorDeen

In recognition successfully completed the industry based training program
VoIP & IP Telecom Master Technical Training

Certificate No : 0A1TFA/000970480048

Date of Certification : December 20th, 2025



Verify this certificate's authenticity at:
<https://training.oneaccess.lk/verifycertificate/online.php>

Ujitha N Rodrigo

Ujitha N Rodrigo
Program Director,
OneAccess Training Institute



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RECOGNIZED



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CAREER



PRACTICAL SKILLS
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OUR GOAL

Our goal is to provide a strong foundation in VoIP and IP Telephony through structured learning and hands-on practical training. We aim to equip participants with the knowledge, confidence, and real-world skills required to design, deploy, manage, and troubleshoot modern VoIP solutions.



HOW YOU WILL BENEFIT

- ✓ Build a strong foundation in VoIP and IP Telephony technologies
- ✓ Gain hands-on experience with Asterisk, FreePBX, and Grandstream UCM
- ✓ Develop practical skills in VoIP deployment, configuration, and troubleshooting
- ✓ Improve your understanding of SIP, PBX systems, and enterprise communication networks
- ✓ Increase your confidence in planning and managing VoIP solutions
- ✓ Enhance your career opportunities in the IT, Networking, and Telecommunications industries



DON'T MISS THIS OPPORTUNITY!

Take the next step in your IT and Telecommunications career. Gain the knowledge, practical skills, and hands-on experience needed to succeed in the rapidly growing VoIP and IP Telecom industry.



Limited Seats Available –
Register Today and Start Your Journey
to Becoming a Professional VoIP Engineer!

THANK YOU

Thank You for Your Interest in Our
VoIP & IP Telecom Training Program!



We sincerely appreciate your interest in our
VoIP & IP Telecom Training Program.



Our mission is to provide real-world, hands-on technical training that helps IT professionals build practical skills, gain industry-recognized knowledge, and advance their careers in VoIP and IP Telecommunications.



We look forward to helping you achieve your goals through quality training, expert guidance, and continuous support.



GRANDSTREAM
UCM IPPBX



IP TELEPHONY
SOLUTIONS



IP PBX
SYSTEMS



SECURE
COMMUNICATIONS



RELIABLE
CONNECTIVITY



SCALABLE
SOLUTIONS



Empowering IT Professionals with the Skills
to Build Smarter **VoIP Solutions.**

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Empowering IT Professionals
Through Real-World
Technical Training

