3. Purpose of the Course

This course prepares the students for implementing a Cisco Unified Communications Manager solution at a single-site environment. This course focuses primarily on Cisco Unified Communications Manager Version 10 and 11 which is the call routing and signalling component for the Cisco Unified Communications solution, the implementation of the NATO Voice Over Secure IP (VOSIP) and the configuration of the V2 (Voice and Video Router) for VOSIP. Students will perform post-installation tasks, configure Cisco Unified Communications Manager, implement Media Gateway Control Protocol (MGCP), H.323 gateways and CUBE, and build dial plans to place on-net and off-net phone calls. You will also implement media resources, Cisco IP Phone Services, Cisco Unified Communications Manager native presence, and Cisco Telepresence. Call Admission Control (CAC), Extension Mobility, NATO Voice Over Secure IP (VOSIP), Call Manager Virtualization and Disaster Recovery (DRS) all this configuration should be implementing according to NATO configurations.

4. Learning Objectives

Upon completion of the course, the qualified student will be able to:

1. **NATO Voice Systems**
   - Outline the different Voice Solutions used within the various NATO networks.

2. **Intro to Call Manager (CUCM)**
   - Understanding Cisco Unified Communications Manager.
   - Understanding CUCM Deployment and redundancy options.

3. **Administering CUCM**
   - Managing Services and Initial Configuration and Initial Configuration.
   - Managing User Accounts in CUCM, Manual, BAT integration with LDAP.

4. **IP Phones**
   - Understanding Endpoints in CUCM.
   - Implementing IP Phones and URI Dialing.
   - Implementing Video Phones.
5. **Dial Plan implantation, Call Routing and Class of Services**
   a. PSTN access Gateways and Session Border Controllers.
   b. Implementing PSTN Gateways MGCP, H.323.
   c. Implementing CUBE, Protocols interworking, Media Flows, Codecs and URI dialing
   d. Implementing SIP trunks.
   e. Implementing Gateway Selection and PSTN Access Features.
   f. Dial Plan implementation.
   g. CUCM Call-Routing components, Addressing Methods and Digit Analysis.
   h. Digit Manipulation.
   i. Implement Calling Privileges using Partitions and CSS,s.
   j. Call Coverage in CUCM, Hunting Groups, Call queuing and Pick Groups.

6. **Media Resources**
   a. Implementing Media Resources Audio and Video Conferences, Meet-me Conferences, Transcoders, MTP, Music and Video on Hold, Annunciators.
   b. Implementing Media Resources Access control MRG and MRGL.

7. **Audio and Video Conferencing**
   a. Overview Audio or Video conferencing.
   b. Implementing software CUCM Conference Bridges and Cisco IOS-based Audio and Video Conference Bridges.
   c. Telepresence Overview.
   d. Integration Cisco Telepresence Server and CUCM.
   e. Telepresence Conferencing Resources Cisco TPS and Telepresence Conductor.

8. **Quality of Services**
   a. QoS requirements.
   b. Overview QoS components.
   c. Implementing Marking, Policing and Shaping.

9. **Feature and Application Implementation**
   a. Configuring CUCM Native Presence.
   b. Implementing Presence Policies.

10. **Implementing Call Admission Control (CAC)**
    a. Configuring Regions.
    b. Create and Configuring Device Pools for CAC.
    c. Configuring Locations and Links.
    d. Implementing Enhance CAC, Location Bandwidth Manager servers
11. Implementing Cisco Extension Mobility (NATO Configuration)
   a. Configuring Cisco IP Phone Services.
   b. Activate Cisco Extension Mobility.
   c. Create a Device Profile.
   d. Create ad device Profile for End Users.
   e. Add Cisco Extension Mobility and Subscribe the IPhones.

   a. Secure Voice overview.
   b. Install CTL Client and Install CTL File.
   c. Configure a V2 Router to deploy Secure MTP.
   d. Import and Export Certificates to CUCM.
   e. Configure SCCP on V2 router.
   f. Add Secure MTP in CUCM.
   g. Configure MRG and MGRL in CUCM.
   h. Add Common Device Configuration and create SIP Security profile and assigned to phones.
   i. Install LSC Certificates in the IPhones for Secure Voice.

13. Configuring a V2 Router for Secure Conference Bridge SCFB and Voice Over Secure IP (VOSIP)
   a. Add Secure CFB in V2 Router.
   b. Import SCFB certificate to CUCM.
   c. Create a Secure Configuration Profile.
   d. Add SCFB in CUCM.

14. Configuring a V2 Router for Secure SRST and Voice Over Secure IP (VOSIP) (NATO Configuration)
   a. Enroll Secure SRST to Root CA.
   b. Export CUCM certificate.
   c. Add Secure SRST Reference in CUCM.
   d. Configure a V2 Secure SRST fall-back mode.

15. Cisco UCS Servers and VMWARE
   a. Cisco UCS Servers.
   b. VMware in UCS Servers.
   c. Export and Import OVF files templates for CUCM.
   d. Deploy CUCM in Virtual environment (VMWARE)

16. Disaster Recovery (DRS)
   a. Install a SFTP Server.
   b. Add Backup Devices.
   c. Performance Backups.
   d. Restore Call Manager Database (DRS).
5. Qualification

NATO Call Manager Technician

6. Student Criteria

1. The course is designed for military and civilian staff assigned within a NCISG, or within in a technical field in NATO (such as NCIA Sectors, NCIA CLD or similar).
2. Have met the Background Knowledge Prerequisites for this course.

7. Rank

- Selected officer
- Enlisted
- Civilian equivalent

8. Language Proficiency

according to STANAG 6001: English SLP 3232

9. Security Clearance

NATO UNCLASSIFIED

10. Course Length

10 working days (two weeks)

11. Special Instructions

i. The student must achieve a grade of 70 % on a comprehensive final practical exercise
ii. The student must achieve a grade of 70 % on a comprehensive final written examination
After daily lessons the students need access to Internet to review the on line documentation and practise using Packet Tracer.

12. Class Size

- Maximum
- Recommended
- Minimum

8/8/4

13. Nomination Procedures

www.nciss.nato.int
14. Pre-course Study Material
N/A

15. Location
The course is conducted at the NATO Communications and Information Systems School (NCISS), Latina Italy.

16. Background Knowledge Prerequisites
Students must have completed/passed NCISS VoIP foundation Course, NCISS Course ID 095.