Course code: JSP-EU230

JSP TELECONSULTANCY

Implementation of Next Gen. Voice Over IP Comm systems, functionality and features over an Air Traffic Service Ground Network

Course Background

The introduction of Voice over IP Telephone, Radio and Recorder interfaces to ATS Ground Networks is now causing an evolution with Air Navigation Service Providers (ANSPs) now starting the migration from analogue/TDM digital circuit switched communication infrastructures in operation today towards a future converged common multimedia IP network infrastructure for both voice and data services. Some ANSP's have fully completed migration to this next generation technology.

ANSPs, VCS, Radio and Recorder vendors have confirmed that the VoIP protocol as defined by the EUROCAE WG67 ED documents is now the strategic direction for ATM Voice communications. The published ED-137B documents are now cited in requirement specifications globally both by civil and military organisations and further enhancement towards global interoperability of systems through these common interfaces is demonstrated through technical contributions towards WG67 from many ANSPs and companies around the world now finalising the ED-137C edition of the documents.

The ICAO Aeronautical Communication Panel (ACP) WG-I are citing EUROCAE ED documents within the Manual for the Aeronautical Telecommunication Network (ATN) using Internet Protocol Suite (IPS) Standards & Protocols (Doc9896), implying that the way is clear for global adoption of the VoIP interfaces. A European Single Sky Implementation (ESSI) objective has commitment of European ANSPs as well defining timeframe for the migration.

Interoperability events for these VoIP-in-ATM interfaces with multi-vendor participation were completed 5 years ago, superseded by EUROCONTROL VOTER conformance test phase, also now completed by the majority of vendors, who are now dedicated to achieve the objective of moving towards a VoIP ground network. These interoperability and conformance test phases have demonstrated that IP technology has the capability to fulfil operational and technical ATM communication requirements and that the interfaces are robust enough to be deployed in an operational environment

The Work Package 15.2.10 within the SESAR JU work programme executed a comprehensive VoIP validation test programme for IP based Ground communications (voice/signalling) over a Pan European Network. This demonstrated that a correctly configured edge access and backbone network for voice transport by the provider working hand-in-hand with the next generation telephone and radio interfaces can achieve performance figures often better than those measured while using current analog/TDM network infrastructures.

The use of the Session Initiation Protocol (SIP) for creating sessions makes the process of accessing Radio Stations within Functional Airspace Blocks (FABs) from multiple VCS's easier to implement and works as an enabler for the Dynamic Airspace resectorization, Sector delegation between adjacent ACCs and also provides solutions for the management of communications between Local and Remote Towers. It is seen from working implementations in the field today that new concepts for Main/Standby VCS switchover as well as establishment of redundant sessions to remote main and standby radios through diverse paths increases availability figures while improving the safety of last resort communications in case of end equipment system failure/outages as well as failures core network.

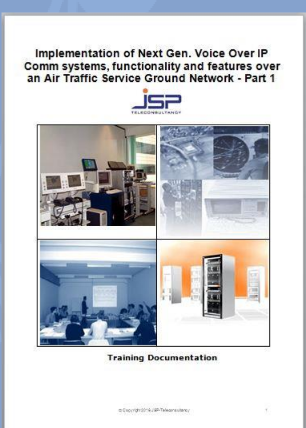
As we enter the implementation phase for the majority of ANSPs, portable Radio and Telephone applications have been developed in order to fully emulate ED-137B telephone and radio interfaces (at VCS) as well as radio interfaces (at Radio). These tools are vital for in-the-field configuration of VCS's and Radios, allowing all call types to be made and received, PTT-types to be keyed/unkeyed, SQU to be set ON/OFF and speech transparency checked.

John Palmer has been a telecommunications consultant to EUROCONTROL in Brussels for more than 6 years followed by 13 years with a role of external consultant to the greater ATC community. He also has over 15 years of experience in the private telecoms industry working for a number of large and medium sized telecoms companies in the UK, Italy and Belgium.

As a participant in the EUROCAE WORKING GROUP 67 that have defined Voice over IP specifications for the future ATS Ground Network, he has contributed to the EUROCAE documents defined by the Telephone Interoperability and the Qualification and Validation Test Sub-groups, while also assisting the Requirements Sub-group.

He successfully organised the FAA Global VoIP Interoperability Event in Washington DC in May 2011 and was actively involved in SESAR WP15.2.10 VoIP performance test activities between VoIP interfaces over the PENS test bed until 2013. Besides satisfying the high global demand for courses in this field, recently he has assisted in development and test of ED-137 Radio and Telephone Emulator applications seen as vital test instruments as VoIP-in-ATM technology is being rolled-out into an operational environment.

Contact us for more details or to arrange a course suitable for you at a location chosen by you



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Course Description

The first 1½ days of the course has the scope of providing the broad picture of VoIP technology and its application to ATS Ground Voice Networks starting with an explanation of work performed by the EUROCAE Working Group 67 in the definition of documents for VoIP in ATM, the course lays a foundation by explaining the basics of Voice over IP functionality before building on this with a technology and functionality overview of today's operational telephone and radio networks tied in with enhancements offered by ED-137B and soon-to-be published ED-137C editions . A module on next generation architectures and how this technology enables sector delegation, dynamic resectorization concepts to be implemented operationally as well as last resort communications. EUROCONTROL and FAA VoIP initiatives followed by an overview of the Network and SESAR WP15.2.10 work for VoIP validation are then reported.

Day 2 PM session provides in-depth details about the Added benefits and functionality that VoIP brings to ATS Ground Telephone network that also introduces the functionality of existing as well as new ED-137C call types and features. Day 3 will provide in-depth details about the added benefits and functionality that VoIP brings to ATS Ground Radio network and Recorder equipment, which also includes Radio Gateway overview. The GL MAPS ED-137B Radio and Telephone Emulator, tools are demonstrated during these modules, today chosen by the majority of ANSPs as the benchmark emulator and configuration test tool for the implementation phase.

The final DAY 4 moves on to examine network performance aspects, VoIP network architecture, security, the completed interoperability and conformance test phases. During the course practical examples of SIP Telephone calls using Soft phones through a WAN emulator, allow network impairments to be introduced, while examining SIP signalling and RTP audio using a wireshark.

Benefits for course participants

After completing the course, the participant would have acquired a deeper understanding of Global requirements and solutions relating to the next generation of Voice Communication Systems, Radios & Recorders and their operation over a Core IP network etc.

- VoIP functionality and work performed by EUROCAE Working Group 67
- Application of VoIP to aeronautical ground communications
- Added benefits VoIP technology brings to Telephone, Radio and Voice Recording functionality with a private IP network
- ATM-VoIP network architecture
- Network performance and security aspects
- Test and validation activities performed to date examined in detail
- Course includes practical demonstrations using SIP User agents endpoints, WAN emulator (for network impairments), wireshark (for protocol analysis) as well as the benchmark GL MAPS Radio and Telephone Emulators being used during implementation.

Course suitable for

Professionals within various sectors of the aeronautical industry who need to develop a greater understanding of the functionality expected from the next generation of Voice Communication Systems, Ground Radio Stations, Telephone and Radio Gateways, Voice Recorders and how the core and edge network designs will be configured to ensure a high voice call performance and voice quality over a secure network.

These past 5 years have seen over 800 participants globally attend this course, making it one of the most popular courses in the field being delivered today.

Course duration:
6 hours per day with 10 or 20 minute breaks every hour as defined below:

HOUR 1	09.00 - 10.00
BREAK	10.00 - 10.20
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BREAK	11.20 - 11.30
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LUNCH	12.30 - 13.30
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BREAK	14.30 - 14.40
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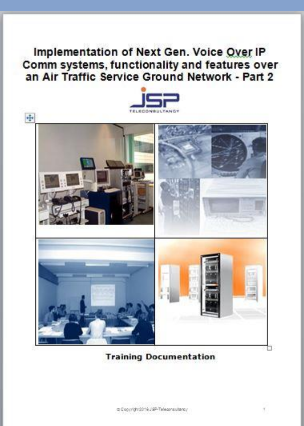

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Course code: JSP-EU230

JSP TELECONSULTANCY

Detailed outline DAY 1

A typical day starts at 09h00 and finishes at 17h00.

1	EUROCAE Working Group 67	2	Voice over IP functionality	3	Current Telephone operation /features plus ED-137 enhancements	4	Current Radio operation /features plus ED-137 enhancements
	30 minutes		60 minutes		120 minutes		150 minutes

Module 1: EUROCAE Working Group 67

- What is EUROCAE?
- EUROCAE WG67 background ?
- Mission Statement & Vienna Agreement
- ED deliverables
- ICAO ACP WG-I consideration of EUROCAE ED docs
- Current status and new structure of ED-137C docs
- Companies/ ANSPs working with this technology

Module 2: Voice over IP functionality

- What is Voice over IP?
- Packet Switched Connections & Efficiency
- IP Core Network
- VoIP deployed in corporate networks
- Signalling in an IP ATS Ground Voice Network
- How is Voice transported over an IP network?
- Why IP version 6?
- IPv4 & IPv6 Packet Header fields
- User Datagram Protocol (UDP)
- Real-time Transport of Voice using (RTP)
- RTP media streams
- What is Session Initiation Protocol (SIP)?
- SIP History
- Why SIP signalling?
- What is Session Description Protocol (SDP)?
- VoIP network architecture elements and their functionality
- Example of SIP telephone technology today
- Demonstration of SIP Telephone calls between SIP User agent soft phones & wireshark capture .

Module 3: Current Telephone operation /features plus ED-137 enhancements

- Telephone network infrastructure overview
- Impacts of VoIP on Voice Communication Systems/Telephone calls
- Current and future DA/IDA Telephone Call operation
- Multi-Destination DA/IDA Enhancement
- Current and future IA/Hotline Telephone Call operation
- Override call operation
- Voice Call operation
- Current and future Priority call operation
- Positioning monitoring
- Subscribe-Notify
- Presence Service-Dialogue Event package enhancement
- End user connectivity checks.
- Conference-Broadcast, Pre-set, Meet-me.
- Telephone supplementary services/features
- Voice codecs and future enhancements
- Sunset dates for analogue & digital leased lines
- Migration of IP through Telephone Gateways

Module 4: Current Radio operation /features plus ED-137 enhancements

- Radio network infrastructure overview
- SIP Session establishment /SDP negotiation to Radios
- Normal/ Emergency session permissions tables
- Traffic/ Coupling/Rx-only & Idle Radio Access modes
- PTT-id assignment by Radio Tx & Rx for Traffic/Coupling sessions
- Radio session establishment from multiple endpoints
- Radio session mix from same endpoint
- Main/Standby VCS and Main/Standby Radio switchover examples – with ED-137 solution
- RTP Stream establishment with Radios- Audio/Keep-alive packet functionality
- RTP Header Extension for Radio Signalling
- Real Time Session Supervision (R2S) operation
- Current & Future Push-To-Talk (PTT) operation – PTT level hierarchy
- Current & Future Aircraft call detection operation
- Non-IP source PTT keying operation
- Cross-coupling (Retransmission), Simplex & Duplex Cross-coupling operation;
- Resolution of blocked frequencies caused by Cross-coupled chains
- Cross-coupling operation enhancement
- Best Signal Selection (Rx Voting), operation and enhancement
- Differential path delay measurement to Remote Radio Receivers-used when comparing measured signal quality values from receivers in BSS group.
- Mixed Received Signal echo compensation
- Best Transmitter Selection
- Multicarrier Offset Transmission (Climax) operation
- Dynamic delay measurement and compensation methodology
- Stepped-on transmissions scenarios– overlapping aircraft call transmissions, controller/aircraft call transmissions.
- Missed transmissions due to dual transmissions in Cross-coupled group and Best Signal Selection
- Off-air side tone- Local Side tone enablers
- WG67 KEY-IN event package
- Lock-out/Summation configuration modes
- Instantaneous control of Cross-coupled groups by Standby VCS
- Remote Radio Gateway Node equipment operation
- Next Generation VoIP Recorder overview
- Migration of IP through Radio Gateways

Summary of first day- Questions and Answers session

JSP TELECONSULTANCY

Course duration:
6 hours per day with 10 or 20 minute breaks every hour as defined below:

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BREAK	10.00 - 10.20
HOUR 2	10.20 - 11.20
BREAK	11.20 - 11.30
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HOUR 4	13.30 - 14.30
BREAK	14.30 - 14.40
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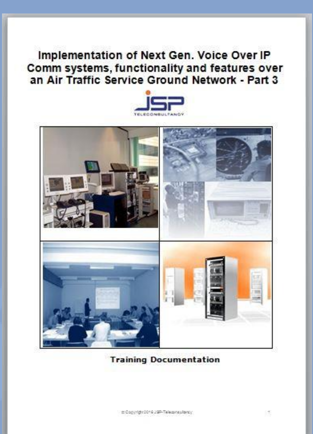

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Detailed outline DAY 2

A typical day starts at 09h00 and finishes at 17h00.

5	ED-137 enablers for ATC operation	6	Eurocontrol & FAA VoIP initiatives	7	Network Overview	8	SESAR 15.2.10 VoIP validation work	9	Added benefits VoIP brings to ATS Ground Telephone Network
	90 minutes		30 minutes		90 minutes		30 minutes		120 minutes

Module 5: ED-137 enablers for operational ATC.

- Circuit to packet switching interworking
- Working towards end-to-end VoIP
- Session Border Controller (SBC) functionality
- Radio Server functionality
- Architecture with Radio Server Integrated
- Next Generation end-to-end IP architectures
- Functional Airspace Blocks (FABs)
- Sector Delegation between ACCs or Local/Remote Towers (Radio)
- Last Resort communication between ACCs (Radio)
- Civil/Military sharing of sectors (Radio)
- Sector Delegation between ACCs or Local/Remote Towers (Telephone)
- Last Resort communication between ACCs (Telephone)
- What are Roles/Missions?
- What is sector delegation?
- Role/Sector/Mission relationship
- Foreseen Network Management sector delegation protocol
- Address scheme for Sector Delegation with examples
- Example of sector delegation from ACC1 to 2.

Module 6: EUROCONTROL & FAA VoIP initiatives

- EUROCONTROL VoIP in ATM Test Specs
- EUROCONTROL VoIP in ATM T VOTER test suite development
- FAA Addendums to ED-137
- VOTER interoperability test suite
- Proposed European Single Sky Implementation (ESSI) Objective 2012-2020
- VoIP in ATM implementation & Transition Expert Group –Overview of work

Module 7: Network overview

- Network domain concept
- Pan-European Network –Core Network & services
- Guaranteeing IP network availability – Built-in redundancy
- PENS Network Architecture
- Virtual Private Networks available on PENS
- Multiprotocol Label Switching (MPLS)
- MPLS path establishment through network
- Layer 2 Virtual leased line (VLL) over MPLS core network
- Layer 2 Virtual Private LAN service (VPLS) over MPLS core network
- E-Line and L-LAN
- Guaranteeing Real Time Voice over network

Module 8: SESAR WP 15.2.10 VoIP validation work

- SESAR Programme /Master Plan for CNS activities
- SESAR Work Packages
- Validation of VoIP in ATM part of WP 15.2.10
- SESAR JU Work Programme/Description of Work
- SESAR Members & 15.2.10 Members
- WAN test bed network topology
- SESAR 15.2.10 VoIP related deliverables
- SESAR 15.2.10 D11 and D12 performance tests performed with result overview /analysis

Module 9: Added benefits VoIP brings to ATS Ground Telephone network

- SIP Request/Response Messages
- SIP User Agents
- SIP User Agent Registration/Authentication procedures
- SIP Network Servers/DNS servers
- SIP session establishment between SIP User Agents
- Example of a SIP Direct Access Call
- Call types (DA, IDA, IA, Position monitoring) –Header info
- WG67-Version Header field
- Proposed VCS address scheme
- SDP Media session attribute negotiation
- G.711.1 Wideband Codec (RFC5391)
- SIP Protocol message detail example for call establishment
- Multi-destination DA/IDA Call Implementation
- Instantaneous Access /Hotline call implementations
- Override Call Implementation
- Radio Intercom Call implementation
- Voice Call Implementation
- Position Monitoring network feature
- Broadcast/Pre-set/Meet-me Conference Call Implementation
- Call Intrusion network feature and examples
- Network wide supplementary services
- Presence, Dialogue event packages., connectivity checks
- Legacy v SIP interworking specifications
- Foreseen stepped migration towards SIP from legacy signalling
- ED-137C Volume 2 enhancement overview.

Summary of 2nd day- Questions and Answers session

JSP TELECONSULTANCY

Course duration:
6 hours per day with 10 or 20 minute breaks every hour as defined below:

HOUR 1	09.00 - 10.00
BREAK	10.00 - 10.20
HOUR 2	10.20 - 11.20
BREAK	11.20 - 11.30
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Detailed outline DAY 3

A typical day starts at 09h00 and finishes at 17h00.

10	**Added benefits VoIP brings to ATS Ground RadioNetwork**	11	**Added benefits VoIP brings to ATS Ground Recording equipment**
	240 minutes		**120 minutes**

Module 10: Added benefits VoIP brings to ATS Ground Radio network

- Radio System Baseline reference model and Performance requirements
- SIP session establishment (VCS & GRS Transceiver endpoints)
- SIP session establishment (VCS & separated GRS Tx and Rx endpoints)
- GRS Reject & WG67 Reason Header values
- GRS placed in Maintenance Mode
- Radio SIP Headers & SDP message body attributes
- Media Session attribute negotiation (Mandatory)
- SDP attribute – <send-receive mode> example
- SDP codec attributes
- Radio Session Access Modes
- R2S-KeepalivePeriod example
- R2S-KeepaliveMultiplier example-LocalHoldTime expiry at GRS or VCS
- ptt-id:<PTT identity> attribute example
- GRS rejects second coupling session example
- SELCAL selective-calling radio system
- Media Session attribute negotiation (Optional)
- Connection type -txrxmode example (optional)
- Best Signal Selection - quality index method> (optional)
- Sigtime -Signalling Info Time Period (optional)
- PTT-rep example for PTT ON/OFF and SQU ON/OFF transition (optional)
- Frequency ID -fid example (optional)
- Multimode radio Frequency change
- WG67 KEY-IN Event package
- SDP attribute – rtphe:<RTPHE Version> example
- What is Linked Session Management?
- Linked session Protection and Execute attributes
- Linked-Session Management-Main/Standby Switchover (optional)
- Linked and Non-linked sessions
- Cross coupled group problem for Main to Standby Switchover
- Backup Cross Coupling Solution
- Multicast solution
- Max No. of SIP sessions per GRS –PTTid assignment by Tx and Rx
- Non-coupling session pre-emption-Tx and Rx examples
- Coupling session pre-emption –Tx and Rx
- Transmitter PTT Lockout/ Summation
- Proposed GRS Address scheme
- RTP Header & Header Extension structure/ types
- RTP Header Extension principle
- PTT-types and PTT-id echo back
- GRS audio lockout mode using Normal/Priority/Emergency PTT-types
- GRS audio summation mode using Normal/Priority/Emergency PTT-types
- Summation of multiple RTP audio streams in the VCS,
- Incoming Aircraft call to GRS receiver with Rx-only
- PTT-ON for Radio-TxRx session to GRS receiver
- Incoming Cross-coupled retransmission distinguished from Aircraft Call
- Aircraft call detection by multiple GRS Transceivers & Best Signal Selection
- Simultaneous Aircraft Call Transmission detection by GRS Receiver
- Stepped on transmissions-A/C call +Normal PTT activation
- Off-air Squelch detection with separated GRS Transmitter and Receiver
- Local side-tone generation enablers
- One-way audio delay estimation
- Request for Measurement message (RMM) & Measurement Answer Messages (MAM)
- Dynamic Delay Compensation method for Climax operation over an IP network
- Multi-carrier offset transmissions (Climax) operation
- PTT Mute-Signalling PTT -ON at inactive transmitters
- Remote Radio Node (RRN) operational modes
- RRN Single channel/Dual Control mode,
- RRN Single/Paired frequency configuration
- RRN Priority/Non-Priority modes.
- RTP Header Extension applied to Remote Radio Control (RRC) operation
- RRN Shared /Non-shared variables.
- ED-137C Volume 1 enhancement overview.

Module 11: Added benefits VoIP brings to Recording equipment

- ED-136 Recording Requirements
- ED-137B Recording Requirements
- Recording Telephone/Radio Speech
- Recording architecture redundancy
- RTSP Control messages
- Methodology for Voice Recording
- RTSP/RTP transport methods
- Referencing communications
- Call Record Data (Event logging)
- Properties and Operations
- Event logging examples
- Replay Functionality
- Recorder Interoperability Test Configuration
- Recorder Change request enhancements for ED-137C.

Summary of third day- Questions and Answers session

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BREAK	10.00 - 10.20
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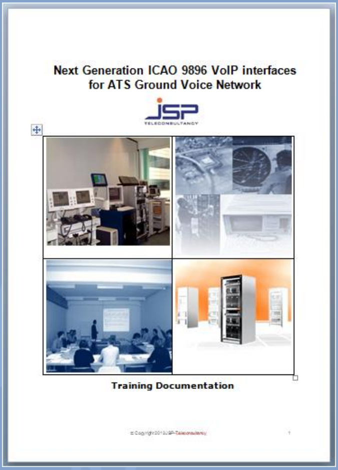

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Detailed outline DAY 4

A typical day starts at 09h00 and finishes at 17h00.

12	**Network Performance & Supervision aspects** **60 minutes**	13	**ATM – VoIP network architecture** **30 minutes**	14	**VoIP in ATM network security** **60 minutes**	15	**GL MAPS ED-137B Radio and Telephone Emulator demo** **60 minutes**
16	**Interoperability Events overview** **45 minutes**	17	**Network validation /Performance measurements** **45 minutes**	18	**Future concepts Summary debate** **60 minutes**		

Module 12: Network performance aspects

- Call performance criteria for DA and IA calls
- One way Voice Delay for Radio/Telephone
- Voice Quality of A/G and G/G communications
- Transmitter Activation Delay – A/C call indication delay
- Precedence level assignment of voice services
- Voice Packet size to IP Bandwidth
- Equivalent Bandwidth examples
- Demonstration of network impairment effects on voice using a WAN emulator tool.
- Supervision Model (ED-137B Volume 5)
- Supervision/Monitoring Functions
- Supervision SNMP Data Flow
- VCS Supervision Structure within MIB
- Basic Monitoring/Control of GRS's

Module 13:VoIP in ATM network architecture

- Technical Spec for provision of PENS
- ED-138 Quality of Service Requirements
- Platinum Service for Voice Transport
- Network Connectivity, Quality and Performance, Availability Issues
- Class of Service (CoS)
- PENS network architecture
- Multicast enabled Network infrastructure
- Multicast configuration for optimum use of bandwidth.

Module 14: VoIP in ATM network security

- ED-137 Security considerations
- ED-138 Security Recommendations
- Security Considerations for VoIP systems
- ED-138 Network Security Model
- PENS core network security requirements
- SIP User Agent authentication
- Transport Layer Security (TLS)
- Secure RTP
- What is IPsec/ IPSec overview
- IP security over Core network
- IP security end-to-end
- Voice encryption aspects
- End-to-End delay budget for Real time voice with IPSec

Module 15: GL MAPS ED-137B Radio & Telephone Emulator demo

- App download and dongle license installation
- Radio Emulator configuration / Start Processor
- Radio Session establishment and protocol analysis
- PTT-type keying and send speech test
- SQU ON/OFF and receive speech test
- Demo of PTT Mute/ PTT summation/ Stepped-on transmission generation and count
- TYPE 1 BSS setting/ measurement, TYPE 2 Climax delay and TYPE 3 RRCE operation
- Full ED-137B Volume 1 feature emulation
- Telephone Emulator configuration / Start Processor
- DA/IA & Positioning Monitoring call types & speech tests
- Bulk Call Generation feature
- Impairments feature for Radio/Telephone

Module 16: Interoperability Events overview

- What are Interoperability Events?
- Industry Driven VoIP interface interoperability testing
- FAA Global Interoperability Event- Overview, tests, procedures
- FAA Event pre-conditions and procedure overview
- FAA Telephone, Radio, Radio Gateway, Recorder interoperability test specifications
- Network Test Bed Topology
- Telephone, Radio, Radio Gateway and recorder tests performed

Module 17: Network emulator /Performance Measurements

- WAN emulator and Network simulation
- Practical demonstration of WAN emulator with network impairments introduced
- Impairments, modifiers, interfaces, filters
- Send and Receive Bandwidth utilization measurements:
- Packet Delay, Loss, R-factor measurements
- Packet Loss Rate and MOS measurements
- Impairments relative to network delay, jitter, packet drop rates, burst packet drops, bandwidth throttle (throughput impairment), packet duplication rate, packet out-of-sequence etc
- Measurement of Voice Quality & latency over an IP network
- Impact on Voice Quality with introduction of network impairments (i.e. jitter)
- Measuring average bandwidth consumption for Radio R2S-Keepalive and RTP audio packets
- PTT latency measurements over LAN and WAN.

Module 18: Future concepts summary/debate

- WAN emulator and Network simulation

Summary of forth day- Questions and Answers session