Course code: JSP-EU232

ED-137C Telephone /Radio theory plus Practical Lab using ED-137B Emulators & ED-138 Performance Monitoring Tool

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 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 followed by a conformance test phase using the EUROCONTROL VOTER test tool have now been completed by the majority of vendors. Also 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.

These test phases have 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. IP technology therefore has the capability to fulfil operational and technical ATM communication requirements. ANSPs are now deploying them in an operational environment

Importance of having validated test tools ready-to-go prior to an implementation phase is important for ANSPs and for this reason ED-137B Radio and Telephone Emulators have been developed in order to assist with Factory Acceptance Tests (FAT) as well deployment of equipment in the field, vital to verify the configuration of VCS's and Radios as they are being installed. Initially released in April 2016 these emulator tools are now deployed by many ANSPs during their migration. Tracking of the standard as well as feedback from users, leads to continued enhancement of the emulator tools. The next Radio Emulator release will allow the independent emulation of 100's of aeronautical radios from a single PC, essential for ANSPs during a Factory Acceptance Test phase.

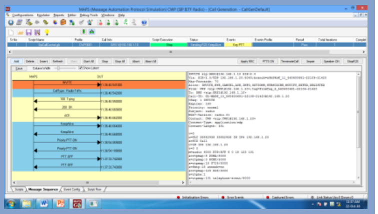
An ED-138 Network Performance Monitoring Tool able to passively tracking every individual phone call and radio call, measure a series of key performance parameters for voice communications over the network, listen in to the voice quality and raise alarms in case of transgression outside the ED-138 specification is also a essential requirement for ANSPs.

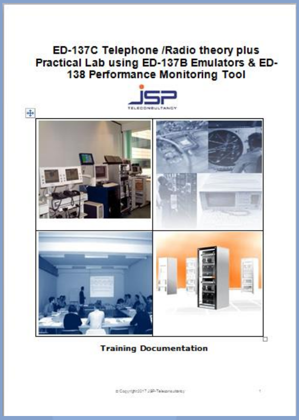
John Palmer has been a telecommunications consultant to EUROCONTROL in Brussels for more than 6 years followed by 14 years with a role of external consultant to the greater ATC community. He also has over 15 years of experience in the private telecoms industry working for a number of large and medium 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, he has assisted the development and test of the in-the-field ED-137 Radio and Telephone Emulator applications now seen as vital test instruments as VoIP-in-ATM technology is being rolled-out in an operational environment.

He currently supports the FAA in the NextGen Segment 2 Radio program as well as assisting Global ANSPs in development of VCS/Radio/Network requirement specifications.




ED-137C Telephone /Radio theory plus Practical Lab using ED-137B Emulators & ED-138 Performance Monitoring Tool

Training Documentation

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

Course Description

Day 1 starts with a technology and functionality overview of today's operational telephone and radio networks. It then provides in-depth details about the added benefits and functionality that VoIP brings to the ATS Ground Telephone network including all the new functionality and Call Types introduced within ED-137C.

Day 2 provides in-depth details about the Added benefits and functionality that VoIP brings to ATS Radio Equipment including all the enhancements introduced by ED-137C as well as the Radio Gateway overview. It also examines network performance aspects, Network configuration and a review of the SESAR 15.2.10 performance measurements.

Day 3 is a Lab Session using for the ED-137B Radio Emulator tool as well as the ED-138 network performance monitoring tool. A series of networked PCs running the Radio Emulator application are configured as either CWP/VCS or GRS Radio, allowing the participants to execute a series of test scenarios in order to learn how to configure the emulator tool and its functionality. If the client has Aeronautical ED-137B radios, then test scenarios with the tool (configured as VCS/CWP) connected to the radio can be performed.

Day 4 will be a Lab Session using for the ED-137B Telephone Emulator tool as well as the ED-138 network performance monitoring tool. A series of networked PCs running the Telephone Emulator application allow the participants to execute a series of test scenarios in order to learn how to configure the emulator tool and its functionality.
A Wide Area Network (WAN) emulator application allows impairments to be introduced over the network, allowing the ED-138 Performance measurement tool to observe the effects introduced on both signalling and Voice.

Benefits for course participants

This course has been delivered successfully to staff of several ANSPs in 2016 and explains the Radio and Telephone theory considered essential prior to the hands-on practical experience in the configuration and use ATC voice equipment using emulators. The Emulator tools are being used during Factory Acceptance Tests as well as during the in-the-field installation phase. The 4-day course includes the following modules:

- ☎ ATS Ground Voice Network Overview
- ☎ Added benefits VoIP technology brings to ATS Radio/Telephone Network
- ☎ Network Performance & Supervision aspects
- ☎ Network Emulator/Performance measurements
- ☎ 2 –day ED-137B Radio/Telephone Emulator Lab also using ED-138 Performance Measurement Tool

Course suitable for

Professionals within various sectors of the aeronautical industry who need to develop a greater understanding of the functionality of the next generation of Voice Communication Systems, Ground Radio Stations and at the same time gain essential practical experience through use of the emulators and network performance monitoring tools. Due to the practical nature of this course, a maximum class size possible is 10 participants.

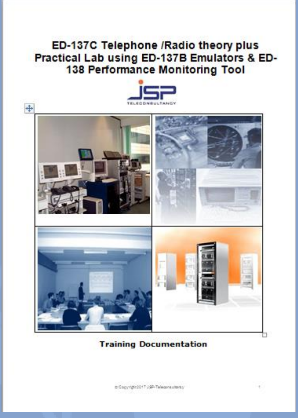
Course duration:
6 hours per day with 10 or 20 minute breaks every hour as defined below:

HOUR 1	09.00 - 10.00
BREAK	10.00 - 10.20
HOUR 2	10.20 - 11.20
BREAK	11.20 - 11.30
HOUR 3	11.30 - 12.30
LUNCH	12.30 - 13.30
HOUR 4	13.30 - 14.30
BREAK	14.30 - 14.40
HOUR 5	14.40 - 15.40
BREAK	15.40 - 16.00
HOUR 6	16.00 - 17.00

The times can be altered to suit client's particular requirements.

The Course language is English. Course participants supplied with a course book in two parts containing coloured slides & accompanying technical notes, a USB key/ lanyard containing course material.

Certificates awarded on full attendance and successful completion of course.



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JSP TELECONSULTANCY

Detailed outline DAY 1

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

1	ATS Ground Voice Network Overview	2	Added benefits VoIP brings to ATS Ground Telephone Network
	120 minutes		240 minutes

Module 1: ATS Ground Voice Network overview

- Radio network infrastructure overview
- Radio Access Modes of Operation
- Push-To-Talk
- Incoming Aircraft Call Detection/Squelch Signal
- Radio Features – Cross-coupling (Retransmission)
- Radio Features – Cross-coupling-Duplex and Simplex transmission
- Radio Features - Cross Coupling chains cause frequency blocking
- Radio Features - Best Signal Selection (Rx Voting)
- Best Transmitter Selection (Transmitter voting)
- Radio Features - Multicarrier Offset Transmission (Climax)
- Stepped-on radio transmissions
- Main/Standby Radio switchover
- Telephone network infrastructure overview
- Single Destination Direct Access (DA/IDA) call operation
- European Instantaneous Access (IA) call operation
- Priority DA call operation
- Multi-destination DA/IDA call
- Single & Multi-destination Voice Call
- Single & Multi-destination Override Call
- Airservices Australia IA call operation
- Single Common Frequency Radio Intercom Call
- Multiple Common Frequency Radio Intercom Call
- Broadcast Conference Call
- Pre-set Conference Call
- Meet-me Conference Call

Module 2: Added benefits VoIP brings to ATS Ground Telephone network

- SIP Request Messages/ SIP Responses Messages (classes)
- SIP User Agents/ SIP User Agent Registration
- SIP Network Servers
- SIP session establishment between VCSs
- Example of a SIP Direct Access Call
- Telephone Call & Conference Call types (SIP Subject and Priority Headers)
- WG-67 Version Header Field
- Proposed VCS address scheme and its operation
- SDP Media session attribute negotiation
- G.711.1 Wideband codec (RFC 5391)
- G.711.1 (PCM-WB) Offer- Answer
- Source Identifier (SID) list and SID sdp attribute
- Example of sid exchanges & local sid lists for OVR calls established in forwards direction downstream
- Echo Prevention Identifier (EPID) list and EPID SDP attribute
- Echo Prevention Identifier example (epid)
- SIP Call Detail –INVITE/180 Ringing/ 200(OK)/ ACK/ BYE/ 200(OK)
- Network wide supplementary services
- Multi-destination DA/IDA call (New Call Type –ED-137C)
- Single and Multi-destination Voice Call (New Call Type –ED-137C)
- Single and Multi-destination Override Call
- Single and Multiple Common Frequency Radio Intercom Call
- Presence Event package: (RFC 3856)
- Dialog Event package: (RFC 4235)
- Endpoint connectivity verification -OPTIONS ping
- Position Monitoring
- Conference- ED-137 Volume 1 Radio document
- Broadcast, Pre-set and Meet-me Conference
- Call Intrusion

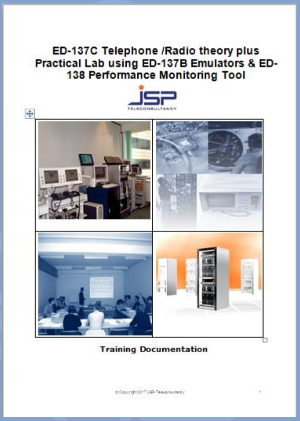
Module 3: Added benefits VoIP brings to ATS Ground Radio network

- Call Establishment Pilot – Controller (Air – Ground with Best Signal Selection)
- SIP session establishment to Transceiver (TxRx)
- SIP session establishment to Transmitter (Tx) & Receiver (Rx)
- GRS Reject & WG67 Reason Header values
- GRS placed in maintenance mode
- Radio SIP Headers & SDP message body attributes
- Media Session attribute negotiation (Mandatory SDP attributes)
  - SDP attribute <send-receive mode> example
  - Codec negotiation example
  - Unsupported Voice Codec example
  - type:<call type> example
  - R2S-KeepalivePeriod example
  - R2S-KeepaliveMultiplier example- LocalHoldTime expiry at GRS and VCS
  - ptt-id:<PTT identity> attribute example
- GRS rejects second coupling session
- SELCAL selective-calling radio system (ED-137C)
- SELCAL tone transmission (ED-137C)
- Media Session attribute negotiation (Optional SDP attributes)
  - Connection type -txrxmode example
  - bss:<BSS quality index method>

Summary of first day- Questions and Answers session

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
HOUR 3	11.30 - 12.30
LUNCH	12.30 - 13.30
HOUR 4	13.30 - 14.30
BREAK	14.30 - 14.40
HOUR 5	14.40 - 15.40
BREAK	15.40 - 16.00
HOUR 6	16.00 - 17.00



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Detailed outline DAY 2

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

3	Added benefits VoIP brings to ATS Ground Radio network	4	Network Performance & Supervision aspects	5	Network Emulator/Performance measurements
	270 minutes		45 minutes		45 minutes

Module 3 continued: Added benefits VoIP brings to ATS Ground Radio network

- Sigtime -Signalling Info Time Period
- PTT-rep example for PTT ON/OFF transition
- PTT-rep example for SQU ON/OFF transition
- Frequency ID -fid example
- Multimode radio Frequency change <DisconnectMode> SDP attribute
- WG67 KEY-IN Event package
- SDP attribute – rtphe:<RTPHE Version> example
- What is Linked Session Management?
- Linked Session protection SDP attribute
- Linked Session execute SDP attribute
- Linked-Session Management-Main/Standby Switchover
- Linked and Non-linked sessions
- Cross coupled group problem for Main to Standby Switchover
- Backup Cross Coupling Solution
- Multicast (ED-137C)
- Successful Multicast Rx session establishment between VCS & GRS transceiver/receiver endpoints
- Multicast session establishment with GRS example

- Max no. of SIP Sessions–PTT-id assignment
- Non-coupling session pre-emption
- Coupling Session pre-emption
- Transmitter- PTT lockout/ Summation
- Proposed GRS Address scheme
- RTP Header and RTP Header extension structure
- RTP Header Extension principle
- PTT-type and PTT-id echo back
- PTT ON/OFF transmission timing diagram
- GRS audio lockout mode using Normal/Priority/Emergency PTT-types
- GRS audio summation mode using Normal/Priority/Emergency PTT-types
- Incoming Aircraft call to GRS receiver
- PTT-ON for Radio-TxRx session to GRS receiver
- Incoming Cross-coupled frequency retransmission distinguished from Aircraft Call
- Coupling PTT Interruption/Summation
- A/C 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 and Measurement Answer Messages
- Dynamic Delay Compensation for Climax/ Rx Voting Groups
- Climax Time-Delay –Compensation mechanism
- 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 4: Network performance aspects

- Call performance criteria for DA and IA calls
- On-way Voice Delay for Telephone
- One way Voice Delay for Radio
- 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 duration to IP Bandwidth
- Calculating Equivalent Bandwidth required in IPv6 networks
- Calculating IPv6 Equivalent Bandwidth required to Radio
- Supervision Model- (ED-137B-Volume 5 Supervision)
- ED-137B/5 Supervision- Monitoring functions
- Supervision –SNMP data flow
- VCS Supervision Structure within MIB
- Basic monitoring and control of radios

Module 5: Network Emulator/Performance Measurements

- WAN emulator and Network simulation
- Network Impairment emulation
- Voice delay measurement of WAN
- Call performance test infrastructure
- SESAR 15.2.10 D12 Telephony/Radio result overview

Summary of 2nd day- Questions and Answers session

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
HOUR 3	11.30 - 12.30
LUNCH	12.30 - 13.30
HOUR 4	13.30 - 14.30
BREAK	14.30 - 14.40
HOUR 5	14.40 - 15.40
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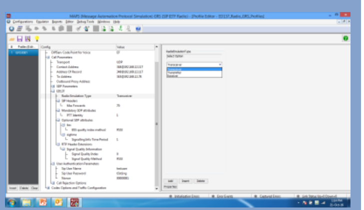

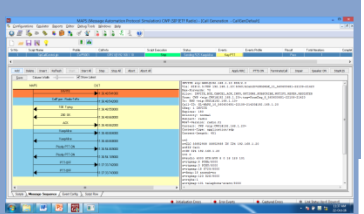

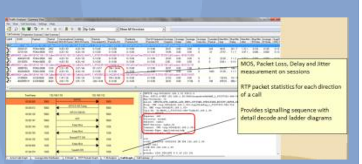

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Detailed outline DAY S 3 and 4

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

6	ED-137B Radio Emulator Lab	7	ED-137B Telephone Emulator Lab	8	ED-138 Performance Measurement Tool Lab
	300 minutes		300 minutes		120 minutes

Module 6: ED-137B Radio Emulator Lab.

- Overview of ED-137B Radio Emulator's features and functionality
- Test Bed Setup
- Configuring the Radio Profile
- Radio Emulator Lab Exercises
- **TC1**- Normal Session start to GRS – Key PTT - Talk – Unkey PTT – Terminate Session
- **TC2**- Emergency Session start to GRS – Key PTT –Talk- Unkey PTT – Terminate Session
- **TC3**- Normal Session start to GRS – Start SQU - Talk – Stop SQU – Terminate Session
- **TC4** - Session Start - StopR2S by CWP, Session Start - Key PTT–StopR2S by CWP
- **TC5** - Session Start - StopR2S by GRS, Session Start - Start SQU –StopR2S by GRS
- **TC6** – Set PTT-id value verification
- **TC7** – Start Session to GRS in maintenance mode. GRS placed in maintenance mode after session established.
- **TC8** – GRS placed in maintenance mode during PTT keying or SQU-ON
- **TC9** – Enable TxRxmode SDP optional attributes in CWP profile editor, Start Session to GRS
- **TC10** – Start Session to non matching Fid- Rejected. Start Session to matching fid- Change GRS Frequency – GRS Terminates Session
- **TC11** – Session request from Unrecognised/ Recognised address rejected/accepted. Session request from unrecognised address with GRS permissions disabled accepted
- **TC12** – Session request. CWP sends PTTS-ON/OFF. GRS sends PTT-ON/OFF, GRS sends SCT-ON/OFF
- **TC13** – Session request. CWP sends PTTM-ON.
- **TC14** – Session request. CWP sends G.729 codec
- **TC15** – CWP sends RTP HE – Climax Relative/ Absolute Time Delay
- **TC16** – GRS sends RTP HE – Best Signal Selection method and SQI information.
- **TC17** – Call Rejection Options- Auto Failure Options
- **TC18** – Multiple Session establishment up to Global max session limit – further normal sessions rejected, Emergency session cause pre-emption
- **TC19** – Two normal coupling sessions to GRS. Emergency coupling session pre-empts normal coupling session.
- **TC20** – Multiple Sessions – PTT Hierarchy
- **TC21** – WG67 KEY-IN event package
- **TC22** – User Defined Traffic Action – Send File- Send Tones- Send Digits- Talk
- TC23 – Impairments – Packet Loss- Latency and Packet Effects from CWP to GRS
- TC24 – Impairments – Packet Loss- Latency and Packet Effects from GRS to CWP
- TC25 – Load Generation from CWP to GRS using Fixed Statistical distribution
- TC26 – Load Generation from CWP to GRS using Ramp Statistical distribution

Module 7: ED-137B Telephone Emulator Lab.

- Overview of ED-137B Radio Emulator's features and functionality
- Test Bed Setup
- Configuring the Telephone Profile
- Telephone Emulator Lab Exercises
- TC1- Instantaneous Access Call from CWP1 to CWP 2 – Terminate Session
- TC2- Priority DA/IDA call from CWP1 to CWP- Terminate Session
- TC3- Tactical/Strategic/General Purpose DA/IDA calls from CWP1 to CWP2– Terminate Session
- TC4- Position Monitoring – Combined A/G+G/G, G/G only, A/G only from CWP1 to CWP2– Terminate Session
- TC5 – Session request. CWP sends G.729 codec
- TC6 – OPTIONS Ping
- TC7 – Call Rejection Options- Auto Answer Failure Options- Manual Answer failure options
- TC8- Multiple Call Types from CWP1 to CWP 2 – Terminate Sessions
- TC9 – User Defined Traffic Action – Send File- Send Tones- Send Digits- Talk
- TC10 – Impairments – Packet Loss- Latency and Packet Effects from CWP1 to CWP2
- TC11 – Load Generation from CWP1 to CWP2 using Fixed Statistical distribution
- TC12 – Load Generation from CWP1 to CWP2 using Ramp Statistical distribution

Module 8: ED-138 Performance Measurement Tool Lab.

- Performance Monitoring Tools
- Solutions for VoIP in ATM
- Application- Architecture
- Passive Recording
- Verify adherence to Protocol Standards
- Signalling and Traffic Summary
- Record Voice to File, Play to Speaker
- Monitors QoS on voice calls
- Measures Network Impairments
- Triggers Action based on call parameters
- Multiple Probes for Centralized QoS Monitoring and Reporting
- Web based Centralized Monitoring Tool
- Centralized Remote Monitoring- Reports
- Centralized Remote Monitoring- Alarm and Logging

Summary of 3rd /4th day- Questions and Answers session