ETSI CTI Plugtests Report <1.0.2> (2009-04)

2nd EUROCAE Plugtests™ Interoperability Event on VoIP for ATM; Sophia Antipolis, France; 25th March to 3rd April 2009

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2 References

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area**.** For non-specific references, the latest version of the referenced document (including any amendments) applies.

[i.1] **EUROCAE** ED-136: "Operational and Technical Requirements", December 2008

[i.2] **EUROCAE** ED-137 "Interoperability Standards for VoIP ATM components, Part 1 : Radio", December 2008

[i.3] **EUROCAE** ED-137 "Interoperability Standards for VoIP ATM components, Part 2 : Telephone", December 2008

[i.4] **EUROCAE** ED-137 "Interoperability Standards for VoIP ATM components, Part 3 : Recorder", December 2008

[i.5] **EUROCAE** ED-137 "Interoperability Standards for VoIP ATM components, Part 4 : Supervision", December 2008

[i.6] **EUROCAE** ED-138 "Network Requirement and Performance for VoIP ATM systems Part 1: Network Specification", December 2008

[i.7] **EUROCAE** ED-139 "Qualification tests for VoIP ATM Components and Systems", April 2008

3 Abbreviations

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ANSP	Air Navigation Service Provider
ATM	Air Traffic Management
ATS	Air Traffic Services
CFE	Conference Focus Entity
CWP	Controller Working Position
ED	EUROCAE Document
ETSI	European Telecommunications Standards Institute
GRS	Ground Radio Station
HE	Header Extension
IP	Internet Protocol
fid	frequency identification
LSB	Least Significant Bit
MD5	Message-Digest algorithm 5)
MYSQL	Relational database management system
MSB	Most Significant Bit
NA	Not Applicable
NO	Not OK
OK	OK
OT	Out-of-Time
PCM	Pulse Code Modulation
PHP	Personal Home Page
PT	Payload Type
PTT	Push-To-Talk
PTT-id	Payload Type- Identification
Rx	Receive
RFC	Request For Comments

RTP	Real Time Protocol
RTSS	Real Time Session Supervision
RTSSP	Real Time Session Supervision Protocol
SDP	Session Description Protocol
SES	Single European Sky
SIP	Session Initiation Protocol
SQL	Squelch
SSL	Secure Sockets Layer
TSR	Test Session Report
Tx	Transmit
UA	User Agent
URI	Uniform Resource Indicator
UTC	Universal Time Coordinate
VCS	Voice Communication System
VF	Visibility Flag
WG67	Working Group 67

4 Second EUROCAE Plugtests™ Interoperability Event on VoIP for ATM

The second EUROCAE Plugtests™ Interoperability Event on VoIP for ATM (Air Traffic Managment) held at the ETSI headquarters in Sophia Antipolis between 25th March and 3rd April 2009 had the scope of progressing onwards from the first Plugtests™ event that covered the simpler Stage 1 interoperability tests held in April 2008 to the more complex Stage 2 interoperability tests that had the scope of performing further mandatory SIP Telephone and Radio interface interoperability tests as defined by the draft EUROCAE ED-139 document in order to verify their correct functionality over a Local Area Network.

Following on from the formal approval of EUROCAE Documents (ED) 136, 137 and 138 documents by the EUROCAE council in February 2009, the second Plugtests™ event also had the scope of providing feedback to issues regarding signalling protocol definition, parameter configuration and Air Traffic Service feature functionality defined within ED Telephone and Radio documents that appear to require further clarification in order to make the interworking between systems more robust.

Several new companies which had not participated in the previous Stage 1 event held in April 2008 were given the opportunity to perform Stage 1 tests in the week prior to the second Plugtests™ event, from 25th March to 27th March.

The results of the multiple interoperability test scenarios achieved by the European (7 Voice Communication System and 4 Ground Radio Station) suppliers have demonstrated a high rate of success:

- Interoperability VCS-VCS : 95,7% (423 tests OK for 442 run)
- Interoperability VCS-GRS : 95,6% (483 tests OK for 505 run)

These results show that the VoIP call types and the wide range of ATS (Air Traffic Services) features specified by the ED 137 interoperability documents, supporting the Operational and Technical Requirements defined by the ED 136 document have now been developed and implemented by the main European VCS and Radio Suppliers with a high level of interoperability achieved. This will lead to ATM VoIP VCS and GRS deployment by ANSPs (Air Navigation Service Providers) in the very near future for operational use in the framework of the Single European Sky (SES).

A list of recommendations to be considered for the enhancement of the ED137 Part 1 Radio and ED137 Part 2 Telephone documents have also been produced. These recommendations will be examined by the EUROCAE WG67 review team with the scope of enhancing future editions of the document.

4.1 Build-up to the event

There was active involvement by all VCS and GRS players in the build-up to 2nd Plugtests™ event. This was achieved through a series of several teleconferences organised by the ETSI Plugtests™ team and also through a very active Plugtests™ email reflector.

Initially a “Proposed test case document” with high level descriptions for each test extracted from the EUROCAE ED139 document made it easier to agree on which tests should be included within the Plugtests™ event. There was democratic involvement of all players during Test selection process eventually leading to agreement by all participants of both Mandatory and Optional tests to be included in the 2nd Plugtests™ event. This agreement was achieved 3 months prior to the Plugtests™ event itself in order to allow the vendors sufficient development time.

All the Stage 1 Telephone and Radio Plugtests™ specifications were reviewed and updated by December 2008 and all the new Stage 2 Plugtests™ specifications were developed and consigned to Participating companies in early January 2009. The feedback process and the teleconferences led to a series of new editions of the specifications in the following weeks until they were considered stable.

In order to make it easier to decipher the individual bits within the RTP header extension fields used for sessions between VCSs and Radios, PARK AIR SYSTEMS developed a Wireshark dissector plug-in and made the plugin available to other companies participating in the event, which made the task of testing much easier. PARK AIR SYSTEMS is thanked by all the participants for allowing them to use the plug-in.

4.2 Reflections on the event

The 2nd Plugtests™ event had much more industry involvement than seen in the first event, with 7 VCS vendors and 4 GRS vendors attending. The tests extracted from the ED139 document were more complex than the simpler tests included in the 1st Plugtests™ event held in April 2008. Due to there being a high number of participating companies, the testing schedule was also much tighter. Every company was allocated a test session with all the other companies during the week. A VCS-VCS test session lasted 4 hours and required that up to 40 test scenarios were executed, while a VCS-GRS test session allowed 2 hours and required that up to 19 test scenarios were executed. The participating vendors worked very hard to complete the tests in the timeframe allocated. In the few cases when the test session time ran out, it was also possible to continue testing on into the evening until 9pm, thanks to ETSI keeping the facilities open in order to ensure that companies were given every chance possible to perform updates/modifications to their existing code in order to complete the programmed test sessions and increase the rate of interoperability.

Participants seemed to be happy with the test facilities available, test network and test tools (wiresharks) made available by ETSI. Each evening a wrap-up session was organised by ETSI during which it was possible to discuss a series of points-of-the-day noted either in the forum or wrap-up session web page. The Wrap-up session also used the ETSI Testing Reporting Tool to allow progress to be monitored, and see comments entered by companies as to why certain tests or test steps were having problems.

It was realised during the course of the event that the companies have really made huge progress in their implementations since the previous Plugtests™ event held in April 2008. Also further progress was also made by all Suppliers during event as tests were performed. A great group spirit was observed during the event with all players working towards a common objective to obtain interoperability. Many useful comments were also accumulated during the event about the test specifications and feedback was also collected on the EUROCAE base specifications. These points are explained in further detail later on in this report.

4.3 Test Session Schedule

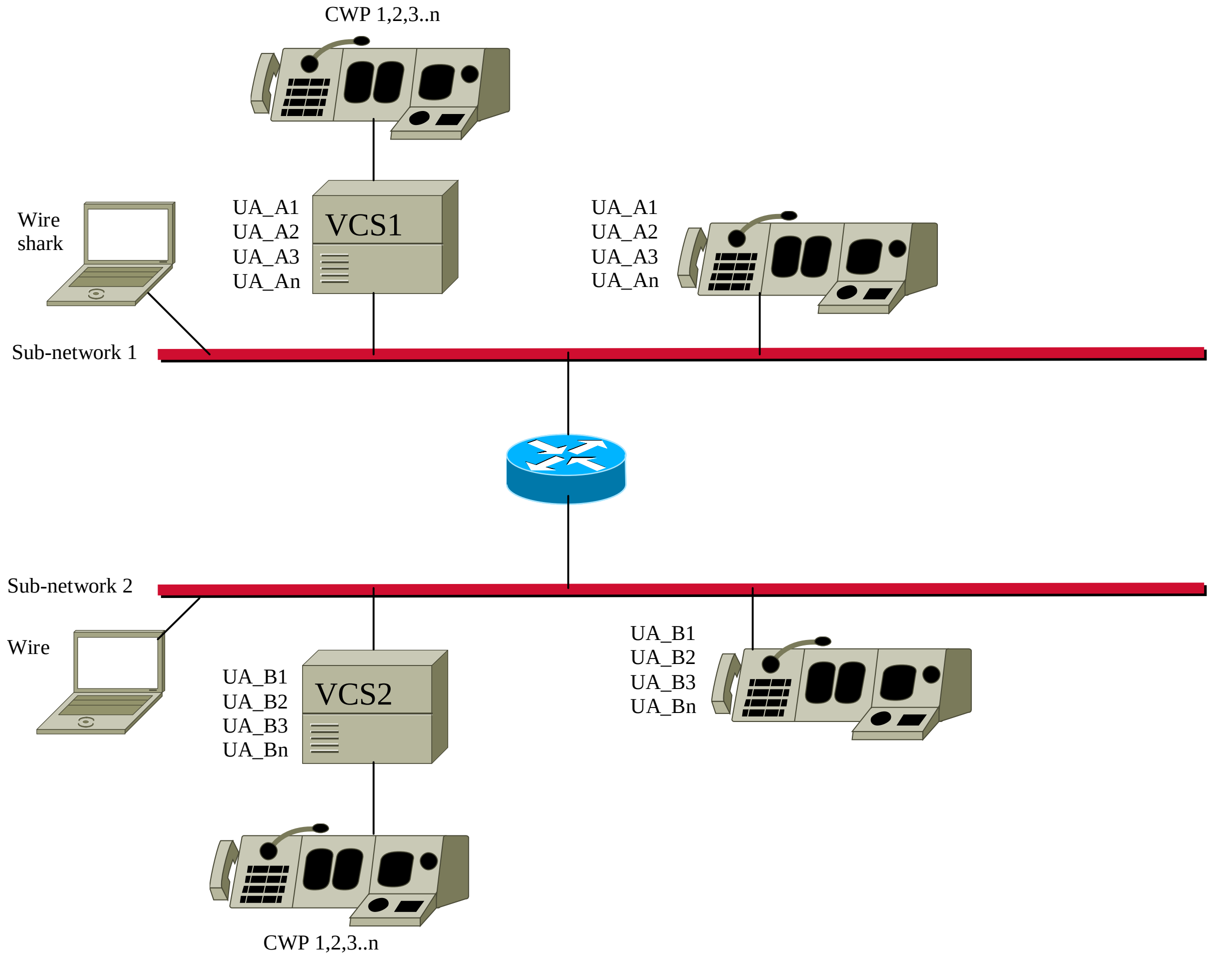
The test schedule indicated which companies were testing with which, when, and where. Each vendor tested once in a test session against every other vendor. It was assumed that at least one test engineer was accompanying each scheduled equipment who knew how to operate it. The Test Schedule was subject to revision at the end of every day and was updated in the case that vendors needed more time in the evenings to complete test sessions.

4.4 Interoperability Test Sessions

The objective of each test session was to execute as many tests from the EUROCAE test specification as possible. The Test execution focused on test execution and observation. For each test case execution traces were captured but these were not analyzed during the test session. The results of each interoperability test session were recorded by the participants themselves. Prior to each test session between two vendors one person in the participating teams was selected to be the test session secretary. After each test execution the interoperability result was agreed amongst both vendors and was then recorded by the secretary. After each test session was completed the Test Session Report was submitted to ETSI.

4.5 Ground Telephone Interoperability Test configuration (Stage 1 & 2)

The test configuration in the Figure below shows up to four SIP Telephone User Agents in sub-network 1 and up to four SIP Telephone User Agents in sub-network 2. The User Agents can either be integral to the VCS (using a Single or Multiple IP address scheme) or as separate SIP Telephone User Agents i.e. CWPs (using a multiple IP address scheme). It should be noted that for the latter case these test procedures treat a CWP as a VCS.

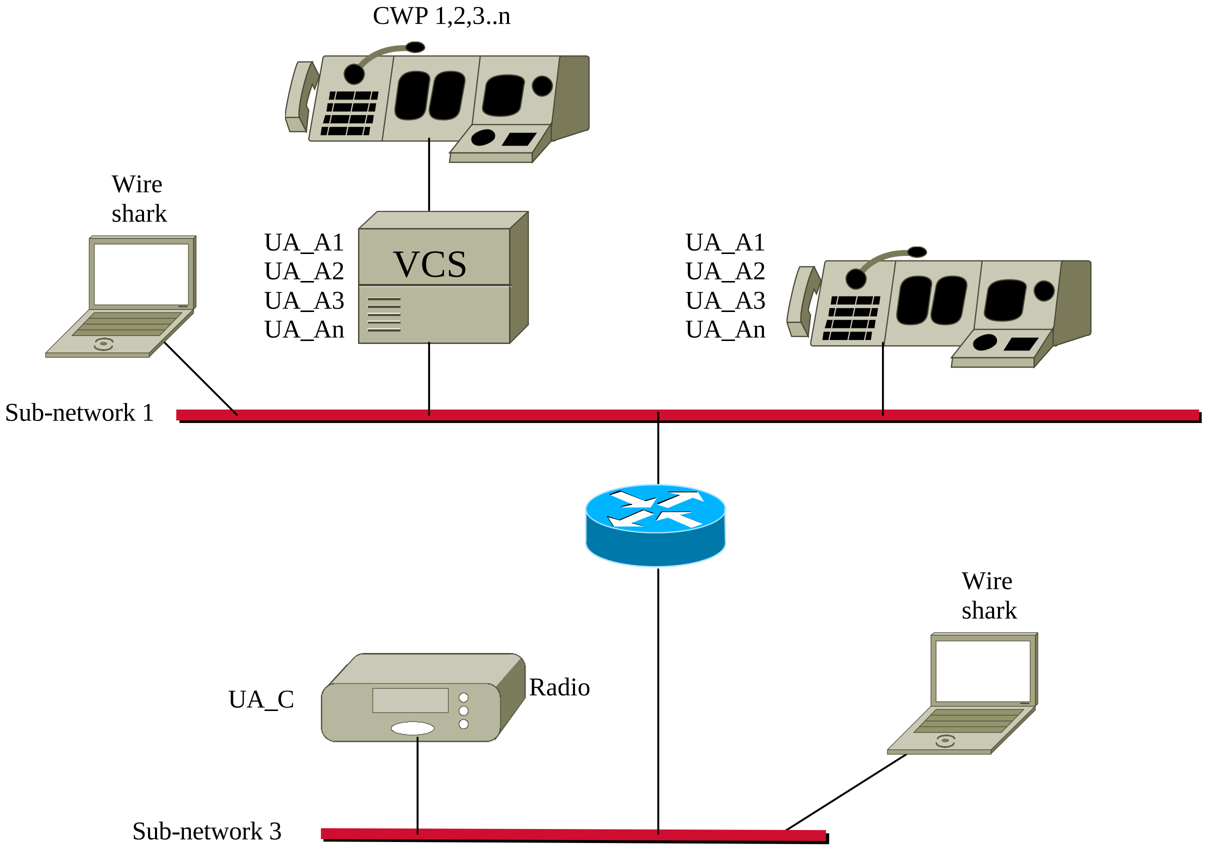


4.6 Ground Radio Interoperability Test configuration (Stage 1)

The test configuration in the Figure below shows up to three SIP Telephone User Agents in sub-network 1 that can either be integral to the VCS (using a Single or Multiple IP address scheme) or as separate SIP Telephone User Agents i.e. CWPs (using a multiple IP address scheme). It should be noted that for the latter case these test procedures treat a CWP as a VCS.

The test configuration also shows one radio in Sub-network 3, having one SIP Radio User Agent assigned a single SIP URI address.

It should be noted that the same test configuration applies to all tests.



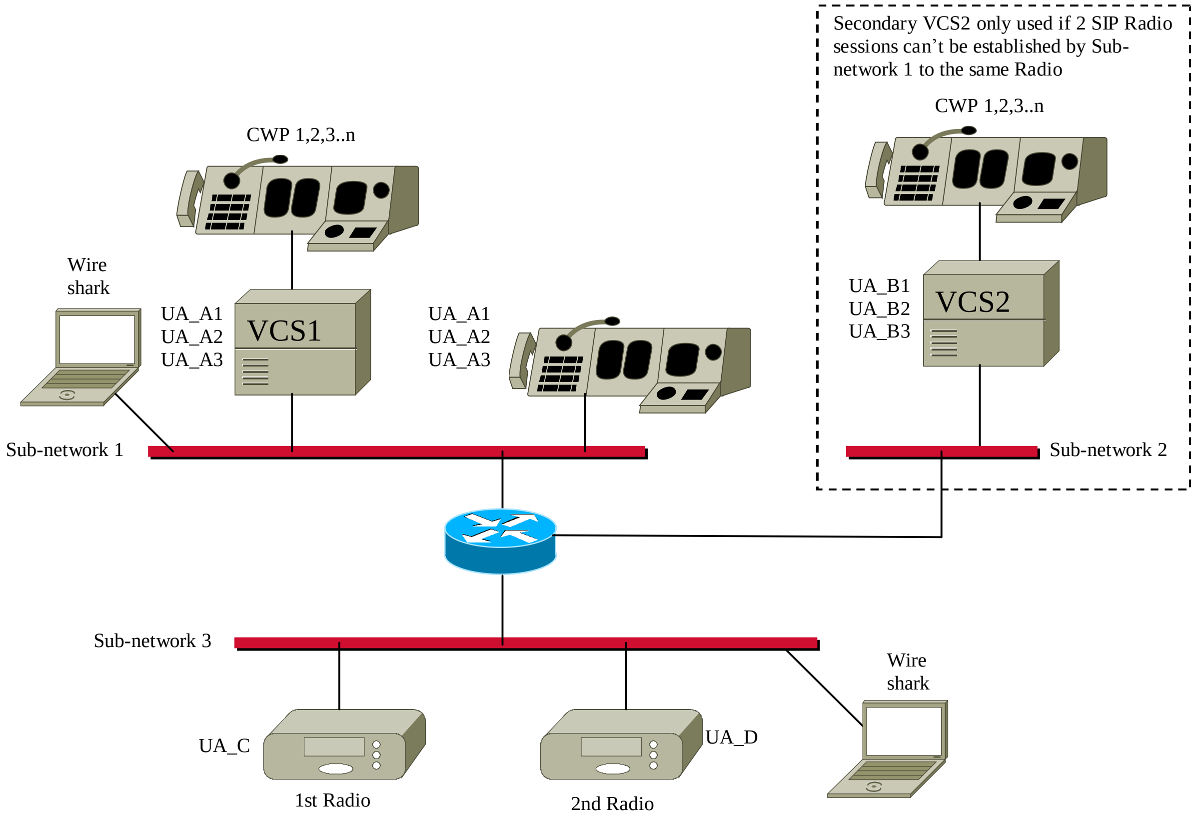
4.7 Ground Radio Interoperability Test configuration (Stage 2)

The test configuration in the Figure below shows up to three SIP Telephone User Agents in sub-network 1 that can either be integral to the VCS (using a Single or Multiple IP address scheme) or as separate SIP Telephone User Agents i.e. CWPs (using a multiple IP address scheme). It should be noted that for the latter case these test procedures treat a CWP as a VCS.

The SIP Telephone User Agents defined in Sub-network 2 will only be employed in the case that 2 SIP Radio sessions can't be established from Sub-network 1 to the same radio ,

The test configuration also shows two radios in Sub-network 3, each with one SIP Radio User Agent assigned a single SIP URI address. Each of the two Radios used will have different IPv4 host addresses.

It should be noted that the same test configuration applies to all tests.



4.8 Scope of Ground Telephone Stage 1 interoperability tests

This section defines a series of SIP signalling interoperability tests between 2 systems (and multiple systems) located in the same room across a LAN.

All the tests should be performed twice with each participant having the role of A side.

The tests have been divided into the following groups:

§ Direct Access Routine Call tests over LAN coverage list (6)

§ Direct Access Priority Call tests over LAN coverage list (2)

§ Instantaneous Access Call tests over LAN coverage list (8)

§ SDP codec description tests (1)

§ SIP Call combination tests (3)

4.9 Scope of Ground Telephone Stage 2 interoperability tests

This section defines a series of SIP signalling interoperability tests between 2 systems (and multiple systems) located in the same room across a LAN.

All the tests should be performed twice with each participant having the role of A side.

The tests have been divided into the following groups:

- Session Description Protocol (SDP) tests (3)
- SIP Supplementary Service tests (4)
- SIP Broadcast conference Supplementary Service tests (2)
- SIP Call Intrusion supplementary service tests (9)

4.10 Scope of Ground Radio Stage 1 interoperability tests

This section defines a series of SIP-SIP signalling tests between 2 systems located in the same room across a LAN.

A summary of the 9 **Stage 1 SIP Ground Radio interface interoperability tests** to be performed by each VCS supplier and Radio Supplier are provided below:

§ Send/Receive SIP Radio session establishment/disconnection at nominated frequency, voice codec etc from VCS to Radio

§ Receive only SIP Radio session establishment/disconnection at nominated frequency, voice codec etc from VCS to Radio

§ Transmit only SIP Radio session establishment/disconnection at nominated frequency, voice codec etc from VCS to Radio

§ Send/Receive SIP Radio session request to an invalid frequency

§ Send/Receive SIP Radio session request from unknown calling party address

§ Normal PTT activation, Voice transmission, PTT deactivation

§ PTT deactivation on PTT keep-alive timer expiry at Radio side

§ Squelch activation, Voice transmission, Squelch deactivation

§ Squelch deactivation on Squelch keep-alive timer expiry at VCS side

4.11 Scope of Ground Radio Stage 2 interoperability tests

This section defines a series of SIP-SIP signalling tests between 2 systems (and multiple systems) located in the same room across a LAN.

The tests have been divided into the following groups:

§ SIP Radio session establishment interoperability tests over LAN (2 tests)
§ Session Description Protocol (SDP) interoperability tests over LAN (2 tests)
§ Push to Talk (PTT) and SIP Radio session pre-emption interoperability tests over LAN (11 tests)
§ Best Signal Selection (BSS) tests over LAN (2 tests)
§ Simultaneous Call Detection tests over LAN (1 test)
§ PTT identity notification test (1 test) - Optional
§ Climax Time Delay test (1 test)- Optional

4.12 EUROCAE PLUGTEST event network topology

Eurocae Plugtests Event
Network Topology

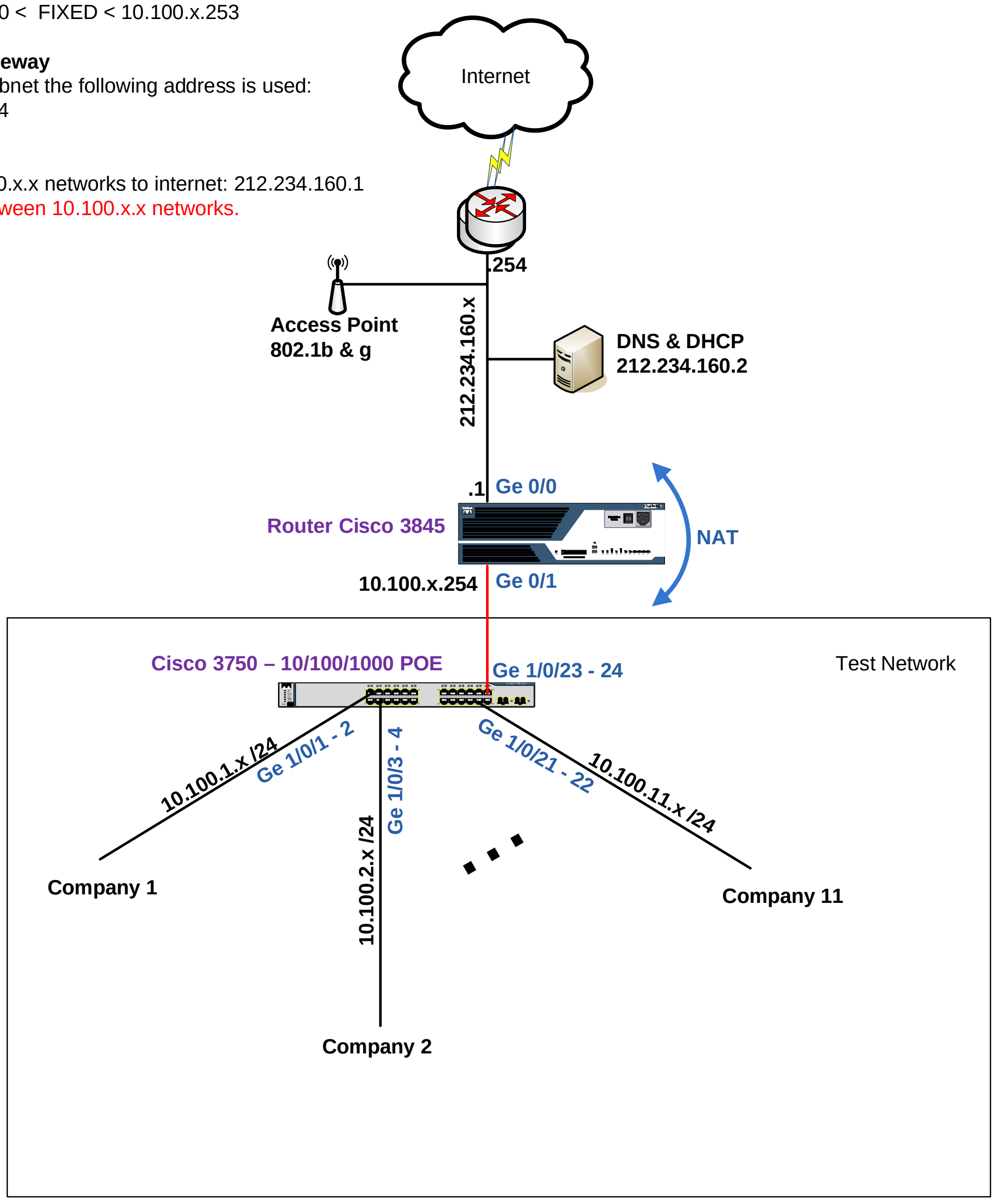
DHCP
Wifi: 212.234.160.50 to 212.234.160.95
For each subnet the following range of IPv4 address is distributed:
10.100.x.1 to 10.100.x.99

DNS
Domaine plugtests.net

FIXED
For each subnet the following range is available:
10.100.x.100 < FIXED < 10.100.x.253

Default Gateway
For each subnet the following address is used:
10.100.x.254

NAT
From 10.100.x.x networks to internet: 212.234.160.1
No NAT between 10.100.x.x networks.



4.13 PLUGTEST™ Result summaries

During the event it was observed that Interoperability was well achieved. In a few cases however there were conformity issues with the ED and relevant RFCs observed.

This section has the scope of providing a summary of the results recorded for each of the 4 PLUGTEST™ specifications used by participants during the event.

4.13.1 Stage 1 Telephone Plugtests™ result summary

Number of tests per test session: 20
Number of Sessions: 6
Of the 6 reported sessions 6 were agreed (100.0%)

All results in the following includes non-agreed sessions

Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
106 (88.3%)	14 (11.7%)	0 (0.0%)	0 (0.0%)	120 (100.0%)	120

Total: 120

Results per Group

	Interoperability		**Not Executed**		**Totals**	
Group	**OK**	**NO**	**NA**	**OT**	**Run**	**Results**
DA Routine	36 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	36 (100.0%)	36
DA Priority	12 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	12 (100.0%)	12
SIP IA	42 (87.5%)	6 (12.5%)	0 (0.0%)	0 (0.0%)	48 (100.0%)	48
SDP	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
SIP Call Combi	10 (55.6%)	8 (44.4%)	0 (0.0%)	0 (0.0%)	18 (100.0%)	18

Results per Test

		Interoperability		Not Executed		Totals	
Group	Test Id	OK	NO	NA	OT	Run	Results
DA Routine	LAN-BC-R1	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-R2	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-R3	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6

	LAN-BC-R4	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-R5	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-R6	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
DA Priority	LAN-BC-PC1	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-PC2	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
SIP IA	LAN-BC-IA1	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA2	5 (83.3%)	1 (16.7%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA3	2 (33.3%)	4 (66.7%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA4	5 (83.3%)	1 (16.7%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA5	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA6	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA7	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-BC-IA8	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
SDP	LAN-SDP-R5	6 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
SIP Call Combi	LAN-MIX-R1	4 (66.7%)	2 (33.3%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-MIX-R2	2 (33.3%)	4 (66.7%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6
	LAN-MIX-R3	4 (66.7%)	2 (33.3%)	0 (0.0%)	0 (0.0%)	6 (100.0%)	6

4.13.1.1 Stage 1 Telephone Plugtests™ result summary analysis

Result analysis of the Stage 1 Telephone Plugtests™ event have indicated that fairly good results have been achieved with a 88.3% overall interoperability result.

Result analysis of the Stage 1 Telephone Plugtests™ event have indicated that Direct Access Routine Call, Direct Access Priority call and SDP related tests performed extremely well. There have been interoperability problems identified with respect to Instantaneous Access Call implementations (87.5% interoperability) and the tests relevant to a mixture of routine DA calls, priority DA calls and IA calls (only 55% interoperability achieved). The main problems identified relate to audio not being monitored correctly by the IA calling user or a second IA call no longer having the capability of monitoring.

4.13.2 Stage 2 Telephone Plugtests™ result summary

Number of tests per test session: 16
Number of Sessions: 30
Of the 30 reported sessions 30 were agreed (100.0%)

All results in the following includes non-agreed sessions

Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
423 (95.7%)	19 (4.3%)	0 (0.0%)	38 (7.9%)	442 (92.1%)	480
Total:	442				

Results per Group

	Interoperability		Not Executed		Totals	
Group	OK	NO	NA	OT	Run	Results
SDP	87 (100.0%)	0 (0.0%)	0 (0.0%)	3 (3.3%)	87 (96.7%)	90
SIP Suppl	68 (84.0%)	13 (16.0%)	0 (0.0%)	9 (10.0%)	81 (90.0%)	90
Broadcast Conf	45 (91.8%)	4 (8.2%)	0 (0.0%)	11 (18.3%)	49 (81.7%)	60
Call Intrusion	223 (99.1%)	2 (0.9%)	0 (0.0%)	15 (6.3%)	225 (93.8%)	240

Results per Test

		Interoperability		Not Executed		Totals	
Group	Test Id	OK	NO	NA	OT	Run	Results
SDP	LAN-SDP-R4	29 (100.0%)	0 (0.0%)	0 (0.0%)	1 (3.3%)	29 (96.7%)	30
	LAN-SDP-R14	29 (100.0%)	0 (0.0%)	0 (0.0%)	1 (3.3%)	29 (96.7%)	30
	LAN-SDP-R15	29 (100.0%)	0 (0.0%)	0 (0.0%)	1 (3.3%)	29 (96.7%)	30
SIP Suppl	LAN-SS-CT1	22 (81.5%)	5 (18.5%)	0 (0.0%)	3 (10.0%)	27 (90.0%)	30
	LAN-SS-CT2	19 (70.4%)	8 (29.6%)	0 (0.0%)	3 (10.0%)	27 (90.0%)	30
	LAN-SS-LINKLOSS1	27 (100.0%)	0 (0.0%)	0 (0.0%)	3 (10.0%)	27 (90.0%)	30
Broadcast Conf	LAN-SS-CONF1	25 (100.0%)	0 (0.0%)	0 (0.0%)	5 (16.7%)	25 (83.3%)	30
	LAN-SS-CONF2	20 (83.3%)	4 (16.7%)	0 (0.0%)	6 (20.0%)	24 (80.0%)	30
Call Intrusion	LAN-SS-CI1	29 (100.0%)	0 (0.0%)	0 (0.0%)	1 (3.3%)	29 (96.7%)	30

	LAN-SS-CI3	28 (100.0%)	0 (0.0%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30
	LAN-SS-CI4	27 (96.4%)	1 (3.6%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30
	LAN-SS-CI5	28 (100.0%)	0 (0.0%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30
	LAN-SS-CI6	28 (100.0%)	0 (0.0%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30
	LAN-SS-CI7	27 (96.4%)	1 (3.6%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30
	LAN-SS-CI8	28 (100.0%)	0 (0.0%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30
	LAN-SS-CI9	28 (100.0%)	0 (0.0%)	0 (0.0%)	2 (6.7%)	28 (93.3%)	30

4.13.2.1 Stage 2 Telephone Plugtests™ result summary analysis

Result analysis of the Stage 2 Telephone Plugtests™ event have indicated that good results have been achieved with a 95.7% overall interoperability result.

The SDP and Call Intrusion Tests performed the best, while the Supplementary service tests (84% interoperability) and Multi-user Conference test performed less better (91% interoperability). The main problems identified relate to the Attended Call Transfer feature not being implemented according to the relevant RFC and the Conference focus causing the conference to be terminated when withdrawing early from a conference.

4.13.3 Stage 1 Radio Plugtests™ result summary

Number of tests per test session: 9
Number of Sessions: 9
Of the 9 reported sessions 9 were agreed (100.0%)

All results in the following includes non-agreed sessions

Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
78 (96.3%)	3 (3.7%)	0 (0.0%)	0 (0.0%)	81 (100.0%)	81

Total: 81

Results per Group

	Interoperability		Not Executed		Totals	
Group	OK	NO	NA	OT	Run	Results
Rad	78 (96.3%)	3 (3.7%)	0 (0.0%)	0 (0.0%)	81 (100.0%)	81

Results per Test

		Interoperability		Not Executed		Totals	
Group	Test Id	OK	NO	NA	OT	Run	Results
Rad	LAN-RAD-R1	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R2	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R3	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R5	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R6	8 (88.9%)	1 (11.1%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R7	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R8	7 (77.8%)	2 (22.2%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R9	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9
	LAN-RAD-R10	9 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	9 (100.0%)	9

4.13.3.1 Stage 1 Radio Plugtests™ result summary analysis

Result analysis of the Stage 1 Radio Plugtests™ event have indicated that good results have been achieved with a 96.3% overall interoperability result.

Nearly all tests performed well, but the test related to Send/Receive SIP Radio session request from unknown calling party address (88.9% interoperability) and the test related to PTT deactivation on PTT keep-alive timer expiry at Radio side (77.8% interoperability) performed less well.

The main problems identified relate to a call from an unknown SIP User agent not be rejected and the PTT activation indication still being transported to the Radio following a link disconnection/re-establishment instead of being deactivated until the subsequent selection of PTT.

4.13.4 Stage 2 Radio Plugtests[TM] result summary

Number of tests per test session: 20
Number of Sessions: 28
Of the 28 reported sessions 28 were agreed (100.0%)

All results in the following includes non-agreed sessions

Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
483 (95.6%)	22 (4.4%)	55 (9.8%)	0 (0.0%)	505 (90.2%)	560

Total: 505

Results per Group

	Interoperability		Not Executed		Totals	
Group	OK	NO	NA	OT	Run	Results
SIP Radio session	55 (98.2%)	1 (1.8%)	0 (0.0%)	0 (0.0%)	56 (100.0%)	56
SDP	56 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	56 (100.0%)	56
PTT	285 (93.8%)	19 (6.3%)	4 (1.3%)	0 (0.0%)	304 (98.7%)	308
BSS	52 (96.3%)	2 (3.7%)	2 (3.6%)	0 (0.0%)	54 (96.4%)	56
Simultaneous Trans	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
PTT identity notification	2 (100.0%)	0 (0.0%)	26 (92.9%)	0 (0.0%)	2 (7.1%)	28
Climax Time Delay	5 (100.0%)	0 (0.0%)	23 (82.1%)	0 (0.0%)	5 (17.9%)	28

Results per Test

		Interoperability		Not Executed		Totals	
Group	Test Id	OK	NO	NA	OT	Run	Results
SIP Radio session	LAN-RAD-CA1	27 (96.4%)	1 (3.6%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28

Results per Test

		Interoperability		Not Executed		Totals	
	LAN-RAD-CA2	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
SDP	LAN-RAD-SDP1	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-SDP2	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
PTT	LAN-RAD-PTT1	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT2	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT3	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT4	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT5	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT6	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT7	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT8	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-PTT9	21 (77.8%)	6 (22.2%)	1 (3.6%)	0 (0.0%)	27 (96.4%)	28
	LAN-RAD-PTT10	13 (52.0%)	12 (48.0%)	3 (10.7%)	0 (0.0%)	25 (89.3%)	28
	LAN-RAD-PTT11	27 (96.4%)	1 (3.6%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
BSS	LAN-RAD-BSS1	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
	LAN-RAD-BSS2	24 (92.3%)	2 (7.7%)	2 (7.1%)	0 (0.0%)	26 (92.9%)	28
Simultaneous Trans	LAN-RAD-SCT1	28 (100.0%)	0 (0.0%)	0 (0.0%)	0 (0.0%)	28 (100.0%)	28
PTT identity notification	LAN-RAD-PTTid	2 (100.0%)	0 (0.0%)	26 (92.9%)	0 (0.0%)	2 (7.1%)	28
Climax Time Delay	LAN-RAD-CLM	5 (100.0%)	0 (0.0%)	23 (82.1%)	0 (0.0%)	5 (17.9%)	28

4.13.4.1 Stage 2 Radio Plugtests™ result summary analysis

Result analysis of the Stage 2 Radio Plugtests™ event have indicated that good results have been achieved with a 95.6% overall interoperability result.

Most of the Stage 2 Radio tests resulted in a good interoperability rates (100% interoperability). The PTT related tests (93.8% interoperability) and the Best Signal Selection Tests (96.3% interoperability) performed worse.

The main problems identified relate to some Radios being unable to perform summation of the radio in the case that two sessions with equal ptt-types are used. Also tests relating to the pre-empting of SIP Radio session established to Radio with SIP Priority header “Normal” by SIP Radio session request with SIP Priority header “Emergency” also caused problems for some Radio vendors.

4.14 Non-interoperability (NO) reasons resolved during Plugtests™

The following list defines the main reasons for non-interoperability that were resolved during the Plugtests.

- Real Time Session Supervision Protocol not being implemented correctly due to the mandatory use of the Real Time Session Supervision Protocol being decided by Plugtests™ participants at a late stage and a series of unclear issues not being clearly understood by the Vendors when reading the ED 137 Part 1 Radio specification. A clearer method for the use of the Real Time Session Supervision was defined during the Plugtests.
- The behaviour of a system with regards to the disconnection of the Real Time Session Supervision and SIP session. Some vendors were initially disconnecting a SIP session within 60ms of a RTP packets not being received. Other vendors did not disconnect the SIP session at all on expiry of the LocalHoldTime at the end-points. It was decided that a SIP session SHALL only be disconnected following expiry of the LocalHoldTime at the endpoints following disconnection of the RTSS.
- The use of non-incrementing timestamps proved a problem for some vendors as their RTP stack required an incrementing timestamp. It was decided to use incrementing timestamps in order to resolve the problem.
- The fact that Session identities didn’t increment by 1 initially also caused problems until the issue was resolved.
- Use of incorrect SIP port numbers through negotiation or checking of Source port numbers at receive side prior to allowing a session to be established also caused problems for some vendors. It was decided not to perform a check on the source port number in order to resolve the problem.
- The inversion of the LSB and MSB bits in the RTP Header Extension by some vendors was also determined to be reason as to why interoperability could not be achieved. Once MSB and LSB bits had been swapped over this problem was resolved.
- Initially some Radio vendors did not allow two sessions to be established to the Radio. Following modification of the code by the Radio vendor, this problem was resolved.
- In the case of two User Agents with sessions established to the same radio, activating different PTT levels at the same time, initially some Radio vendors were not sending the PTT-id and ptt-type of the User Agent accessing the Radio Transmitter to the User Agent that was not given access.

4.15 Main non-interoperability reasons during Plugtests™

The following list defines the main reasons for non-interoperability that were not resolved by all vendors during the Plugtests.

- Not being able to perform audio summation at radio with when equal ptt-types received
- Unable to perform pre-emption of established sessions by Radio

- Call intrusion feature not implemented according to ED137 Part 2 Telephone
- Call Transfer Attended feature not implemented according to RFC 5359
- Focus leaving conference while at least 2 other participants present, should cause conference to remain (as per ED136 definition).

4.16 Recommendations to ED137 Part 1 Radio for consideration

4.16.1 Change of Real Time Session Supervision acronym

It is noted that use of the acronym ”R2S” to imply “REAL TIME SESSION SUPERVISION PROTOCOL” can be confusing due to this term being used in previous editions of the ED137 Part 1 Radio document to imply “Raw RTP session” defining a different protocol. It is therefore recommended that the acronym “RTSS” is used to represent the “REAL TIME SESSION SUPERVISION” and the acronym “RTSSP” us used to represent the “REAL TIME SESSION SUPERVISION PROTOCOL”.

4.16.2 Mandatory use of Real Time Session Supervision Protocol

The employment of the REAL TIME SESSION SUPERVISION PROTOCOL (RTSSP) should be defined as mandatory between VCS and Radios. The REAL TIME SESSION SUPERVISON PROTOCOL (RTSSP) SHALL be used to establish sessions between SIP USER AGENTS at the VCS endpoint and the Radio Endpoint.

In the case of a Radio Transceiver containing both transmitter and receiver in the same unit, a send/receive RTSS SHALL be established between VCS endpoint and Radio Endpoint.

In the case of a Radio Transmitter only a send/receive RTSS SHALL be established between VCS endpoint and Radio Endpoint.

In the case of a Radio Receiver only a send/receive RTSS SHALL be established between VCS endpoint and Radio Endpoint.

4.16.3 Use of only send/recv SDP attribute to establish Real Time Session Supervision

Due to the SIP User Agent of the Radio Transceiver, Radio Transmitter or Radio Receiver requiring a two-way session to be established with the SIP User Agent at the VCS endpoint, the use of the **<send-receive mode**> SDP attribute set to **“**sendrecv” becomes mandatory. The use of the **<send-receive mode>** SDP attribute set to **“**sendonly” or “recvonly” SHALL not be used. The <send-receive mode> SDP attribute currently defines 3 values (i.e. recvonly, sendonly and sendrecv) should now only defined one value of sendrecv. The Table 6 defined in ED137 Part 1 Radio should be modified to reflect these changes.

It should also be stated that in the case a **<send-receive mode>** SDP attribute is not present, it SHALL be assumed that its value is set to “Sendrecv”.

4.16.4 Extension of CALL-TYPE attribute to described type of session

Due to the fact that the **<send-receive mode> SDP attribute** SHALL be set to “sendrecv” only, this implies that there is no longer information being sent to the Radio that describes the type of session (i.e. Transmit only, Receive only or Transmit/Receive).

It is recommended that the EUROCAE SG2 Radio should decide if the **type: <call-type>** SDP attribute be extended to include the following values:

Attribute	Parameter	Values
a=	type:<call-type>	radio-Rxonly radio-Txonly radio-TxRx coupling

Or in order to avoid ambiguity regarding coupling in receivers/transmitters is it better to leave the **type: <call-type>** SDP attribute unchanged in ED137 Part 1 Radio with the following values:

Attribute	Parameter	Values
a=	type:<call-type>	radio coupling

and introduce a new dedicated attribute for “transmission/reception mode” with the following values:

Attribute	Parameter	Values
a=	txrxmode:<mode-type>	TXonly RXonly TX-RX

This information is required in order to prevent a Transmit/Receive Radio session from being established to a Receive only or Transmit only Radio for example. A radio SHALL therefore check the “type” or the new “txrxmode” SDP attribute and reject the session establishment request if they do not match with the following criteria.

§ RXonly session to Radio Receiver or Radio Transceiver
§ TXonly session to Radio Transmitter or Radio Transceiver
§ TX-RX session to Radio Transceiver

Note that in the case where a VCS wants to use a transceiver as a receiver (ie RXonly) or as a transmitter (TXonly) then a Radio Transceiver should accept any type of session

4.16.5 Incrementing timestamp definition

The ED137 Part 1 Radio document shall explain the fact that the “The Real Time Session Supervision (RTSS) packets + RTP audio packets SHALL be sent as one RTP stream. Therefore there SHALL be only one SSRC and an incrementing timestamp during transitions from RTSS keep-alives to RTP audio and vice versa.

Currently the ED137 radio specification incorrectly defines the timestamp as zero. It has been demonstrated during the Plugtests™ event some RTP protocol stacks require the mandatory use of an incrementing timestamp and will reject a session if the timestamp is not providing an increasing value. As the ED137 Part 1 Radio document previously defined that the timestamp was set to zero, it is now necessary to specify the method to be employed in order to increment the timestamp in the RTP stream.

It should be noted that a proposal to correlate the timestamp to a UTC source is not in line with RFC3550 which states that "If RTP packets are generated periodically, the nominal sampling instant as determined from the sampling clock is to be used, not a reading of the system clock.
So unless the sampling clock is directly derived for the UTC source, UTC source should not be used for timestamping RTP packets.

It is therefore proposed that the timestamp generation should be defined as follows:

A VCS or Radio should employ an incrementing timestamp value equivalent to the number of voice sample in the timeperiod.

For a G.711 codec for example the timestamps with voice should be equal to the number of samples per voice packet (i.e. 160)

§ Samples=Packet Duration/Sampling rate =20ms/0.125ms (G.711 codec) =160 voice samples

While the Timestamps without Voice should be equal to the equivalent number of samples between two RTSS-keepalive periods (i.e. 1600).

§ Samples=RTSS-keepalive period/Sampling rate =200ms/0.125ms (G.711 codec) =1600 voice samples

For a G.729 codec for example the timestamps with voice should be equal to the number of samples per voice packet (i.e.1)

§ Samples=Packet Duration/Sampling rate =10ms/10ms (G.729 codec) =1 voice sample

While the Timestamps without Voice should be equal to the equivalent number of samples between two RTSS-keepalive periods (i.e. 20).

§ Samples=RTSS-keepalive period/Sampling rate =200ms/10ms (G.729 codec) =20 voice samples

4.16.6 Changes to R2S-KeepalivePeriod and R2S-KeepaliveMultiplier values

It is recommended that the term nominated R2S-KeepalivePeriod is changed to RTSS-KeepalivePeriod
It is recommended that the term nominated R2S-KeepaliveMultiplier is changed to RTSS-KeepaliveMultiplier

It is recommended that the ED137 Part 1 Radio document defines the use of default values as defined in ED137 Radio Part1 spec to be configured at VCS endpoint and Radio endpoint.

R2S-KeepalivePeriod 200ms (default value)
R2S-KeepaliveMultiplier = 10
R2S-LocalHoldTime= R2S-KeepalivePeriod x R2S-KeepaliveMultiplier = 200ms x 10 = 2 seconds

During the Plugtests™ it was observed that the Real Time Session Supervision is established between two end-points using the Real Time Session Supervision Protocol, following establishment of the SIP session between the two endpoints. If the R2S-KeepalivePeriod is too small (i.e. 20ms), it was observed that there is insufficient time to establish the Real Time Session Supervision between the endpoints following SIP session establishment. The use of the R2S-KeepalivePeriod set to 200ms however always allowed sufficient time to establish the RTP session, due to the first Keepalive packet being sent after 200ms.

The keep-alive packets SHALL only be sent every 200ms when there is no voice. This will also lead to a reduction in RTP packet activity and less demand on bandwidth and processing time.

In the case of no packets arriving at opposite endpoint (i.e. VCS endpoint or Radio endpoint) for a period greater than the R2S-LocalHoldTime (i.e. 2 seconds default), it SHALL lead to a SIP session disconnection at each of the end-points following expiry of the R2S-LocalHoldTime timer (counting down to zero). It should be noted that a 2 second timeout set at both the VCS and Radio endpoints is still within the 3 second link loss requirement as defined by ED136 Requirements specification.

4.16.7 Received RTP packets with bit VF=0 SHALL NOT cause session disconnection

The EUROCAE SG2 Radio has to decide if a Visibility Flag (VF) in the RTP Header Extension is needed or not. In the case that the VF remains the EUROCAE SG2 has to decide whether RTP packets received with a Visibility Flag VF=0 SHALL or SHALL NOT cause a disconnection of the RTP session as currently defined in ED137 Part 1 Radio.

Argument in favour of deleting use of VF flag:

The important issue is that RTP packets are being received. It is not important whether the VF flag is set to a 1 or 0. If RTP packets are received the R2S-LocalHoldTime is reset, while if RTP packets are not received then the count-down of the R2S-LocalHoldTime will begin. A RTP session SHALL only be disconnected therefore in the case of a physical disconnection between the endpoints implying that the RTP packets are no-longer being received by an end-point.There is therefore no reason to maintain the Visibility Flag (VF) field in the RTP Header Extension as the User Agent shouldn't care about the Visibility Flag value. Receiving RTP packets with a bit VF=0 should not cause the RTP session to be disconnected.

In order to diminish problems with RTP session establishment between 2 end-points, many vendors during the Plugtests adopted a procedure of sending the RTP packets with bit VF=1 always and disabled functionality that caused a timeout and RTP session disconnection if bit VF=0 was being received.

Argument in favour of continuing use of VF flag:

Receiving continuously packets with VF=0 is a sign of one way physical disconnection, something like a cable problem or a routing problem or a firewall problem where packets go only in one direction. If an endpoint is receiving RTP packets with a bit VF=0, this implies that the RTP packets it is sending are not being received by the corresponding endpoint. It is not possible to rely on only one endpoint to release the connection in the case it doesn't receive packets. Receiving RTP packets with bit VF=0, should cause the count-down of the R2S-LocalHoldTime to begin and after expiry of the R2S-LocalHoldTime will cause the RTP session to be disconnected.

4.16.8 Re-insertion of a=interval attribute with extension to include codec payload type

It is necessary to reintroduce the a=interval packet-size attribute in Table 6 of ED137 Radio Part 1.

i.e. a=interval: 20 (in case of 20ms packets).

It is also recommended to extend the use of this attribute to include the associated codec payload type to be used with the nominated packet size.**(i.e. a=interval: codec ms), a=interval 8 20-** implying payload type 8 (G.711 with 20ms packets).

Example: a=interval: codec ms

a=interval:8 20
a=interval:15 20
a=interval:18 10

Although the default codec has been defined as ITU-T G.711 A-law using 20ms packetization size, it should also be possible to define a different packet size in the case that another codec (i.e. G.728 (PT=15) or G.729 (PT=18)) is used.

The "rtpmap" attribute is currently used to negotiate codec type used, but present method of using the a=interval attribute could result in a packerization interval non-ideal for the codec type.

4.16.9 Clarification on use of Frequency Identification “fid” attribute

Although Frequency Identity “fid” is an optional attribute, the ED137 should make it clear that when receiving a fid value as part of the SDP message body, it should be mandatory for the Radio to verify its value against the frequency configured for the Radio and reject session establishment request if SDP fid attribute does not match the Frequency configured at the radio.

4.16.10 Introduction of new PTT-id attribute assigned by Radio in 200OK response

It is necessary to clarify the link between the introduction of the new PTT-Id attribute and the WG67 KEY IN Event package specified in ED 137 Part 1. The PTT-Id attribute in 200 OK was originally introducted temporarily for the plugtest only. The WG67 KEY-IN Event package is currently defined in ED137 Part 1 radio as mandatory in which case the use of this PTT-id attribute in a 200OK becomes redundant.

As the PTT-id attribute functionality worked well during the Plugtest, it should be evaluated by EUROCAE SG2 Radio if both WG67 KEY-IN Event package and PTT-id allocated by Radio in a 200OK response can remain.

It should be noted that the KEY IN Event package provides more information than the PTT-ID in 200 OK as it provides information to all VCSs that has subscribed to the package and not just those with sessions established to the Radio.

If the PTT-id attribute remains then the ED137 Part 1 radio specification SHALL indicate that a Radio is responsible for allocating a PTT-id (i.e. 0001 to 0007) to a SIP_UA (VCS endpoint). It will be necessary to add a new SDP attribute nominated “PTT-id” to the SDP Types and Parameters in Table 6.

The method executed during the Plugtests™ event to implement this functionality is described as follows:

When a SIP Radio session is established between a SIP UA_A (VCS endpoint) and a SIP UA_B (Radio endpoint), a PTT-id is assigned to this session. The PTT-id is communicated to the VCS endpoint in the 200OK response from the Radio endpoint during session establishment in the following format:

a=PTT-id:0001

where PTT-id can take following numbers: 0001, 0002, 0003, 0004, 0005, 0006, 0007 dependent of an internal radio order.

4.16.11 Suggestion on allocation of PTT-id values

The EUROCAE SG2 Radio group should evaluate a new method of allocating PTT-id values to a session. Rather than using a fixed value (i.e PTT-id: 0001, 0002, 0003, 0004, 0005, 0006, 0007) dependent when several VCS are transmitting on a radio, assigning values like 1, 2, 4, 8, 16, 32, 64 to a VCS and then summing the PTT-id would provide more explicit information on which User Agent or User Agents are transmitting (in the case of PTT summation). For example if PTT-id 8 and 32 are transmitting at the same time, then the transmitter should echo back a PTT-id value of 40 (in case of PTT summation configured at Radio).

It should be noted however that as the PTT-id field in the RTP Header extension currently has only 4 bits, it would be necessary to determine if 4 bits would be sufficient or whether more bits would be required (i.e. 7 bits in the case that a binary notation were used).

The need for a new PTT-id (i.e. 1111) value to be echoed back by Radio to User Agents in case of summation occurring at the Radio as defined by 4.16.14 would therefore no longer be required.

4.16.12 Tx and Rx path PTT-id value description

A description should be provided explaining that when "PTT" is activated at the SIP_UA (VCS endpoint) the Tx path RTPTx Header Extension should contain the assigned PTT-id in its PTT-id field towards the radio. The Rx path RTPRx Header Extension sent from the Radio endpoint confirms the PTT-id (in its PTT-id field) of the SIP_UA at the VCS endpoint that currently has the transmitter activated

In the case of just one SIP_UA (VCS endpoint) activating PTT, it should receive confirmation in the PTT-id field of its Rx path RTPRx header extension of its own PTT-id previously assigned by the radio.

In the case of a Priority-PTT or Emergency-PTT from SIP_UA2 interrupting a Normal-PTT from SIP_UA1 for example, the SIP_UA1 will receive confirmation in the PTT-id field within its Rx path RTPRx header extension, that UA_A2 PTT-id currently has Priority_PTT or Emergency-PTT activated.

Likewise in the case of a Normal-PTT from a SIP_UA2 being locked-out due to a previous Normal-PTT activation by SIP_UA1 for example, the SIP_UA2 will receive confirmation in the PTT-id field of its Rx path RTPRx header extension that SIP_UA1 PTT-id currently has Normal_PTT activated.

4.16.13 Review of PTT-id=0 value

The ED137 Part 1 Radio states that the PTT-Id in the Tx Path can be set with 0 if not yet known by the VCS. In this case, it is unclear what PTT-id should be echoed back by the Radio on the Rx path. Probably the real PTT-Id of the keying VCS which is known by the radio. Further clarification is required in the ED137 Part 1 radio document about this aspect.

It should be made clear that as a PTT-id is assigned by the Radio endpoint on session establishment, (through a 200OK response) every VCS should know its PTT-id and the case when a PTT-id=0 should therefore not occur.

4.16.14 Need for a new PTT-id values to be echoed back from Radio in case of Audio summation of User Agents with same ptt level

The Test case LAN-RAD-PTT9 - Normal v Normal PTT activation test on given frequency (PTT summation configured at Radio) has indicated the need for a new PTT-id (i.e. 1111) value to be echoed back by Radio to User Agents in case of summation occurring at the Radio.

In the case of multiple SIP User Agents activating the same level of PTT (i.e. all Normal-PTT, all Priority-PTT or all Emergency-PTT) towards a radio configured for PTT summation, the present definition within “ED137 Part 1 Radio” requires that the Radio echoes back the same PTT-id being sent by the User Agent (i.e. UA_A1 and UA_A2 sending “ptt-id=0001” and “ptt-id=0002” respectively would each receive an identical value echoed back from the Radio). This implies that the SIP User Agent is unable to determine if it has sole use of the Radio Transmitter or if there is Audio summation of multiple users in progress.

It is therefore recommended that ED137 Part 1 Radio should be updated to indicate that a new PTT-id value SHALL be echoed back to multiple SIP User Agents simultaneously activating PTT at same level, in the case that Audio Summation is configured at the radio. The suggested value for PTT-id indicating summation is 15 or ptt-id=1111.

UA_A1 and UA_A2 with the same PTT level activated towards the same Radio and sending ptt-id=0001 and ptt-id=0002 respectively for example would now both receive back a ptt-id=1111 from the Radio, implying more than one User Agent is currently transmitting at the Radio Transmitter.

4.16.15 Tx and Rx path ptt-type value description

A description should be provided explaining that when "PTT" is activated at the SIP_UA (VCS endpoint) the Tx path RTPTx Header Extension should contain the PTT type in its ptt-type field towards the radio. The Rx path RTPRx Header Extension sent from the Radio endpoint confirms the PTT type (in its ptt-type field) of the SIP_UA at the VCS endpoint that currently has the transmitter activated

In the case of just one SIP_UA (VCS endpoint) activating PTT, it should receive confirmation in the ptt-type field of its Rx path RTPRx header extension of its own ptt-type previously sent to the radio.

In the case of a Priority-PTT or Emergency-PTT from SIP_UA2 interrupting a Normal-PTT from SIP_UA1 for example, the SIP_UA1 will receive confirmation in the ptt-type field within its Rx path RTPRx header extension of the ptt-type (i.e. Priority_PTT or Emergency-PTT) being used by UA_A2 to activate the Radio transmitter.

Likewise in the case of a Normal-PTT from a SIP_UA2 being locked-out due to a previous Normal-PTT activation by SIP_UA1 for example, the SIP_UA2 will receive confirmation in the ptt-type field of its Rx path RTPRx header extension of the ptt-type being used by SIP_UA1 to activate the transmitter.

4.16.16 Inversion of bit order within RTP Header Extension “bss-qidx” field

With reference to ED137 Part 1 Radio paragraph 5.10.5.1 – Best Signal Selection

It is necessary to modify the text defined in this paragraph to the following:

*The SQP value normalized between 0 and 15 **SHALL** be coded in the **bss-qidx** field as the bits b18 to b25 **where b25 is the lower significant bit and b18 the upper significant bit.***

With respect to the RTP header Extension “SQP value” it is recommended that SG2 should consider using all the 8 bits in field to provide SQP information. Using a linear scale instead of a level partitioning classes and using a **two's complement** notation, all RSSI values from **-**128 to +127 can be represented.

4.16.17 Inversion of bit order within RTP Header Extension “CLD” field

With reference to ED137 Part 1 Radio paragraph 5.10.5.2 – Climax**-** Time Delay

It is necessary to modify the text defined in this paragraph to the following:

*The CLD value evaluated between 0 and 64 **SHALL** be coded in the **CLD** field as the bits b18 to b23 **where b23 is the lower significant bit and b18 the upper significant bit***

4.16.18 Radio protection from unauthorised calls

It is recommended that instead of using the validity of the URI address defined in the From: header within an INVITE method as a means to verify if the call is unauthorised, it is more appropriate to protect a radio from unauthorized or unknown SIP calls by either checking the source IP signalling access list or using INVITE authentication based on digest MD5 credentials.

It would apparently be extremely easy for abusive users that know a valid SIP URI to make a false call as if it came from a valid user. Checks made on the IP address would prevent this from happening. INVITE authentication should also be considered based on digest MF5 credentials.

The EUROCAE SG2 should evaluate the use of MD5 identification as a suitable way forward. In a global network where many VCSs can get access to radios and taking into account that this access may be done through proxys, maintaining access lists would become difficult.

4.16.19 Clarification on how Real Time Session Supervision and SIP session is cleared/ re-established while PTT is activated

Feedback received following the Plugtests™ (relative to test LAN-RAD-R8) has shown that the event of PTT deactivation on PTT timeout at Radio endpoint needs further clarification within the ED137 Part 1 Radio document as to when and how the Real Time Session Supervision and SIP session are actually taken down following a link disconnection while PTT is active.

When the link is physically disconnected for more than 2 seconds while PTT is activated and transmission is in progress at the nominated frequency, there will be two timers that will expire:

1. the PTT timeout at the Radio Transmitter causing the Radio Transmission to be switched-off.

2. The R2S-LocalHoldTime (i.e.R2S-KeepalivePeriod (200ms) x R2S-KeepaliveMultiplier (default 10)) will also expire after 2 seconds, due to no RTP packets being received causing the R2S-LocalHoldTime to be decremented by 20ms for each 20ms RTP voice packet not received or by 200ms for each 200ms RTP KeepAlive packet not received until zero has been reached.

When the R2S-LocalHoldTime has expired it should be assumed that the Real Time Session Supervision (R2S) protocol is no longer active and the SIP session SHALL therefore be released by sending the SIP BYE method. A SIP BYE method SHALL be sent from the VCS endpoint and RADIO endpoint on its R2S-LocalHoldTime value reaching the value zero. As there is no physical connection however between the endpoints a SIP BYE method sent will never arrive at the corresponding end-point, but the SIP session will remain in the unacknowledged state until the SIP stack timers expire at which point the SIP session will be cleared internally.

The established SIP session SHALL therefore be torn down when link is physically disconnected after 2 seconds.

When the link is then physically re-connected (after 2 seconds minimum), the session between VCS and Radio SHALL be automatically re-established as quickly as possible. The frequency key at the CWP SHALL also return to its previous frequency select state.

Following SIP session establishment the Send-Receive Real Time Session Supervision SHALL then be re-established between VCS and Radio.

In the case that the PTT-ON remains continuously activated during the period that the link was disconnected, following link re-establishment the Tx path RTP Header Extension SHALL now transport PTT-OFF towards the Radio and as a result only RTP Keepalive packets SHOULD be sent to the Radio transmitter instead of RTP packets containing voice. The Radio Transmitter SHALL therefore remain in a deactivated state.

The reason for this is that the link disconnection has caused part or all of the controller message to be lost., in which case the pilot would not have understood the message and asked that it repeated. In these circumstances it is important that only the receive path from RADIO towards the controller is active in order that the controller can hear the pilots response to the aborted message.

On the first PTT activation following R2S session re-establishment, the Tx path RTP Header Extension SHALL again be transporting RTP packets containing voice and PTT-ON indication towards the Radio. The Radio Transmitter SHALL then be reactivated.

A continuous high packet loss for a sustained time interval with 200 x 20ms packets lost could also therefore lead to Real Time Session Supervision and SIP session disconnection.

When the link is physically disconnected for less than 2 seconds while PTT is activated and transmission is in progress at the nominated frequency, it SHALL cause the R2S-LocalHoldTime at the VCS endpoint and Radio endpoint to be reset to its default value (i.e. 2 seconds) following re-connection and the arrival of RTP packets with bit VF=0 or 1. It is therefore recommended that the arrival of any RTP packets at an endpoint SHALL cause the R2S-LocalHoldTime to remain at 2 seconds.

4.16.20 Clarification on how Real Time Session Supervision and SIP session is cleared/ re-established while SQUELCH is activated

Feedback received following the Plugtests™ (relative to test LAN-RAD-R10) has shown that the event of Squelch deactivation on Squelch keep-alive timer expiry at VCS endpoint needs further clarification within the ED137 Part 1 Radio document as to when and how the Real Time Session Supervision and SIP session are actually taken down following a link disconnection while SQUELCH is active.

When the link is physically disconnected **for more than 2 seconds while SQUELCH is activated and reception is in progress at the nominated frequency**, there will be two timers that will expire:

1. the SQUELCH keep-alive timer at the VCS causing the SQUELCH indication at the CWP to be stopped.

2. The R2S-LocalHoldTime (i.e.R2S-KeepalivePeriod (200ms) x R2S-KeepaliveMultiplier (default 10)) will also expire after 2 seconds, due to no RTP packets being received causing the R2S-LocalHoldTime to be decremented by 20ms for each 20ms RTP voice packet not received or by 200ms for each 200ms RTP KeepAlive packet not received until zero has been reached.

When the R2S-LocalHoldTime has expired it should be assumed that the Real Time Session Supervision (R2S) protocol is no longer active and the SIP session SHALL therefore be released by sending the SIP BYE method. A SIP BYE method SHALL be sent from the VCS endpoint and RADIO endpoint on its R2S-LocalHoldTime value reaching the value zero. As there is no physical connection however between the endpoints a SIP BYE method sent will never arrive at the corresponding end-point, but the SIP session will remain in the unacknowledged state until the SIP stack timers expire at which point the SIP session will be cleared internally.

The established SIP session SHALL therefore be torn down when link is physically disconnected after 2 seconds.

When the link is then physically re-connected (after 2 seconds minimum), the session between VCS and Radio SHALL be automatically re-established as quickly as possible. The frequency key at the CWP SHALL also return to be indicating SQUELCH.

Following SIP session establishment the Send-Receive Real Time Session Supervision SHALL then be re-established between VCS and Radio.

In the case that the SQUELCH remains continuously activated during the period that the link was disconnected, following link re-establishment the Rx path RTP Header Extension SHALL now transport SQUELCH-ON towards the VCS and as a result the VCS endpoint is now receiving again.

A continuous high packet loss for a sustained time interval with 200 x 20ms packets lost could also therefore lead to Real Time Session Supervision and SIP session disconnection.

When the link is physically disconnected for less than 2 seconds while SQUELCH is activated and transmission is in progress at the nominated frequency, it SHALL cause the R2S-LocalHoldTime at the VCS endpoint and Radio endpoint to be reset to its default value (i.e. 2 seconds) following re-connection and the arrival of RTP packets with bit VF=0 or 1. It is therefore recommended that the arrival of any RTP packets at an endpoint SHALL cause the R2S-LocalHoldTime to remain at 2 seconds.

For a Radio receiver it is therefore absolutely essential that the link and the corresponding frequency select status is re-established automatically after a link loss.

4.16.21 Clarification on Protocol definition for Real Time Session Supervision Establishment

The protocol definition for Real Time Session Supervision establishment needs to be clarified by ED137 Part 1 Radio document.

It is recommended that both VCS and Radio endpoints SHALL have the capability to initiate the process of R2S establishment with the corresponding endpoint.

Within 200ms of SIP session establishment with the Radio, the VCS and Radio SHALL send RTP packets with VF=0 or 1 to the Radio.

Following establishment of a SIP session the Radio and VCS SHALL expect to receive RTP packets with bit VF=0 or 1 within 200ms. It is recommended that the LocalHoldTime at the VCS and Radio endpoints is set to 2 seconds in order to allow a window for R2S session establishment and to ensure that the SIP session is not disconnected within 2 seconds in the case that RTP packets do not arrive.

If a Radio receives RTP packets with bit VF=0 or 1, it SHALL always reply to VCS with RTP packets with bit VF=1.

When a VCS receives RTP packets with a VF=0 or 1, it SHALL always reply to Radio with RTP packets with bit VF=1

When the bit RTP packets with VF=1 or 0 is set, the **R2S-LocalHoldTime** at both VCS and radio endpoint is reset to 2 seconds.

From this point onwards any RTP packets received with bit VF=0 or 1 SHALL NOT cause a decrement in the RS2-LocalHoldTime set at the end-points.

The VCS or Radio endpoint SHALL therefore only disconnect a SIP session in the case of no RTP packets arriving over a period of 2 seconds and SHALL be triggered by the R2S-LocalHoldTime reaching zero.

The text in this paragraph is influenced by the decision taken by SG2 with respect to paragraph 4.16.7.

4.16.22 Distinction needed between a Squelch Signal detected by a Radio Receiver as a result of an Aircraft call and that due to an off-air squelch signal

The Test case **LAN-RAD-PTT11** –Automated radio check during PTT (off-air audio/squelch method) and PTT failure indication test has indicated the need for further clarification in the ED137 Part 1 Radio document.

When a User Agent sends PTT-ON to a Radio Transmitter on its Tx path it will receive the same value echoed-back on the Rx path from the Radio Transmitter. With feedback between Radio Transmitter and Radio Receiver simulating off-air audio/squelch, the Radio Receiver will detect a Squelch signal and indicate this on its Rx path towards the User Agent.

In order to distinguish between a Squelch Signal detected by a Radio Receiver as a result of an Aircraft call and that due to an off-air squelch signal due to PTT activation by a User Agent towards a Radio Transmitter, it is also necessary to inform the Radio Receiver each time PTT is activated towards a Radio Transmitter.

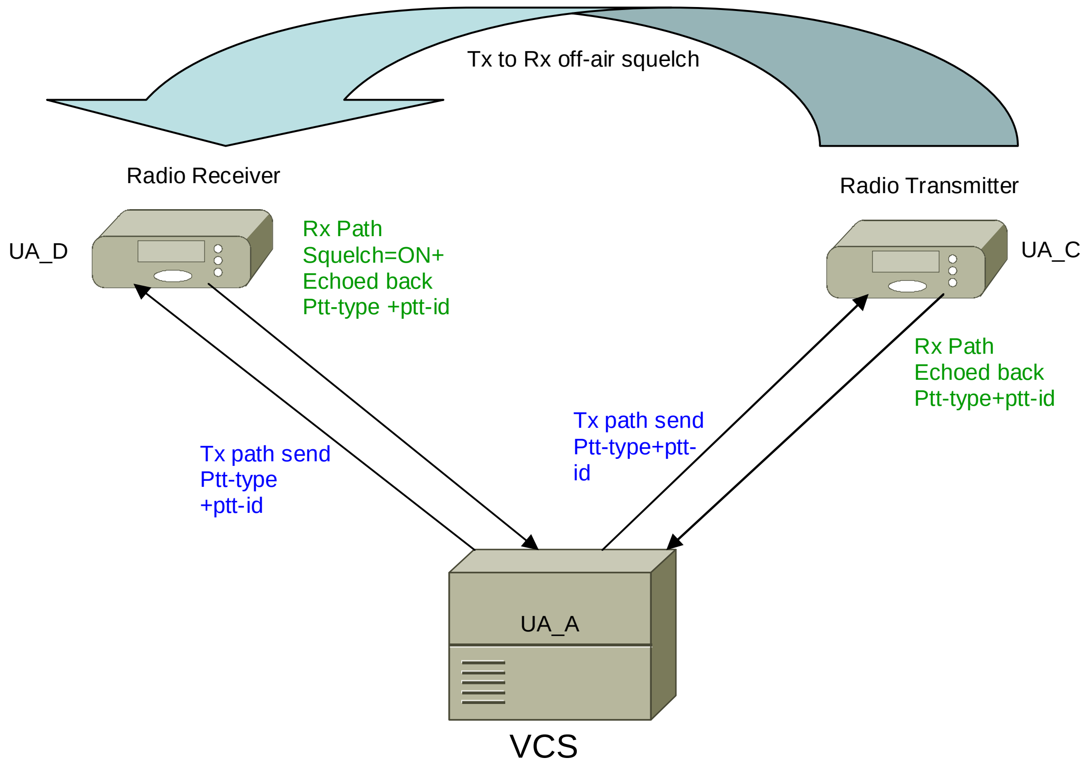
The ED137 Part 1 Radio document should indicate that when PTT is activated by a SIP User Agent this should cause ptt-type and ptt-id to be sent towards the Radio Transmitter, but should also cause the same ptt-type and ptt-id to be sent towards the Radio Receiver.

In this way a Radio Receiver detecting a squelch signal and simultaneously receiving a ptt-type + ptt-id from a User Agent will recognise that the squelch signal is infact off-air squelch.

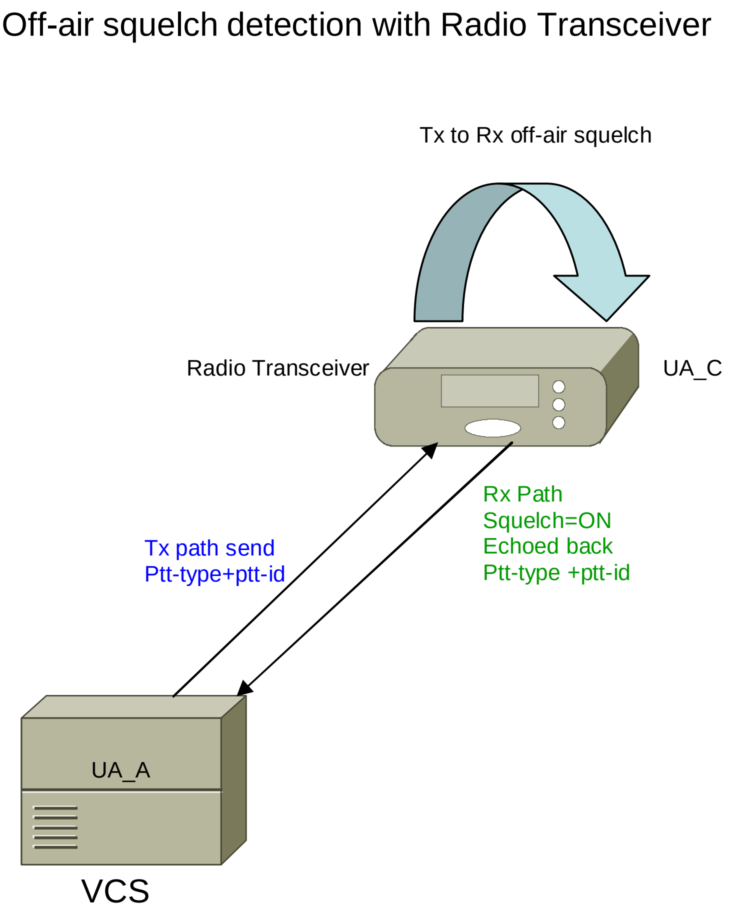
It should also be noted that a Radio transceiver is not always associated with one and only one receiver. When PTT is activated by a SIP User Agent, in the case of multiple Radio receivers therefore it would be necessary to send ptt-type + ptt-id to each of the Radio receivers. It is also recommended that this solution should probably be analysed more in details before implementing what is proposed here in the report.

Refer to diagram below for a further explanation.

Off-air squelch detection with separated Radio Transmitter & Receiver



A Radio Transceiver detecting a squelch signal and simultaneously receiving a ptt-type + ptt-id on Tx path from a User Agent will recognise that the squelch signal is infact off-air squelch. The Squelch, ptt-type + ptt-id will be sent on Rx path back to User Agent. Refer to the diagram below.



4.16.23 Harmonization of SDP <encoding name>/<clock rate> attributes used for Telephone and Radio

It is recommended that the method used to define SDP <encoding name>/<clock rate> attributes for Telephone and Radio in performed in the same way. At present the ED137 Part 1 Radio document inserts a X-PTT prefix;

- X-PTT- PCMU/8000 (for PCM-µ)
- X-PTT-PCMA/8000 (for PCM-A)
- X-PTT- G728/8000 (for G728)
- X-PTT- G729/8000 (for G729)

(X-PTT-PCM-A:/8000 Default Value)

It should be made clear in ED137 Part 1 Radio document the reason for the X-PTT prefix and why it is needed just for Radio. (i.e. the X-PTT suffix indicates that extended header for radio signalling is used and supported).

While the ED137 Part 2 Telephone document defines these attributes as follows:

- PCMU/8000 (for PCM-µ)
- PCMA/8000 (for PCM-A)
- G728/8000 (for G.728)
- G729/8000 (for G.729)

It is recommended that this X-PTT prefix is removed in order to harmonize the definition of this parameter.

4.16.24 Audible beep to pilot when Radio Switches-over session

It is observed that when a radio switches-over transmission between two sessions on arrival of a higher ptt-type level, there is no indication provided to the pilot that this event has occurred. The voice towards the pilot on one session is just replaced by voice from another session. The switchover is instant with no clicks or noise introduced, implying that a pilot will not know there has been a switchover of the sessions. It should be evaluated if future operational procedures will require that the pilot is informed of a switchover through a short audible tone inserted in the voice path towards the Aircraft or not.

In a future operational environment, the switching over of sessions by the Radio can only be performed in the case of a higher ptt-type being received and it is assumed that this has occurred due to a emergency situation occurring.

In the case of multiple SIP User Agents as endpoints within the same ATS unit, it is also possible for the switchover between sessions to occur when the sessions have been established by SIP User Agents within the same ATS unit. Today there is already the situation where a Mentor is able to override transmission of a trainee controller in case of an incorrect verbal instruction being transmitted. In these circumstances it is not used to warn the pilot of this as the whole point of the interruption is to correct the previously transmitted vocal message.

4.16.25 Importance of sending a=coupling attribute to Radio when Cross-coupled group has been configured at a VCS.

Further clarification is required in the ED137 Part 1 Radio document about the procedures used to inform a Radio that it is apart of a cross-coupling group.

The procedure to configure a Cross-coupling group of frequencies normally follows the establishment of sessions for each of the frequencies to the respective radios. It is rare that the inverse occurs (i.e. that a cross-coupling group is defined before the sessions to the radio are established).

Normally when a VCS establishes a session with a Radio it will use the type: <call type> attribute set to “radio”. Once sessions have been established with the respective radios then the process of configuring a cross-coupling group normally occurs.

It is therefore recommended that definition of a cross-coupling group should cause a RE_INVITE to be sent to each of the Radios forming the Cross-coupling group. The Re-INVITE should then use type: <call type> attribute set to “coupling”. The Radio is then able to acknowledge the RE_INVITE by sending a 200K and know that it’s frequency is now part of a cross-coupled group. A radio receiving a second INVITE or RE_INVITE request to be included in a cross-coupled group, SHALL therefore reject the request.

The ED 136 Requirements document states:

1 [REQ RADIO FUNCTIONAL] IDENTICAL FREQUENCIES NOT ALLOWED IN TWO CROSS-COUPLING SESSIONS

In order to prevent cross-coupling chains, means **SHALL** be provided to ensure that a particular frequency can only be included in one cross-coupling session.

2 [REQ RADIO FUNCTIONAL] CROSS-COUPLING FREQUENCY SELECTION REFUSAL INDICATION

In the event that a frequency is already cross-coupled on one CWP the system **SHALL** prevent it being cross coupled at another CWP – regardless of its location (except if the other CWP is on the same Sector Suite – depending on ANSP implementation). The system **SHALL** also present clear information, on the CWP of the Controller being prevented from cross coupling the frequency, that the cross-coupling procedure has been refused and thus not executed.

4.16.26 Clarification about use of PT=8 and PT=123 on Tx path with PTT ON and OFF

Further clarification is required in the ED137 Part 1 Radio document with respect to when PT=8 and PT=123 should be used on the transmit path with PTT activated and deactivated. The following conditions are considered valid:

- VCS activates PTT for audio: Sends RTP packets with RTP Header Extension set to ptt_type=PTT ON, PT=8 every period=20ms
- VCS deactivates PTT: Sends RTP packets (i.e 3 RTP packets maximum) with RTP Header Extension set to ptt_type=PTT OFF, PT=8 every period=20ms
- VCS not transmitting audio (keepalive packets):Sends RTP packets with RTP Header Extension set to ptt-type=PTT OFF, PT=123 every R2S-KeepalivePeriod (default value: 200ms).

As a Radio Transmitter should be able to detect PTT ON/OFF state, the use of Voice Activity Detection should not be used on the RTP session established between the VCS and Radio endpoints. The ED137 part 1 radio document should make it clear therefore that sending PTT OFF and PT=8 in RTP packets every 20ms continuously SHALL not be allowed.

4.16.27 Clarification about use of PT=8 and PT=123 on Rx path with SQUELCH ON and OFF

Further clarification is required in the ED137 Part 1 Radio document with respect to when PT=8 and PT=123 should be used on the receive path with SQUELCH activated and deactivated. The following conditions are considered valid:

- Radio activates SQUELCH for audio: Sends RTP packets with RTP Header Extension set to SQL ON, PT=8 every period=20ms.
- Radio deactivates SQUELCH: Sends RTP packets (i.e. 3 RTP Packets max) with RTP Header Extension set to SQL OFF, PT=8 every period 20ms
- Radio not receiving audio (keepalive): Sends RTP packets with RT Header Extension set to SQL OFF, PT=123 every R2S-KeepalivePeriod (default value: 200ms).

As a VCS should be able to detect SQUELCH ON/OFF state, the use of Voice Activity Detection should not be used on the RTP session established between the Radio and VCS endpoints. The ED137 part 1 radio document should make it clear therefore that sending SQUELCH OFF and PT=8 in RTP packets every 20ms continuously SHALL not be allowed.

4.16.28 Clarification about LocalHoldTime reset for RTP packets with PT=8 and PT=123

The ED137 Part 1 Radio document should make it clear that when RTP packets from corresponding Endpoint are received with PT=8 or PT=123, the local timer R2S-LocalHoldTime SHALL always be reset, due to RTP packets being received (i.e. link is up)

4.16.29 SIP URI format recommendation.

The ED137 Part 1 Radio document should make a Recommendation of the format to be employed for the SIP URI recognised by a radio (i.e.: sip:txrx.frequency.atsu@radio_site_id.local_domain") as defined by ED137 Part 1 Radio document. This format as defined by ED137 Part 1 Radio was not tested during the Plugtests and many Radio vendors were incapable of accepting this format as they had already implemented the SIP URI address plan defined for the Plugtests.

Also, it must be clarified how it will be done in the future, whether radios will be adapted to support this format or whether the ED137 will be modified.

4.16.30 Clarification needed if Real Time Climax Delay compensation is to be deployed.

The ED137 Part 1 Radio Specification currently defines Climax delay as a timing delay parameter to be sent in the Tx RTP HE from the VCS to each Radio Transmitter in the multi-carrier offset group of transmitters. This timing delay parameter can be set to a different value for each transmitter.

On the receive path of a Climax transmission, it is recommended to use BSS. (i.e. each radio receiver sends its received signal to VCS and VCS decides which is best quality signal).

Currently from the ED 137 Part 1 Radio definition it appears that the CLD parameter value in RTP HE is statically configured. The Climax Time Delay test implemented by Vendors during the Plugtests was implemented using static values configured without any feedback mechanism.

Clarification needed if Real Time Climax Delay compensation is to be deployed. It is foreseen that the delay calculation could be based on ground-ground segment only (least precise) or on both a combination of ground-ground + air-ground segments (more precise)

1. Using Ground-segment only, where VCS able to calculate ground-delay to each Radio transmitter and adjusts CLD values to each Radio Transmitter in Climax group automatically.
2. Using Ground-segment only, where VCS is able to measure the relative delay in Squelch signal reception from each of the Radio Receivers used in a multi-carrier climax group. It can then use this measurement to calculate necessary voice delay to be sent to each of the transmitters in a multi-carrier climax group of transmitters.
3. Using Ground + Air segments where Radio can measure real time delay to aircraft (maybe possible in future with digital radios) and send info in a new field within Rx RTP HE to VCS, allowing CLD values sent to Radio Transmitters in Climax group to be adjusted automatically .

4.17 Recommendations to ED137 Part 2 Telephone for consideration

4.17.1 Clarification about conference focus leaving an active conference

Further clarification is required in the ED137 Part 2 Telephone document with respect to whether the SIP User Agent with the conference focus role has the option to transfer the focus role to another Conference participant before leaving the conference.

The Stage 2 Telephone Plugtests™ Test case **LAN-SS-CONF2** – "Establishment of a 5 party conference using "A" or "Conference Focus Entity" as Focus. Parties either eliminate themselves or are eliminated from conference one at a time by "A" or "CFE" has raised an issue about what should occur when a conference focus leaves the conference.

The test scenario requires that when the Conference focus leaves the conference while other User Agents are still present, the conference itself is not terminated, the remaining users stay connected with the conference focus now having a mute role. The conference remains connected until the pen-ultimate User agent leaves the conference.

There is a request to EUROCAE SG2 Telephone to re-consider this functionality such that when the Conference Focus leaves the conference, the whole conference will be terminated. It should be decided if it should be optional or mandatory for the Conference Focus to have the capability of transferring the role of focus to another user in the conference prior to withdrawing from the conference in progress. In this way the conference could continue with a new conference focus and all remaining users stay connected.

The main objection to the current Conference implementation being tested during the Plugtests™ is that why should a VCS or SIP User Agent as conference initiator continue to be used as a focus after it has withdrawn from the conference. It could lead to the SIP user agent being prevented from hosting another conference while it is still being used as the focus for a previous conference that it initiated. It also implies that the SIP user agent resources will continue be employed following withdrawal by the focus.

In the case that a vendor has implemented the Conference feature using an autonomous Conference Focus Entity (CFE) however it should still be possible to maintain the conference when the initiator withdraws from the conference. SG2 should note however that introducing an option that allows the role of focus to be transferred to another user in the conference, introduces an additional difficulty for conference participants since they have to recognise and accept the focus entities URIs, in addition to the CWPs URIs.

4.17.2 Clarification should be provided that a Call can only be transferred within a dialogue.

Further clarification is required in the ED137 Part 2 Telephone document in order to clarify that a call can only be transferred inside a dialogue. The current description for Call Transfer assumes that when using the REFER method, the same Call-id will be used as the original call to be transferred (i.e. inside dialogue), but this is not always necessary as a different Call-id could also be used for the call transfer (outside dialogue).

It should be noted that the *INVITE/Referred-by* request from Transferee to Target (as used by some vendors) is a brand new call that has no connection with Transferor-Transferee call, while the *REFER* request from Transferor to Transferee must be transmitted in the existing dialog between Transferor and Transferee. This is the choice cited in RFC 5359/2.5 (other RFCs or drafts discuss the merits of in-dialog and out-dialog REFER requests).

Further clarification is therefore required citing the RFC 5359 that a call can only be transferred within a dialogue.

4.17.3 Clarification to INVITE responses used during call intrusion

Further clarification is required in the ED137 Part 2 Telephone document about the response to an INVITE in the case of an unsuccessful intrusion due to a Protected User or by attempting to intrude into a Priority call in progress.

It was observed that during intrusion tests that resulted in the intrusion not being successful [i.e. Intrusion Protected or Two Priority calls simultaneously] different responses [182 , 180] to the INVITE packet are sent from different vendors. It is recommended that clarification is therefore provided.

4.17.4 Proposal to implement SUBSCRIBE/NOTIFY mechanism for dialog and presence events

EUROCAE SG2 should conisider the implementation of SUBSCRIBE/NOTIFY mechanism for dialog and presence events in next edition of document. This is useful if you want to see the status of a SIP URI (i.e if it is online or not, if it has an active call-in-progress or not) from a remote SIP URI.

4.17.5 Proposal to implement Replaces header RFC3891 in order to be able to perform call pick-up.

EUROCAE SG2 should conisider the implementation of Replaces header RFC3891 in order to be able to perform a call pickup. This service requires that Proposal 4.17.4 is also implemented). Refer also to RFC5359 Call pickup.

Recommended updates to PLUGTESTS™ Specifications

A number of minor errors were identified with both the Plugtests™ Stage 1 and Stage 2 test specifications during test case execution.

The following list the errors identified in the test specifications:

- The Stage 1 Radio Plugtests™ specification still refers to valid <send-receive mode> SDP attributes as sendonly/recvonly. As it became mandatory to use the Real Time Session Supervision protocol during this event for both Stage 1 and Stage 2 Radio Tests, this now implied mandatory use of the <send-receive mode> set to sendrecv. It was observed that the Stage 1 Radio Plugtests™ document had not been updated.
- The Stage 1 and Stage 2 Radio Plugtests™ specifications still referred to a R2S-KeepAlivePeriod period of 20ms and a R2S-KeepaliveMultiplier of 3, implying a LocalHoldTime of 60ms. As the default values were used for these parameters during the Plugtests™ it is necessary to reflect this in the test specifications. The test specifications should therefore be changed to the following default values:
R2S-KeepAlivePeriod=200ms
R2S-KeepaliveMultiplier=10
LocalHoldTime=2 seconds
- The Stage 1 Radio Plugtests™ LAN-RAD-R5 "Send/Receive SIP Radio session request to an invalid frequency", should be changed to "Send/Receive SIP Radio session request to an invalid URI". The frequency identification information is no longer present in the SIP URI as this is now an optional SDP attribute nominated "fid". The test steps should also reflect this change.
- The Stage 1 Radio Plugtests™ LAN-RAD-R1, R2 and R3 relate to establishing a send/recv session, a sendonly session and receive only session. These tests should now be reviewed as it was no longer possible to establish a sendonly or recvonly session with the Radio due to the mandatory use of the Real Time Session Supervision protocol requiring the sendrecv sessions only are established. It is recommended that these tests are re-defined as they were during the Plugtests™ event in order to verify that sessions can be established to a Radio Transceiver, a Radio Transmitter and Radio Receiver.
- The Stage 1 Radio Plugtests™ LAN-RAD-R4 "Send/Receive SIP Radio session request and immediate cancellation" should be deleted from the specification. It was seen that the response times of the systems was very high, making it impossible to send an INVITE method following by a CANCEL method through manual intervention. This test can only be performed by a conformance test system with the ability to send the two methods at the same time towards the Implementation under Test.
- The Stage 2 Radio Plugtests™ LAN-RAD-PTT1 "Coupling PTT activation, Voice transmission, Coupling PTT deactivation" is currently sending ptt-coupling directly to the radio on cross-coupling group PTT activation by the controller. Although this test has proven that coupling-ptt can be sent to the radio, in an operational environment the only time coupling-ptt would be sent to a radio by a SIP User agent is for retransmission of an incoming aircraft call towards other radios configured in the cross-coupled group at the SIP User agent itself. It should be noted that when a controller activates PTT on a cross-coupled group configured at its position, the normal-ptt type would be sent to all radios forming the cross-coupled group.
- The Stage 2 Telephone Plugtests™ LAN-SS-CI1 and LAN-SS-CI2 are very similar tests requiring either the manual or automatic answer of an incoming Priority call while there is a Routine call in progress. During the Plugtests™ these two tests were assigned an either/or status implying that it was mandatory for the supplier to only execute one of the two tests. It is recommended that these tests are combined into a single test allowing the user to configure the system for either manual or automatic priority call answer.
- It was observed that both Stage 1 and Stage 2 Telephone Plugtests™ had scenarios relating to "Call intrusion", but the method employed for performing the call intrusion was different between the Stage 1 and Stage 2 Telephone tests. It was understood that Stage 1 tests were simpler tests when compared to the Stage 2 and didn't have the scope of testing all aspects of the Call Intrusion feature, but it is recommended that the procedure used by Stage 1 tests should now be made identical to the procedure used by Stage 2 (i.e. as defined in the ED137 Part 2 Telephone document).

The following list the suggestions to improve the format of the test specification and aid the supplier when executing the tests:

- It is recommended that each test defined in the Plugtests™ specifications should have a reference par. related to the ED requirements to make it easy to check, find percentage test coverage etc. If it were possible to also add the relevant RFCs to each test then this would be a plus. This will make it easier to determine which ED136 requirements have been tested. It is also possible that more than one ED136 requirement is covered by a single test and in this case the relevant paragraph numbers should be indicated.

- It is recommended that all the Plugtests™ pre-conditions are reviewed and updated, leaving only important interoperability related configuration details and facts.

- It would have made test case execution easier if each test also defined a message sequence chart detailing the message exchanges between the end-points. The present specifications provide only textual definitions, whereas a pictorial definition would have aided test scenario execution. It was appreciated however that there was little time or resources dedicated to test specification development and more time would have been required to add Message Sequence Charts. Following the first few test sessions however it was observed that companies were more at ease with the test specifications and the execution of the individual test steps.

- The test steps defined for many test scenarios appear to be a mixture of “functionality” + “message checks”. The true interoperability specifications normally check for functionality interoperability only, but as there has never been an official conformance test phase, it has resulted in test cases having a mix of both functionality and conformity checks. It is recommended that either all tests just check functionality or all tests check functionality and conformity (in the case that no conformance test phase is performed).

- Following on from the previous point above, it was observed that for some test scenarios it was possible to pass the test but not have correct message flows. For example the Attended Call Transfer test indicates that the call should be placed on hold, but doesn’t verify the correct message sequence for placing the call-on-hold. It was possible for a supplier to have passed the test and not have implemented correctly the “Call-on-hold” message exchange between the endpoints.

4.18 What will happen after this event?

- Following the comments to the Plugtests™ specifications, new versions of both the Telephone & Radio Plugtests™ specifications shall now be produced. It is likely that Stage 1 and Stage 2 tests shall now be combined into single specifications for Radio and Telephone.

- The Plugtests™ report will be compiled containing all the feedback accumulated during the event and this will be distributed to all participating companies. This report will also contain a series of recommendations noted prior and during the event following feedback provided by the participating vendors.

- Each participating company will still have external access to the Test Reporting Tool in the coming weeks allowing them to review their own test session reports and download their traces relevant to all tests that they have performed during the event. Companies are asked to remember the NDA that they have signed.

- The Plugtests™ report, updated test specifications, and feedback from this event will be uploaded to the event wiki

- It is foreseen that this Plugtests™ report will also be presented to the next WG67 meeting taking place on the 21st April in Brussels.

4.19 Proposed Next Steps

- It is proposed to proceed in the development of EUROCAE Conformance Test Suites for Telephone and Radio (and maybe Recorder). This DELTA test suite will define test cases that cover the differences between IETF RFC 3261 and the EUROCAE ED137 specifications, in order to test added SIP/SDP protocol functionality not covered by the RFC 3261 test suite. The development of this test suite is seen as fundamental in the validation of the SIP Telephone and Radio User agents and will cover a series of valid behaviour as well as non-valid behaviour tests.

- It is proposed to have a 3rd Telephone/Radio/Recorder Plugtests™ event in 2010 should there be sufficient demand from other VCS, Radio and Recorder suppliers in the European and Global markets. It will also be possible for VCS and Radio suppliers who have already participated in previous events to also take part in order that they can repeat tests that were not successful or for which they ran out of time. It is also anticipated that Plugtests™ specifications would be expanded to include the all remaining tests for ED requirements not previously covered by the previous Plugtests™ events.
- It is also planned to survey Voice Recorder vendors in order to understand their intentions with respect to the current state of developing a SIP User Agent interface for their equipment according to the ED137 Part 1 Recorder document. Sufficient interest could lead to Recorder vendors also attending the Telephone and Radio Plugtests™ event in 2010.
- A EUROCAE IOP SIP gateway Plugtests™ event is planned by ETSI from 7th to 11th September. There are currently 6 VCS companies that have committed to attend the event. It shall be possible to perform tests relating to SIP-ATS MFC-R2 and SIP –ATS-QSIG gateway interworking as defined by the ED137 Part 2 Telephone document.

4.20 Conclusion

The second EUROCAE Plugtests™ Interoperability Event on VoIP for ATM (Air Traffic Managment) held at the ETSI headquarters in Sophia Antipolis between 25th March and 3rd April 2009 has resulted in the significant quantity of feedback relating to the ED137 documents being accumulated both prior and during the execution of tests. The importance of a Plugtests™ event in demonstrating that specifications are robust and that they contain sufficient clarity in order to achieve interoperability between multiple vendors has been confirmed during the event. The information collected has led to a series of recommendations being proposed in order to improve the robustness of the EUROCAE ED 137 interoperability documents that specify the interworking between Voice Communication Systems (VCS) as well as the interworking between Voice Communication Systems and Ground Radio Stations (GRS). The numerous test scenarios performed by the vendors participating in the event has demonstrated the readiness of these VoIP interfaces in their deployment within the framework of the Single European Sky (SES).

Progressing onwards from the first Plugtests™ event that covered the simpler Stage 1 interoperability tests held in April 2008 and following on from the formal approval of EUROCAE Documents (ED) 136, 137 and 138 documents by the EUROCAE council in February 2009, the second Plugtests™ event involving more complex Stage 2 interoperability tests had the scope of performing further mandatory SIP Telephone and Radio interface interoperability tests defined by the draft EUROCAE ED-139 document in order to verify their correct functionality over a Local Area Network.

Several new companies which had not participated in the previous Stage 1 event held in April 2008 were given the opportunity to perform Stage 1 tests in the week prior to the second Plugtests™ event, from 25th March to 27th March.

The results of the multiple interoperability test scenarios achieved by the European (7 VCS and 4 GRS) vendors have demonstrated a high rate of success:

- Interoperability VCS-VCS : 95,7% (423 tests OK for 442 run)
- Interoperability VCS-GRS : 95,6% (483 tests OK for 505 run)

These results show that the VoIP call types and the wide range of ATS (Air Traffic Services) features specified by the ED 137 interoperability documents, supporting the Operational and Technical Requirements defined by the ED 136 document have now been developed and implemented by the main European VCS and Radio Suppliers with a high level of interoperability achieved. This will lead to ATM VoIP VCS and GRS deployment by ANSPs (Air Navigation Service Providers) in the very near future for operational use in the framework of the Single European Sky (SES).

A further EUROCAE Plugtests™ Interoperability Event for SIP –MFC-R2 and SIP-ATS-QSIG gateway testing is planned for September 2009 and a 3rd Plugtests™ event could take place in 2010 should there be sufficient demand from the European and Global vendors in the market. It is proposed that the event includes a new series of Optional feature tests as defined by EUROCAE documents and is also expanded to include interested Recorder vendors resulting in a Plugtests™ specification for SIP Voice Recorders developed according to the ED137 Part 3 Recorder document.

Many VCS and Radio vendors expressed their opinions during the event and have indicated that Industry has now invested heavily in the development and testing of the SIP User Agents for their products according to the EUROCAE documents. These have now been shown to be at a very advanced point in their development. Many vendors now see the next steps should relate to Voice Quality, Call Performance tests etc over a Wide Area Network. Resources for the implementation of such tests couldn't be funded by Industry themselves and would have to be funded by a central European Institute or through a central European Programme.

History

Document history		
<Version>	<Date>	<Milestone>