ETSI CTI Plugtests Report <1.0.0> (2011-06)

4th EUROCAE Plugtests™ Interoperability Event on VoIP for ATM; Sophia Antipolis, France; 20th June to 24th June 2011

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1 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.

[ED 136] **EUROCAE** ED-136: "Operational and Technical Requirements", February 2009

[ED 137A-1] **EUROCAE** ED-137A "Interoperability Standards for VoIP ATM components, Part 1: Radio", September 2010

[ED 137A-2] **EUROCAE** ED-137A "Interoperability Standards for VoIP ATM components, Part 2: Telephone", September 2010

[ED 137A-3] **EUROCAE** ED-137A "Interoperability Standards for VoIP ATM components, Part 3: Recorder", February 2009

[TEL-TEST] "Telephone Test Case Specification for Voice over IP in ATM" **– Eurocontrol –** Edition 1.0 – 28 February 2011

[RAD-TEST] "Radio Test Case Specification for Voice over IP in ATM" – **Eurocontrol** – Edition 1.0 – 28 February 2011

[REC-TEST] **FAA** "VoIP-IE-Recorder Test Specification" – Draft V1.2 – 10 April 2011

[REC-GUIDE] **FAA** "VoIP-IE Guidelines for Recorder tests" – Draft V1.3 – 14 April 2011

[Plugtest2] **ETSI** "2nd EUROCAE Plugtests Interoperability Event on VoIP for ATM – Report" – V 1.0.0 – April 2009

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
ENAC	Ecole Nationale de l'Aviation Civile
GRS	Ground Radio Station
HE	Header Extension
IP	Internet Protocol
NA	Not Applicable
NO	Not OK
OK	OK
OT	Out-of-Time
PCM	Pulse Code Modulation
PT	Payload Type
PTT	Push-To-Talk
PTT-id	Payload Type- Identification
REC	Recorder system
Rx	Receive
RfC	Request For Comments
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol

SDP	Session Description Protocol
SES	Single European Sky
SIP	Session Initiation Protocol
SQL	Squelch
Tx	Transmit
UA	User Agent
URI	Uniform Resource Indicator
VCS	Voice Communication System
WG67	Working Group 67

3 Fourth EUROCAE Plugtests™ Interoperability Event on VoIP for ATM

The fourth EUROCAE Plugtests™ Interoperability Event on VoIP for ATM (Air Traffic Management) held at the Sophia Country Club between 20th June and 24th June 2011 had the following scopes:

- perform interoperability tests with the new version of the EUROCAE Standard
- perform interoperability tests between VCS, GRS and Recorders.

The fourth 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, Radio and Recorder documents that appear to require further clarification in order to make the interworking between systems more robust.

The results of the multiple interoperability test scenarios achieved by the suppliers have demonstrated a high rate of success:

- Interoperability VCS-VCS : 100 % (300 tests run in 12 sessions)
- Interoperability VCS-GRS : 99,6% (509 tests run in 20 sessions, 2 tests failed in two separated sessions)
- Interoperability VCS-REC : 100 % (48 tests run in 8 sessions)
- Interoperability GRS-REC : 100 % (20 tests run in 10 sessions)

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] 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 [ED137A-1], [ED137A-2] and [ED137A-3] has also been produced. These recommendations will be examined by the EUROCAE WG67 review team with the scope of enhancing future editions of the document.

3.1 Participants

12 companies contributed to the test results with 4 VCS, 5 GRS and 5 REC, as listed in the table below.

Company Name	Type of system
ATIS	REC
Frequentis	VCS
	REC
JOTRON	GRS
Nice	REC
Park Air System	GRS

Company Name	Type of system
Rhode & Schwarz	GRS
Schmid Telecom	VCS
SELEX	GRS
SITTI	VCS
Telerad	GRS
TOPEX	VCS
	REC
Ultra Electronic	REC

Table 1: participants list

3.2 Acknowledments

This is to acknowledge the effort of

- Andrew Lake, Software Consultant, Park Air Systems UK, for the provision of Wireshark ED137 RTP dissector.
- Christophe Guerber, Voice and Data Aeronautical Networks team, Ecole Nationale de l'Aviation Civile France, for the provision of Wireshark ED 137 RSTP and RTP over TCP dissectors

4 Test overview

4.1 Test Plan

4.1.1 VCS-GRS test plan

During the regular conference call, the participants agreed on a selection of 25 mandatory tests and 2 optional tests extracted from [RAD-TEST]. The following table shows the summary of the test objectives, grouped by specific features.

Group	Test case reference	Status	Summary
Radio SIP session establishment	LAN-RAD-R1	MAN	Normal SIP Radio session establishment/clearing from VCS to GRS for Radio Access key configured in Traffic mode
	LAN-RAD-R2	OPT	Normal SIP Radio session establishment/clearing from VCS to GRS for Radio Access key configured in Rx-only mode
	LAN-RAD-R3	MAN	Normal SIP Radio session establishment/clearing from VCS to GRS for Radio Access key configured in Coupling mode
	LAN-RAD-R6	MAN	Emergency SIP Radio session establishment/clearing from VCS to GRS for Radio Access key configured in Coupling mode
	LAN-RAD-R7	MAN	SIP Radio session request with SIP "From" header address unrecognised by GRS

Group	Test case reference	Status	Summary
	LAN-RAD-R8	MAN	SIP Radio session request with SIP "To" header address unrecognised by GRS
	LAN-RAD-R10	MAN	Two consecutive SIP Radio session establishment requests from 2 User Agents at the same VCS for session establishment to same GRS. Multiple streams from same VCS permitted by GRS
Real Time Session Supervision	LAN-RAD-R2S3	MAN	SIP session clearing from GRS endpoint when R2S-KeepAlive packets are not received
	LAN-RAD-R2S4	MAN	SIP Radio session cleared by GRS when placed in Maintenance mode
	LAN-RAD-R2S5	MAN	SIP Radio session cleared by GRS when its frequency changed
Push to Talk	LAN-RAD-PTT2	MAN	Priority PTT activation, Voice transmission, Priority PTT deactivation
	LAN-RAD-PTT4	MAN	Incoming aircraft call on cross-coupled group frequency f1 triggering Coupling PTT activation on cross-coupled group frequency f2
	LAN-RAD-PTT5	MAN	Multiple SIP Radio session establishment and simultaneous transmission
	LAN-RAD-PTT6	OPT	Normal v Priority PTT activation test on given frequency
	LAN-RAD-PTT8	MAN	Priority v Emergency PTT activation test on given frequency
	LAN-RAD-PTT10	MAN	Normal v Normal PTT activation test on given frequency (PTT summation configured at GRS)
	LAN-RAD-PTT12	MAN	Coupling v Normal/Priority/Emergency PTT activation test on given frequency (Coupling PTT interruption configured at GRS)
	LAN-RAD-PTT13	MAN	Coupling v Normal/Priority/Emergency PTT activation test on given frequency (Coupling PTT summation configured at GRS)
	LAN-RAD-PTT14	MAN	Pre-empting of established Normal SIP Radio session without PTT active by Emergency SIP Radio session
	LAN-RAD-PTT17	MAN	PTT-ON arbitration between transmitters using same frequency without offset
Squelch activation	LAN-RAD-SQ1	MAN	Squelch activation, Voice transmission and Squelch deactivation
Session Description Protocol	LAN-RAD-SDP7	MAN	SDP Best Signal Selection attribute negotiation for SIP Radio session establishment request from VCS to GRS (RSSI method default)
	LAN-RAD-SDP9	MAN	Incorrect SDP frequency identity attribute for SIP Radio session establishment request from VCS to GRS
	LAN-RAD-SDP10	MAN	To verify that a GRS can only be configured as part of one cross-coupled group.
BSS	LAN-RAD-BSS2	MAN	BSS test on multiple receiver frequencies
Simultaneous transmission detection	LAN-RAD-SCT1	MAN	Detection of Simultaneous transmissions

Group	Test case reference	Status	Summary
PTT Id notification	LAN-RAD-PTTid	MAN	PTT identity notification

Table 2: VCS-GRS test plan

4.1.2 VCS-VCS test plan

During the regular conference call, the participants agreed on a selection of 25 mandatory tests extracted from [TEL-TEST]. The following table shows the summary of the test objectives, grouped by specific features.

Group	Test case reference	Summary
Routine direct access call	LAN-BC-R2	Routine Direct Access call from A1 to B1 and B1 clears call
	LAN-BC-R3	Routine Direct Access call from A1 to B1. Call cleared before being answered
	LAN-BC-R4	Routine Direct Access call from A1 to B with non-valid called party address
	LAN-BC-R5	Routine Direct Access call from A to B1 with non-valid calling party address
	LAN-BC-R6	3 consecutive Routine Direct Access calls from A1, A2 and A3 to B1.
Priority Direct Access call	LAN-BC-PC1	Priority Direct Access call from A1 to B1 and A1 clears call
Instantaneous Access call	LAN-BC-IA2	Instantaneous Access call from A1 to B1, B1 responds, A1 deactivates and then re-activates IA key, B1 deactivates IA key, A1 deactivates IA key (check that call is cleared)
	LAN-BC-IA3	2 consecutive Instantaneous Access calls from A1 and A2 to B1 with monitoring enabled at B1
	LAN-BC-IA4	2 consecutive Instantaneous Access calls from A1 and A2 to B1 with monitoring disabled at B1
	LAN-BC-IA5	Instantaneous Access call from A1 to B1 with A1 physically disconnected from network. Verify after timer IA timer T0 expiry (2 seconds) that “IA call failure” is indicated to A1.
SDP/Direct Access	LAN-SDP-DA2	Direct Access call from A1 to B1 with media description defined as audio, RTP/AVP 8 (PCM A-law)
	LAN-SDP-DA6	SDP Media description codec negotiation for Direct Access call from A1 to B1
SDP/instantaneous Access	LAN-SDP-IA5	Instantaneous Access call from A1 to B1 with non-valid media description included in the SDP Message Body
SIP call combination test	LAN-MIX-R2	Routine Direct Access call from A1 to B1. Instantaneous Access call from A2 to B1. Priority Call from A3 to B1.
SIP supplementary service	LAN-SS-POSMON1	Position monitoring by A2 of all transmitted and received audio for G/G and A/G communications in progress at B1
	LAN-SS-POSMON2	Position monitoring by A2 of all transmitted and received audio for G/G communications only in progress at B1

Group	Test case reference	Summary
	LAN-SS-POSMON3	Position monitoring by A2 of all transmitted and received audio for A/G communications only in progress at B1
Broadcast conference supplementary service	LAN-SS-CONF1	Migration of an established 2 party SIP call to a 3 party conference using “A1” or “Conference Focus Entity” as focus
	LAN-SS-CONF3	Establishment of a 5 party conference using "A1" or “Conference Focus Entity” as Focus. Parties either eliminate themselves or are eliminated from conference one at a time by “A1” or “CFE”.
Call Intrusion supplementary service	LAN-SS-CI1	Priority Call from A2 to Busy user B1 answered automatically after B1 clears its active routine call with A1
	LAN-SS-CI2	Priority Call from A2 to Busy user B1 answered manually by B1 after clearing its active routine call with A1
	LAN-SS-CI4	Successful Immediate Priority Call intrusion from A2 to Busy unprotected User B1
	LAN-SS-CI5	Priority Call from A2 to Busy protected User B1 answered manually after B1 clears its active routine call with A1
	LAN-SS-CI6	Priority Call from A2 to Busy unprotected User B1 with another Priority call in progress answered manually after B1 clears its first active Priority call with A1
	LAN-SS-CI7	Priority call intrusion effective between A1, B1 and A2. Normal Call Clearing by A2 (Intruding party)

Table 3: VCS-VCS test plan

4.1.3 GRS-REC test plan

During the regular conference call, the participants agreed on a selection of 2 mandatory tests extracted from [REC-TEST]. To allow the tests to be performed with coherent settings, the participants agreed on following [REC-GUIDE]. The following table shows the summary of the test objectives.

Test case reference	Summary
LAN-REC5	Outgoing Radio Call Voice media and Call Record Data recording test
LAN-REC6	Incoming Radio Call Voice media and Call Record Data recording test

Table 4: GRS-REC test plan

4.1.4 VCS-REC test plan

During the regular conference call, the participants agreed on a selection of 6 mandatory tests extracted from [REC-TEST]. To allow the tests to be performed with coherent settings, the participants agreed on following [REC-GUIDE]. The following table shows the summary of the test objectives.

Test case reference	Summary
LAN-REC1	Direct Access Routine Call Voice media and Call Record Data recording test
LAN-REC2	Direct Access Priority Call Voice media and Call Record Data recording test
LAN-REC3	Instantaneous Access Call Voice media and Call Record Data recording test

LAN-REC4	Override Call Voice media and Call Record Data recording test
LAN-REC7	DA, IA or Radio Call Voice media replay at Replay client (optional)
LAN-REC10	Direct Access Priority Call Intrusion Voice media and Call Record Data recording test

Table 5: GRS-REC test plan

4.2 Test Scheduling

Mo 0700 - 1000	**Welcome session / Set-up of equipment**					
Mo 1000 - 1300	**Pre-Testing: Basic interconnection checks for REC - GRS, REC - VCS, GRS – VCS, VCS - VCS**					
Mo 1400 - 1600			VCS - VCS SITTI - Schmid	VCS - GRS Frequentis - JOTRON	VCS-GRS Topex - Park	
Mo 1600 - 1800	GRS - REC JOTRON – ATIS	GRS - REC Park – Ultra		VCS - GRS Frequentis – R&S	VCS-GRS Topex - SELEX	Frequentis/Topex support for GRS - REC
Tu 0900 - 1100			VCS - GRS SITTI - TELERAD	VCS - GRS Frequentis - Park	VCS-GRS Topex - JOTRON	VCS - GRS Schmid – R&S
Tu 1100 - 1300	GRS - REC JOTRON - Ultra	GRS - REC Park - Nice	SITTI support for GRS - REC	VCS - GRS Frequentis – TELERAD	VCS - GRS Topex – R&S	VCS - GRS Schmid - SELEX
Tu 1400 - 1600	VCS - REC Topex - Frequentis	VCS- REC Frequentis - Topex	VCS - GRS SITTI - Park	VCS – VCS Frequentis – Topex		VCS - GRS Schmid - JOTRON
Tu 1600 - 1800	GRS - REC JOTRON - Nice	GRS - REC Park - ATIS	VCS - GRS SITTI – R&S			Schmid support for GRS - REC
We 0830 - 1030	VCS - REC Topex - Nice	VCS- REC Frequentis - ATIS	VCS - GRS SITTI - JOTRON			VCS - GRS Schmid - Park
We 1030 - 1230	VCS - REC Topex - Ultra	VCS- REC Frequentis - Nice	VCS - GRS SITTI - SELEX			VCS - GRS Schmid - TELERAD
We 1330 - 1530	VCS- REC Topex - ATIS	VCS- REC Frequentis - Ultra				VCS - GRS ad-hoc Schmid/SITTI – any GRS
We 1530 - 1730				VCS - GRS Frequentis – SELEX	VCS - GRS Topex –TELERAD	

<table>
<tr><td>Th 0900 - 1100</td><td>GRS – REC
Park - Topex</td><td>GRS - REC
JOTRON – Frequentis</td><td rowspan="2">VCS – VCS
Topex - Schmid</td><td rowspan="2">VCS – VCS
Frequentis – SITTI</td><td></td><td></td></tr>
<tr><td>Th 1100 - 1300</td><td>GRS – REC
JOTRON - Topex</td><td>GRS - REC
Park – Frequentis</td><td></td><td></td></tr>
<tr><td>Th 1400 - 1600</td><td rowspan="2">GRS - REC ad-hoc
Park/JOTRON –
any REC</td><td rowspan="2"></td><td rowspan="2">VCS – VCS
Topex - SITTI</td><td rowspan="2">VCS – VCS
Frequentis – Schmid</td><td></td><td></td></tr>
<tr><td>Th 1600 - 1800</td><td></td><td></td></tr>
<tr><td>Fr 0900 - 1100</td><td colspan="6" rowspan="2">Reserved for Re-Testing (To be scheduled during the week)</td></tr>
<tr><td>Fr 1100 - 1300</td></tr>
<tr><td>Fr 1400 - 1600</td><td colspan="6">Final wrap-up session / Dismantling of equipment</td></tr>
</table>

4.3 Test infrastructure

The test infrastructure was based on separate physical networks, build on switches. All the physical networks were interconnected through a switch router.

On each physical network, a logical network was defined, based on a subnet of the private IP addresses network 10.0.0.0/8.

Each company had then a dedicated physical and logical network, with a prefix of 10.200.X.0/24, where X allows the identification of the company. The first one hundred IP addresses were reserved for DHCP allocation to allow the connection of specific machines used by the company. The rest of the subnetwork addresses was reserved for fixed attribution for systems under test.

The test infrastructure is depicted in the following schema.

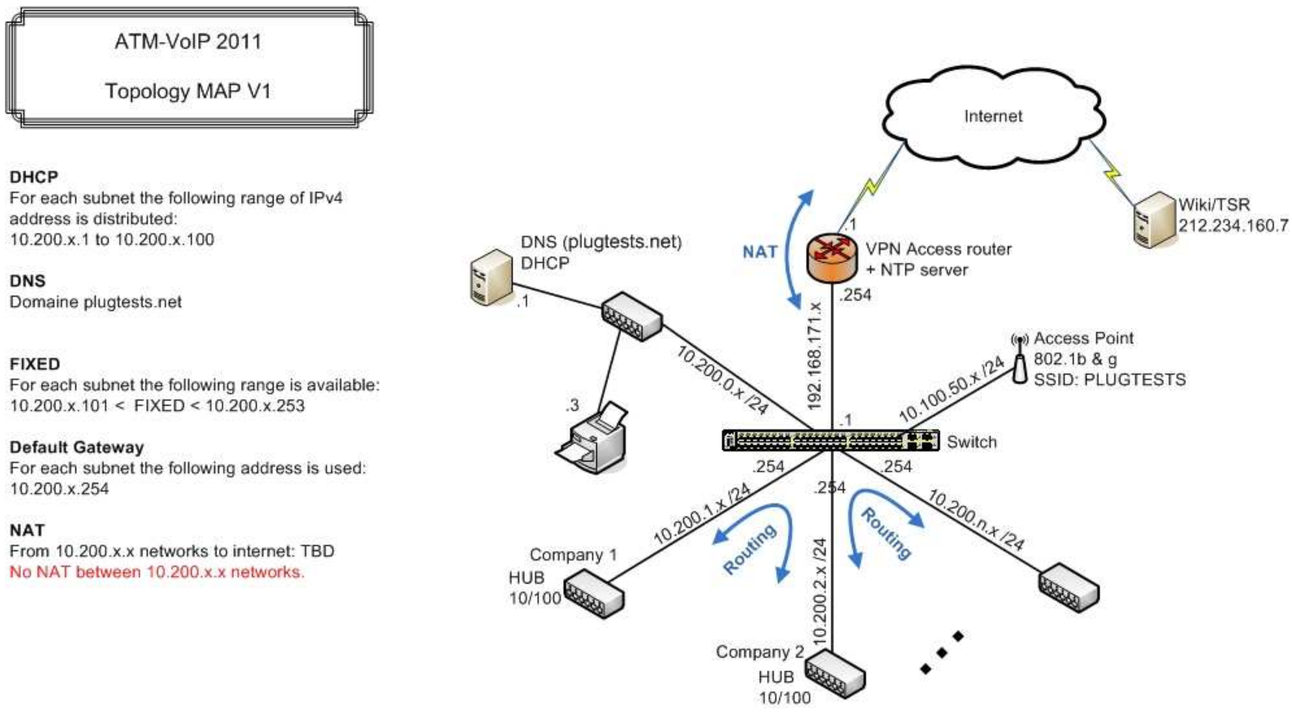


Figure 1: test infrastructure

5 Result summary

5.1 VCS-GRS

5.1.1 Overall results

Interoperability	
OK	NO
507 (99.6%)	2 (0.4%)

Not executed	
NA	OT
31 (5.7%)	0 (0.0%)

Totals	
Run	Results
509 (94.3%)	540

5.1.2 Results per group

	Interoperability		Not Executed		Totals	
Group	OK	NO	NA	OT	Run	Results
Radio SIP Session establishment	**138 (100.0%)**	**0 (0.0%)**	2 (1.4%)	0 (0.0%)	138 (98.6%)	140
Real Time Session Supervision (R2S)	**60 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	60 (100.0%)	60
Push to Talk (PTT)	**186 (99.5%)**	**1 (0.5%)**	13 (6.5%)	0 (0.0%)	187 (93.7%)	200
Squelch activation/deactivation	**20 (100.0%)**	**0(0.0%)**	0 (0.0%)	0 (0.0%)	20 (100.0%)	20
Session Description Protocol (SDP)	**57 (100.0%)**	**0(0.0%)**	3 (5.0%)	0 (0.0%)	57 (95.0%)	60
Best Signal Selection (BSS)	**16 (100.0%)**	**0(0.0%)**	4 (20.0%)	0 (0.0%)	16 (80.0%)	20
Simultaneous Transmissions Detection	**16 (100.0%)**	**0(0.0%)**	4 (20.0%)	0 (0.0%)	16 (80.0%)	20
PTT identity notification	**14 (93.3%)**	**1 (6.7%)**	5 (25.0%)	0 (0.0%)	15 (75.0%)	20

5.1.3 Comments on results

Interoperability defaults were found on tests:

- LAN-RAD-PTT17: PTT-ON arbitration between transmitters using same frequency without offset : PTT Lockout Problem is underlined
- LAN-RAD-PTTId: PTT identity notification: VCS is able to SUBSCRIBE, but radio sends every new NOTIFY with different from tag - VCS declines it.

For the 'not applicable' marks, they are located on few tests:

- PTT group
    - LAN-RAD-PTT12 "*Coupling v Normal/Priority/Emergency PTT activation test on given frequency (Coupling PTT interruption configured at GRS)*": 4 times. The current version of GRS does not support this functionality.
    - LAN-RAD-PTT13 "*Coupling v Normal/Priority/Emergency PTT activation test on given frequency (Coupling PTT summation configured at GRS)*": 4 times. The current version of GRS does not support this functionality.
    - LAN-RAD-PTT17 "*PTT-ON arbitration between transmitters using same frequency without offset*": 5 times. The current version on VCS does not support this functionality.

- Other group: functionality is not implemented in the current version of the VCS

5.2 VCS-VCS

5.2.1 Overall results

Interoperability		Not executed		Totals	
OK	NO	NA	OT	Run	Results
258 (100.0%)	0 (0.0%)	42 (14.0%)	0 (0.0%)	258 (86.0%)	300

5.2.2 Results per group

	Interoperability		Not Executed		Totals	
Group	OK	NO	NA	OT	Run	Results
Routine Direct Access call tests	**60 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	60 (100.0%)	60
Priority Direct Access call tests	**12 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	12 (100.0%)	12
Instantaneous Access call tests	**48 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	48(100.0%)	48
Session Description Protocol/Direct Access	**24 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	24 (100.0%)	24
SDP/Instantaneous Access	**12 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	12 (100.0%)	12
SIP Call combination tests	**12 (100.0%)**	**0 (0.0%)**	0 (0.0%)	0 (0.0%)	12 (100.0%)	12
SIP Supplementary Service Tests	**9 (100.0%)**	**0 (0.0%)**	**27 (75.0%)**	0 (0.0%)	9 (25.0%)	36
Broadcast Conference supplementary service	**12 (100.0%)**	**0 (0.0%)**	**12 (50.0%)**	0 (0.0%)	12 (50.0%)	24
Call Intrusion supplementary service	**69 (100.0%)**	**0 (0.0%)**	3 (4.2%)	0 (0.0%)	69 (95.8%)	72

5.2.3 Comments on results

No interoperability default was found.

For the ‘not applicable’ marks, they are located on few tests:

- SIP Supplementary Service Tests
  - LAN-SS-POSMON1 “*Position monitoring by A2 of all transmitted and received audio for G/G and A/G communications in progress at B1*”: 9 times. The current version of VCS does not support this functionality.

  - LAN-SS-POSMON2 "*Position monitoring by A2 of all transmitted and received audio for G/G communications only in progress at B1*": 8 times. The current version of GRS does not support this functionality.
  - LAN-SS-POSMON3 "*Position monitoring by A2 of all transmitted and received audio for A/G communications only in progress at B1*": 10 times. The current version on VCS does not support this functionality.
- Broadcast Conference supplementary service
  - LAN-SS-CONF1 "*Migration of an established 2 party SIP call to a 3 party conference using "A1" or "Conference Focus Entity" as focus*": 6 times. The current version on VCS does not support this functionality.
  - LAN-SS-CONF3 "*Establishment of a 5 party conference using "A1" or "Conference Focus Entity" as Focus. Parties either eliminate themselves or are eliminated from conference one at a time by "A1" or "CFE"*." 6 times. The current version on VCS does not support this functionality.
- Call Intrusion supplementary service
  - LAN-SS-CI1 "*Priority Call from A2 to Busy user B1 answered automatically after B1 clears its active routine call with A1*": 3 times. The current version on VCS does not support this functionality.

5.3 VCS-REC

5.3.1 Overall results

Interoperability	
OK	NO
35 (100.0%)	0 (0.0%)

Not executed	
NA	OT
13 (27.1%)	0 (0.0%)

Totals	
Run	Results
35 (72.9%)	48

5.3.2 Comments on results

No interoperability default was found.

For the 'not applicable' marks, they are located on 2 tests:

- LAN-REC4: Override call not in Europe
- LAN-REC7: RTSP Playback not implemented (optional test)

5.4 GRS-REC

5.4.1 Overall results

Interoperability	
OK	NO
20 (100.0%)	0 (0.0%)

Not executed	
NA	OT
0 (0.0%)	0 (0.0%)

Totals	
Run	Results
20 (100.0%)	20

5.4.2 Comments on results

No interoperability default was found.

6 Post meeting traces analysis

During the plugtest event, 389 wireshark traces files were uploaded distributed as follows:

- 218 traces for VCS-GRS tests
- 142 traces for VCS-VCS tests
- 18 traces for REC-VCS tests
- 11 traces for REC-GRS tests

As it was impossible to analyse this amount of traces during the meeting, a post meeting trace analysis has been performed. Only comments related to general behaviours will be provided.

6.1 VCS-GRS traces analysis results

The VCS-GRS tests traces were analysed with the help of portable Wireshark 1.4.7 and the plugins provided by Park Air System.

In this part, the term "**coupled sessions entity (CSE)**" defines the couple (SIP Tx session) and (SIP Rx session) used to handle the transmitted and receiving communications between a single VCS and a single GRS, with separated Tx and Rx. In the following picture, the "**coupled sessions entity**" is the couple SIP session 1 and SIP session 2.

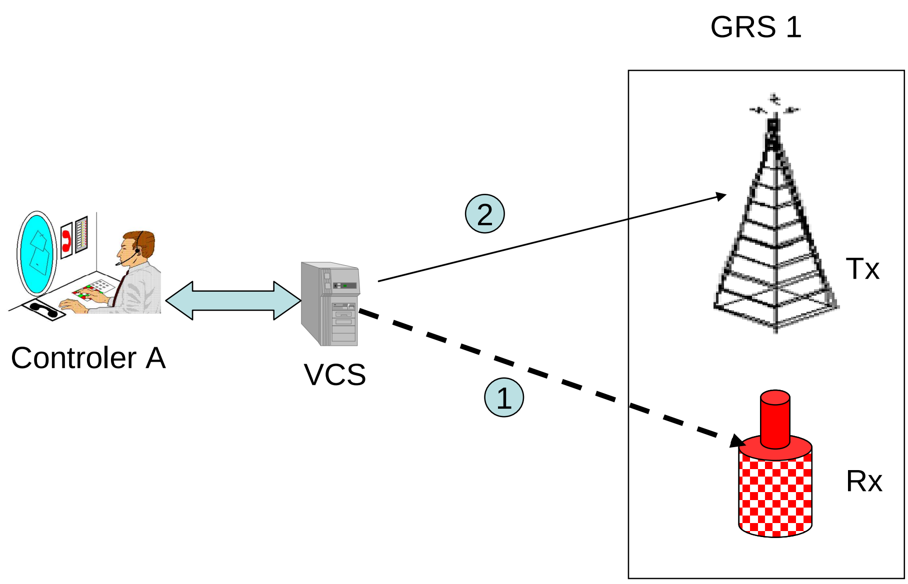
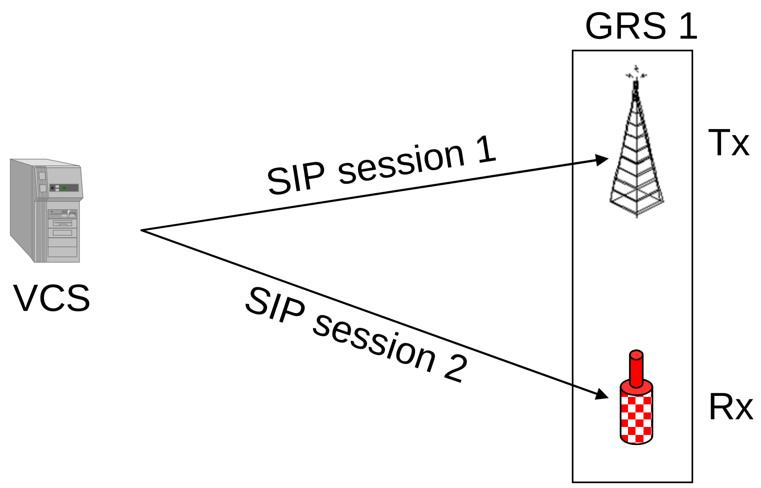


Figure 2: coupled sessions entity

A "**coupled sessions entity**" will be said as established when both SIP sessions are established.

6.1.1 SIP sessions with separate Tx-Rx

The case is the following one:



To establish both sessions, the following methods have been found in the traces:

- try to establish Rx session first. As long as Rx session was not established, no try to establish the Tx session is performed.
- try to establish Rx session first. Whatever the result is, try to establish the Tx session.
- try to establish Tx session first. Whatever the result is, try to establish the Rx session.

When the sessions are closed, no relationship is kept between both sessions.

On an operational point of view, in this case, the communication can be considered as operational only if both SIP sessions are established, in other words, if a "**coupled sessions entity**" is established. As no link is defined between the Tx and the Rx sessions in the EUROCAE Standards, some side effects may occur in case of lack of resources (e.g. with pre-emption case).

As requirement is not set on a "**coupled sessions entity**" but on a single SIP session establishment, the delay for the establishment of both sessions belonging to a single "**coupled sessions entity**" may be rather long (average value around 2.5 seconds). In fact, each SIP session establishment are really compliant with the establishment delay expressed inside [ED 136], but the delay between the first session establishment and the request of the second one takes around 2.3 seconds as an average value.

6.1.2 Pre-emption case

The initial status is described in the following schema.

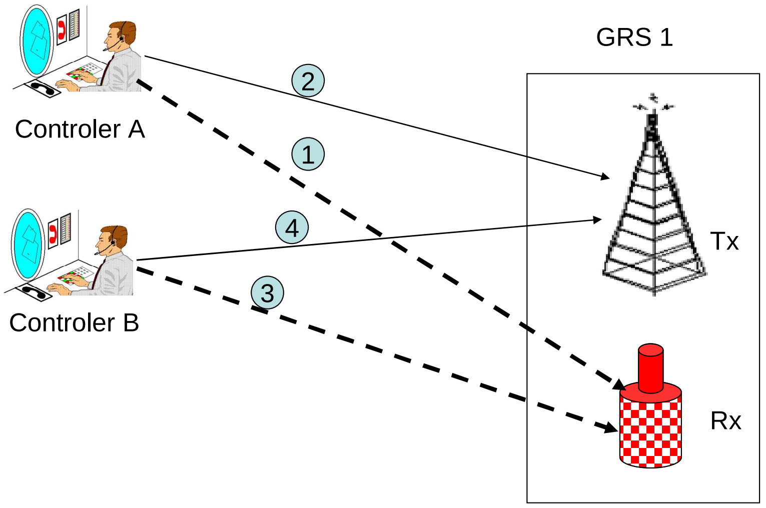


Figure 3: initial state before pre-emption

A first "**coupled sessions entity**" is established between controller A and the GRS, with SIP session 1 for Rx and SIP session 2 for Tx. A second "**coupled sessions entity**" is established between controller B and the GRS, with SIP session 3 for Rx and SIP session 4 for Tx. Both relationships are established with a normal SIP priority.

If a third controller C wants to establish a relationship with the GRS and if the GRS does not allow an additional session, pre-emption will occur following [ED137A-1] §3.8.1.3 provided controller C request an emergency SIP session establishment (Test case: LAN-RAD-PTT14).

By looking at the traces, pre emption really occurred on both Tx and Rx sides. As no real rule is defined inside the EUROCAE standard to select the way pre emption has to be done, the following occurred (half of the traces):

- disconnection of SIP Tx session of one "**coupled sessions entity**"
- disconnection of SIP Rx session of the other "**coupled sessions entity**"

The result of this behaviour is that both **CSE** for controller A and for controller B are no more established. Only controller C has an established "**coupled sessions entity**".

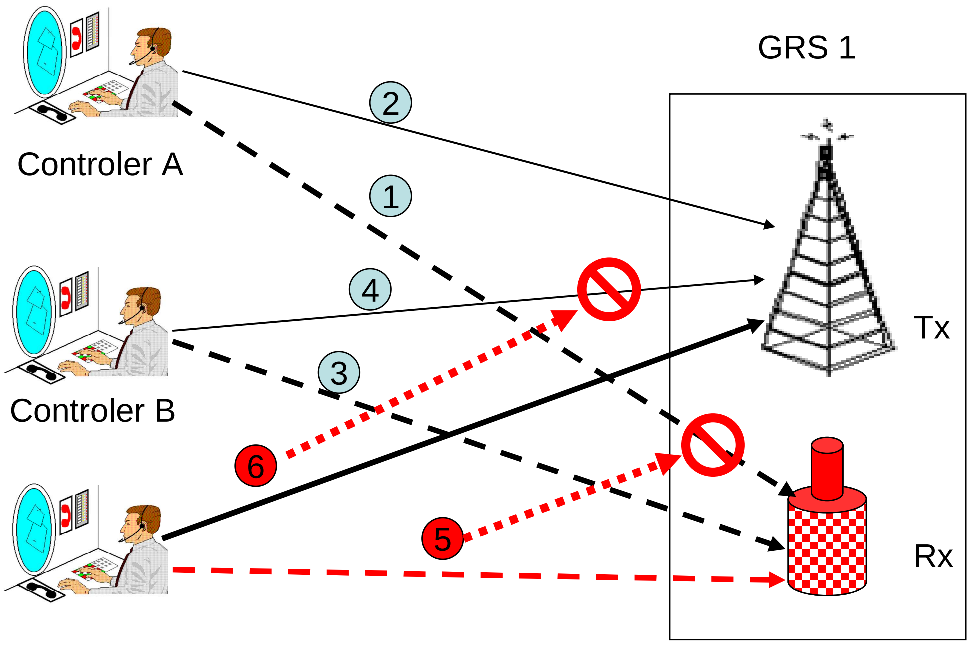


Figure 4: pre-emption result

6.1.3 RTPTx HE

PTT Id is not always set inside RTPTx HE (Keepalive messages).

6.1.4 RTPRx HE with audio transmission

The behaviour of the systems differs provided the system is a combined GRS or a GRS with separate Tx and Rx. To explain it, we will just take a simple case, with one relationship between a VCS and a GRS.

When voice is applied at the VCS:

- In case of combined GRS, GRS receives it, acknowledges the first PTT-ON message immediately. Then GRS sends back an RTP message with:
  - SQU ON
  - PTT type set
  - PTT Id set
  - No audio voice
  - On a keepalive timer basis
- In case of separated Tx – Rx GRS, Tx and most of the time Rx systems immediately acknowledge the first PTT-ON message. Then
  - Rx sends to VCS
    - SQU ON
    - PTT type: quite never set
    - PTT Id: quite never set
    - On a voice packet time basis
  - Tx sends to VCS
    - PTT type
    - PTT Id
    - On a keepalive timer basis

This point was already discussed inside [Plugtest2], §4.16.20.

6.1.5 Coupling cases

During tests using cross coupled frequencies, some loops were detected inside the traces with separated Tx Rx. This loop was not detected during the test, due to the way the test was performed.

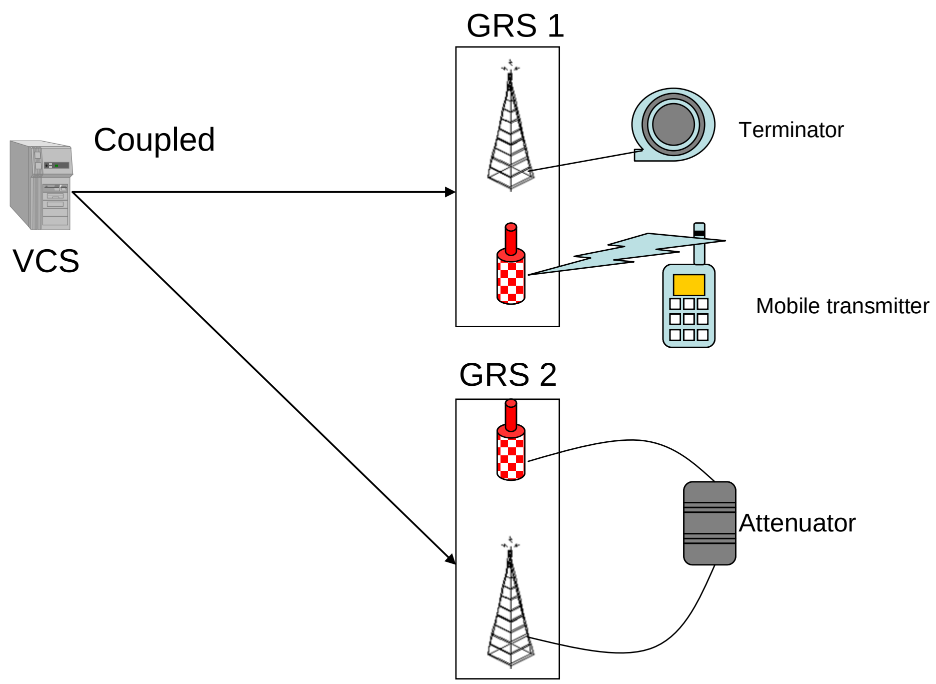


Figure 5: test coupling infrastructure

Voice is applied through a mobile transmitter. The receiver of GRS 1 sends the received audio stream to the VCS. Due to the coupling setting, this audio stream is forwarded to GRS2 transmitter. Then receiver of GRS 2 sends it back to the VCS. If VCS forwards this audio stream back to the transmitter of GRS1, this transmission will not been detected, as the transmitter is connected to a terminator.

This event does not occur during all the transmission, but only at the end of it. When VCS received the squelch OFF from Rx system of GRS1, it sends to Rx GRS2 a keepalive message, with PTT OFF. If the forwarding of the audio messages was not complete at this time, Rx of GRS2 will send back to VCS the remaining audio messages. Then these audio messages are forwarded back to GRS1.

Only few voice packets are impacted (around 3 to 4 packets), for a duration of about 100 ms. This behaviour should have been avoided by the implementation of the XC2 timer, as defined in [ED136] §2.1 and Figure 7.

This case underlines also another problem, in relationship with the network latency on a “**coupled sessions entity**” that is not described inside the standards. Let’s take the case where the latency on the VCS->Tx path is around 80 ms and the latency on the VCS->Rx path is around 10 ms. This case is illustrated in the following schema.

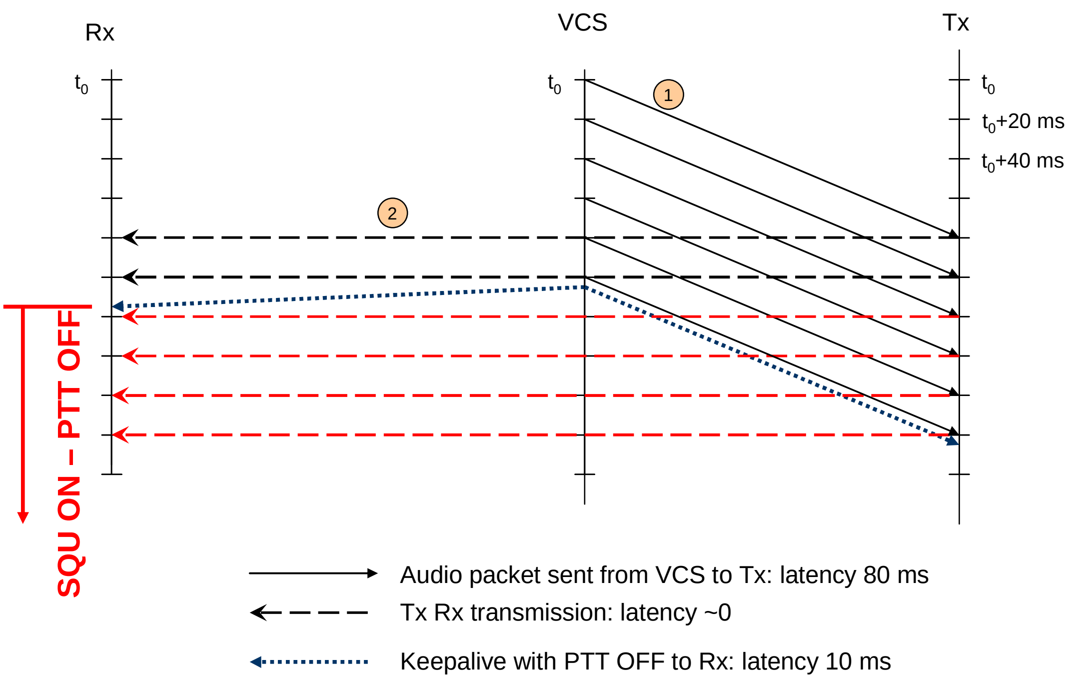


Figure 6: latency effect on separated GRS

The VCS sends its transmission to the transmitter Tx, which forwards it (1). The signal is received by the Rx system (2). At the end of the transmission, the VCS sends the PTT OFF value. This piece of information (blue dash line) is received by the Rx system whilst 4 RTP packets (due to the figures of the example) were not sent. Then the last 4 RTP packets will be sent back to the VCS with a PTT off value. This may happen as soon as the latency on the Rx path will be smaller than the latency on the Tx path.

6.1.6 SIP CANCEL

The CANCEL request is not always handled as defined inside RfC 3261. The RfC 3261 clearly expresses that a CANCEL request shall be discarded by the server if a final response has already been sent. If the final response is a rejection of the session establishment through an error message, transaction is considered as closed. If a positive final response (200 OK) has already been sent, the following diagrams shall apply:

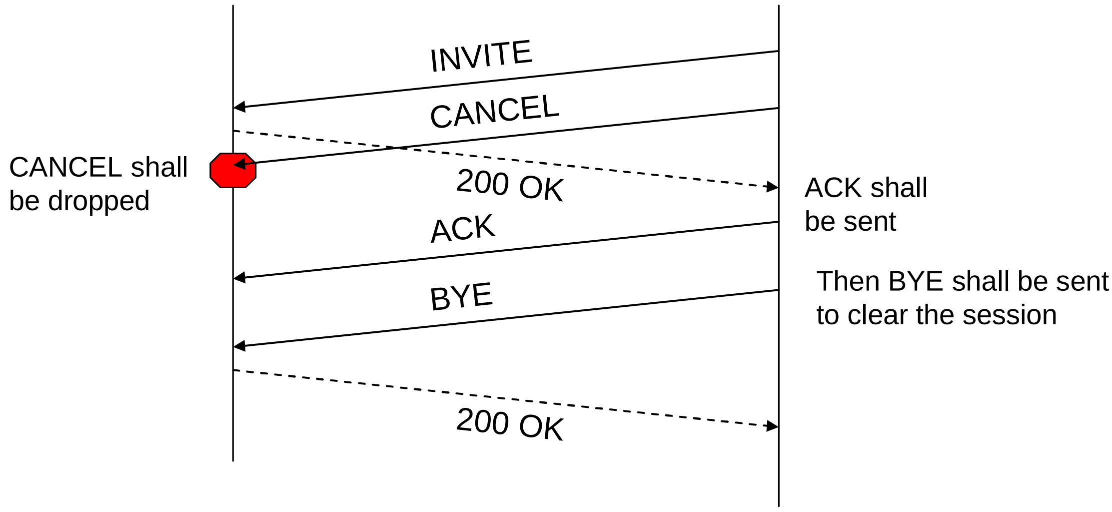


Figure 7: CANCEL request reception after final response to INVITE

6.1.7 LAN-RAD-PTT6

The purpose of this test is to check what may happen when a transmission is activated with a PTT type set to priority (UA2 transmitting) while another transmission with a normal PTT type is on going (UA1 transmitting).

For this test, pre-emption of UA2 transmission always is always done. The difference between the implementations occurred when UA2 stops its transmission. Some VCS does not allow UA1 to transmit again when UA2 transmission is finished unless UA1 releases its PTT key and presses it again.

[ED137A – 1] §5.5.6 (Multiple RTP audio stream management at GRS transceiver/transmitter) defines what shall be done when the second audio stream is received at the GRS side. But no clear definition is done on what shall be done neither at the VCS side nor when this second audio stream stops. The use of the verb "interrupt" in this case may lead to different understandings.

6.1.8 Detected non compliance/bad settings with no impact

6.1.8.1 URI

When sending the SIP ACK message, the request URI is sometimes in lower cases (e.g.: tx instead of TX in the other messages). RfC 3261, §19.1.4 clearly expresses the fact that the user part of an URI is case sensitive (page 154, example :

```
SIP:ALICE@AtLanTa.CoM;Transport=udp             (different usernames)

   sip:alice@AtLanTa.CoM;Transport=UDP
```

Then, even if no interoperability issue was found with this, it seems better to correct this point to be compliant with RfC3261.

6.1.8.2 Separate Rx Tx devices – SDP attributes txrxmode

Sometimes the SDP attribute txrxmode is set in an improper way (TxRx instead of Rx only) in the SIP INVITE request. In these cases GRS always follow the standards by answering with a proper txrxmode set to Rx.

6.1.8.3 SDP bss attribute

BSS attribute is set in INVITE sent to Transmitter only. Even if this may not lead to interoperability issues, this attribute has no link with the transmitter and should not be set.

6.1.8.4 SDP ptime attribute

The SDP ptime attribute is used sometimes. This parameter is ignored, due to [ED137A-1] §3.6:

> "The SDP types and parameters indicated in Table 7 **SHALL** be supported; those received SDP types and parameters not included in Table 7 **SHALL** be ignored."

6.2 VCS-VCS traces analysis results

The VCS-VCS tests traces were analysed with the help of Wireshark 1.6.0 and the integrated plugins (Telephony plugins).

6.2.1 Call Intrusion

The following behaviour was often found:

- intrusion occurred, the intrusive call receives a first temporary response Ringing
- before sending the reINVITE to the already connected UA, a temporary response "intrusion in progress" is sent to the new comer
- then reINVITE is performed.

This behaviour is not compliant with [ED137A-2], figure 20. The new comer shall receive first a temporary 182 Queued response, which was never found in the traces. Then conference is built with a reINVITE sent to the already established session. Once this exchange is acknowledged, a final response is sent to the new comer.

6.2.2 Monitoring

During IA call, when monitoring is enabled, echo is audible especially once the called party pressed its IA key. This may due to both RTP channel sending back the users voice. This side effect is consistent with [ED137A-2] §4.3.3:

> "In this case, since two sessions (A -> B & B -> A) coexist for a single IA call, VCS systems SHOULD remove the calling party's voice and the called party's voice from respective monitoring channels to avoid possible problems derived from the same audio coming to a party's terminal through two different RTP paths."

6.2.3 Performances

The RTP Wireshark plugins computes the following mean jitter values:

- at the sending side, the mean jitter value is less than 0.5 ms
- at the receiving side, the mean jitter value is around 3 to 5 ms.

As the network is rather small and the load is not heavy, it is impossible to draw conclusion.

6.2.4 Voice quality with conference/monitoring

Conference and monitoring often lead to a degradation of the voice quality, especially with the emergence of echo effects.

6.3 REC traces analysis results

The captured files were analysed using a modified Wireshark, version 1.4.7, implementing ED137 specificities concerning RTSP and RTP, in particular Call Record Data dissection and RTP over independent TCP framing (ENAC implementation). Some of the RTP streams could be extracted and replayed.

Some comments are common to all the tests. Some comments are linked only with the VCS-REC tests, some others only with GRS-REC tests.

6.3.1 Common comments

6.3.1.1 SDP messages in the ANNOUNCE

They should contain the Audio Media Codec description defined as “m=audio 5004 RTP/AVP/**TCP** 8”. TCP is often forgotten.

This is to be compliant with RfC 3551/2327 which underlines that without transport field, it is assumed that UDP is implicit when RTP/AVP is used.

Nevertheless, this will not lead to interoperability issue, as the SETUP message will provide the transport information.

6.3.1.2 SETUP and CRD

CRD are often sent inside the SETUP message. This should be avoided. The SETUP message, as defined in RfC 2326, is reserved for transport settings only, so as to give firewalls and other intermediate network systems the information needed to perform their work properly.

This is also reminded in [ED137A-3] §2.6.4: SET_PARAMETER

> “This message **SHALL** be used to set the value of a parameter (call record data) for a presentation or stream specified by the URI (request and response)”

6.3.1.3 Voice quality

The quality of the voice is sometime rather bad. This has no link with the recorder by itself (direct listening to RTP stream).

6.3.2 REC-GRS tests analysis

The test traces show a procedural compliance of the systems except on the common remarks.

6.3.2.1 PTT and SQU operation

The value for squ/ptt off properties is generally not compliant with [ED137A-3] Table4: a value of 0 is transmitted whereas the ED document defines a value of 2 for ptt/squ off.

6.3.2.2 Performance issue

The signalling and the data uses separate TCP connections. Some desynchronisations have been found in the traces. Let’s illustrate this comment with an example from one trace.

For the signalling channel:

- at t=0, a PAUSE message is sent to the recorder. 3.60 seconds after, a RECORD message is sent.
- With the timestamp inside the RTSP messages, the delay between both event is 3 seconds, but the timestamp was not precise enough (range of 1 second).

For the RTP channel:

- at t= -105 ms, a TCP message is sent, which carries the last part of an RTP packet, with an RTP timestamp value of 62400.
- At t=95 ms, TCP, the recorder acknowledges the TCP segment carrying the previous RTP packet
- At t= 298ms, the last RTP packet is sent inside a TCP message carrying 5 RTP packets. The last RTP packet timestamp value is 63200.
- At t=3.888 sec, the first RTP packet is sent, with a first RTP timestamp value of 63840.

In this example, the halt in the recording has a duration of 3.6 seconds, while the RTP stream is only halted during 640 ticks (80 ms, as one tick is added by sampling time).

No further investigation was possible to find where the origin of the problem was.

6.3.3 REC-VCS tests analysis

6.3.3.1 CallingNr and CalledNr

The CallingNr and CalledNr are not always coded with a proper format (URI).

6.3.3.2 Beginning of recording session

The recording session is not always started (ANNOUNCE message) as soon as the phone call is initiated. Some traces show an ANNOUNCE message sent after the SIP session establishment (SIP ACK receipt/sent).

6.3.3.3 Timestamp

The timestamp does not always have a resolution down to milliseconds.

7 Recommendations to WG 67

As no trouble was detected during the event, the following proposals have been done after the meeting. Two WEB conferences were held after the meeting to find a common agreement on these proposals. Nevertheless, they may not reflect the wanted of all the participants. It is then up to the EUROCAE WG 67 to deal with these proposals.

7.1 Recommendations for ED 137A part1 – Radio

7.1.1 Definition of a coupled sessions entity for separated GRS

The “coupled sessions entity” is defined in §6.1.1. It has been underlined that the lack of this definition inside the EUROCAE Standards avoids the clear definition of what shall be done in several cases. Then some effects appears (e.g. when session pre emption occurs). It is then proposed to add the following definition in [ED137A-1] § 2.1 Definitions

> *CSE: Coupled Sessions Entity: when a VCS uses a separate GRS to both send and receive audio messages, the two separated SIP sessions established between VCS and Tx one side and VCS and Rx on the other side are grouped together in a logical entity called “Coupled Session Entity”. This entity is defined to ease the way the standard is written. It does not imply the implementation of any specific structure inside any component: the way the behaviour is implemented when CSE concept is used is implementation specific.*

During SIP session pre-emption tests, it happened that two already established CSE was affected in the same time (one CSE loose the Tx session while the other one loose the Rx session). It is then recommended to investigate if a solution is available to avoid this.

7.1.2 Definition of the Rx behaviour

When a VCS sends a message to a GRS, the only place where a definition exists of what shall be done is found in [ED137A-1] Figure 15. By looking at this figure, it seems that the receiver sends back to the VCS the received message. But no explicit requirement is defined. Furthermore, as the figure is inside a note, it looks like a clarification only.

To explain the proposition, it seems important to provide some illustration. Inside the following figures, the GRS can be a combined or a separated one.

Case 1: 2 controllers using a single frequency. An aircraft transmission occurs.

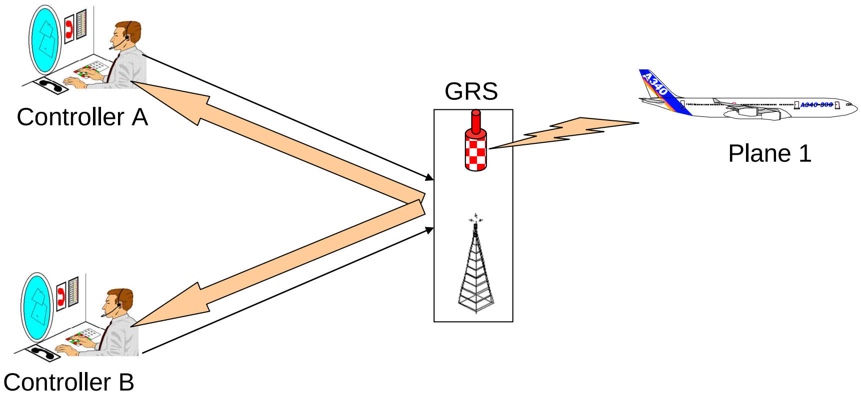


Figure 8: 2 controllers using a single frequency - example 1

This is a trivial case. Rx receives a transmission and forwards it to both controllers without any thing else to do.

Case 2: 2 controllers using the same frequency. Controller A speaks.

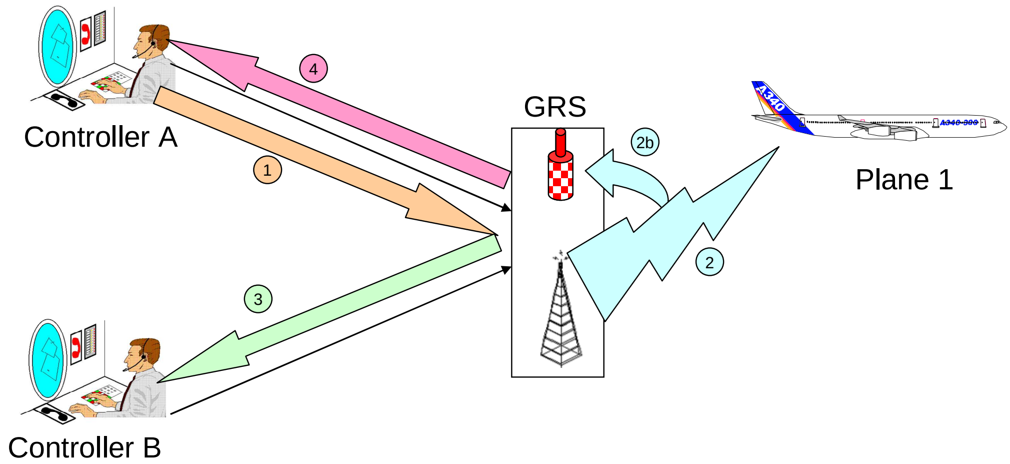


Figure 9: 2 controllers using a single frequency - example 2

Controller A applies its speech (1); Tx sends it to the plane (2). At an operational level, it is difficult to understand this case if both controllers are not responsible of adjacent sectors. Then, it seems to be an operational need that controller B receives also this message.

This case is taken into account in [ED137A-1] §5.5.3.2:

> "When a GRS Transceiver/Receiver endpoint detects an incoming RF signal at the antenna it activates the SQUELCH signal and **SHALL** send an RTP audio packet with RTP Header Extension set to SQL ON and PT=Codec Number type (i.e. PT=8 for G.711 codec) towards the VCS endpoint every RTP audio packet period (i.e. default value 20ms)."

As no distinction is made on the incoming RF signal, this requirement applies to controller A transmission. Then RTP packets shall be sent to both Controller A and controller B with audio samples. This implies 3 and 4 in example 2.

Furthermore, with [ED136] 40[REQ RADIO FUNCTIONAL], controller B shall know who is speaking. This requirement is covered in [ED137A-1] § 3.6.1.12, p28:

> "Likewise a User Agent at a VCS endpoint with a SIP session established with the GRS Transceiver/Transmitter endpoint **SHALL** also be notified of the ptt-id of the User Agent currently transmitting at the GRS Transceiver/Transmitter through information sent by the GRS endpoint in the ptt-id field of the Receive Path RTP Header Extension (see 5.5.4 RTP Header extension)."

In this requirement, it may be understanden that only RTPRx sent back from Tx in case of separated GRS shall send this piece of information back to the VCS. This remark also applies to [ED137A-1]§5.5.5 and §5.5.7 for PTT type and PTT-Id. Further more [ED137A-1]§5.5.7 - PTT type says:

> "In case of an Aircraft Call (A/C call) reception, the **ptt-type value** sent in the RTPRx HE **SHALL** be set to PTT OFF. If this field is not used, it **SHALL** be set to **0. "**

Taking into account that A/C call is defined in [ED137A-1] as "a transmission being received", it implies that RTPRx Information field from the transceiver/receiver endpoint shall always have a PTT Id of 0.

This is not consistent with [ED137A-1] figure 15. This non consistency relies on the use of the same wording for two separated events: real aircraft transmission on one side, squelch activity on the other side. It is then proposed to have two separated definitions to avoid this:

> *Aircraft call: transmission received from an aircraft*
>
> *Squelch activity: transmission received on the Rx. It may be either an aircraft transmission or the receipt of a VCS transmission from the associated Tx.*

7.2 Recommendations to ED 137A part2 – Telephone

No real recommendation may be issued from the event results.

7.3 Recommendations to ED 137A part3 - Recorder

7.3.1 Separate or combined IN/OUT channels

[ED137A-3] §3.1 requires the VCS provides a summarized audio signal (IN and OUT) as a single coded PCM stream that is sent to the Recorder. This requirement does not rely on an operational requirement neither listed in [ED136] nor in ICAO –Annex 10, Volume II, Chapter 3.5 relating to recording of ATC communications.

The legal recording allows the analyse of what may happen on a CWP. It is then important to be able to separate the input stream from the output stream. This will not be possible with a combined analogical audio signal adding both input and output signal at the CWP. It seems then more relevant to have a separate recording, with one channel for the input signal and one channel for the output signal. Further more a separate recording of input and output channels seems more compliant with [ED136] §5.3 REQ 1:

> "a true and faithful representation of the audio signals being presented at the points detailed in 4"

Finally, when a controller is transmitting, its speech is sent back by the receiver. Then the same signal will be mixed with an offset equal to the sum of the latency of the Tx and Rx path (between 100 to 200 ms). This will induce a degradation of the recorded voice, which is not compliant with [ED136] §5.3 REQ1:

> "Audio quality **SHALL** neither be degraded nor improved".

> ***It is then proposed to remove the requirement of a single coded PCM stream, summarizing the IN and OUT channels. The possibility of configuring the recording of either separated IN/OUT streams or a summarized IN/OUT streams should be provided***.

7.3.2 true and faithful representation of the audio signals

[ED136] §5.6 REQ1requires a "true and faithful representation of the audio signals". This requirement is not understanden in the same way by all the participants.

Some participants interpret this requirement in term of RTP data loss. They reject then the use of UDP as an unreliable protocol.

Some others interpret this requirement at the audio signal level. With this interpretation, the loss of some RTP packets is allowed, provided it does not reach too high a level and are in favour of UDP usage.

As no common agreement exists on this point and as it is a key point to go further, no firm technical proposal will be done on this topic.

As a key point, it is then proposed to EUROCAE WG67 to fix this issue first before studying any technical solution.

However, the technical solution used during the event (RTSP over TCP and RTP over independent TCP) shows weaknesses. As it was defined, its use in an operational context is questionable.

Depending on the decision taken by WG67, ideas being considered could be:

- if any data loss shall be avoided:
    - then RTP over UDP is precluded
    - as TCP is not designed to handle real time communication, the recording shall not be designed as a real time recording as expressed in [ED137A-3] §2.4.2 : "User Terminals SHALL use RTSP to enable controlled, on-demand delivery of real-time data" (a pseudo real time recording design may be enough).
        - E.g. real time memory at the recording points inside a file and sending this file on TCP to the recorder as soon as the recorded communication is finished or after a certain delay
        - …
- If data loss is not precluded
    - Then RTP over UDP can be used
    - The maximum value the data loss rate may reach shall be defined
    - …

In both cases, tests shall be performed to validate the proposed solution.

.

7.3.3 RTSP transport

As RTSP is an acknowledged protocol, it may be played over either TCP or UDP, without any loss. As only few data are exchanged between the client and the server, the use of TCP will not show as many default as its use with the RTP stream.

7.3.4 SIP usage for recording

During the event, SIP was not used for recording purpose. This did not lead to any defect.

It is then recommended to remove all the reference to SIP in [ED137A-3]

7.3.5 Phone call record data

7.3.5.1 The priority property

The priority property is defined inside [ED137A-3], with an unclear mapping between the given levels and their real meaning. [REC-GUIDE] provided the following mapping:

SIP Priority header setting	***SET_PARAMETER Priority value***
Emergency	*1*
Urgent	*2*

Normal	3
non-urgent	*4*

Table 6: mapping between SIP priority header and priority property value

As this proposition is consistent with the actual implementations, it is proposed to include it in [ED137A-3]

7.3.5.2 SetupTime-AlertTime-ConnectTime- DisconnectTime

[ED137A-3] does not provide a clear way of recording the status of a specific phone communication. During the tests, except the mandatory properties, only the optional property "DisconnectTime" has been used. It seems relevant to clearly define the meaning of the following phone properties:

- SetupTime: proposition: *the time of the initial communication request of the user (time of the SIP invite request)*
- DisconnectTime: proposition: *the time of the SIP BYE request in case of point to point communication – the last user SIP BYE request in case of conference*

It is also proposed to make these properties mandatory.

For the following properties, agreement was found on the following definition:

- AlertTime: *proposition: the time of the Ringing temporary response (time of the SIP 180 Ringing answer)*
- ConnectTime: *proposition: the time of the call established status (time of the SIP 200 OK answer)*

7.3.6 Radio call record data

7.3.6.1 PTT operation

Given to [ED137A-1] § 5.5.5, 5 levels of PTT have been defined. Depending on the selected PTT level, a controller speech may be blocked or transmitted. It seems then relevant to record this level to understand what happen to a controller communication.

It is then proposed to code and record the PTT value as it appears inside the RTP Tx Header extension inside the PTT operation.

PTT operation value	Description
0	PTT OFF
1	Normal PTT ON
2	Precluded (see Note)
3	Priority PTT ON
4	Emergency PTT ON

Table 7: PTT operation values

Note: PTT value of 2 codes the COUPLING PTT. This value cannot be requested by the controller when he presses its PTT key.

Remark: the way PTT was coded in [ED137A-3] Table 4 suggests that this operation is only in link with an action of the controller. In fact, at the exit of the VCS (Rx channel), this information is also relevant provided it is associated with the PTT Id, which is provided to the controller following [ED136] §2.4 REQ40 RADIO FUNCTIONAL.

7.3.6.2 SQU operation

According to [ED137A-1] §5.5.7, the squelch indication is coded on a single bit, with the value 0 for Squelch OFF and the value of 1 for Squelch ON. It seems more relevant to adopt the same coding for the squelch operation, rather than the one proposed in [ED137A-3] table 4.

7.3.6.3 Simultaneous Transmission operation

According to [ED137A-3] Table 4, this operation is optional. Most of the time, this piece of information is not necessary, as the simultaneous transmissions may be heard. But that may not be the case. If the GRS was able to detect it, it would be a pity not uses it. On the other hand, the meaning of this operation may be reduced to the meaning of the SCT bit of the RTP Rx information field ([ED137A-1]§5.5.7).

As for PTT operation, this operation applies to both Tx and Rx recording.

It is then proposed to define the Simultaneous transmission operation as mandatory, with only 2 values, identical to the SCT bit setting and meaning.

8 Conclusion

The fourth EUROCAE Plugtests™ Interoperability Event on VoIP for ATM (Air Traffic Management) has shown a real increase of the interoperability between the VCS and GRS provided by various suppliers in comparison with the previous events. This result should be particularly underlined that the tests used a new updated version of the EURACAE standards and that several newcomers attended this event.

The VCS-VCS interoperability tests and the resulting traces analysis show a high level of maturity of the EUROCAE ED137A part 2 document.

The more important improvement of the standard for the radio communication explains a lower maturity level of the EUROCAE ED137A part 1 document. Nevertheless, the problems found rely more of a lack of explanation inside the standards and the interoperability is not really impacted.

Regarding the recorder part of the standardisation, results are more difficult to analyse. At a first sight, they seem rather good. A deeper analysis based on the tests traces in comparison with the standard raises some concerns. Many settings were only based on a mutual agreement not described inside the standards and the technical selected solution seems to have some weaknesses.

Regarding the quality of the improvement of EUROCAE ED137A part 1 and part 2 after the previous Plugtests™ Interoperability Event on VoIP for ATM and the quality of the systems that has been manufactured with these improved standards, it is foreseen that the EUROCAE ED137A part 3 will follow the same improvement process to reach the same level of maturity of the first two parts.

9 History

Document history		
<Version>	<Date>	<Milestone>
0.1	07/23/2011	Draft version
0.2	07/25/2011	Internal ETSI reviewing
0.3	08/04/2011	Event's participants review
1.0	08/12/2011	First public version