ETSI CTI Gateway Plugtests Report <1.0.0> (2010-03)

3rd EUROCAE VoIP for ATM Plugtests™ -Gateway Interoperability Event; Sophia Antipolis, France; 1st March to 5th March 2010

World Class Standards

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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", February 2009

[i.3] **EUROCAE** ED-137 "Interoperability Standards for VoIP ATM components, Part 2 : Telephone", February 2009

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
ATS-QSIG	Air Traffic Services- QSIG signalling protocol
ATS-R2	Air Traffic Services –R2 signalling Protocol
CICL	Call Intrusion Capability Level
CIPL	Call Intrusion Protection Level
CPIPL	Call Priority Interrupt Protection Level
CPICL	Call Priority Interrupt Capability Level
CWP	Controller Working Position
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Service
ED	EUROCAE Document
ETSI	European Telecommunications Standards Institute
IP	Internet Protocol
IUT	Implementation Under Test
LD-CELP	Low-Delay- Code Excited Linear Prediction
LSB	Least Significant Bit
MSB	Most Significant Bit
NAT	Network Address Translation
NA	Not Applicable
NO	Not OK
OK	OK
OT	Out-of-Time
PCM	Pulse Code Modulation
PT	Payload Type
RFC	Request For Comments
RTP	Real Time Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TSR	Test Session Report
UA	User Agent
URI	Uniform Resource Indicator
VCS	Voice Communication System
WG67	Working Group 67

4 Third EUROCAE VoIP in ATM Plugtests™ - Gateway Interoperability Event

The third EUROCAE VoIP in ATM (Air Traffic Management) Plugtests™ Event held at the ETSI headquarters in Sophia Antipolis between 1st March and 5th March 2010 had the scope of testing the Telephone Gateways Interoperability of 6 VCS suppliers that attended the event. Six VCS suppliers tested their Gateway for SIP v ATS-R2 interworking while 4 VCS suppliers also tested their Gateway for SIP v ATS-QSIG interworking.

Tests defined in the EUROCONTROL Gateway test specifications were employed during the event. These define all the relevant SIP v ATS-R2 and SIP v ATS-QSIG gateway interworking tests. Tests defined in these specifications have been divided into two groups for Incoming and Outgoing Gateway and further divided into 3 sub-groups relating to Direct Access Call, Call Interruption and Call Intrusion tests.

The interoperability tests defined in these specifications relate to Signalling Interworking between SIP and ATS-R2 as specified in Chapter 4 of the EUROCAE ED-137 Part 2 Telephone document February 2009. and to Signalling Interworking between SIP and ATS-QSIG as specified in Chapter 6 of the EUROCAE ED-137 Part 2 Telephone document February 2009.

4.1 Reflections on the event

The event had good industry participation with 6 VCS suppliers attending. 6 VCS suppliers performed SIP V ATS-R2 interworking tests while only 4 of the original 6 performed SIP v ATS-QSIG interworking tests. The Gateway interworking test specifications produced by EUROCONTROL were used for event and distributed 3 months prior to the event made it possible for the VCS suppliers to agree on which tests should be implemented for Gateway Plugtests event.

There was active involvement by all 6 VCS players in build-up to Gateway Plugtests through 3 teleconferences organised by the ETSI Plugtests™ team and Plugtests email reflector.

Feedback was also obtained from a number of VCS suppliers who reviewed the Gateway specifications in detail and pointed out errors/ambiguities in time such that they could be corrected prior to the actual event.

Due to limited time being available in a 4 hour test pairing period between two VCS suppliers it was agreed to perform only a selection of tests only extracted from the EUROCONTROL gateway specifications. These selected tests had to be the most important ones however and prove interoperability of signalling protocols. The test schedule was still considered very tough due to quantity of tests to be performed in a 4 hour test session however, but all participants worked very hard to make it work. An extra 2 hours was made available in the evenings to recuperate sessions that ran out-of-time All VCS suppliers had sessions with all other attending VCS suppliers.

Each of the EUROCONTROL Gateway test specifications defined the step-by-step test procedure definition to be executed by the VCS suppliers, but Message Sequence Charts for each test scenario were later added were added in order the aid understanding of message flows during the call scenario.

Each VCS supplier attending the event brought their Gateway as the Implementation Under Test and also the equipment for their SIP end. During one 4 hour test session each company took the role of Gateway and SIP end. During the course of the day, two 4 hour test sessions were completed by each company.

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.

Huge progress was made by all VCS suppliers during event as the tests were performed and a great group spirit was observed during the week. Some time was lost during the 1st Day due to interfacing problems, but by the 2nd day these problems had been resolved with a 4 hour session allowing sufficient time to execute the tests. By the 3rd day all companies were able to complete the nominated tests well within the 4 hour session.

Points were noted during the test case execution, feedback was obtained from each of the companies as the tests proceeded and this will now be feedback into the ED137 Telephone base specification A list of recommendations to be considered for the enhancement of the ED137 Part 2 Telephone document have also been produced. These recommendations will be examined by the EUROCAE WG67 Sub-groups with the scope of enhancing future editions of the document.

4.2 Test equipment supplied for the event

ATS-R2 Emulators were supplied to the event by INDRA, FREQUENTIS, NUCLEO and EUROCONTROL, while ATS-QSIG Emulators were supplied to event by INDRA, FREQUENTIS and JSP-Teleconsultancy. An ATS-QSIG conformance tester/emulator was also supplied to event by JSP-Teleconsultancy and was used by the participants for tests involving QSIG cause responses.

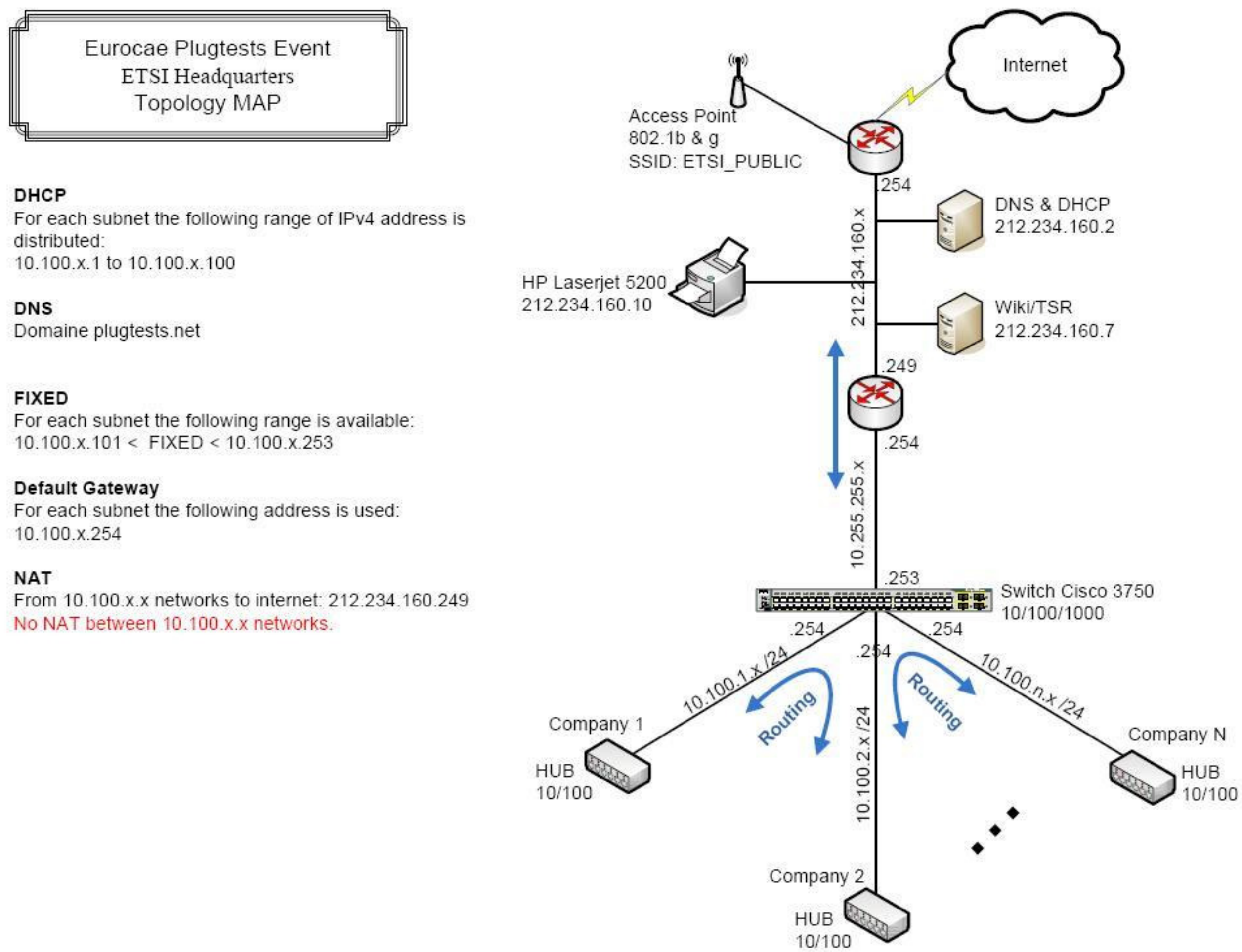
Participants seemed to be happy with the test facilities available, test network and test tools (wiresharks) made available by ETSI.

It was observed that Interoperability with Emulators well achieved by all participants, although it was noted that the EUROCONTROL ATS-R2 emulator had fault with its microphone. Interoperability between Gateway and VCS end well achieved by all participants

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 EUROCAE Gateway Plugtests event network topology



4.5 EUROCAE Gateway Plugtests Pre-conditions

The following pre-conditions were set with respect to the Gateway Plugtests™ event:

1. **Equipment to be brought to event for ATS-R2 v SIP Gateway interworking**
   Each company performing SIP v ATS-R2 gateway tests should bring with them the following equipment:

   - 1 Voice Communication System containing 3 SIP User Agent Interfaces, 3 ATS-R2 interfaces + 3 Controller Working Positions
   - 1 Gateway containing 3 SIP User Agent interfaces, 3 ATS-R2 interfaces

2. **Equipment to be brought to event FOR ATS-QSIG v SIP Gateway interworking**
   Each company performing SIP v ATS-QSIG gateway tests should bring with them the following equipment:

   - 1 Voice Communication System containing 3 SIP User Agent Interfaces + 3 Controller Working Positions

- 1 Gateway containing 3 SIP User Agent interfaces, 1 ATS-QSIG interface

3. **Test Architecture**
Refer to Figure 1 in each of the Gateway Interworking Test Specifications for diagrams relating to the test architecture.

4. **Wiresharks**
ETSI to provide 1 PC with wireshark application per company. This will allow access to the Wiki and permit recorded test files to be uploaded to the Wiki if desired by both companies involved in the test session.

5. **ATS-R2 Emulators**
During a SIPvATS-R2 test session each company needs access to an MFC ATS-R2 Emulator.

If necessary a company should be prepared to lend an emulator to another company for a Test Session.

6. **ATS-QSIG PUMA Emulators**
During a SIPvATS-QSIG test session each company needs access to an ATS-QSIG PUMA 4600E Emulator.

If necessary a company should be prepared to lend an emulator to another company for a Test Session.

7. **Emulator to Gateway interworking**
It is expected that each company will ensure the interworking of their Gateway with the Emulator (ATS-R2 and/or ATS-QSIG PUMA 4600E) prior to attending the Gateway Plugtests™ event. Any interworking layer 1, layer 2 datalink or layer 3 network layer problems between the Gateway and the Emulator should have been resolved prior to the event in order that time is not lost during a test session.

8. **Gateway Interworking test specifications**
The Gateway Plugtests™ specifications show all mandatory tests in Black and Optional Tests in red. Optional tests in red can be performed if both companies agree and there is sufficient time during the session.

9. **Number of mandatory SIP v ATS-R2 tests to be performed in 4 hour session**

Direction Company A (VCS) to Company B (Gateway/Emulator)	
Incoming Gateway R2 to SIP tests:	13
Outgoing Gateway SIP to R2 tests:	10
SUB-TOTAL:	23

Direction Company B (VCS) to Company A (Gateway/Emulator)	
Incoming Gateway R2 to SIP tests:	13
Outgoing Gateway SIP to R2 tests:	10
SUB-TOTAL:	23

TOTAL NUMBER OF TESTS IN 1 FOUR HOUR SESSION : 46

10. **Number of mandatory SIP v ATS-QSIG tests to be performed in 4 hour session**

Direction Company A (VCS) to Company B (Gateway/Emulator)	
Incoming Gateway QSIG to SIP tests:	12
Outgoing Gateway SIP to QSIG tests:	11
SUB-TOTAL:	23

Direction Company B (VCS) to Company A (Gateway/Emulator)	
Incoming Gateway QSIG to SIP tests:	12
Outgoing Gateway SIP to QSIG tests:	11
SUB-TOTAL:	23

TOTAL NUMBER OF TESTS IN 1 FOUR HOUR SESSION : 46

11. **SIP Proxy Servers not to be used during the event**
It has been agreed that testing on the SIP side shall be performed directly between SIP User Agents and that SIP Proxy servers shall not be used during the Gateway test event.

12. **Quantity of SIP User Agents required during the event**
It was also recommended that each participating VCS company should bring 3 SIP User Agents in order to be able to make up to three contemporary DA calls to the 3 SIP User Agents at the gateway of another participating VCS company

13. **Technical Experts on hand**
It is recommended that each participating VCS company has technical experts available on site in order to make on-the-spot changes to their SIP User Agent and Gateway configurations.

14. **Only IPv4**
For the Gateway Plugtests Event it was agreed that only IPv4 shall be used.

15. **Voice over RTP/UDP/IPv4**
On the IP side, voice shall therefore be sent over RTP/UDP/IPv4.

16. **SIP over UDP**
On the IP side, SIP Signalling shall be sent over UDP.

17. **RTP Port numbers**
RTP data is carried on an even UDP port number. The source port numbers to be used by a VCS Suppliers can be decided independently and during a test session each indicates (i.e. through SDP) where they want to receive RTP packets. It is recommended that UDP port numbers (even numbers) in the range 5004 (default) to 5020 are selected during the Gateway Plugtests™.

18. **Voice Codec for SIP v ATS-R2**
For SIP v ATS-R2 interworking only the ITU-T G.711 PCM codec will be used end-to-end.

19. **Voice Codec for SIP v ATS-QSIG**
For SIP v ATS-QSIG interworking only the ITU-T G.729 LD-CELP codec will be used end-to-end.

20. **No SDP Pre-conditions**
There are No SDP preconditions for any of the tests.

21. **Voice Packet size**
During Gateway Plugtests™ it is recommended that the Voice packet duration is set to default 20ms packet size. It is therefore NOT necessary to define the SDP attribute a=interval:20 for any tests. This shall be fixed as 20ms packets.

22. **No IPSec and TLS**
IPSec and TLS shall not be used during either Gateway Plugtests™ event.

23. **No Header Compression**
Header Compression shall not be used shall during the Gateway Plugtests™ event.

24. **No SIP User Agent Authentication**
SIP User Agent authentication with the SIP Proxy Server shall not be implemented. As defined in point 11, Proxy Servers shall not be used.

25. **No DNS Sever**
No DNS will be used during the Gateway Plugtests™ event, which implies that domain names will not be used, instead static IPv4 addresses will be used in place of domain names. Please refer to IP Address Plan version 4, on the Wiki defined in an EXCEL worksheet. This has now been agreed to by each company and should be configured in the systems of each company prior to the event.

26. **No Performance Measurements**
Due to Interoperability of SIP interfaces over a LAN/WAN and Gateway Interworking being the principle scope of these tests, it is also recommended that Call Signalling Performance, Voice delay performance & Voice quality related tests are not performed during the Gateway Plugtests™ Event. Hence it is recommended that no timing measurements will be implemented and no Voice Quality measurement tests will be performed.

27. **No RTCP**
The Real Time Control Protocol (RTCP) shall not be used during the Gateway Plugtests™ Event.

28. **No DHCP**
The Dynamic Host Configuration Protocol (DHCP) shall not be used during the Gateway Plugtests™ Event.

29. **Octet synchronization integrity within the ATS-QSIG 64kbps channel**
The 64kbps connection between the PUMA and Gateway shall have the capability of transporting the octet timing signal end-to-end. Note these are Bipolar octet violations every 8 bits. These octet violations allow identification of the first bit in an octet of 8 bits. The connection should be compliant with EN 300 288/289.

30. **Gateway capability to generate/receive octet timing signal to/from PUMA**
The Gateway should have the capability of generating an octet timing signal towards the PUMA. The Gateway should have the capability of receiving an octet timing signal from the PUMA. It can be assumed that the PUMA will always generate an Octet Timing signal towards the Gateway.

31. **Gateway unable to generate octet timing signal towards PUMA**
In the case when a Gateway is unable to generate an octet timing signal towards the PUMA, it is possible to configure the PUMA to synchronize using a search algorithm for the HDLC opening and closing flags of a frame. Once the flags have been identified, then it is possible to locate the position of the D-signalling sub-channel and each of the 3 voice sub-channels within the octet.

32. **G.728 LD-CELP encoders and decoders**
It is assumed that each Gateway and ATS-QSIG interface containing 3 LD-CELP encoders and 3 LD-CELP decoders (i.e. 1 for each 16kbps voice channel).

33. **In-band synchronization within each 16kbps voice sub-channel**
The G.728 LD-CELP coding algorithm as defined by ECMA 312 ed.3 (June 2003) shall be used. This ensures in-band synchronization of each 16kbps voice sub-channel between an LD-CELP encoder and LD-CELP decoder by creating an inband synchronization channel. By Bit-robbing the MSB of each 10 bit codeword every 16th 10 bit codeword, an inband synchronization channel is created. This implies that 1 bit every 160 bits is used as the synchronization bit instead of the MSB of the codeword. The synchronization pattern is always a 101010……pattern and has a rate of 100bits/sec.

34. **Voice channel from PUMA to Gateway will always have inband synchronization bit inserted.**
When a Gateway outputs the G.728 coded RTP packets for the G.728 stream read from one of the PUMA's three 16kbps voice sub-channels, the synchronization bit shall always be present. The gateway SHALL NOT remove synchronization bits prior to insertion of the G.728 10 bit codevectors in the RTP stream towards the SIP end.

35. **Voice channel from SIP end to Gateway will most probably not have inband synchronization bit inserted.**
When a Gateway receives the G.728 coded RTP packets read from the SIP end the synchronization bit is most probably not present in the 10 bit codevectors. It is therefore necessary for the Gateway to insert the synchronization bit within the 10 bit codevectors being sent in each of the G.728 voice sub-channels.

IMPORTANT

Each VCS company must declare if their RTP voice stream containing the 10 bit codevectors being sent towards the Gateway will contain Inband Voice Sync bits or not.

36. Insertion of Synchronization bit by gateway.

The Gateway shall perform the action of synchronization bit insertion by identifying the MSB of a 10 bit codeword coming from the RTP stream and replacing this by a synchronization bit. It shall then count sixteen 10bit codevectors before repeating the same action. One bit each 160 bits is then used as the inband synchronization channel.

37. Gateway enabled/disabled from insertion of Synchronization bit

It should be possible to enable/disable a Gateway for synchronization bit insertion.

38. SIP end does not include Synchronization bit within MSB codevectors.

It is assumed that the SIP end supplied by each VCS supplier will not generate the G.728 synchronization bits within the RTP stream transporting the 10 bit codevectors. This implies that the Gateway itself will have the task of performing the action of inserting the synchronization bits towards the PUMA.

39. Gateway Network/User configuration

It shall be possible to configure that Gateway for Network Side or User side for each ATS-QSIG link with the PUMA. If the Gateway is configured as Network side the PUMA is configured as User side and vice versa.

40. Gateway A/B configuration for ATS-QSIG interface

It shall be possible to configure that Gateway for A Side or B side for each ATS-QSIG link with the PUMA. If the Gateway is configured as A side the PUMA is configured as B side and vice versa. The Gateway when configured as an A-side shall select the 3-voice channels in an ascending order relative to their availability. When configured as a B-side, the Gateway shall select the 3-voice channels in a descending order relative to their availability.

41. Gateway A/B configuration for ATS-R2 interface

It shall be possible to configure that Gateway for A Side or B side for each ATS-R2 circuit with the ATS-R2 emulator. If the Gateway is configured as A side the ATS-R2 emulator is configured as B side and vice versa. The Gateway when configured as an A-side shall select the voice circuits in an ascending order relative to their availability. When configured as a B-side, the Gateway shall select voice circuits in a descending order relative to their availability.

42. Gateway Routine/Priority for ATS-QSIG interface

The Gateway shall be capable of initiating ATS-QSIG routine and priority calls towards the PUMA. The distinction between routine and priority calls is defined by ECMA 312 ed.3 June 2003.

The Gateway shall be capable of receiving routine and priority calls towards the PUMA. The distinction between routine and priority calls is defined by ECMA 312 ed.3 June 2003.

The ATS-QSIG routine call shall always containing the following information elements:

Sending Complete
Bearer Capability
Channel Identification
Call Priority Interruption Protection with Protection =3/2/1/0
Calling Party Address
Called Party Address
Shift Codeset 4
Transit counter

The ATS-QSIG priority call shall always containing the following information elements:

Sending Complete
Bearer Capability
Channel Identification
Call Intrusion Capability Level (CICL=3)
Call Priority Interrupt Capability Level (CPICL=3)
Call Priority Interruption Protection (CPIPL =3)
Calling Party Address
Called Party Address
Shift Codeset 4
Transit counter

4.6 Interoperability Test Sessions

The objective of each test session was to execute the group of pre-agreed selected tests that had been extracted from the EUROCONTROL Gateway Test specification. 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.

4.6.1 SIP v ATS-R2 Gateway Interoperability Plugtests Test Configuration

The test architecture employed for the Gateway interworking tests is depicted in the Figure below. This shows the Gateway VCS A as the Implementation Under Test (IUT) positioned in sub-network 1. The Gateway contains three SIP Telephone User Agents (for connection via an Ethernet LAN to three SIP user agents at the SIP end VCS B) and one ATS-R2 analogue line interface for connection via the circuit switched part of the network to an ATS-R2 emulator or an ATS-R2 end VCS.

The SIP End VCS B is positioned in sub-network 2. This contains three SIP Telephone User Agents (using a Single or Multiple IP address scheme) for connection via an Ethernet LAN to 3 SIP user agents within the Gateway. The SIP Telephone User Agents can either be integral to VCS B or treated 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 Sub-networks 1 and 2 simulating a LAN network communicate via an IPv4 router.

When performing the Incoming and Outgoing Direct Access test case scenarios defined in this test specification only one ATS-R2 emulator is required at the circuit switched side to perform the tests. The ATS-R2 emulator requires only one 4 wire analogue circuit to connect the ATS-R2 emulator to the gateway. This circuit will be used to transport both analogue signalling and voice.

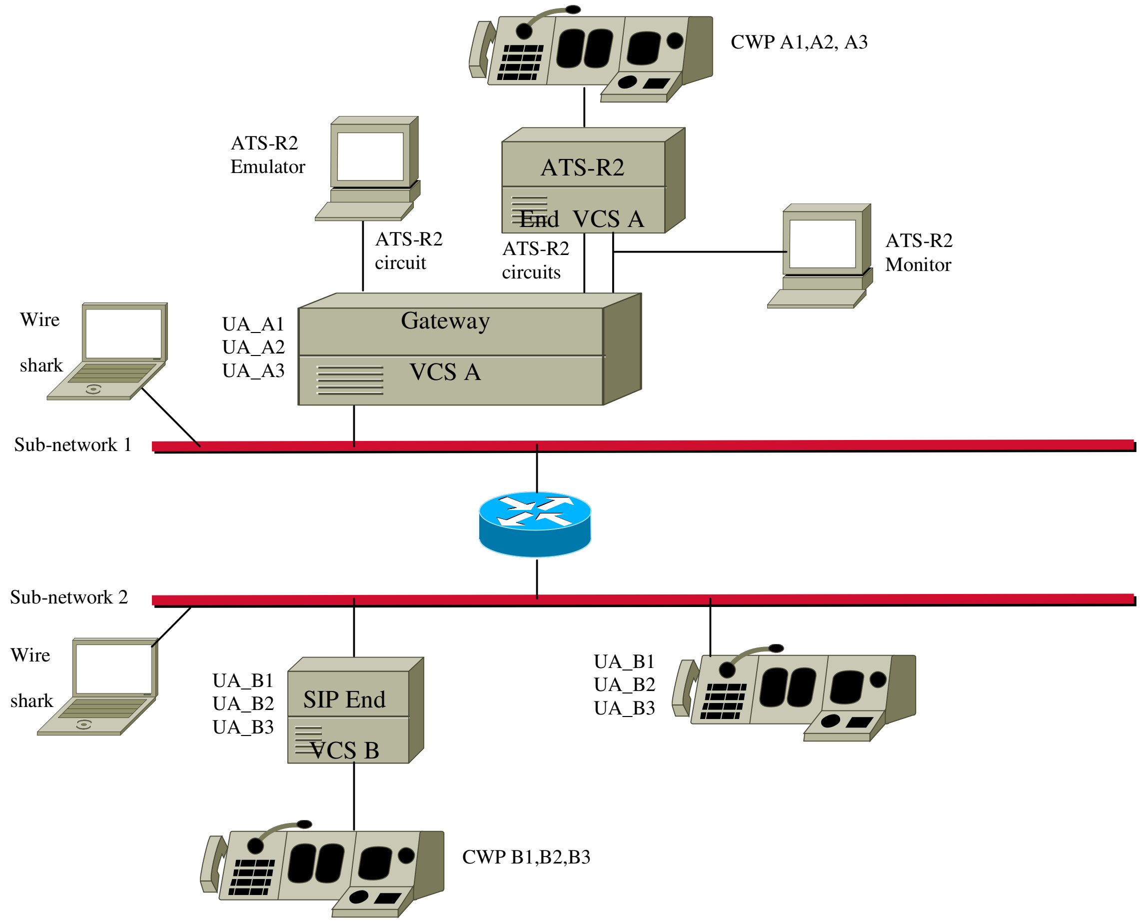
The use of an ATS-R2 Emulator is preferred for the simple Direct Access test case scenarios because it is possible to configure A/B side, Calling and Called Party address and Priority Digit for outgoing ATS-R2 calls and to configure the ATS-R2 status signal response (i.e. Free, Busy, Out-of-service or Congestion) for incoming calls. The emulator is able to send the Release and Blocking line signals on demand. It also allows signal triggers to be set that allow the capture of signalling exchanges from when the trigger event occurs.

Note 1: *It would be possible to replace the ATS-R2 emulator with a VCS containing at least two ATS-R2 interfaces if desired, providing the supplier has the capability of configuring the VCS parameters in real time for the individual test cases.*

The ATS-R2 emulator does not have call interruption or call intrusion functionality. Therefore when performing the Call Interruption and Call Intrusion test case scenarios defined in the test specification, it was sometimes considered more appropriate to use a VCS with at least two ATS-R2 interfaces at the circuit switched side to perform these tests. Each ATS-R2 interface within the VCS requires one 4 wire analogue circuit to connect the gateway. Each circuit will be used to transport both analogue signalling and voice.

It is also possible to configure the ATS-R2 test equipment for a high impedance Monitor mode, allowing the ATS-R2 signalling exchanges to be captured over a single 4-wire analogue circuit between Gateway and ATS-R2 VCS.

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



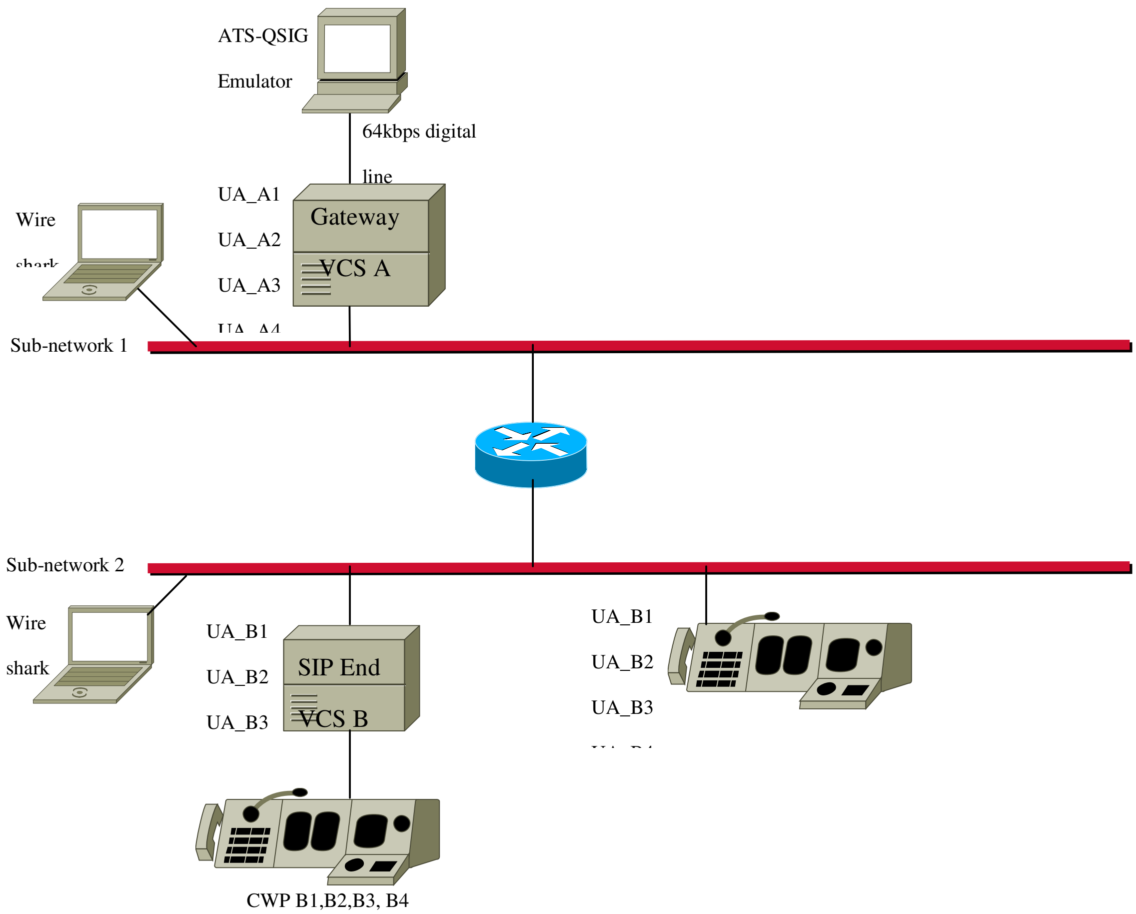
4.6.2 SIP v ATS-QSIG Gateway Interoperability Plugtests Test Configuration

The test architecture employed for the Gateway interworking tests is defined by the Figure below. This shows the Gateway VCS A as the Implementation Under Test (IUT) positioned in sub-network 1. The Gateway contains four SIP Telephone User Agents (for connection via an Ethernet LAN to 4 SIP user agents within SIP end VCS B) and one ATS-QSIG digital line interface for connection via the circuit switched part of the network to an ATS-QSIG emulator.

The SIP End VCS B is positioned in sub-network 2. This contains four SIP Telephone User Agents (using a Single or Multiple IP address scheme) for connection via an Ethernet LAN to 4 SIP user agents within the Gateway. The SIP Telephone User Agents can either be integral to VCS B or treated 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 Sub-networks 1 and 2 simulating a LAN network communicate via an IPv4 router.

When performing the tests only one ATS-QSIG emulator (PUMA 4600E) was required at the circuit switched side to perform all the test case scenarios defined in this specification. Only one 64kbps co-directional digital unrestricted line was required to connect the ATS-QSIG emulator to the gateway. It should be noted that one 64kbps digital unrestricted line with octet integrity (D64U) as required by ATS-QSIG is able to transport 4 x 16 kbps sub-channels as defined by the ECMA 253 standard (one sub-channel used for signalling and 3 used for ITU-T G.728 LD-CELP coded voice).



4.7 Scope and Structure of SIP v ATS-R2/ATS-QSIG interworking tests

This section defines the series of SIP v ATS-R2 and SIP v ATS-QSIG signalling interworking tests performed between SIP and ATS-R2/ATS-QSIG end systems interconnected via a gateway implementation under test.

These tests had the scope of verifying that suppliers had developed their SIP v ATS-R2/ATS-QSIG gateway product as specified in Chapters 4 and 6 of EUROCAE ED137 Part 2.

In one grouping of two suppliers the tests were performed twice in order that each supplier had their system configured with the role of Gateway and with the role of SIP end system.

The interoperability tests to be performed between the gateway SIP User Agents positioned in Sub-network 1 and the SIP end User Agents positioned in Sub-network 2 were divided into Incoming and Outgoing Gateway tests phases.

The incoming gateway tests defined all interworking tests from the ATS-R2/ATS-QSIG circuit switched environment towards the packet switched environment through the gateway.

The outgoing gateway tests defined all interworking tests from the packet switched environment through the gateway towards the ATS-R2/ATS-QSIG circuit switched environment.

Both the incoming and outgoing gateway tests were further sub-divided into Direct Access basic call tests, Call Priority Interrupt Supplementary Service Tests and Call Intrusion Supplementary Service Tests.

As it was not always possible to use the ATS-R2 emulator when performing the supplementary service test case scenarios as some types of this test equipment were incapable of performing call interruption or call intrusion tests. In this case it was necessary that ATS-R2 interfaces within the VCS were used instead.

Participants had to complete all tests during the allocated time-slot for a pairing of two companies. It should be noted that during the pairing of two companies, sufficient time was allocated in order that each company assumed the role of Gateway while the other company assumed the role of SIP end VCS.

As ATS-R2 is a non-symmetrical protocol the ATS-R2 protocol behaviour is different for "A" side and "B" side systems. A mix of tests were therefore performed defining the ATS-R2 end as either an "A" side or a "B" side system.

The test scenarios defined by the EUROCONTROL gateway specifications had been divided into the following groups:

SIP v ATS-R2 Incoming Gateway interworking tests (28 tests)

- ATS-R2 to SIP Direct Access call test scenarios (18 tests) - To be repeated twice with the ATS-R2 emulator configured as A side and B side
- ATS-R2 to SIP Direct Access call interruption Supplementary Service test scenarios (3 tests)
- ATS-R2 to SIP Direct Access call intrusion Supplementary Service test scenarios (7 tests)

SIP v ATS-R2 Outgoing Gateway interworking tests (21 tests)

- SIP to ATS-R2 Direct Access call test scenarios (14 tests) – To be repeated twice with the ATS-R2 configured as "A" side and "B" side
- SIP to ATS-R2 Direct Access call interruption Supplementary Service test scenarios (3 tests)
- SIP to ATS-R2 Direct Access call intrusion Supplementary Service test scenarios (4 tests)

SIP v ATS-QSIG Incoming Gateway interworking tests (29 tests)

- ATS-QSIG to SIP Direct Access call test scenarios (20 tests)
- ATS-QSIG to SIP Call interruption Supplementary Service test scenarios (2 tests)
- ATS-QSIG to SIP Call intrusion Supplementary Service test scenarios (7 tests)

SIP v ATS-QSIG Outgoing Gateway interworking tests (26 tests)

- SIP to ATS-QSIG Direct Access call test scenarios (20 tests)
- SIP to ATS-QSIG Call interruption Supplementary Service test scenarios (2 tests)
- SIP to ATS-QSIG Call intrusion Supplementary Service test scenarios (4 tests)

4.7.1 Test case selection procedure

Prior to the event it was recognised that time constraints would make it difficult to execute all tests defined in EUROCONTROL Gateway test specification in one 4 hour session. Suppliers therefore agreed on selection of most important test cases for execution during a single timeslot involving a pairing of two companies.

The following tests were therefore selected from the specifications:

- R2-SIP Incoming Gateway: 13 from 27 test cases selected (48%)
- SIP-R2 Outgoing Gateway: 10 from 19 test cases selected (52%)
- QSIG-SIP Incoming Gateway: 12 from 28 test cases selected (42%)
- SIP-QSIG Outgoing Gateway: 11 from 26 test cases selected(42%)

This would provide an overall coverage of 46 from 100 test cases to be run (46% coverage)

It was agreed that unselected tests could be performed time permitting.

Tests were repeated twice however with each company having role of Gateway (Implementation Under Test) and SIP end.

There were therefore 46 SIP v ATS-R2 or 46 SIP v ATS-QSIG Tests performed in one 4 hour session.

It is foreseen therefore that the timeslot allocated to a pairing of two companies should allow sufficient time in order to perform the tests.

4.7.2 Test execution procedure

Each of the 49 test scenarios has its own unique identifier code.

For each test it is necessary to capture all SIP/SDP, RTP etc exchanges on the IP segment by using a Test tool monitor (i.e. Wireshark application running on a PC) and save the captured traces relative to the test with a filename identical to the test identifier code (i.e. R2-SIP-DA1.txt).

For each test it is also necessary to capture all ATS-R2 signalling exchanges on the ATS-R2 segment by using a ATS-R2 emulator or ATS-R2 monitor and save the captured traces relative to the test with a filename identical to the test identifier code (i.e. R2-SIP-DA1.txt);

For each of the test scenarios defined in this test specification it is necessary to perform the following checks:

Step	Test description	Verdict	
		Pass	Result
1.	*Check :Are SIP v ATS-R2 signalling interworking compliant with the figures defined in Chapter 4 of EUROCAE ED137 Part 2?*	Yes	
2.	*Check :Are method and response exchanges to and from the gateway compliant with RFC 3261 ?*	Yes	
3.	*Check :Are SIP header fields to and from the gateway compliant with RFC 3261 & EUROCAE ED137 Part 2 ?*	Yes	
4.	*Check :Are SDP message bodies to and from the gateway compliant with RFC 2327, RFC 3264 & EUROCAE ED137 Part 2?*	Yes	

4.7.3 SIP v ATS-R2 Incoming gateway tests

IDENTIFIER	DESCRIPTION
R2-SIP-DA1	Priority call establishment of ATS-R2 call with Priority digit 1 from ATS-R2 emulator to SIP end and ATS-R2 emulator clears call. ATS-Emulator (A-side), Gateway (B-side) (EUROCAE 137 Part 2 - Fig.32 and Fig.34).
R2-SIP-DA2	Priority call establishment of ATS-R2 call with Priority digit 6 from ATS-R2 emulator to SIP end and SIP end clears call. ATS-Emulator (A-side), Gateway (B-side) (EUROCAE 137 Part 2 - Fig.32 and Fig.35)
R2-SIP-DA3	Tactical routine call establishment of ATS-R2 call with Priority digit 2 from ATS-R2 emulator to SIP end and ATS-R2 emulator clears call. ATS-Emulator (B-side), Gateway (A-side) (EUROCAE 137 Part 2 – Fig. 32 and Fig.34).
R2-SIP-DA4	Tactical routine call establishment of ATS-R2 call with Priority digit 7 from ATS-R2 emulator to SIP end and SIP end clears call. ATS-Emulator (B-side), Gateway (A-side) (EUROCAE 137 Part 2 - Fig.32 and Fig.35)
R2-SIP-DA9	Routine call attempt from ATS-R2 emulator to SIP end that results in “Terminal Busy” ATS-R2 status signal being returned by Gateway – Test could be repeated for 4 different SIP response codes. Test performed once with 1 of 4 SIP response codes only (EUROCAE 137 Part 2 - Fig.37)
R2-SIP-DA11	Routine call attempt from ATS-R2 emulator sending invalid calling party address to SIP end. Verify that call attempt is not rejected by gateway.
R2-SIP-DA13	Routine call attempt from ATS-R2 emulator to SIP end that results in “Terminal out of Service” ATS-R2 status signal being returned by Gateway. – Test could be repeated for 11 different SIP response codes.

	(EUROCAE 137 Part 2 - Fig.37)
R2-SIP-DA14	Routine call attempt from ATS-R2 emulator to SIP end that results in "Trunk congestion" ATS-R2 status signal being returned by Gateway. (EUROCAE 137 Part 2 - Fig.37)
R2-SIP-DA15	Routine call attempt from ATS-R2 emulator to SIP end that results in ATS-R2 "RELEASE" line signal being returned by Gateway. – Test could be repeated for 23 different SIP response codes. Test performed once with 1 of 23 SIP response codes only (EUROCAE 137 Part 2 - Fig.38)
R2-SIP-DA18	Routine call attempt from ATS-R2 emulator to SIP end that results in Gateway's internal "Outgoing call answer" timer expiry at the gateway leading to call clearing from the gateway.
R2-SIP-CPI1	**ATS-R2 to SIP Priority Call Interruption** Interruption of established routine (non-priority) call between ATS-R2 end and SIP end by a priority call (priority 1 digit) made from ATS-R2 end towards the SIP end. To verify that both the ATS-R2 and SIP segments of the routine call are interrupted before priority call is executed. ATS-R2 Emulator (A-side), Gateway (B-side) (EUROCAE 137 Part 2– Fig.43)
R2-SIP-CI1a	**ATS-R2 to SIP Priority call answered automatically following release of previous call before T1 expiry.** (EUROCAE 137 Part 2– Fig.49) Optional if R2-SIP-CI1b is selected
R2-SIP-CI1b	**ATS-R2 to SIP Priority call still presented as priority call following release of previous call before T1 expiry.** (EUROCAE 137 Part 2– Fig.49) Optional if R2-SIP-CI1a is selected
R2-SIP-CI2	**ATS-R2 to SIP Successful Priority Call Intrusion and clearing by Intruding user A2** (EUROCAE 137 Part 2– Fig.50 and Fig.55)

4.7.4 SIP v ATS-R2 Outgoing gateway tests

IDENTIFIER	DESCRIPTION
SIP-R2-DA1	Priority call establishment from SIP end to ATS-R2 emulator and ATS-R2 emulator clears call ATS-R2 emulator (A-side), Gateway (B-side) (EUROCAE 137 Part 2– Fig.33 and Fig.34)
SIP-R2-DA2	Tactical routine call establishment from SIP end to ATS-R2 emulator and SIP end clears call ATS-R2 emulator (B-side), Gateway (A-side) (EUROCAE 137 Part 2– Fig.33 and Fig.35)
SIP-R2-DA5	Routine call attempt from SIP end to ATS-R2 emulator that results in "Terminal Busy" ATS-R2 status signal being returned to Gateway and mapped by Gateway to "Busy Here" SIP response code towards SIP end (EUROCAE 137 Part 2– Fig.40)
SIP-R2-DA8	Unsuccessful Routine call attempt from SIP end to ATS-R2 emulator resulting in call clearing at the gateway (EUROCAE 137 Part 2– Fig.42)
SIP-R2-DA9	Routine call attempt from SIP end to ATS-R2 emulator that results in "Terminal out of Service" ATS-R2 status signal being returned to Gateway and mapped by Gateway to "Not found" SIP response code towards SIP end

	(EUROCAE 137 Part 2– Fig.40)
SIP-R2-DA10	Routine call attempt from SIP end to ATS-R2 emulator that results in “Trunk congestion” ATS-R2 status signal being returned to Gateway and mapped by Gateway to the “Service Unavailable” SIP response code towards SIP end (EUROCAE 137 Part 2– Fig.40)
SIP-R2-DA11	Routine call attempt from SIP end to ATS-R2 emulator that results in ATS-R2 “RELEASE” line signal being returned to Gateway and mapped by Gateway to the “Server Internal Error” SIP response code towards the SIP end. (EUROCAE 137 Part 2– Fig.41
SIP-R2-DA12	Routine call from SIP end to ATS-R2 emulator and SIP end clears call while Gateway is dialling number towards ATS-R2 emulator. (EUROCAE 137 Part 2– Fig.39)
SIP-R2-CPI1	**SIP to ATS-R2 Priority Call Interruption** Interruption of established routine (non-emergency) call between SIP end and ATS-R2 end by a SIP call (SIP Priority header-emergency) made from SIP end towards the ATS-R2 end. To verify that both the ATS-R2 and SIP segments of the routine call are interrupted before priority call is executed. ATS-R2 Emulator (B-side), Gateway (A-side) (EUROCAE ED-137 Part 2 - Fig.46)
SIP-R2-CI1	**SIP to ATS-R2 Successful Priority Call Intrusion and clearing by Intruding user B2** (EUROCAE ED-137 Part 2 - Fig.57 and Fig.62)

4.7.5 SIP v ATS-QSIG Incoming gateway tests

QSIG-SIP-DA1	Priority call establishment of ATS-QSIG call with CPIPL=3, CPICL=3, CICL=3 from ATS-QSIG emulator to SIP end and ATS-QSIG emulator clears call. ATS-QSIG Emulator (A-side), Gateway (B-side) ATS-QSIG Emulator (Network), Gateway (User) Refer to EUROCAE ED137 Part 2 – Fig.108 and Fig.110
QSIG-SIP-DA4	General purpose routine call establishment of ATS-QSIG call with CPIPL=0 from ATS-QSIG emulator to SIP end and SIP end clears call. ATS-QSIG Emulator (B-side), Gateway (A-side) ATS-QSIG Emulator (User), Gateway (Network) Refer to EUROCAE ED137 Part 2 – Fig.108 and Fig.111.
QSIG-SIP-DA9	Routine call attempt from ATS-QSIG emulator to a busy user at SIP end resulting in SIP response “486 Busy Here” mapped by gateway to ATS-QSIG Cause value 17 “User Busy” towards emulator. Refer to EUROCAE ED137 Part 2 – Fig.108 and Fig.11
QSIG-SIP-DA11	Routine call attempt from ATS-QSIG emulator to an out-of-service or unconfigured address at SIP end resulting in SIP response “404 Not Found” mapped by gateway to ATS-QSIG Cause value 1 “Unallocated (Unassigned) number” towards emulator.
QSIG-SIP-DA13	Routine call attempt from ATS-QSIG emulator sending invalid calling party address to SIP end. Verify that call attempt is not rejected by gateway. Verify that the gateway includes a SIP URI identifying the gateway in the SIP From header.
QSIG-SIP-DA14	Routine call attempt from ATS-QSIG emulator to SIP end with Transit Counter value set to 0. Verify that SIP Max-forwards Header from the gateway is 19
QSIG-SIP-DA18	Routine call attempt from ATS-QSIG emulator to SIP end with Transit Counter value set to 4. Verify that SIP Max-forwards Header value from the gateway is 3
QSIG-SIP-CPI1	**ATS-QSIG to SIP Priority Call Interruption** Interruption of established routine (CPIPL=0) call between ATS-QSIG end and SIP end by a priority call (CPIPL=3, CPICL=3 and CICL=3) made from ATS-QSIG end towards the SIP end. To verify that both the ATS-QSIG and SIP segments of the routine call are interrupted before priority call is executed.

	EUROCAE ED137 Part 2 Fig.117.
QSIG-SIP-CI1a	**ATS-QSIG to SIP Priority call answered automatically following release of previous call before T1 expiry.** EUROCAE ED137 Part 2 - Fig.121.
QSIG-SIP-CI1b	**ATS-QSIG to SIP Priority call still presented as priority call following release of previous call before T1 expiry.** EUROCAE ED137 Part 2 - Fig.121.
QSIG-SIP-CI2	**Successful ATS-QSIG to SIP Priority Call Intrusion and clearing by Intruding user.** EUROCAE ED137 Part 2 - Fig.122 and Fig. 127.

4.7.6 SIP v ATS-QSIG Outgoing gateway tests

IDENTIFIER	DESCRIPTION
SIP-QSIG-DA1	Priority call establishment of SIP call with SIP priority header “emergency” from SIP end to ATS-QSIG emulator. Verify that ATS-QSIG call from gateway has CPIPL=3, CPICL=3, CICL=3. SIP end clears the call. ATS-QSIG Emulator (A-side), Gateway (B-side) ATS-QSIG Emulator (Network), Gateway (User) EUROCAE ED137 Part 2 Fig.109 and Fig.111.
SIP-QSIG-DA3	Strategic routine call establishment of SIP call with SIP priority header “normal” from SIP end to ATS-QSIG emulator. Verify that ATS-QSIG call from gateway has CPIPL=1. ATS-QSIG emulator clears the call. ATS-QSIG Emulator (B-side), Gateway (A-side) ATS-QSIG Emulator (Network), Gateway (User) EUROCAE ED137 Part 2 Fig.109 and Fig.111.
SIP-QSIG-DA8	Routine call attempt from SIP end to ATS-QSIG emulator that results in Gateway’s QSIG T.301 Timer expiry leading to call clearing from the gateway.
SIP-QSIG-DA9	Routine call attempt from SIP end to ATS-QSIG emulator that results in ATS-QSIG cause value 17 “User Busy” value being sent to the Gateway. Verify that this is mapped by the gateway to a SIP response “486 Busy Here” towards the SIP end. EUROCAE ED137 Part 2 Fig.115
SIP-QSIG-DA10	Routine call attempt from SIP end to ATS-QSIG emulator incurring channel congestion, causes gateway to abandon call and respond with a SIP response “503 Service Unavailable” towards the SIP end.
SIP-QSIG-DA11	Routine call attempt from SIP end to an out-of-service or unconfigured address at ATS-QSIG emulator resulting in ATS-QSIG Cause value 1 “Unallocated (Unassigned) number mapped by gateway to SIP response “404 Not Found” towards SIP end. EUROCAE ED137 Part 2 Fig.115
SIP-QSIG-DA12	Routine call attempt from SIP end to invalid Request URI or SIP To header address towards ATS-QSIG emulator. Verify ATS-QSIG call is not made from Gateway to ATS-QSIG emulator. Verify that call attempt is rejected by the gateway through a SIP 4xx, 5xx or 6xx response towards the SIP end.
SIP-QSIG-DA14	Routine call attempt from SIP end to ATS-QSIG emulator with SIP Max-forwards Header value set to 20. Verify that Transit Counter value from the gateway towards the ATS-QSIG emulator is 0
SIP-QSIG-DA18	Routine call attempt from SIP end to ATS-QSIG emulator with SIP Max-forwards Header value set to 6. Verify that Transit Counter value from the gateway towards the ATS-QSIG emulator is 4
SIP-QSIG-CPI1	**SIP to ATS-QSIG Priority Call Interruption** Interruption of established routine (non-emergency) call between SIP end and ATS-QSIG Emulator by a SIP call (SIP Priority header-emergency) made from SIP end towards the Emulator. To verify that both the ATS-QSIG and SIP segments of the routine call are interrupted before priority call is executed. EUROCAE ED137 Part 2 Fig.119
SIP-QSIG-CI1	Successful SIP to ATS-QSIG Priority Call Intrusion and clearing by Intruding user **EUROCAE ED137 Part 2 Fig.129 and Fig.132**

4.8 Gateway Plugtests™ Result summaries

4.8.1 SIP v ATS-R2 Gateway Plugtests™ Result summaries

This section has the scope of providing a summary of the results recorded for each of the SIP v ATS-R2 PLUGTESTS™ specification used by participants during the event. During the event it was observed that Interoperability was well achieved.

Number of tests per test session: 24

Number of Sessions: 30

Of the 30 reported sessions 30 were agreed (100.0%)

650 Tests performed in total

Interoperability		Not executed		Totals	
OK	**NO**	**NA**	**OT**	**Run**	**Results**
645 (99.2%)	**5 (0.8%)**	**27 (3.8%)**	**43(6.0%)**	**650 (90.3%)**	**720**

4.8.2 SIP v ATS-QSIG Gateway Plugtests™ Result summaries

This section has the scope of providing a summary of the results recorded for each of the SIP v ATS-QSIG PLUGTESTS™ specification used by participants during the event. During the event it was observed that Interoperability was well achieved.

Number of tests per test session: 23

Number of Sessions: 12

Of the 12 reported sessions 12 were agreed (100.0%)

214 Tests performed in total

Interoperability		Not executed		Totals	
OK	**NO**	**NA**	**OT**	**Run**	**Results**
202 (94.4%)	**12 (5.6%)**	**10 (3.6%)**	**52 (18.8%)**	**214 (77.5%)**	**276**

4.9 SIP v ATS-R2 Non-interoperability reasons resolved during Gateway Plugtests™

The following list defines the main reasons for SIP v ATS-R2 non-interoperability that were resolved during the Gateway Plugtests.

- Call Intrusion timer T1 was defined as 10 seconds by ED137 but often the ATS-R2 transit call timer P22 (default 12 seconds) was found timed out prior to timer T1, causing the call to be terminated immediately. The companies changed T1 <10 seconds (i.e. 5 to 8 seconds) (P22 is timeout value between the 13th Acknowledge signal (ATS R2) and the start of a register type status signal for general transit interworking).

- A SIP User Agent as the wanted party of a call intrusion has to act as focus and mix audio to all other parties (i.e. Unwanted User and Intruding user) during the call intrusion. It was observed that in some cases the ATS-R2 2280Hz Release line signal sent by one of the parties towards the gateway in order to leave the call has to be cut-off immediately by Gateway otherwise it is also mixed with audio and sent to other parties. This obviously has the result that all parties are disconnected from the call. Those VCS suppliers experiencing this problem corrected their implementations.

- It was observed that on call intrusion occurring at the SIP User agent of a wanted party, some company's gateways were not sending back the mixed RTP audio stream towards the intruding party. This was corrected by all companies that experienced this problem.

- It was observed that some company's Gateways did not send a Message indicating call interruption warning towards the SIP side when an active ATS-R2/ATS-QSIG v SIP call had been selected for interruption at the ATS-R2/ATS-QSIG circuit switched side. This implied that there as no interrupt warning tone played to the user at the SIP end warning them of an impending interruption of their call-in-progress by a Priority call initiated from the circuit switched side. This was corrected by all companies that experience this problem.

- Some companies Gateways didn't send Release line signal on receiving Trunk Congestion tone from the ATS-R2 end. This was corrected by all companies that experienced this problem.

- Some companies did not always have the mappings between ATS-QSIG cause numbers and SIP response numbers correctly implemented their gateways. These were corrected by all companies that experienced these problems.

4.10 SIP v ATS-QSIG Non-interoperability reasons resolved during Gateway Plugtests™

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

- Sometime there were interoperability problems regarding SIP-ATS-QSIG end-to-end synchronization of audio. The Gateway when receiving incoming RTP stream from the SIP end sometimes had problems to insert the inband synchronization bit within the RTP stream in order that the ATS-QSIG end could synchronize to the inband sync pattern and obtain transparent audio. It was often observed that audio was only obtained in the direction ATS-QSIG end to SIP end and not vice versa. This problem was resolved by the VCS suppliers and each of the VCS suppliers Gateways can now be configured to enable/disable G.728 inband sync bit insertion in all incoming RTP streams from the SIP end that did not contain the sync bit. It was observed that the SIP end did not always send an RTP stream containing inband sync bits especially if the SIP end happened to be a SIP telephone using the G.728 LD-CELP codec.

- All companies agreed to send two 20 byte frames in one 20ms IP packet. Each frame therefore contains 16 x G.728 10 bit Codevectors. The first bit of each of the two frames contained in the 20 ms packet therefore becomes the bit for the synchronization channel.

- The gateway when receiving an RTP stream can therefore easily insert the 1010…inband sync pattern as it knows exactly where the sync bits have to be inserted.

- In the case of 10ms IP Packets it would be possible to insert just one 20byte frame in the packet, while in the case of 30ms IP packets it would be possible to insert three 20byte frames in the packet. It can be assumed that the one sync bit exists every 160 bits received and this bit will always be the first bit in a 20 bytes frame.

4.11 Recommendations for enhancement of ED137 Part 2 Telephone- Chapter 4

4.11.1 Modification of default value of the Call Intrusion Timer T1 to 8 seconds

In case of ATS-R2, allow Call intrusion timer T1 to have a maximum value of 8 seconds instead of 10 as often ATS-R2 P22 timer (12 seconds) expired first.

4.11.2 Modification of SIP protocol for management of Call Interruption at Gateway

When a SIP call with an Emergency Priority header encounters circuit/channel congestion at the Gateway, there is a long period of silence heard by the SIP user, due to the 5 second interrupt warning period on the circuit switched side followed a 5 second Blocking signal of the ATS-R2 protocol (in total a 10 second delay). The SIP caller during this time is not aware that the call is actually proceeding because there is no indication given by the gateway that this is the case.

It is recommended that the ED137 Part 2 Telephone document is modified such that the gateway responds with a 183 Progress response that indicates the reason for the delay. In that way it would be possible for the SIP end to provide an indication to the SIP end user that the call is proceeding.

4.11.3 Modification to ED137 Part 2 Telephone - Tones table –section 3.9

The Table in ED137 Part 2 Telephone - section 3.9 incorrectly shows call intrusion warning tone relating to Call Interruption. It is recommended that this is corrected.

4.11.4 Clarification on Call Intrusion/ Call Interruption tones and messages at the Gateway

It is recommended that the following general statement should be added below tones table in section 3.9:

Tones generated at the analogue side shall be transparently forwarded to the SIP end over the IP network.

Tones generated at the gateway shall be injected into the voice path towards the analogue end and shall be announced via SIP INFO over the IP network towards the SIP end.

Any INFO messages received from the SIP end shall result in audible tones being injected into the voice path towards the analogue end (refer to 3.8.8.1 Pt. 4).

4.11.5 Recommended use of new SIP INFO message indicating "Interruption Cancelled"

A SIP call with "Emergency" header made towards the gateway encountering circuit/channel congestion in the circuit switched part of the network, according to 4.11.2 will receive a 183 Progress response informing them that a Call Interruption is in progress.

If another SIP v ATS-R2 active call in progress is selected by the gateway for interruption then both of these users will hear an Interrupt Warning Tone informing them that their call is about to be interrupted. In the case however that the Call Interruption is abandoned by the gateway due to another circuit/channel becoming available or due to the SIP Emergency Call request being cleared, within the Interrupt Warning Time period, currently the user at the SIP end is not informed of the fact that the call interruption has been abandoned.

It is recommended to introduce an INFO message containing a message that informs the remote SIP user at the SIP end (whose call was about to be interrupted) that the Call Interruption has been abandoned.

4.11.6 Modification to ED137 Paragraph 4.3.1.3. for 182 response to ATS-R2 status 6 translation

It is recommended to modify ED137 part 2 Telephone - paragraph: 4.3.1.3. in order to exclude a 182 response from being translated into a ATS-R2 Status 6 “Terminal Free”. A 182 response shall not cause any ATS-R2 signal to be sent towards the ATS-R2 end.

The proposed text should read as follows:

Receipt of 18x Provisional Response
The Gateway **SHALL** map a received SIP 180 or SIP 181 response to an ATS-R2 status signal no. 6 “Terminal Free” and supply Ringing tone on the inter-VCS link. The gateway shall ignore a received SIP 182 response and shall not generate any ATS-R2 signal towards the ATS-R2 end.

4.11.7 Modification of ED137 Para. 4.3.3.2. Priority Call Intrusion from SIP to ATS-R2

ED137 paragraph 4.6.3.2 Priority Call Intrusion from SIP to ATS-R2 is incorrect. The Text “The Gateway SHALL be configured to operate with T1 = 0 and automatic Priority call answer” should be deleted. The text should therefore read as follows:

On receipt of a SIP INVITE(emergency) request from the IP network, the Gateway **SHALL** attempt to establish a Priority call (Priority digit = 1 or 6) towards the ATS-R2 network. On receipt of an ATS-R2 status signal no. 6 “Terminal Free”, the Gateway **SHALL** send SIP 200 (OK) response towards the IP-network. Other possible ATS-R2 responses **SHALL** be handled in accordance with subclause 4.3.3.1.

Once a priority call is answered at the ATS-R2 end, the Gateway User Agent **SHALL** be ready to receive a dialog subscription from the SIP_End User Agent (the user who requested Call Intrusion); on receiving the dialog subscription, the Gateway **SHALL** send a notification about the intruded party.

4.12 Recommendations for enhancement of ED137 Part 2 Telephone- Chapter 6

4.12.1 Clarification about Insertion of G.728 inband sync bits

It is recommended that the ED137 Part 2 Telephone –Chapter 6 provides clarification about the following:

- Transcoding of voice at a gateway shall not be necessary as the SIP end shall always send G.728 frames (i.e. the G.728 LD-CELP codec is negotiated using SDP during SIP session establishment).
- Definition of a new optional SDP parameter nominated G.728 sync bits: Y/N, allowing Gateway to be informed if it has to insert a sync pattern or not towards the ATS-QSIG end. In this way it would not be necessary for a Gateway to perform a search for sync bits as it will know if they are present or not during the SIP session establishment.
- a Gateway should have capability to insert G.728 inband Sync Bits on RTP streams sent from the SIP end without sync bits included.
- Definition that one 20 byte frame should be sent in a 10ms IP packet.
- Definition that two 20 byte frames should be sent in a 20ms IP packet.
- Definition that three 20 byte frames should be sent in a 30ms IP packet.
- One frame shall contain sixteen G.728 10 bit codevectors. The first bit in each frame shall always be assumed by gateways to be the synchronization bit.
- The gateway shall know where the synchronization bits are located in an incoming RTP stream and shall not have to perform a search for the bits.

It should be noted that the G.728 LD-CELP codec has been designed to provide good quality voice even though one bit every sixteen 10 bit codevectors is being robbed for inband synchronization purposes.

4.13 General Recommendations for enhancement of ED137

4.13.1 Change from BYE message with Text/Plain body to Reason Header

Delete use of BYE message body with Text/Plain body indicating "Emergency - Forced Release as defined by ED137 par.4.6.2.2 according to the definitions listed below:

- Replace with BYE message containing Reason header (as this is standardized method for indicating the reason why the call was not successful or released).
- The Reason header to include text indicating the reason why the call was released (i.e. "Emergency –Forced Release")
- Necessary to include SIP Reason Header in ED137 par 3.4.1 and 3.4.2.
- Table 2 to include a "Reason" line as optional "o" under "BYE and CANCEL".
- Table 3 to include a "Reason" line for "All" status codes as optional "o".

The BYE message in general to always contain a Reason header indicating why the call was released. This would also allow for example an error response to an OPTIONS/INVITE request containing a more detailed description about the current state of a gateway (e.g. in general Gateway operational but currently has no free circuit/channel of a trunk);

The Reason header is defined in RFC3326 and shall have the following syntax:

Reason = "Reason" HCOLON reason-value *(COMMA reason-value)
reason-value = protocol *(SEMI reason-params)
protocol = "SIP" / "Q.850" / token
reason-params = protocol-cause / reason-text / reason-extension
protocol-cause = "cause" EQUAL cause
cause = 1*DIGIT
reason-text = "text" EQUAL quoted-string
reason-extension = generic-param

e.g. for SIP:

Reason: SIP ;cause=200 ;text="Call completed elsewhere"

In order to specify new cause values, it is mandatory to define a new protocol – It is suggested to use "WG-67" with cause values 1xxx for phone and 2xxx for radio.

For instance:

Reason: WG-67 ; cause=1200 ; text="No channel available"

Reason: WG-67 ; cause=2001 ; text="Missing R2S- keepalive"

It is therefore necessary that the ED137 contains a new table that lists all cause values used by WG-67 (Radio and Telephone) because all of them will be optional.

4.13.2 Recommendation that RTP ports used in SIP session between 2 UAs become symmetric

A device supports symmetric RTP if it selects, communicates, and uses port numbers such that, when receiving a bidirectional RTP media stream on UDP port "A", it also transmits RTP media for that stream from the same source UDP port "A". That is, it uses the same UDP port to transmit and receive one RTP stream.

A device that doesn't support symmetric RTP would transmit RTP from a different port, than the port used to receive RTP for that bidirectional media steam.

It should be noted that NATs require that endpoints use symmetric UDP ports to establish bidirectional traffic. Also if firewalls are used, a firewall can expect symmetric RTP ports, then the firewall's dynamic per call port filter list can be more restrictive compared to asymmetric RTP.

It is therefore recommended that the RTP ports used in SIP session between two user agents become symmetric.

The transmit port is already negotiated through SDP during SIP session establishment.

It is proposed that the receive port should have an identical value to the transmit port and that this should apply to both Radio and Telephone sessions.

When updating the ED137 document, reference should be made to RFC4961 which provides a good definition on the use of symmetric ports.

Above proposal should also be evaluated by SESAR 15.2.10 D2 dealing with Security policy and procedures for A/G-G/G communications and applications;

4.14 Recommended updates to EUROCONTROL Gateway Test Specifications

A number of minor errors were identified with both EUROCONTROL gateway test specifications during test case execution.

The following list the errors identified in the test specifications:

- R2-SIP-DA13 Step 4: incorrectly defines SIP cause value 488. This should be deleted.
- For Priority Call Interruption tests, the Test Specification does not show a BYE message containing a Text/Plain body indicating "Emergency - Forced Release" to the SIP End as defined by ED137. (Note: a recommendation exists to replace the Text/Plain body of a BYE message with the Reason header field);
- The tests QSIG-SIP_CI1a, CI1b and CI1c should make it clear that it is necessary to change the called party number at the ATS-QSIG emulator .

It is also proposed to add the following new tests to future editions of the gateway test specifications:

1. An OPTIONS ping test towards the gateway should be included, in order to check that connection with Gateway is present.
2. To be evaluated if the OPTIONS PING test should be extended in order to determine reachability of each user in the Circuit switched network. The gateway would then convert the OPTIONS PING to a specific user into a test call to that user in the circuit switched part of the network;
3. Instead of using the OPTIONS PING test to determine reachability of end users, it may be more appropriate that test calls to users in circuit-switched network are made as normal calls, with gateway performing only signalling interworking function.

4.15 What will happen after this Plugtests event?

- Following feedback during the Gateway Plugtests event to the EUROCONTROL Gateway specifications in additional to feedback received by EUROCONTROL in their request for comments to industry, both EUROCONTROL Gateway specifications shall be updated and new versions produced.
- Following feedback obtained from participating VCS vendors during the Gateway Plugtests event, all points noted shall be proposed as a series of recommendations to be used in any eventual update of the next ED137 Part 2 Telephone specification, following discussion by the EUROCAE WG56 Telephone Subgroup.
- This Final Gateway Event Plugtests™ report containing all feedback accumulated during the event shall be distributed to all participating companies for comments and also placed on the ETSI wiki tool. The report shall also be used as an input for the next EUROCAE WG67 meeting planned for the 13th April in Romatsa, Bucharest, Romania.
- All VCS companies are kindly asked to remember the NDA that they have signed.

4.16 Proposed future ETSI Plugtests events

- In Dec 2009 a “Call for interest” was sent to all Global VCS, GRS and Recorder market players. To date there are already 5 new comers (2 VCS, 2 Radio and 1 Recorder) that have expressed an interest to participate in future Plugtests events;

- It is therefore proposed to hold an event only for new comers that will repeat all Stage 1 and Stage 2 Telephone/Radio Tests. This event is proposed to be held in the Autumn of 2010.

- Following the update to the ED137 Radio and Telephone documents following feedback from the Plugtests, ICAO and the FAA it is then proposed to hold a 4th Plugtests event in the Spring of 2011. This event would execute tests for the following:

    - Limited set of tests to ensure interoperability of new features etc in the updated version of the ED137 telephone and radio specifications;

    - Some tests related to features of the new RTP Header Extension format proposed for the sessions with Radios.

    - Main/Standby Radio switchover tests;

    - Tests to cover the final 20% of ED137 as present untested. These would include Presence, Dialog, Event Packages and EUROCAE optional features.

    - Use of IPv6 only and SIP Proxy Servers with authentication of SIP User Agents.

    - Series of Tests for VCS/Radio interoperability with the Voice Recorder

- The availability of a conformance tester as proposed by EUROCONTROL, with test suites relevant to SIP Telephone User Agents, SIP Radio User agents and SIP Recorder User Agents should be considered as a pre-condition for the organisation of the next Plugtests event.

4.17 Conclusion

The third EUROCAE Plugtests™ Interoperability Event on VoIP for ATM (Air Traffic Management) had the scope of testing the Telephone Gateways developed by six VCS suppliers. The event held at the ETSI headquarters in Sophia Antipolis between 1st March and 5th March 2010 has resulted in the significant quantity of feedback relating to the ED137 Telephone document being accumulated both prior and during the execution of tests. The importance of a Gateway 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 Telephone documents that specify the interworking between Voice Communication Systems (VCS) using SIP and VCSs using legacy ATS-R2 and ATS-QSIG signalling protocols. 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).

A series of the most important tests were extracted from the EUROCONTROL Gateway Test Specifications and performed during 4 hour sessions between all combinations of company pairings. Each VCS supplier took the role of Gateway and the Role of SIP end during a single 4 hour session.

The results of the Gateway interworking test scenarios achieved by the European (6 VCS) vendors have demonstrated a high rate of success:

- SIP v ATS-R2 Interoperability: 99,2% (645 tests OK for 650 tests run)
- SIP v ATS-QSIG Interoperability: 94,4% (202 tests OK for 214 tests run)

These results show that the Gateway products developed by each of the six participating VCS suppliers are ready to be deployed in operational environments in order to assist the transition from legacy ATS-R2 analogue interfaces and ATS-QSIG digital interfaces towards VoIP using SIP interfaces. These Gateway Products will allow the ANPS to migrate towards VoIP while keeping their existing Voice Communication Systems until the end of their lifecycles. These Gateway products now developed by the VCS suppliers have been proven to support the Operational and Technical Requirements defined by the ED 136 document and have achieved a high level of interoperability. This will lead to ATM VoIP VCS Gateway deployment by ANSPs (Air Navigation Service Providers) in the very near future for operational use in the framework of the Single European Sky (SES).

History

Document history		
<Version>	<Date>	<Milestone>