ED 137 version C

VoIP in ATM

Convergence of voice and data into one multimedia network and its increasing reliability and availability all over the planet made the ATM community consider the evolution of their legacy and/or analogue networks towards a common modern infrastructure.

ED137 is the document issued by EUROCAE for the definition of "Interoperability Standards for VoIP ATM Components". The Voice over IP (VoIP) technology proved to be mature to improve operational and technical communication requirements, Quality of Service (QoS) levels, security and safety needs.

EUROCAE Working Group 67 (WG-67) undertook the mission of assessing the feasibility of using Voice over Internet Protocol (VoIP) for providing ATM voice services. WG67 defined criteria, requirements and guidelines for ground-ground ATM communications and the ground-ground segment of air-ground ATM communications. The result is the issue of the following standards:

- **ED-136 - VoIP ATM System Operational and Technical Requirements**
- **ED-137 - Interoperability Standards for VoIP ATM Components**
- **ED-138 - Network Requirements and Performance for VoIP ATM Systems**

The above-mentioned documents use the so-called "**Vienna Agreement**" as a common reference for the definition of the different components of a VoIP ATM system and their mutual interfaces (see scheme here below).

ED137 Anatomy

The ED137 standard is made of different independent sections (named "Volumes"), each dealing with different aspects concerning VoIP in ATM, namely radio, telephone, recording, supervision. ED137 aims at being a common standard covering all communication options and modes used by the ATM community.

The latest version of the ED137 standard (version C) consists of the following 4 parts:

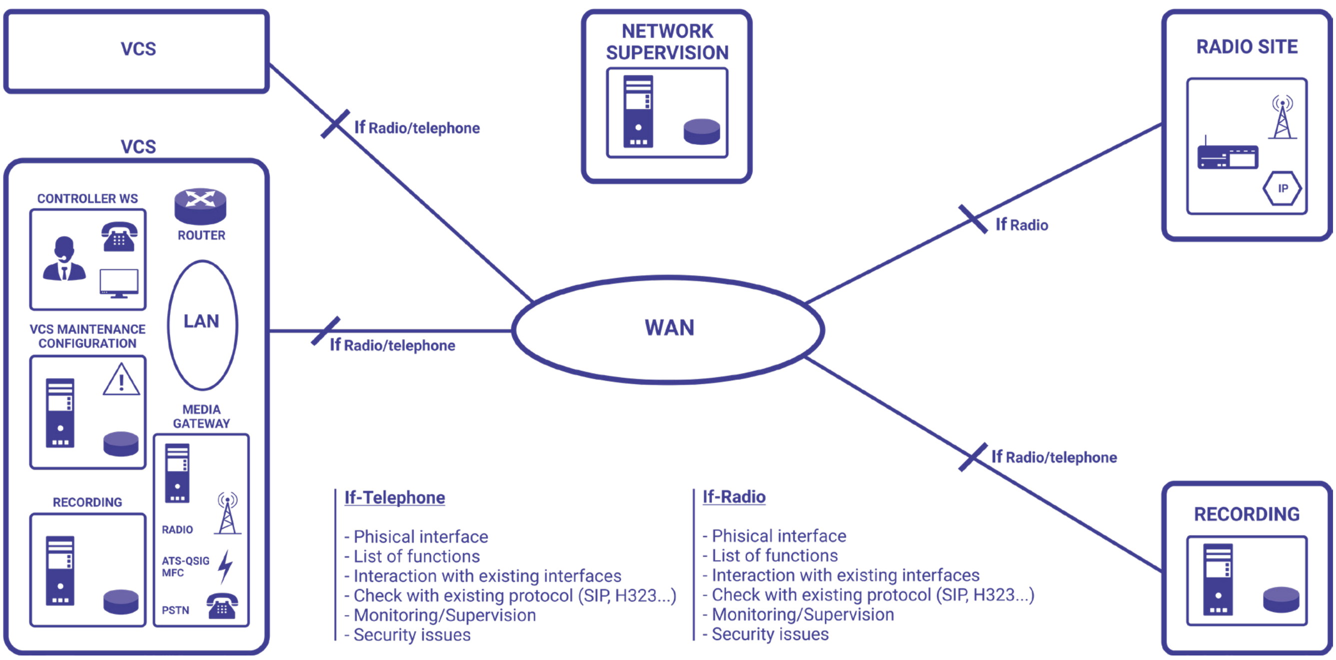
- Volume 1 – Radio
- Volume 2 – Telephone
- Volume 4 – Recording
- Volume 5 – Supervision
  (please note volume 3 does not exist)

The telephone section is made of a core part plus addenda. The core section deals with basic telephone features, while separate addenda were developed for coping with specific telephone call types.

Up to now, there are 8 addenda to the telephone volume, dealing with MFC/ATSQSIG, FAA legacy telephone interworking, Instantaneous access calls, Override calls, Voice calls, Extended call forwarding, Hotline access calls, Radio intercom calls.

SITTI has been giving its contribution to all volumes. In particular, SITTI has been leading the subgroup that brought to the standardization of the supervision section (volume 5).

VIENNA AGREEMENT



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From B to C

Version B of the ED137 standard presents a good level of maturity and can validly be used for implementing a standard compliant system with high level of interoperability also including devices that utilize version C. In fact, this latter introduced a mean to identify the version of each connected component and to negotiate the corresponding behavior. This allows systems implementing different versions of the standard to coexist and interoperate according to a common agreed level.

Version C of the ED137 standard introduced a number of significant changes and additions to the features foreseen by its predecessor. The process that led to the introduction of such changes and new facilities went also through the execution of 6 so-called "plugtests" or "interoperability events" that were run in Europe and USA in order to validate the provisions of the standard.

SITTI has been participating in all such events, significantly contributing to the final result.

Version C - What's new

The following tables list the main changes introduced by version C when compared to version B. It shall be once again underlined that the two versions can work together: systems implementing different versions automatically negotiate the best way of interfacing.

// Radio

- Version identification and negotiation
- RX IDLE mode (in addition to TRX, RXonly, Coupling)
- Multicast transmission for radio receivers
- SELCAL tones via SIP messages
- TEST PTT used by VCS to perform a transmission test during inactivity periods
- New error codes to better identify the reason of unsuccessful connection or transmission
- Frequency ID from non-VoIP keying sources
- Best Signal Selection (BSS) delay difference compensation prior to signal quality comparison
- No SIP session disconnection if multimode radio changes frequency
- Improvement in the algorithm for multiple transmitter dynamic delay compensation

// Telephone

- Version identification and negotiation
- Independent focus in multi-party calls
- Preset conference call
- New call types:
  - Extended call forward
  - Hotline access call
  - Voice call
  - Radio intercom call
  - Basic call forward
  - Multi destination call
  - FAA legacy telephone calls
  - Call/answer call
  - Call pickup
- Call type indication in the SIP header
- Loop detection algorithm
- Echo detection algorithm

// Recording

- Recording Clients may use any transport protocol while Recording Servers shall support all of them
- Additional optional modes:
  - Audio only mode (without CRD)
  - R2S Header Extension Recording
  - Radio Selection CRD
  - RTSP Session Keep Alive
  - Recording Server Liveliness
  - Proprietary CRD Metadata
- Ambient recording
- Clarification on the use of playback

// Supervision

- Clear identification of the components being supervised
- Identification of applicable RFCs
- MIB-2 object groups involved
- Clarification on MIB structure and architecture
- VCS, Radio, Recorder, Gateway requirements
- Alarms and events
- Performance monitoring