

ED-137/4C Change 2

INTEROPERABILITY STANDARDS FOR VOIP ATM COMPONENTS VOLUME 4: RECORDING

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COMPLEMENTS ED-137/4C PLUS CHANGE 1

FOREWORD

1. This document was developed by EUROCAE Working Group 67 “Voice over IP (VoIP) for ATM Applications” and was approved by the Council of EUROCAE on [22 January 2025].
2. This document complements ED-137/4C ‘Interoperability Standards for VoIP ATM Components – Volume 4 (Recording)’ (March 2019) plus Change 1 (May 2020).
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CHAPTER 1

CHANGE SUMMARY

Change 2 to ED-137/4C is summarised as follows:

Description of change	Affected chapters, paragraphs
Naming IPv6 in the presentation of the document	Chapter 1.2 ED-137 VOLUME 4 PRESENTATION
Requirement to support IPv6	Chapter 3.1 SESSION
IPv6 support in transport parameters	Chapter 3.2.4 FORMATING DETAILS FOR THE DIFFERENT TRANSPORT PARAMETERS
IPv6 support in RTSP URL	Chapter 3.3 CONTROL MESSAGES
SDP RFC reference update	Chapter 3.3.1 ANNOUNCE AND SETUP
DESCRIBE Response content headers	Chapter 3.3.7 Table 3.2
Timestamp format correction	Chapter 3.5.5 TIMESTAMP FORMAT
Backward compatibility statement	Chapter 3.6 WG67-Version
XML Schema correction	ANNEX A XML XSD SCHEMA
RFC reference update	ANNEX B REFERENCES

CHAPTER 2

CHANGE DETAILS

2.1 CHAPTER 1.2 ED-137 VOLUME 4 PRESENTATION

The scope of the WG-67 ED-137 Documents is to define the rules for VoIP implementations to support ATM communications:

- Radio for the Volume 1
- Telephone for the Volume 2
- Recording for the Volume 4
- Supervision for the Volume 5

This includes the performances requested for the related communications.

The present document, that is the ED-137 Volume 4, proposes a profile standard for the use of RTSP to establish, terminate and control recording sessions of the Ground Telephone Service and the Radio Service in an Air Traffic Services Ground Voice Network (AGVN).

RTSP is an application layer protocol for establishing, terminating and controlling multimedia sessions. It is typically carried over the Internet Protocol (IPv4 or IPv6) (IETF RFC 791 [2] and IETF RFC 2460 [6]). Although this specification is based on RTSP version 1.0 as specified in IETF RFC 2326 [5], some of the **transport** parameters (see 3.2.4 for a list) and **explicit IPv6 support** are extensions taken from RTSP version 2.0 as specified in IETF RFC 7826 [16].

The document ED-137 Volume 4 "Recording" is intended to serve as interoperability guidelines for implementing recording facilities against service recording needs in the "WAN" according to FIGURE 1.1. This document is not intended to propose implementation details to satisfy all the requirements for audio legal recording.

This document proposes a specification for the signaling profile for basic services that provide a bi-directional transfer capability for speech media between recording clients and server in an IP AGVN employing RTSP in support of ATS recording services.

Interworking between an IP AGVN and a public IP network is out of scope of this document.

2.2 CHAPTER 3.1 SESSION

Recording Clients **SHALL** use RTSP in accordance with IETF RFC 2326 [5] to enable controlled, on-demand delivery of real-time data. Although this specification is based on RTSP version 1.0 as specified in IETF RFC 2326 [5], some of the below **transport** parameters (see 3.2.4 for a list) and **explicit IPv6 support** are extensions taken from RTSP version 2.0 as specified in IETF RFC 7826 [16].

Implementations of ED-137 Volume 4 **SHOULD** support IPv6 addresses in URIs and RTSP headers as defined in this document.

Negotiation of the Recording Server address is out of the scope of this document. It is assumed in this document that Recording Clients already know the IP address of the Recording Server.

Recording Clients **SHALL** use ANNOUNCE and SETUP to establish a recording session. Recording Clients, **SHALL** use DESCRIBE and SETUP to establish a replay session. Session setup provides the session description (connection information) and media description (media name and transport address) of each participant.

Recording Clients **SHALL** use TEARDOWN to close a session. Recording Clients **SHALL** terminate an RTSP session upon any RTSP request rejection or timeout. TEARDOWN request **SHALL NOT** be sent from the Recording Server to the Recording Client to terminate an RTSP session.

CHAPTER 3.2.4 FORMATING DETAILS FOR THE DIFFERENT TRANSPORT PARAMETERS

The following format **SHALL** be used for transports parameters:

Transport	= "Transport" ":" transport-spec
transport-spec	= transport-protocol/profile[/lower-transport]*parameter
transport-protocol	= "RTP"
profile	= "AVP"
lower-transport	= "TCP" "UDP"
parameter	= ("unicast") ";" "destination" ["=" address] ";" "source" ["=" address] ";" "interleaved" "=" channel ["-" channel] ";" "dest_addr" "=" addr-list) ";" "src_addr" "=" addr-list) ";" "setup" "=" contrans-setup) ";" "connection" "=" contrans-con) ";" "client_port" "=" port ["-" port] ";" "server_port" "=" port ["-" port] ";" "ssrc" "=" ssrc ";" "mode" = "<" 1\#mode ">"
ssrc	= 8*8 (HEX)
channel	= 1*5 (DIGIT)
address	= host
contrans-setup	= "active" "passive"
contrans-con	= "new "
mode	= "play" "record"
host-port	= "<" host [":" port] ":" port ">"
host	= <As defined in RFC 3986, 3.2.2. Host>, case insensitive
port	= <As defined in RFC 3986, 3.2.3. Port>

General parameters:	
destination	The address to which a stream will be sent.
source	If the source address for the stream is different than can be derived from the RTSP endpoint address (the server in playback or the client in recording), the source MAY be specified.
mode	The mode parameter indicates the methods to be supported for this session. Valid values are PLAY and RECORD. If not provided, the default is PLAY.
interleaved	The interleaved parameter implies mixing the media stream with the control stream. The argument provides the channel number to be used in the # block (see 3.2.1.1) and it SHALL be present. This parameter MAY be specified as an interval, e.g., interleaved=4-5 in cases where the transport choice for the media stream requires it, e.g. for RTP with RTCP. The channel number given in the request is only a guidance from the client to the server on what channel number(s) to use. The server MAY set any valid channel number in the response. The declared channels are bidirectional, so both end parties MAY send data on the given channel.
client_port	This parameter provides the unicast RTP/RTCP port pair on which the client has chosen to receive media data and control information. It is specified as a range, e.g., client_port=3456-3457.