



The European Organisation for Civil Aviation Equipment
L'Organisation Européenne pour l'Équipement de l'Aviation Civile

INTEROPERABILITY STANDARDS FOR VOIP ATM COMPONENTS

VOLUME 4: RECORDING

This document is the exclusive intellectual and commercial property of EUROCAE.

It is presently commercialised by EUROCAE.

This electronic copy is delivered to your company/organisation for internal use exclusively.

In no case it may be re-sold, or hired, lent or exchanged outside your company.

ED-137/4B

January 2012

102 rue Etienne Dolet
92240 MALAKOFF, France
Web Site: www.eurocae.net

Full document details: EUROCAE ED-137/4B 2012

The logo features the letters "AA" in a stylized yellow font, followed by the word "Documents" in a bold blue font. Below this, the text "Document Sharing Hub" is written in a smaller blue font. To the right of the "AA" logo, the contact information is listed: "Tel: 33 1 40 92 79 30", "Fax: 33 1 46 55 62 69", and "Email: eurocae@eurocae.net".

INTEROPERABILITY STANDARDS FOR VOIP ATM COMPONENTS

VOLUME 4: RECORDING

This document is the exclusive intellectual and commercial property of EUROCAE.

It is presently commercialised by EUROCAE.

This electronic copy is delivered to your company/organisation for internal use exclusively.

In no case it may be re-sold, or hired, lent or exchanged outside your company.

ED-137/4B

January 2012

FOREWORD

1. This document “**Interoperability Standards for VoIP ATM Components – Volume 4: Recording**” was prepared by EUROCAE Working Group 67 and was accepted by the Council of EUROCAE on 19 January 2012.
2. EUROCAE is an international non-profit making organisation. Membership is open to manufacturers and users in Europe of equipment for aeronautics, trade associations, national civil aviation administrations and non-European organisations. Its work programme is principally directed to the preparation of performance specifications and guidance documents for civil aviation equipment, for adoption and use at European and world-wide levels.
3. The findings of EUROCAE are resolved after discussion among its members and, where appropriate, in collaboration with RTCA Inc, Washington D.C. USA and/or the Society of Automotive Engineers (SAE), Warrendale, PA, USA through their appropriate committee.
4. This document is part of a series of EUROCAE Documents, composed of 5 volumes:
 - ED-137/1 “Interoperability Standard for VOIP ATM Components – Volume 1: Radio”
 - ED-137/2 “Interoperability Standard for VOIP ATM Components – Volume 2: Telephone”
 - ED-137/3 “Interoperability Standard for VOIP ATM Components – Volume 3: European Legacy Telephone Interworking”
 - ED-137/4 “Interoperability Standard for VOIP ATM Components – Volume 4: Recording”
 - ED-137/5 “Interoperability Standard for VOIP ATM Components – Volume 5: Supervision”.

Together the 5 volumes, edition B, replace ED-137A, published in September 2010.
5. This version of the document represents “the minimum specification required for Manufacturers and Users to assure Interoperability from/to VoIP ATM Recorder Components”. EUROCAE WG-67 has updated the previous version of ED-137:
 - following a series of plug tests, performed by many suppliers in Arlington in May 2011 and in Sophia Antipolis in June 2011, improving ED-137 concerning the understanding of interoperability requirements between the ATM VoIP components,
 - following discussions with FAA for taking into account US requirements agreeable within the European context.
6. This version is taking into account the EUROCONTROL VoIP in ATM Test Case Specification.
7. EUROCAE performance specifications are recommendations only. EUROCAE is not an official body of the European governments; its recommendations are valid statements of official policy only when adopted by a particular government or conference of governments.
8. Copies of this document may be obtained from:

EUROCAE
102 rue Etienne Dolet
92240 MALAKOFF
France
Tel: 33 1 40 92 79 30
Fax: 33 1 46 55 62 65
Email: eurocae@eurocae.net
Web Site: www.eurocae.net

TABLE OF CONTENTS

FOREWORD	i
CHAPTER 1 INTRODUCTION	1
1.1 BACKGROUND	1
1.2 ED-137 VOLUME 4 PRESENTATION	3
1.3 TERMINOLOGY FOR REQUIREMENTS, RECOMMENDATIONS AND OPTIONS	3
1.4 VERSION TRACKING	4
CHAPTER 2 RECORDING MODEL	5
2.1 ACTIVE RECORDING	5
2.2 RECORDING PHONE COMMUNICATION	5
2.3 RECORDING RADIO COMMUNICATION	6
2.4 SESSION SETUP	6
2.4.1 SIP	6
2.4.2 RTSP	8
2.5 TRANSPORT	9
2.5.1 EMBEDDED BINARY DATA	9
2.5.2 RTP OVER INDEPENDENT TCP	10
2.5.3 RTP OVER UDP	11
2.6 RTSP CONTROL MESSAGES	12
2.6.1 ANNOUNCE AND SETUP	12
2.6.2 RECORD	12
2.6.3 PAUSE	13
2.6.4 SET_PARAMETER	13
2.6.5 TEARDOWN	15
2.6.6 REPLAY (optional)	15
2.7 CALL RECORD DATA FORMAT	16
2.7.1 Properties	16
2.7.2 Operations	17
2.7.3 Time stamp format	17
2.8 REFERENCING CALL SCENARIOS	18
2.8.1 CRD CONNECTION REFERENCE GENERATION	19
2.8.2 CRD TELEPHONY CALLREF GENERATION	19
2.8.3 CRD XML BODY IN RECORD, PAUSE, SET_PARAMETER AND TEARDOWN	19
2.8.4 SETUP WITHOUT CRD	19
2.9 WG67-Version	19
CHAPTER 3 PHONE	21
3.1 AUDIO SOURCE AND CODING	21
3.2 CALL RECORD DATA	21
3.3 FLOW EXAMPLES FOR TELEPHONY CALLS	22
3.4 RTSP SESSION FOR CALL INTRUSION AND CONFERENCE	24
3.5 HOLD OPERATION	26

CHAPTER 4	RADIO.....	28
4.1	AUDIO SOURCE AND CODING.....	28
4.2	CALL RECORD DATA	29
4.2.1	PTT OPERATION	30
4.2.2	SQU OPERATION	30
4.2.3	VOTING OPERATION	31
ANNEX A	REFERENCES	33
ANNEX B	ACRONYMS	35
ANNEX C	LIST OF EUROCAE WG-67 CONTRIBUTORS	37

FIGURE INDEX

FIGURE 1 – VIENNA AGREEMENT	2
FIGURE 1 – RECORDING SESSIONS	5
FIGURE 2 – RECORDING PHONE COMMUNICATION.....	5
FIGURE 3 – RECORDING RADIO COMMUNICATION	6
FIGURE 4 – SIP REGISTRATION	7
FIGURE 5 – SIP SESSION SETUP	7
FIGURE 6 – SIP SESSION SETUP: MESSAGE SEQUENCE.....	7
FIGURE 7 – RTSP RECORD AND REPLAY SESSION.....	8
FIGURE 8 – EMBEDDED BINARY DATA FORMAT	10
FIGURE 9 – TCP FRAME FORMAT	11
FIGURE 10 – RTSP RECORD SESSION SETUP	12
FIGURE 11 – RTSP RECORD SESSION SETUP: MESSAGES AND SEQUENCE	12
FIGURE 12 – RECORD: MESSAGES AND SEQUENCE	13
FIGURE 13 – PAUSE: MESSAGES AND SEQUENCE	13
FIGURE 14 – SET_PARAMETER: MESSAGES AND SEQUENCE	14
FIGURE 15 – TEARDOWN: MESSAGES AND SEQUENCE.....	15
FIGURE 16 – RTSP REPLAY SESSION: MESSAGES AND SEQUENCE.....	16
FIGURE 17 – CALL SCENARIO	18
FIGURE 18 – AUDIO SOURCE AT USER TERMINAL (G/G).....	21
FIGURE 19 – AUDIO SOURCE AT RADIO (A/G)	28
FIGURE 20 – AUDIO SOURCE AT USER TERMINAL (A/G)	28

TABLE INDEX

TABLE 1 – LIST OF PHONE PROPERTIES	21
TABLE 2 – LIST OF PHONE OPERATIONS.....	25
TABLE 3 – LIST OF RADIO PROPERTIES.....	29
TABLE 4 – LIST OF RADIO OPERATIONS	29

CHAPTER 1

INTRODUCTION

1.1 BACKGROUND

Ground-Ground (G-G) ATM voice systems have been based upon analogue and more recently, digital Time Division Multiplexing / Pulsed Code Modulation (TDM/PCM) technologies for many years.

Nowadays, however, convergence of voice and data into one multimedia network is a popular trend with a variety of technical solutions available on the market. Following in this direction ATM communication networks are adopting, by a process of gradual evolution, a common infrastructure for voice and data services.

As the technology has developed IP Technology which has now the true potential to fulfil operational and technical ATM communication requirements - including those of voice / data convergence, Quality of Services (QoS), security and safety. There is also the possibility that IP may deliver solutions that will, over time, bring about true savings in investment and operating costs.

EUROCAE Working Group 67 (WG-67) undertook the mission to assess the feasibility of using Voice over Internet Protocol (VoIP) for providing ATM voice services. The group defined criteria, requirements and guidelines based upon the following operational needs and constraints:

- Operational and Technical Air-Ground (A-G) and Ground-Ground (G-G) ATM Voice system requirements
- Existing IP Voice protocols and signalling standards
- IP network capabilities for Voice services
- Security, Quality of Service (QoS), and Convergence (infrastructure, protocol, applications)
- Existing IP Voice ATM system capabilities and service interfaces.

The following tasks were identified to fulfil the WG-67 mission:-

- Define ATM Systems and identify their components (Voice Communication System / VCS, Ground-based Radio Station / GRS)
- Determine possible additional operational and technical ATM requirements for new ATM voice systems, also taking into consideration A-G communications.
- Make recommendations to upgrade current standardisation documents.
- Develop a Technical Specification for a VoIP Voice ATM System including:
 - o Minimum performance and safety/security requirements for the system and, if appropriate, for components;
 - o Interoperability requirements between IP components of the VoIP ATM system;
 - o Minimum performance requirements of an IP Network to support ATM Voice services;

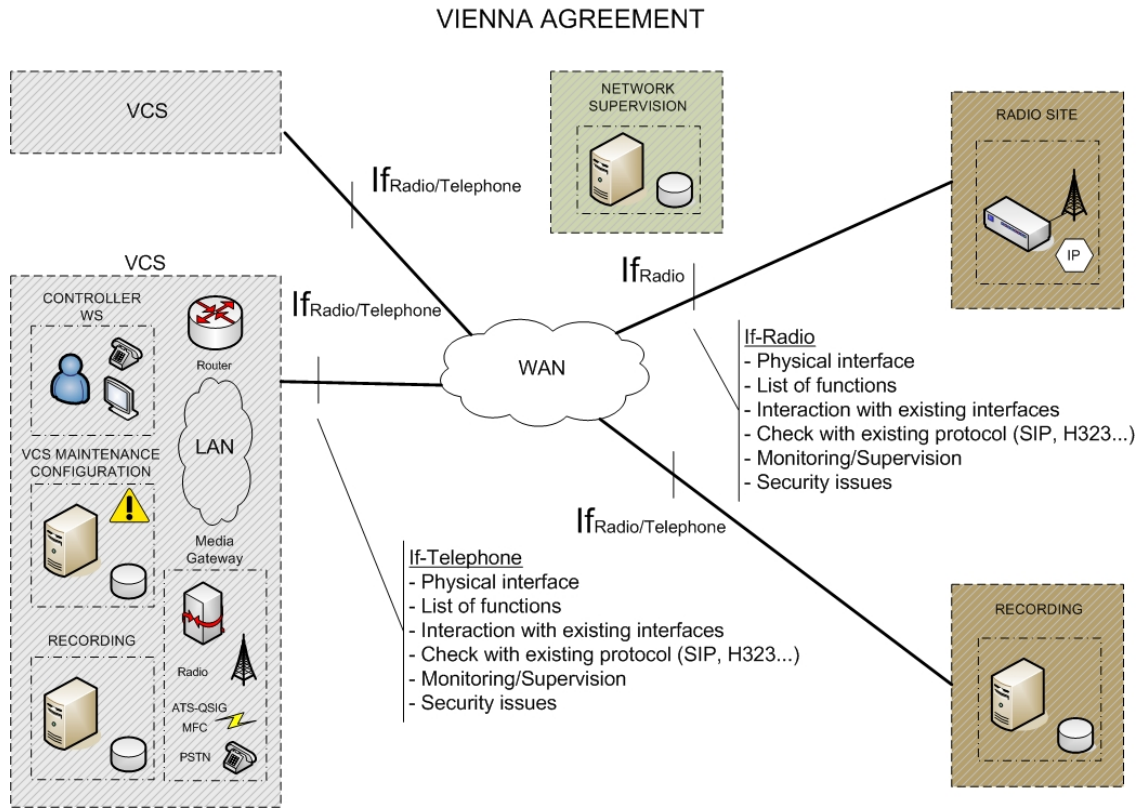
Consequently the following three documents were delivered:

ED-136 - VoIP ATM System Operational and Technical Requirements

ED-137 - Interoperability Standards for VoIP ATM Components

ED-138 - Network Requirements and Performances for VoIP ATM Systems

The contents of all three documents are premised on the “**Vienna Agreement**” which defines the different components of a VoIP ATM system and their mutual interfaces as depicted in Figure 1.



VoIP components are interconnected through an IP network and suppliers are free to define their internal architecture (IP/Ethernet, TDM/PCM - Time Division Multiplexing / Pulsed Code Modulation, ...). Between VoIP components, required interfaces are defined to guarantee their functional and technical interoperability.

Therefore, VoIP ATM Systems are composed of:

- **VoIP VCS Components** performing Radio and / or Telephone functions, including:
 1. Controller Working Positions, assuring HMI including voice devices (microphone and loudspeaker);
 2. Possible local VCS Maintenance and Configuration stations;
 3. Possible local Recording devices;
 4. Possible LAN for local interconnection;
 5. Possible Media Gateways to legacy systems (ATS-QSIG, ATS-R2, ATS-No.5, PSTN, Radio analogue lines, ...).
- **VoIP Ground Radio Station Components** performing AM VHF and UHF Radio functions.
- **VoIP Supervision System Components** performing monitoring and control functions.
- **VoIP Recording Components** performing recording functions.
- **IP WAN Components** performing interconnection services between two or more different physical components.

1.2 ED-137 VOLUME 4 PRESENTATION

The scope of the WG67 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
- European Legacy Telephone Interworking for the Volume 3
- 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 modify 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 modifying multimedia sessions. It is typically carried over the Internet Protocol (IP) (IETF RFC 791 [2] and IETF RFC 2460 [6]). RTSP is defined in IETF RFC 2326 [5].

The document ED137 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 "Vienna Agreement". 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 signalling profile both for basic services that provide a bi-directional transfer capability for speech media between user terminals, radios and a recorder in an IP AGVN employing SIP and RTSP in support of ATS recording services.

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

1.3 TERMINOLOGY FOR REQUIREMENTS, RECOMMENDATIONS AND OPTIONS

The terminology for requirements, recommendations and options in this document is based on RFC 2119 [4], which specifies Best Current Practice regarding the use of Key Words for the Internet Community. As such, the following terminology is applied:

- The word **SHALL** denotes a mandatory requirement;
- The word **SHOULD** denotes a recommendation;
- The word **MAY** denotes an option.

To avoid confusion with their natural meanings in the English language, the words **SHALL**, **SHOULD**, and **MAY** take on the meaning stated above only where printed in boldface. When printed in normal (Arial) typeface, the natural English meaning is meant.

Detailed description of terminology:

1. **SHALL** This word has the same meaning as the phrase "REQUIRED" and means that the definition is an absolute requirement of the specification.
2. **SHALL NOT** This phrase means that the definition is an absolute prohibition of the specification.
3. **SHOULD** This word, or the adjective "RECOMMENDED", means that there may exist valid reasons in particular circumstances to ignore a particular item, but the full implications must be understood and carefully weighed before choosing a different course.

4. **SHOULD NOT** This phrase, or the phrase "NOT RECOMMENDED" mean that there may exist valid reasons in particular circumstances when the particular behaviour is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behaviour described with this label.
5. **MAY** This word, or the adjective "OPTIONAL", mean that an item is truly optional.

1.4

VERSION TRACKING

The present document, ED-137 Volume 4, **SHALL** be referenced as recorder.01 in the WG67-Version RTSP header field as described in [2.9](#).

CHAPTER 2

RECORDING MODEL

2.1

ACTIVE RECORDING

1 [RECORDING] Active Recording

Recording **SHALL** be based on active sessions opened from clients (User Terminal, Radio Transmitter/Receiver or specific 3rd party devices) to one recording device (or two devices required for redundancy). Active means that any client that sends or receives media streams (i.e. audio) takes the responsibility to send a copy of either stream to the recorders. The used time source **SHALL** be synchronized to the ATSU time source. This is assumed to be Universal Time Coordinated (UTC) to the accuracy specified by ICAO.

NOTE: *In order to simplify drawings, the following just mentions a single recording device. All described mechanisms are valid for two or a defined number of recorders.*

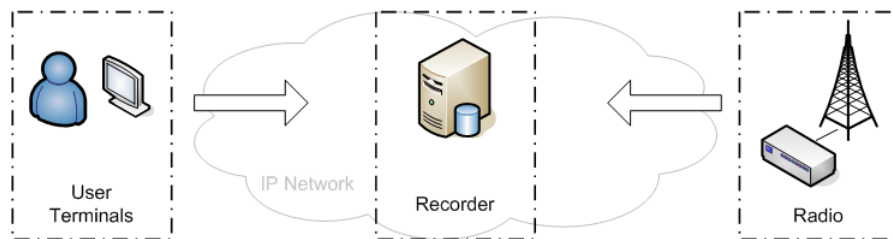


FIGURE 1: RECORDING SESSIONS

2.2

RECORDING PHONE COMMUNICATION

2 [RECORDING] Phone Communication

User Terminals participating in a G/G communication session **SHALL** provide a single audio stream that summarizes all incoming (IN) and outgoing (OUT) audio streams.

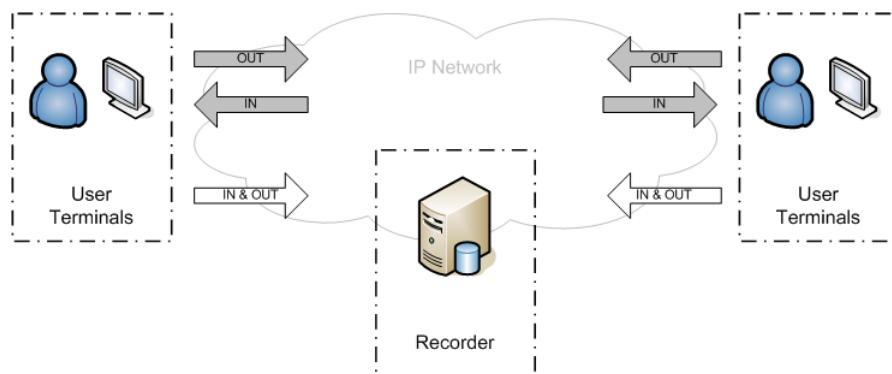


FIGURE 2: RECORDING PHONE COMMUNICATION

2.3 RECORDING RADIO COMMUNICATION

3 [RECORDING] Radio Communication

User terminal client **SHALL** provide a single audio stream towards its nominated recorder server that contains summed audio of TX and/or RX on all frequencies selected (enabled). If local side tone (generated by VCS) or remote side tone (RX audio that contains off-air transmitted audio) is enabled from configuration, the original TX audio **SHALL** be summed twice. Due to the voting algorithm it is possible to exclude some Rx streams on the same frequency from summed audio stream Radios (or Gateways connecting legacy radios to an IP network) **SHALL** provide a single audio stream that contains the received (RX) audio stream related to a single radio channel.

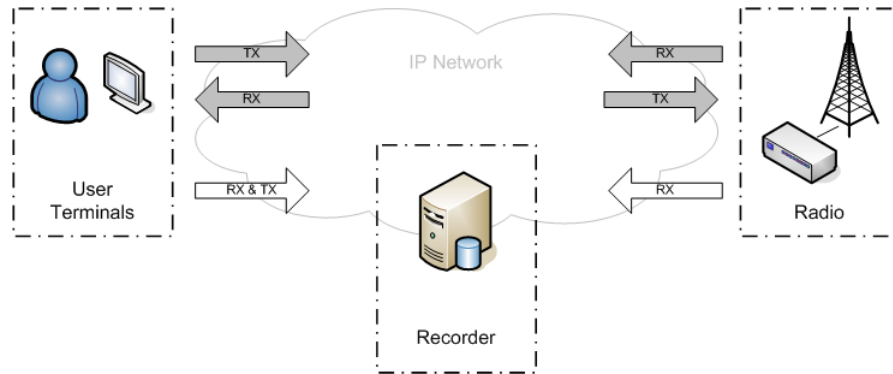


FIGURE 3: RECORDING RADIO COMMUNICATION

2.4 SESSION SETUP

Active recording requires an established session (i.e. a certain number of parameters that are exchanged between entities prior to any recording). User Terminal, Radio and Recorder **SHALL** use RTSP for such sessions. As RTSP relies on a transport layer protocol (TCP or UDP), these entities **MAY** use SIP to exchange capabilities and connection information (i.e. IP address, port number, and transport protocol). The following section describes the session setup using SIP and RTSP.

2.4.1 SIP

Note that this section assumes that SIP is used for session setup hence the terminology for requirements, recommendations and options is only valid for this case.

Any entity involved in a recording session (User Terminal and Recorder) **SHOULD** register with a SIP Registrar using the REGISTER method according RFC3261. It **SHOULD** be possible to register multiple contacts for a single Address of Record (AOR).

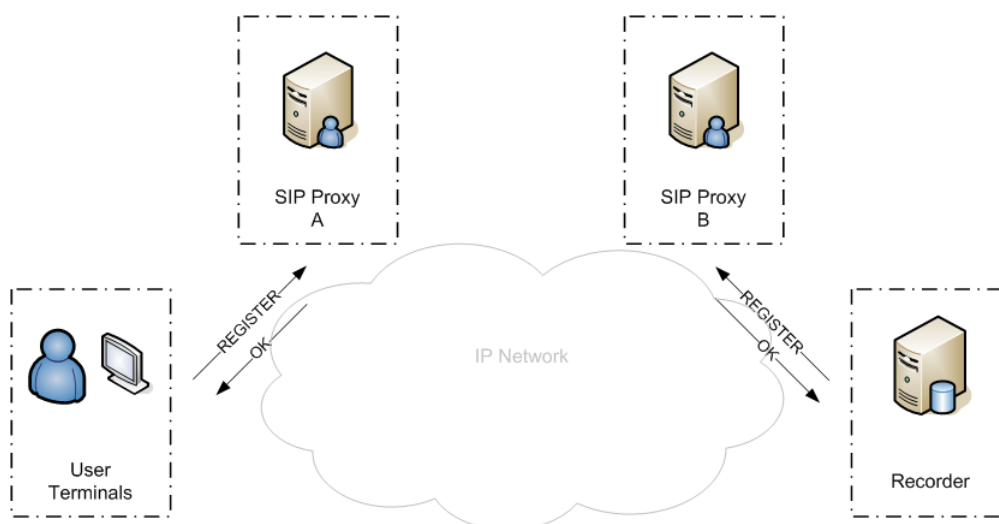


FIGURE 4: SIP REGISTRATION

Participants (User Terminal, Radio) **SHALL** use INVITE to establish a session. This session setup provides the session description (connection information) and media description (media name and transport address) of each participant.

A SIP session between the point of voice presence and point of recording **MAY** be set-up using public key encryption technology to verify identity of point of voice presence to be recorded and point of recording.

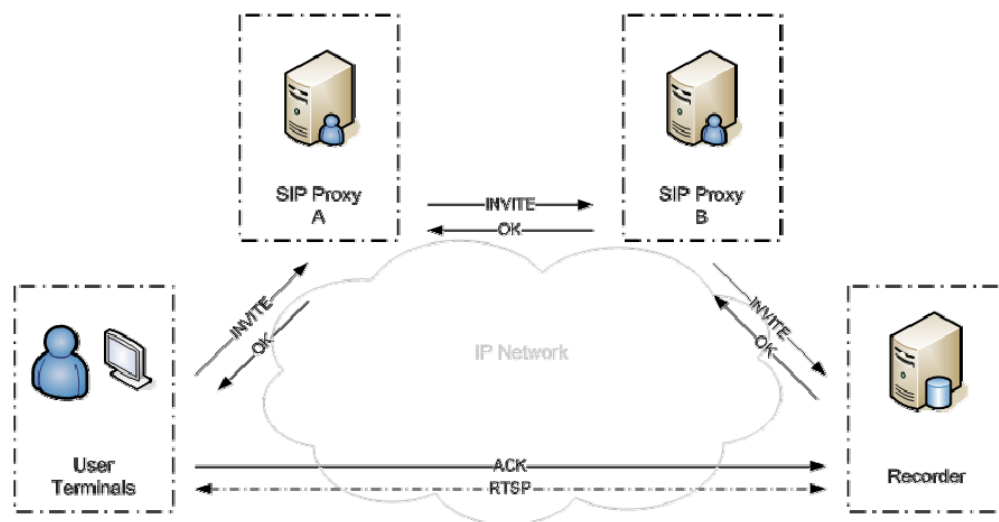


FIGURE 5: SIP SESSION SETUP

Recorder	Terminal	Radio	Recorder
	INVITE		
<-----			
200 OK		INVITE	
>-----		<-----	
ACK		200 OK	
<-----		<-----	
TCP		ACK	
>-----		>-----	
		TCP	
		<=====	

FIGURE 6: SIP SESSION SETUP: MESSAGE SEQUENCE