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

Interoperability Standards for VoIP ATM Components

Part 2: Telephone

ED-137 Part 2
"Month Year"

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FOREWORD

1 The document ED-137 “**Interoperability Standards for VoIP ATM Components**” was prepared by EUROCAE Working Group 67 and was accepted by the Council of EUROCAE on “Month Year”.

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 The document represents “the minimum specification required for Manufacturers and Users to assure Interoperability between VoIP ATM Components”.

5 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.

6 Copies of this document may be obtained from:

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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 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:
    - Minimum performance and safety/security requirements for the system and, if appropriate, for components;
    - Interoperability requirements between IP components of the VoIP ATM system;
    - Minimum performance requirements of an IP Network to support ATM Voice services;
    - Guidelines for qualification tests of VoIP ATM systems and their components.

Consequently the following four 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

ER-139 - Qualification tests for VoIP ATM Components and Systems

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

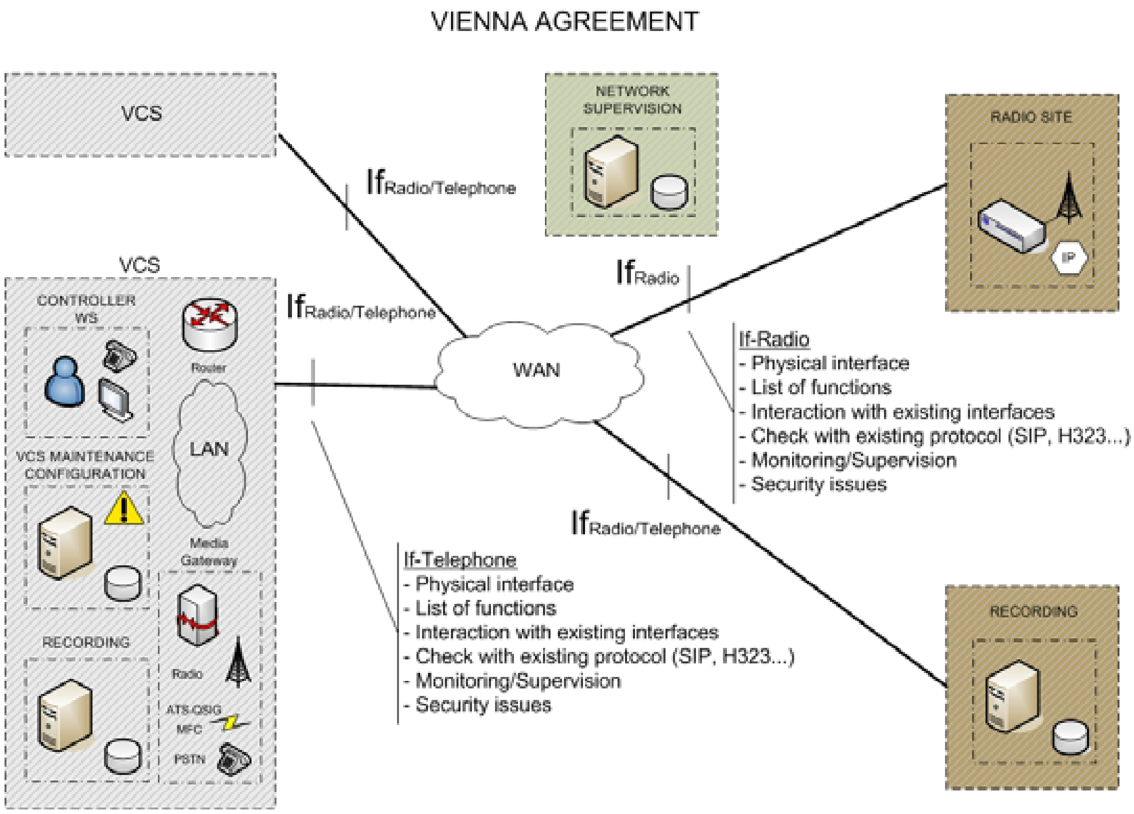


Fig. 1 – Vienna Agreement

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 PRESENTATION

The scope of the WG67 ED-137 Document is to define the rules for VoIP implementations to support ATM communications. This includes the performances requested for radio communications (ED-137 Part 1), telephone communications (ED-137 Part 2), audio recording (ED-137 Part 3) and system supervision (ED-137 Part 4).

The present document, ED-137 Part 2, proposes a profile standard for the use of SIP to establish, terminate and modify speech media sessions of the Ground Telephone Service in an Air Traffic Services Ground Voice Network (AGVN).

SIP is an application layer protocol for establishing, terminating and modifying multimedia sessions. It is typically carried over the Internet Protocol (IP) (RFC 791 [2] and RFC 2460 [6]). Telephone calls are considered as a type of multimedia session where just audio is exchanged. SIP is defined in RFC 3261 [8].

This document specifies the signalling profile both for basic services, that provide a bi-directional transfer capability for speech media between user terminals in an IP AGVN employing SIP, and for call-related signalling in support of ATS supplementary 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 [5], 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 <u>requirement</u>;
- The word **SHOULD** denotes a <u>recommendation</u>;
- The word **MAY** denotes an <u>option</u>.

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.

CHAPTER 2

COMMUNICATION MODEL

2.1 DEFINITIONS

The following terms are used in this document:

- **Gateway:** An entity that performs interworking between a circuit-switched AGVN employing ATS-R2, ATS-No.5 or ATS-QSIG and an IP AGVN employing SIP. It also acts as a Media Gateway that terminates circuit-switched AGVN facilities (trunks, loops), packetizes the media stream and delivers packetized traffic to the IP AGVN. It performs these functions in the reverse order for media streams flowing from the IP AGVN to the circuit-switched AGVN.

- **Incoming Gateway:** A gateway that routes an incoming call from a route employing the ATS-R2, ATS-No.5 or ATS-QSIG signalling system on to an IP network employing the SIP signalling system.

- **Outgoing Gateway:** A gateway that routes an incoming call from an IP network employing the SIP signalling system on to a route employing the ATS-R2, ATS-No.5 or ATS-QSIG signalling system.

- **SIP:** SIP is the session initiation protocol defined by the IETF in RFC 3261 [8]. It defines the control messages to create and terminate communications sessions between multiple endpoints on IP networks.

- **SIP Headers:** SIP Headers are a set of parameters that could be assigned specific values inside a SIP message. They convey information about the SIP request or response.

- **SIP Messages:** SIP Messages are the basic language elements in SIP. Each SIP message contains SIP headers and may contain a message body. There are two types of SIP Messages: Request and Response.

- **SIP Network Entity:** A SIP Network Entity is any network component that supports SIP signalling.

- **SIP Profiles:** Profiles in SIP define headers to be used as well and values restrictions and definitions. It also defines which SIP Internet Draft to use and how to use them. Profiles may be defined by any organization. This document defines one of such profiles.

- **SIP Request Methods:** SIP Request Methods are messages, typically, sent by the SIP client to initiate a transaction. For example, an INVITE method starts a new call. A CANCEL method cancels the request.

- **SIP Responses:** SIP Responses are messages received by the SIP client during a transaction that was initiated by a request. One or more responses can take place in answer to a single request.

- **User Agent Client:** A User Agent Client (UAC) is the logical entity within each network component that creates new requests.

- **User Agent Server:** A User Agent Server (UAS) is the logical entity within each network component that generates a response to a SIP request.

2.2 PROTOCOL STACK

The specifications provided hereafter **SHALL** concern the interface between two VCSs for the following subjects:

- Signalling
- Audio

All requirements in this document can be met by corresponding IETF RFCs, ITU recommendations, ECMA standards and Eurocontrol specifications and recommendations, as indicated in each case.

2.2.1 Signalling

The object of the present document is the specification of the use of SIP in an IP AGVN, as well as the interworking between an IP AGVN and an ATS-R2/No.5 (analogue) or ATS-QSIG (digital) circuit-switched AGVN.

2.2.1.1 Signalling in an IP AGVN

Signalling in an IP AGVN **SHALL** be based on the Session Initiation Protocol (SIP) (RFC 3261, [8]).

SIP is an application-layer transaction protocol that provides advanced signalling and control functionality for a large range of multimedia communications. SIP is an important component in the context of other protocols to enable a complete multimedia architecture, as shown in Fig. 2.

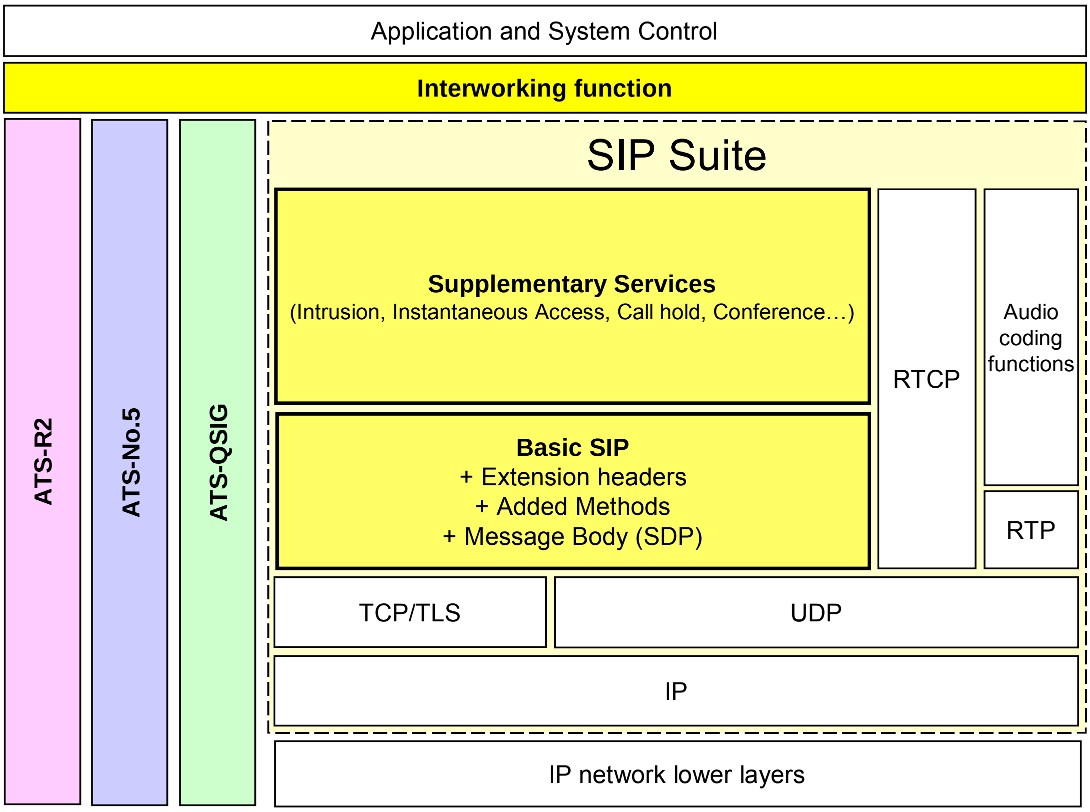


Fig. 2 – Protocol Stack

In addition to the signalling functionality specified in this document, it is assumed that the IP AGVN components also includes the following functionality:

- one or more physical interfaces on the IP network side supporting, through layer 1 and layer 2 protocols, IP as the network layer protocol and UDP (RFC 768 [1]) and TCP (RFC 793 [3]) as transport layer protocols, these being used for the transport of SIP signalling messages and, in the case of UDP, also for media information;

- the support of TLS (RFC 4346 [24]) as additional transport layer secure protocol on the IP network side, this being used for the transport of SIP signalling messages; and

- a means of transferring media information.

2.2.1.2 Interworking with ATS Legacy Systems

The SIP environment **SHALL** need to interwork with ATS-R2/ATS-QSIG-based AGVN in order to support calls originating in one environment and terminating in the other. Interworking with ATS-No.5-based AGVN is **RECOMMENDED**.

Interworking is achieved through gateways. The term “Gateway” is used in the present document, within the context of a call, as an entity that performs interworking between one signalling system and another. This definition is specifically applied here to interworking between the following signalling systems:

- SIP *vs.* ATS-R2
- SIP *vs.* ATS-No.5
- SIP *vs.* ATS-QSIG

This covers all scenarios for incoming gateway calls from legacy circuit-switched AGVN to SIP on a packet-switched IP AGVN. Likewise, it covers all scenarios for outgoing gateway calls from SIP in a packet-switched IP AGVN to legacy circuit-switched AGVN. The functionality required for the different gateways has been defined in accordance with the Vienna Agreement to guarantee interoperability between IP components of IP Voice ATM systems and the current non-IP voice ATM systems.

The use of signalling “tunnelling” over an IP network, which means replacement of dedicated (analogue or digital) links by an IP network using two “converters” (that encapsulate, for instance, analogue signalling in RTP frames) connected at opposite ends of the IP network to interconnect two legacy VCSs, comes out of the original Vienna Agreement and, therefore, is outside the scope of the present document.

2.2.1.3 Interworking with Public Telephone Networks or Private Branch Exchanges

Interworking with PSTNs, ISDNs, PABXs or analogue terminals is outside the scope of this specification; informative references are RFC 3398 [11], RFC 3666 [16] and ECMA-339 [34]

2.2.2 Audio Specifications

The audio transport **SHALL** be supported by the real-time transport protocol (RTP) (RFC 3550 [14]), augmented by its associated control protocol (RTCP) to allow monitoring of the audio packets delivery.

The Voice **SHALL** be coded in accordance with ITU-T Rec. G711 A-law (in Europe) or μ-law (in USA/Japan) coding. If compression is needed ITU-T Rec. G728 **SHALL** be used without Packet Loss Concealment.

For interoperation with ATS-QSIG, ITU-T G.728 (LD-CELP) **SHALL NOT** be transcoded, but forwarded to avoid codec transcoding losses; in this case, ITU-T G.728 **SHALL** be used end-to-end.

Use of ITU-T Rec. G.729 audio coding is **OPTIONAL**.

CHAPTER 3

PROFILE STANDARD FOR THE USE OF SIP IN AN AGVN

An Air Traffic Services VoIP Communications System **SHALL** support SIP version 2 as specified in RFC 3261 [8]. The requirements of RFC 3261 [8] **SHALL** apply with modifications as specified in the following sections.

3.1 LOGICAL ATS-SIP ENTITIES

3.1.1 User Agents

User Agents in an IP AGVN **SHALL** support the following services and procedures:

1. Registration
   1.1. Registration Discovery
      1.1.1. Configured registrar address (RFC 3261 [8], section 10.2.6)
      1.1.2. Using address-of-record (RFC 3261 [8], section 10.2.6)
   1.2. Adding Bindings (RFC 3261 [8], section 10.2.1)
   1.3. Removing Bindings (RFC 3261 [8], section 10.2.2)
   1.4. Refreshing Bindings (RFC 3261 [8], section 10.2.4)
   1.5. Querying Bindings (RFC 3261 [8], section 10.2.1)
   1.6. Ordering contacts (RFC 3261 [8], section 10.2.1.2)

2. Call Control
   2.1. Establishing a session
      2.1.1. UAC procedures (RFC 3261 [8], section 13.2)
      2.1.2. UAS procedures (RFC 3261 [8], section 13.3)
      2.1.3. Based on Location server messages (RFC 3261 [8], section 8.1.3.4)
   2.2. Terminating a session with BYE
      2.2.1. UAC procedures (RFC 3261 [8], section 15.1.1)
      2.2.2. UAS procedures (RFC 3261 [8], section 15.1.2)
   2.3. Cancelling a session
      2.3.1. UAC procedures (RFC 3261 [8], section 9.1)
      2.3.2. UAS procedures (RFC 3261 [8], section 9.2)

3. Querying for capabilities
   3.1. Asking for capabilities (RFC 3261 [8], section 11)
   3.2. Answering to a capabilities query (RFC 3261 [8], section 11)

3.1.2 Registrar

Registrars in an IP AGVN **SHALL** support the following services and procedures:

1. Registration
   1.1. Maintaining Bindings (RFC 3261 [8], section 10.3)
   1.2. Ordering contacts (RFC 3261 [8], section 10.2.1.2)
   1.3. Unicast Registration (RFC 3261 [8], section 10.3)

3.1.3 Proxy Server

SIP Proxy Servers in an IP AGVN **SHALL** support the following services and procedures:

1. Call Control
   1.1. Establishment of a session
      1.1.1. Statefull procedures (RFC 3261 [8], sections 16 and 8.2)
      1.1.2. Based on Location server messages (RFC 3261 [8], sections 16 and 8.2)
   1.2. Terminating a session with BYE
      1.2.1. Statefull procedures (RFC 3261 [8], sections 16 and 8.2)

1.3. Cancelling a session
 1.3.1. Statefull procedures (RFC 3261 [8], sections 16 and 8.2)

2. Querying for capabilities
 2.1. Answering to a capabilities query (RFC 3261 [8], section 11)

3.1.4 Redirect Server

Redirect Servers in an IP AGVN **SHALL** support the Redirection service as specified in RFC 3261 [8], section 8.3.

3.2 SUPPORTED REQUESTS

Logical SIP entities in an IP AGVN **SHALL** support requests (methods) indicated in Table 1.

Method	Logical SIP Entity							
	User Agents		Registrar		Proxy Server		Redirect Server	
	Support Sending	Support Receiving	Support Sending	Support Receiving	Support Sending	Support Forwarding	Support Sending	Support Receiving
INVITE (RFC 3261 [8])	m	m	x	n/a	m	m	x	m
ACK (RFC 3261 [8])	m	m	x	n/a	m	m	x	n/a
CANCEL (RFC 3261 [8])	m	m	x	n/a	m	m	x	m
BYE (RFC 3261 [8])	m	m	x	n/a	m	m	x	m
REGISTER (RFC 3261 [8])	m	n/a	x	m	x	m	x	n/a
OPTIONS (RFC 3261 [8])	m	m	x	n/a	x	m	x	m
SUBSCRIBE (RFC 3265 [10])	o	o	x	n/a	x	m	x	n/a
NOTIFY (RFC 3265 [10])	o	o	x	n/a	x	m	x	n/a
REFER (RFC 3515 [13])	m	m	x	n/a	x	m	x	n/a
INFO (RFC 2976 [7])	m	m	x	n/a	x	m	x	n/a
MESSAGE (RFC 3428 [12])	o	m	x	n/a	x	m	x	n/a

m: mandatory; o: optional; x: prohibited; n/a: not applicable

Table 1 – Supported Requests

The requirements of RFC 3261 [8], RFC 2976 [7], RFC 3265 [10], RFC 3428 [12] and RFC 3515 [13] **SHALL** apply with modifications as specified in the following sub-clauses.

3.2.1 INVITE

INVITE requests **SHALL** include an SDP body for all type of calls; nevertheless, UAS **SHOULD** support receiving INVITE requests with no SDP body.

3.3 SUPPORTED RESPONSES

Logical SIP entities in an IP AGVN **SHALL** support the reception of all 1xx Provisional Responses, 2xx Successful Responses, 3xx Redirection Responses, 4xx Request Failure Responses, 5xx Server

Error Responses and 6xx Global Failure Responses, as defined in RFC 3261 [8].

3.4 SUPPORTED HEADER FIELDS

The requirements of RFC 3261 [8], RFC 2976 [7], RFC 3265 [10], RFC 3428 [12], RFC 3515 [13], RFC 3891 [21] and RFC3911 [22] **SHALL** apply with modifications as specified in the following sub-clauses.

As indicated in RFC 3261 [8], header field values and parameter values **SHALL** be case-insensitive.

Table 2, Table 3, Table 4 and Table 5 compile and summarize the minimum set of header fields that **SHALL** be supported in requests and responses.

> ***Note 1.***
> *According to the indicated RFCs, all the mandatory header fields appear as mandatory in the tables. Besides, Content-Length, Priority and Subject header fields (indicated as optional in the RFCs) appear here as mandatory for specific requests; this is indicated by an '**m**' in bold face type.*

3.4.1 User Agent Request Headers

SIP UAs in an IP AGVN **SHALL** be capable of sending and receiving the SIP request header fields indicated in Table 2. SIP Proxy Servers in an IP AGVN **SHALL** be capable of receiving the SIP request header fields indicated in Table 2.

UA Request Header Field	Requests										
	ACK	BYE	CAN	INF	INV	MES	NOT	OPT	REF	REG	SUB
Accept	---	o	---	o	o	---	o	m	o	o	o
Allow	---	o	---	---	o	o	o	o	o	o	o
Allow-Events (RFC 3265 [10])	o	o	---	---	o	---	o	o	---	o	o
Authorization	o	o	o	o	o	o	o	o	o	o	o
Call-ID	m	m	m	m	m	m	m	m	m	m	m
Contact	o	---	---	o	m	---	m	o	m	o	m
Content-Length	**m**	**m**	**m**	**m**	**m**	**m**	**m**	**m**	o	**m**	**m**
Content-Type	*	*	---	*	*	*	*	*	*	*	*
Cseq	m	m	m	m	m	m	m	m	m	m	m
Date	o	o	o	o	o	o	o	o	o	o	o
Event (RFC 3265 [10])	---	---	---	---	---	---	m	---	---	---	m
Expires	---	---	---	o	o	o	---	---	o	o	o
From	m	m	m	m	m	m	m	m	m	m	m
In-Reply-to	---	---	---	---	o	o	---	---	---	---	---
Join (RFC 3911 [22])	---	---	---	---	o	---	---	---	---	---	---
Max-Forwards	m	m	m	m	m	m	m	m	m	m	m
MIME-Version	o	o	---	---	o	---	o	o	o	o	o
Priority	---	---	---	o	**m**	o	---	---	---	---	o
Proxy-Authorization	o	o	---	o	o	o	o	o	o	o	o
Proxy-Require	---	o	---	o	o	o	o	o	o	o	o
Record-Route	o	o	o	o	o	---	o	o	o	---	o
Refer-To (RFC 3515)	---	---	---	---	---	---	---	---	o	---	---
Replaces (RFC 3891 [21])	---	---	---	---	o	---	---	---	---	---	---
Reply-to	---	---	---	---	o	o	---	---	---	---	---
Require	---	c	---	---	c	c	o	c	c	c	o

UA Request Header Field	Requests										
	ACK	BYE	CAN	INF	INV	MES	NOT	OPT	REF	REG	SUB
Route	c	c	c	o	c	o	c	c	c	c	c
Subject	---	---	---	o	**m**	o	---	---	---	---	---
Subscription-State (RFC 3265 [10])	---	---	---	---	---	---	m	---	---	---	---
Supported	---	o	o	---	m	---	o	o	o	o	o
To	m	m	m	m	m	m	m	m	m	m	m
Via	m	m	m	m	m	m	m	m	m	m	m

c: conditional
m: mandatory
o: optional
---: not applicable (i.e. header should not be included in the request)
* : required if message body is not empty

CAN: CANCEL
INF: INFO
INV: INVITE
MES: MESSAGE
NOT: NOTIFY
OPT: OPTIONS

REF: REFER
REG: REGISTER
SUB: SUBSCRIBE

Table 2 – SIP UA Request Header Fields

3.4.2 User Agent Response Headers

The Status Code column in Table 3 indicates the status code for which a given header may be included in the response. The columns that correspond to the request methods, indicate whether a given header field may (o) or shall (m) be used in a response to that particular type of request.

SIP UAs in an IP AGVN **SHALL** be capable of sending and receiving the SIP response header fields indicated in Table 3. SIP Proxy Servers in an IP AGVN **SHALL** be capable of receiving the SIP response header fields indicated in Table 3.

UA Response Header Field	Status Code	Requests										
		ACK	BYE	CAN	INF	INV	MES	NOT	OPT	REF	REG	SUB
Accept	2xx	---	---	---	---	o	---	---	m	---	o	---
Accept	415	---	c	---	---	c	m	o	c	c	c	o
Accept-Encoding	2xx	---	---	---	---	o	---	--	m	---	o	---
Accept-Encoding	415	---	c	---	---	c	m	o	c	c	c	o
Accept-Language	2xx	---	---	---	---	o	---	---	m	---	o	---
Accept-Language	415	---	c	---	---	c	m	o	c	c	c	o
Allow	2xx	---	o	---	---	m	o	o	m	---	o	o
Allow	405	---	m	---	o	m	m	m	m	m	m	m
Allow	All except 2xx,405	---	o	---	---	o	o	o	o	o	o	o
Allow-Events (RFC 3265 [10])	2xx	o	o	---	---	o	---	o	o	---	o	o
Allow-Events (RFC 3265 [10])	489	---	---	---	---	---	---	m	---	---	---	m
Authentication-Info	2xx	---	o	---	---	o	o	o	o	o	o	o
Call-ID	All	m	m	m	m	m	m	m	m	m	m	m
Contact	1xx	---	---	---	---	o	---	o	---	---	---	o
Contact	2xx	---	---	---	---	m	---	o	o	m	o	m
Contact	3xx	---	o	---	---	o	o	m	o	---	o	m

UA Response Header Field	Status Code	Requests										
		ACK	BYE	CAN	INF	INV	MES	NOT	OPT	REF	REG	SUB
Contact	485	---	o	---	---	o	o	o	o	o	o	o
Content-Length	All	**m**	**m**	**m**	**m**	**m**	**m**	**m**	**m**	o	**m**	**m**
Content-Type	All	*	*	---	*	*	*	*	*	*	*	*
Cseq	All	m	m	m	m	m	m	m	m	m	m	m
Date	All	o	o	o	o	o	o	o	o	o	o	o
Expires	2xx	---	---	---	o	o	---	---	---	---	o	---
From	All	m	m	m	m	m	m	m	m	m	m	m
MIME-Version	All	o	o	---	---	o	---	o	o	o	o	o
Min-Expires	423	---	---	---	---	---	---	---	---	---	m	m
Proxy-Authenticate	407	---	m	---	o	m	m	m	m	m	m	m
Proxy-Authenticate	401	---	o	o	---	o	o	---	o	o	o	---
Record-Route	18x	o	o	o	---	o	---	---	o	o	---	---
Record-Route	2xx	o	o	o	o	o	---	---	o	o	---	---
Record-Route	401,484	---	---	---	---	---	---	o	---	---	---	o
Reply-To	All	---	---	---	---	o	o	---	---	---	---	---
Require	All	---	c	---	---	c	c	o	c	c	c	o
Supported	2xx	---	o	o	---	m	---	o	m	o	o	o
To	All	m	m	m	m	m	m	m	m	m	m	m
Unsupported	420	---	m	---	o	m	o	o	m	o	m	o
Via	All	m	m	m	m	m	m	m	m	m	m	m
Warning	All	---	o	o	o	o	o	o	o	o	o	o
WWW-Authenticate	401	---	m	---	o	m	m	m	m	m	m	m
WWW-Authenticate	407	---	o	---	---	o	o	---	o	o	o	---

c: conditional
m: mandatory
o: optional
---: not applicable (i.e. header should not be included in the response)
* : required if message body is not empty

CAN: CANCEL
INF: INFO
INV: INVITE
MES: MESSAGE
NOT: NOTIFY
OPT: OPTIONS

REF: REFER
REG: REGISTER
SUB: SUBSCRIBE

Table 3 – Mapping between Requests and UA Response Header Fields

3.4.3 Registrar Server Request Headers

The Registrar Server in an IP AGVN **SHALL** be capable of receiving the SIP request header fields indicated in Table 4.

REGISTRAR Request Header Field	Request
	REGISTER
Allow	o
Authorization	o
Call-ID	m
Contact	m
Content-Length	m
Content-Type	*
Cseq	m
Date	o
Expires	m
From	m

REGISTRAR Request Header Field	Request
	REGISTER
Max-Forwards	o
MIME-Version	o
Proxy-Authorization	o
Proxy-Require	o
Require	m
Route	m
Supported	o
To	m
Via	m

m: mandatory; o: optional; *: required if message body is not empty

Table 4 – REGISTRAR Request Header Fields

3.4.4 Registrar Server Response Headers

Registrar Servers in an IP AGVN **SHALL** be capable of sending the SIP response header fields indicated in Table 5.

REGISTRAR Response Header Field	Status Code	Request
		REGISTER
Allow	2xx	o
Authentication-Info	2xx	o
Call-ID	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Contact	2xx	m
Contact	3xx	m
Content-Length	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Content-Type	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	o
Cseq	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Date	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	o
Expires	2xx	o
From	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Min-Expires	423	o
MIME-Version	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	o
Proxy-Authenticate	407	o
Proxy-Authenticate	401	m
Require	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Supported	2xx	o
Unsupported	400, 401, 403, 404, 406, 407, 423	o
To	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Via	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	m
Warning	2xx, 300, 400, 401, 403, 404, 406, 407, 423, 500, 503	o
WWW-Authenticate	401	m
WWW-Authenticate	407	o

m: mandatory; o: optional

Table 5 – Mapping between Requests and REGISTRAR Response Header Fields

3.4.5 Max-Forwards

A VCS **SHOULD** provide a management means of configuring the acceptable (network dependent) Max-Forwards initial value. Nevertheless, it is **RECOMMENDED** that the initial value for the Max-

Forwards header field is less than 20.

Note 2.
The recommended initial value in RFC 3261 is 70 for the Internet.

3.4.6 Priority

The Priority header field **SHALL** take one from the values in Table 6.

In case of the Priority header field is not included in a request or takes a different value to those in Table 6, it **SHALL** be assumed as “non-urgent”.

Type of call		SIP Priority header field value
Priority	Priority Direct / Indirect Access call	emergency
Routine	Instantaneous Access call	urgent
	Tactical Direct / Indirect Access call	urgent
	Strategic Direct / Indirect Access call	normal
	General Purpose Direct / Indirect Access call	non-urgent
Monitoring	Position monitoring	non-urgent

Table 6– Priority Header Field Values

3.4.7 Subject

The Subject header field **SHALL** take one from the values in Table 7.

In case of the Subject header field takes value “Radio” or “Radio call”, the call **SHALL** be rejected. In other cases, should the Subject header field be not included in a request or take a different value to those in Table 7, it **SHALL** be assumed as “DA/IDA call”.

Type of call	SIP Subject header field value
Instantaneous Access call	IA call
Priority Direct / Indirect Access call	DA/IDA call
Routine Direct / Indirect Access call	DA/IDA call
Position monitoring	monitoring

Table 7 – Subject Header Field Values

3.5 MESSAGE BODY

SIP message bodies containing a description of the session, time and media **SHALL** be encoded in the Session Description Protocol (SDP) (RFC 4566 [25]). The SDP types and parameters indicated in Table 8 **SHALL** be supported.

SIP message bodies containing conference state data **SHALL** be encoded according to the SIP Event Package for Conference State (RFC 4575 [26]).

SIP message bodies containing presence information data **SHALL** be encoded in the Presence Information Data Format (PIDF) (RFC 3863 [20]), according to the Presence Event Package for the Session Initiation Protocol (RFC 3856 [19]).

SIP message bodies containing mid-call messages **SHALL** be encoded in the Text/Plain Format (RFC 3676 [17])

Description	SDP Types	SDP Parameters	Values
Session	Protocol version (“v=”)	<SDP version number>	• 0
	Origin (“o=”)	<username>	(Application dependent)
		<sess-id>	(Application dependent)
		<sess-version>	(Application dependent)
		<nettype>	• IN
		<addrtype>	• IP4 (in the interim) • IP6
		<unicast-address>	(Application dependent)
	Session name (“s=”)	<session name>	(Application dependent)
	Connection data (“c=”)	<nettype>	• IN
		<addrtype>	• IP4 (in the interim) • IP6
		<connection-address>	(Application dependent)
Time	Timing (“t=”)	<start-time>	(Application dependent)
		<stop-time>	(Application dependent)
Media	Media descriptions (“m=”)	<media>	• audio
		<port>	(Application dependent)
		<proto>	• RTP/AVP
		<fmt>	• 0 (for PCM-μ) • 8 (for PCM-A) • 15 (for G.728) • 18 (for G.729)
	Attributes (“a=”)	<send-receive mode>	• recvonly • sendrecv • sendonly • inactive
		rtpmap:<payload type>	• rtpmap:0 (for PCM-μ) • rtpmap:8 (for PCM-A) • rtpmap:15 (for G.728) • rtpmap:18 (for G.729)
		<encoding name>/<clock rate>	• PCMU/8000 (for PCM-μ) • PCMA/8000 (for PCM-A) • G728/8000 (for G.728) • G729/8000 (for G.729)

Table 8 – Supported SDP Types and Parameters

3.6 ADDRESS FORMAT

As specified in RFC 3261 [8], the formal syntax for a SIP and SIPS URI is:

SIP-URI = “sip:” [userinfo] host [“:” port] uri-parameters [headers]
SIPS-URI = “sips:” [userinfo] host [“:” port] uri-parameters [headers]

where <userinfo> **SHALL** be coded as:

userinfo = (ATS_telephone_number / user_role) ”@”
ATS_telephone_number = 1*DIGIT
user_role = 1* (unreserved)

Besides, it is **RECOMMENDED** that <host> be coded as:

host = atsu ”.” centre_id ”.” local_domain

atsu	=	1* (unreserved)
centre_id	=	4*ALPHA; /*ICAO identifier for a specific ATS centre
local_domain	=	1* (unreserved); /*ANSP domain

Examples:

```
sip:314002@en_route.lecm.es.ipax.net
sip:planner.zmu@en_route.lecm.es.ipax.net
```

3.7 SECURITY REQUIREMENTS

Logical SIP entities in an IP AGVN **SHALL** support HTTP Authentication with the “Digest” authentication mechanism using the following parameters:

1. digest-uri
2. nonce
3. realm
4. response
5. username

Besides, logical SIP entities in an IP AGVN **SHOULD** support the following additional Security capabilities:

1. S/MIME (Secure / Multipurpose Internet Mail Extensions)
2. TLS (Transport Layer Secure) protocol

3.8 GROUND TELEPHONE FACILITIES

3.8.1 Routine Direct/Indirect Access Call

The Direct Access (DA) facility enables a user to initiate a call by operating on a single key; whereas the Indirect Access (IDA) facility enables a user to enter a complete address on a telephone dialling keypad (or equivalent device) in order to cause a call attempt to be made to the supplied address, this is equivalent to normal dialled telephone operation.

The establishment and clearing of a Routine (Non-priority) DA/IDA call **SHALL** be handled as specified in RFC 3261 [8] and RFC 3665 [15].

The INVITE request for a Routine DA/IDA call **SHALL** include the Priority header field with value “urgent”, “normal” or “non-urgent”, as indicated in Table 6, and the Subject header field with value “DA/IDA call”.

3.8.2 Call Priority

The Priority facility is a means of attaching an indicator to a DA/IDA telephone call to show that it is "emergency" as opposed to "routine". It is intended for use when it is necessary to make a call concerning the safety of aircraft (i.e., an emergency situation).

In addition to the requirements of a Routine call, Priority DA/IDA calls imply execution of the Intrusion supplementary service as specified in sub-clause 3.8.8 below.

The INVITE request for a Priority DA/IDA call **SHALL** include the Priority header field with value “emergency” and the Subject header field with value “DA/IDA call”.

3.8.3 Instantaneous Access Call

An "Instantaneous Access call" is a call with automatic call establishment that enables audio transmission from the calling party to the called party, without the called party having to perform any acceptance action at the called-party’s terminal. Once the IA call is established, the called party can also transmit to the calling party by activating his/her IA key. The Instantaneous Access call remains while there is at least one party transmitting to the other.

In order to make an IA call, the calling party will activate the associated IA key on the IA panel at his/her terminal. The called party is informed of IA call arrival through a change in state of the corresponding IA key on the IA panel at his/her terminal.

If the called party is already involved in an active call, the IA facility **MAY** establish a monitor-type connection between the calling party and the called party.

3.8.3.1 Application

An individual IA call **MAY** only take place exclusively on a point-to-point basis between two pre-defined parties, and the 'System' **SHALL** incorporate facilities to both establish and control this so as to ensure that the integrity of the call is never compromised.

3.8.3.2 Establishment Conditions

The IA Supplementary Service (SS-IA) must be available in the VCSs involved and corresponding IA keys must be configured on both user terminals.

The called party may be free or involved in another active call (e.g. IA, Radio, DA/IDA routine or DA/IDA priority call); the actual status of the called party is irrelevant.

3.8.3.3 Audio

Once the IA call is established, if monitoring is enabled, the calling party **MAY** hear a combination of:

- transmitted and/or received audio from any active DA/IDA telephone voice calls at the called party's CWP; and/or
- transmitted and/or received audio from any other active IA calls at the called party's CWP; and/or
- transmitted and/or received audio from any active radio voice calls at the called party's CWP; and/or
- any other audio signal suitably injected in the audio path to the calling user.

Such audio combination **SHALL** be configurable according to ANSP specific requirements, for the whole system or for specific user terminals. In any case, to prevent either echoing or audio feedback, VCS systems **SHALL** remove the calling party's voice from the monitoring channel.

The called party **SHALL** hear the audio transmitted by the calling party.

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

3.8.3.4 Timing Constraints

The SS-IA **SHALL** meet the requirements of the Instantaneous Controller-Controller Voice Communication (ICCVC) which stipulates that communication be established between non-physically adjacent controllers within 1 second or less in 99% of the time.

3.8.3.5 IA Call Signalling Procedures

An IA call **SHALL** be handled as two independent sessions (A → B & B → A), each session **SHALL** be controlled by its creator. The initial session from A to B **SHALL** be controlled (released) by A and its possible response from B to A **SHALL** be controlled (released) by B.

Note 3.
From a SIP perspective, these two sessions are completely independent; there is only a dependency within the application or Graphical User Interface.

A VoIP VCS implementing SIP SS-IA **SHALL** support the offer/answer model specified in RFC 3264 [9].

3.8.3.5.1 Actions at the Calling-party User Agent

To make an IA call, the Calling-party UA **SHALL** send an INVITE request, including a Priority header field with value "urgent" and a Subject header field with value "IA call"; it **SHALL** start timer T1.

If the call fails for reasons other than those covered below, the Calling-party UA **SHALL** stop timer T1 and indicate "IA call failure" to the Calling-party.

While awaiting IA call confirmation,

a) if the calling party releases the IA call, the Calling-party UA **SHALL** send a CANCEL request and stop timer T1;

b) on receipt of a 200 (OK) final response, the Calling-party UA **SHALL** send an ACK request and establish both-way RTP media, enable Calling-party's Tx & Rx and stop timer T1; "IA-Tx: active, IA-Rx: monitoring-active" **MAY** be indicated on the appropriate IA key on the IA panel of the calling party's terminal;

Note 4.
At this point, a session has been created in which the voice path from the called party to calling party is used for monitoring Ground and Radio calls in progress at the called party's terminal.

c) on receipt of a 200 (OK) final response containing an "a=recvonly" SDP answer attribute, the Calling-party UA **SHALL** send an ACK request and establish one-way RTP media, enable calling party's Tx, stop timer T1 and enter state IA-Tx-Active; "IA-Tx: active, IA-Rx: non-active" **MAY** be indicated on the appropriate IA key on the IA panel of the calling party's terminal;

Note 5.
In this case, a session has been created without monitoring.

d) on receipt of a 180 (Ringing), 182 (Queued) or 183 (Session Progress) provisional responses, or a 4xx-6xx final response, the Calling-party UA **SHALL** indicate "IA call failure" to the calling party and stop timer T1;

e) on expiry of timer T1, the Calling-party UA **SHALL** indicate "IA call failure" to the calling party.

Once the IA call is established,

a) if the calling party releases the IA transmission, the Calling-party UA **SHALL** send a BYE request for its controlled session and disable the calling party's Tx; "IA-Tx: non-active, IA-Rx: non-active", or "IA-Tx: non-active, IA-Rx: active" **MAY** be indicated on the appropriate IA key on the IA panel of the calling party's terminal, depending on the status of the non-controlled IA session.

3.8.3.5.2 Actions at a Proxy

No special actions are required in support of SIP SS-IA.

3.8.3.5.3 Actions at the Called-party User Agent

Upon receipt of an INVITE(Priority="urgent", Subject="IA call") request, the Called-party UA **SHALL** check if there is an application dependent cause to reject the IA call;

1. Should there be a cause for rejection, the Called-party UA **SHALL** send the corresponding 4xx-6xx response; “Incoming call attempt” **MAY** be indicated on the appropriate IA key on the IA panel of the Called-party’s terminal;

2. otherwise, the Called-party UA **SHALL** immediately send a 200 (OK) response and enable the called party’s Rx; if monitoring is enabled, the Called-party UA **SHALL** establish a monitor type connection to the calling party (all Ground and Radio calls in progress at the called party’s terminal **SHALL** be monitored by the calling party), otherwise, an “a=recvonly” SDP attribute **SHALL** be included in the 200 (OK) response; “IA-Tx: non-active, IA-Rx: active” **MAY** be indicated on the appropriate IA key on the IA panel of the called party’s terminal.

Once the IA call is established,

a) on receipt of a BYE request, the Called-party UA **SHALL** send 200 (OK) final response and disable the called party’s Rx; “IA-Tx: non-active, IA-Rx: non-active”, or “IA-Tx: active, IA-Rx: non-active” **MAY** be indicated on the appropriate IA key on the IA panel of the called party’s terminal, depending on the status of the non-controlled IA session.

3.8.3.6 IA Call Parameter Values (Timers)

Timer T1 **SHALL** operate at the Calling-party UA during state IA-Awaiting-Confirmation. It **SHALL** be started on sending INVITE(Priority=“ urgent”, Subject=“IA call”) and stopped on receipt of either 180 (Ringing), 182 (Queued), 183 (Session Progress) provisional responses, or 200 (OK), 4xx-6xx final responses.

Timer T1 **SHALL** have a value of 2 seconds.

3.8.3.7 Interaction with Other ATS Supplementary Services

3.8.3.7.1 Interaction with the Call Priority Interruption Supplementary Service

An IA call **SHALL NOT** be interrupted.

3.8.3.7.2 Interaction with the Call Hold Supplementary Service

Call hold **SHALL NOT** be invoked for an IA call.

3.8.3.7.3 Interaction with the Call Transfer Supplementary Service

Call transfer **SHALL NOT** be invoked for an IA call.

3.8.3.7.4 Interaction with the Call Intrusion Supplementary Service

Intrusion **SHALL NOT** be invoked for an IA call; this means that the Priority header field included in an INVITE for an IA call **SHALL** always take value “urgent” and never “emergency”.

> ***Note 6.***
> *An IA call may monitor all active Ground (DA, IDA and IA) and Radio calls at the called position before talking; anyway, this is not an intrusion because audio is not sent to remote unwanted users.*

Besides, an IA call **SHALL NOT** be intruded by a DA/IDA Priority call.

3.8.3.8 IA Call Message Sequence Charts

The Message Sequence Charts (MSC) in figures below show the information flows between the Calling-party UA and the Called-party UA. Each information flow is named according to the corresponding message sent to or received from respective party.

Dashed lines (---) represent SIP signalling messages that are mandatory to the call scenario. The arrow indicates the direction of message flow.

Double dashed lines (===) represent media paths between network elements.

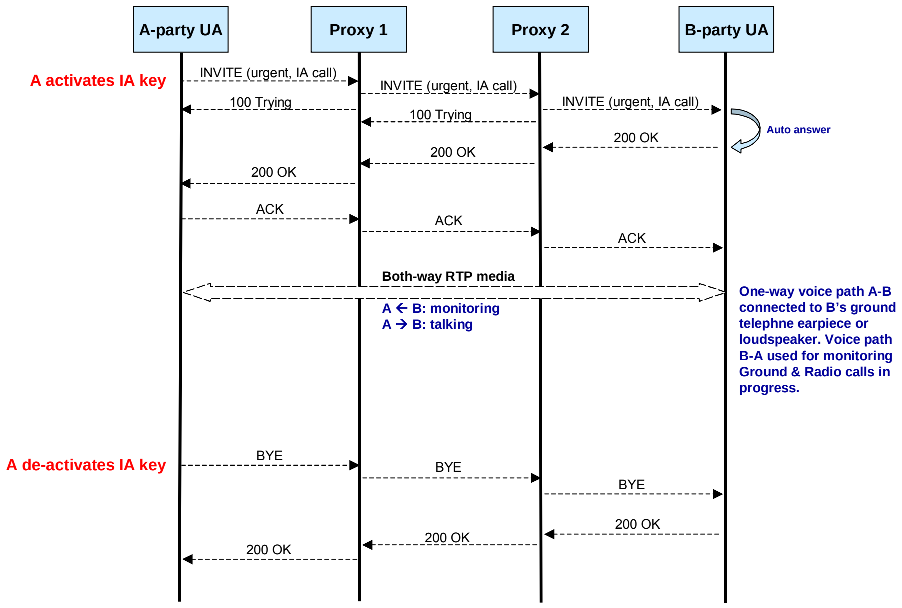


Fig. 3 – IA Call from A-party to B-party.
IA call established and A-party clears call

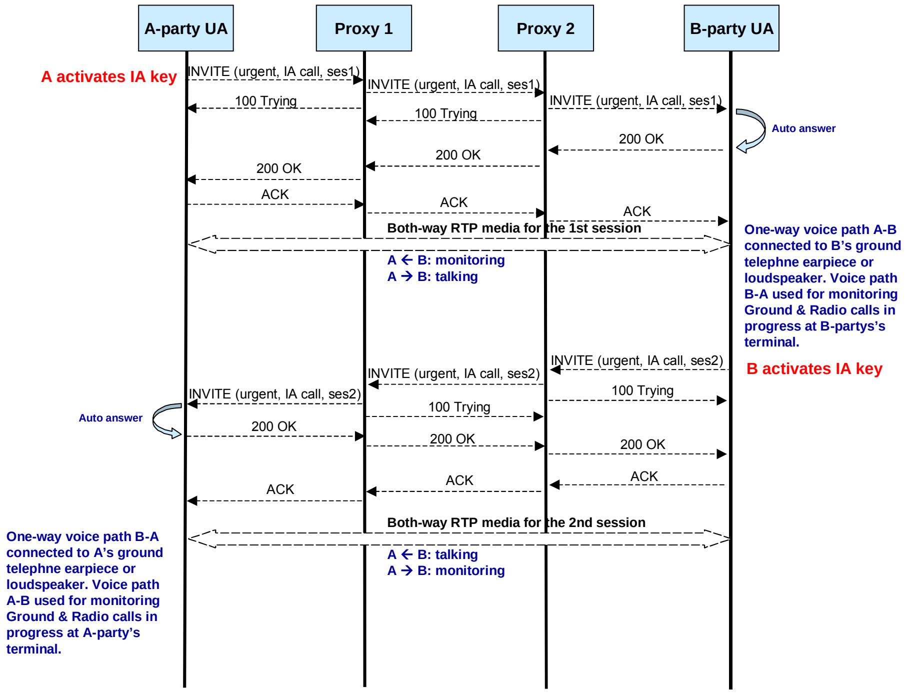


Fig. 4 – IA Call from A-party to B-party.
IA call established, B-party responds

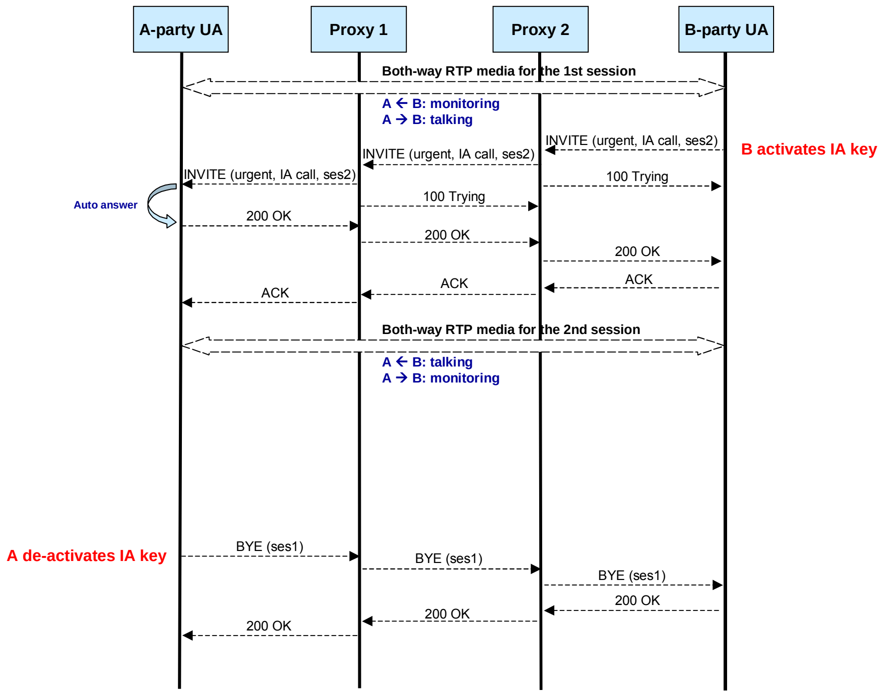


Fig. 5 – IA Call from A-party to B-party.
IA call established, B-party responds, then A-party de-activates IA key

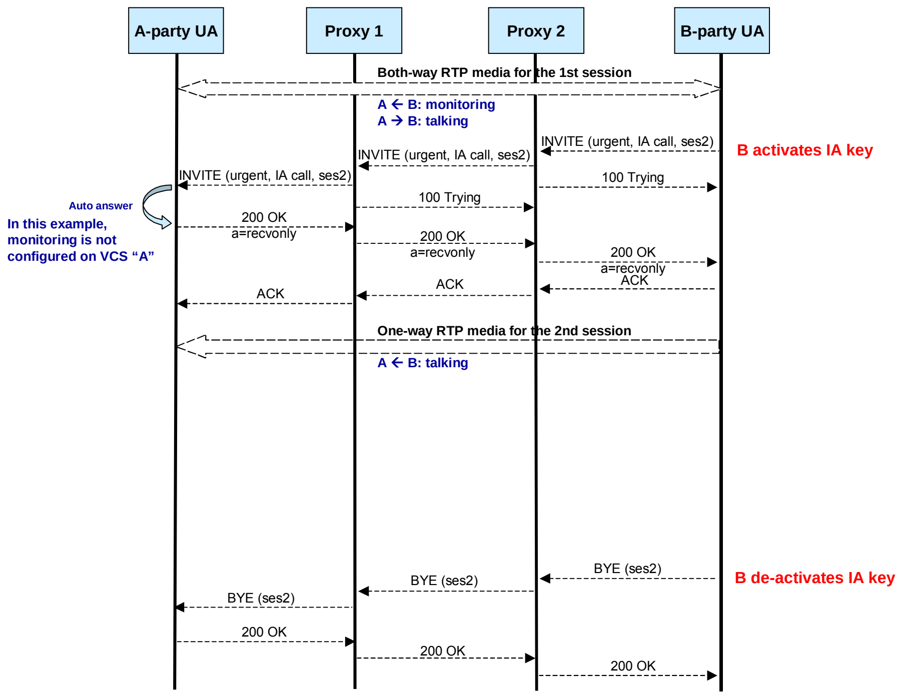


Fig. 6 – Monitoring not configured on A-party's VCS.
IA Call from A-party to B-party. IA call established, B-party responds, then B-party de-activates IA key

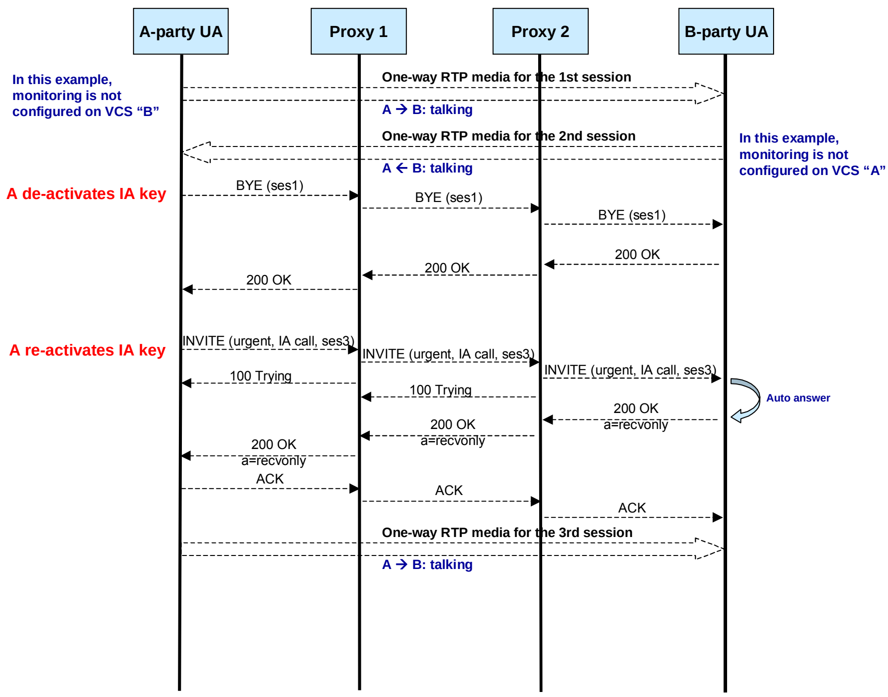


Fig. 7 – Monitoring not configured on both VCSs.
IA Call from A-party to B-party. IA call established, A → B: talking & B→ A: talking, A-party de-activates IA key and then re-activates IA Key

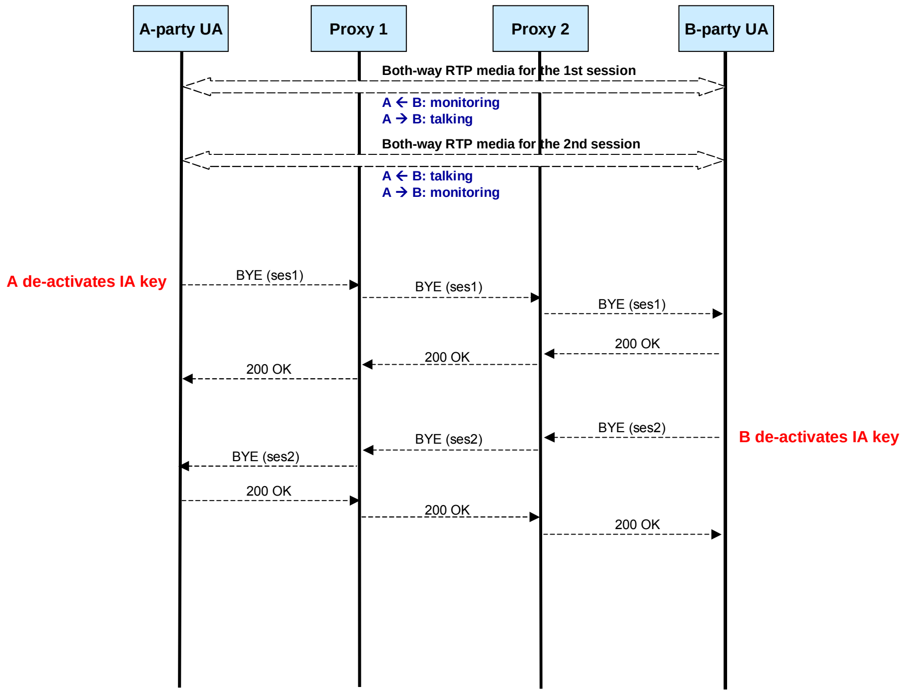


Fig. 8 – IA Call from A-party to B-party.
IA call established, A → B: talking & B→ A: talking, A-party de-activates IA key and B-party de-activates IA key

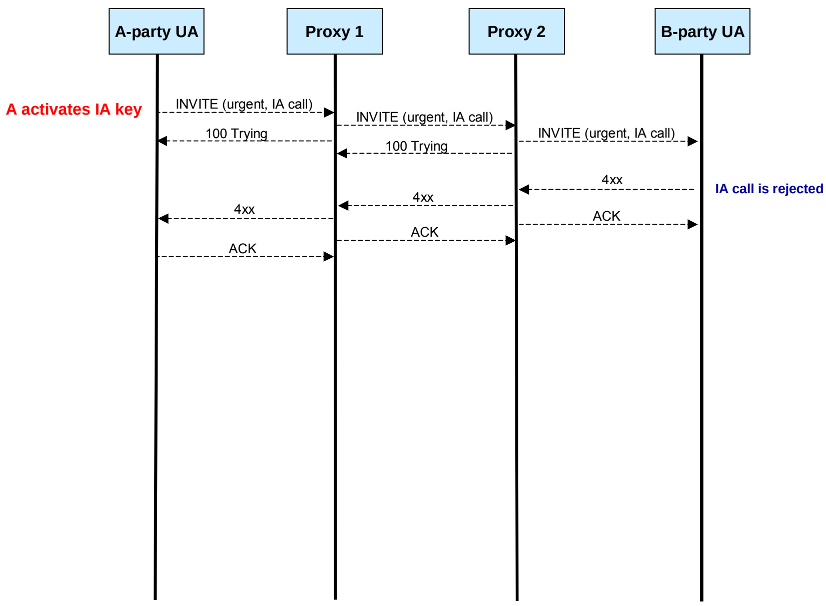


Fig. 9 – IA Call from A-party to B-party.
IA call is rejected

3.8.4 Call Hold

The Hold service allows a user to disconnect temporarily from an established call in order to carry out other telephony functions before returning to the original established call.

Call Hold **SHALL** be handled as indicated in section 2.1 ("Call Hold") and section 2.2 ("Consultation Hold") of draft-ietf-sipping-service-examples-15 [28].

3.8.5 Call Transfer

The Call Transfer service enables a user involved in an active call to establish a new call between the other user in the active call and a third party.

Call Transfer **SHALL** be handled as indicated in section 2.4 ("Transfer – Unattended") and section 2.5 ("Transfer - Attended") of draft-ietf-sipping-service-examples-15 [28].

3.8.6 "Meet Me" Conference

The Conference service enables a user to interconnect a number of parties allowing full speech facilities to all connected parties.

To create a "Meet Me" conference, users **SHALL** call at an agreed time, a pre-determined conference number.

A "Meet Me" conference **SHALL** be created according to section 5.1 of RFC 4579 ("INVITE: Joining a conference using the conference URI – dial-in") [27].

3.8.7 Broadcast Conference

The conference initiator may sequentially call other users to establish the conference. All users in an established conference **SHALL** hear a notification tone when a new participant joins the conference.

3.8.7.1 Operational Requirements

1. The Conference Entry Notification Tone feature **SHALL** be configurable (enable/disable).

2. All conferees **SHALL** hear a Conference Entry Notification Tone when a new participant joins the conference and the Conference Entry Notification Tone feature is active.

3. The initiator of a conference **SHALL** be able to eliminate participants selectively from the conference at any time (drop from conference) as long as the initiator is part of that conference.

4. Any user (including the initiator) **MAY** leave the conference at any stage and remaining users **SHALL** stay interconnected.

3.8.7.2 Interoperability General Requirements

1. A Broadcast conference **MAY** be created according to section 5.4. of RFC 4579 (INVITE: Creating a conference using the ad-hoc SIP methods) [27].

2. The initiator of a conference **SHALL**:

    a. either provide a focus role ('*isfocus*') if it hosts the conference and maintains SIP signalling relationship with each participant;

Note 7.
Note that this method does not deny the capability of leaving the conference, letting the conference 'alive'. In that case, the capability of establishing a new conference depends on internal (VCS, Position) capabilities.

b. either invite a supporting conference URI (focus), possibly selected through a Conference factory, that will host the conference and maintain SIP signalling relationship with each participant.

Note 8.
Note that all possible conference IDs must be known in advance.

3. The focus (‘*isfocus*’) is responsible for the conference state distribution and **SHOULD** support the RFC 4575 “Event Package for Conference State” [26].

4. Each conference aware participant (supporting *isfocus*) **SHOULD** subscribe to the conference URI (*SUBSCRIBE/NOTIFY*) [10].

5. A conference device that occupies the focus role **SHALL** continue hosting the conference even if the conference degrades to a two party communication.

6. On receipt of a notification (NOTIFY request) of a new participant joining the conference, each conferee UA **MAY** display the participants name on a user interface.

7. A conference device that occupies the focus role **SHALL** generate the “Conference Entry Notification” audible tone if a new participant is joining the conference, whenever this tone feature is enabled by configuration.

3.8.7.3 Execution of a Conference

This section focuses on the initial move from a simple two-party connection to a conference and in addition the adding of further parties. Conference creation is based on SIP ad-hoc methods according to RFC 4579 [27].

Note 9.
Please note that conference factory and focus are logical roles.

The execution of a conference **SHALL** be based on RFC 3891 (“Replaces Header”) [21], RFC 3515 (“Refer Method”) [13], RFC 4575 (“Event Package for Conference State”) [26] and RFC 4579 (“Call Control - Conferencing for User Agents”) [27].

3.8.7.3.1 Conference Factory

The conference factory is a possible solution to maintain conference URIs dynamically. The Conference factory is not required if the conference URI is known at the SIP UA (for example by parameterization) or if the SIP UA itself has the capability to behave as a focus.

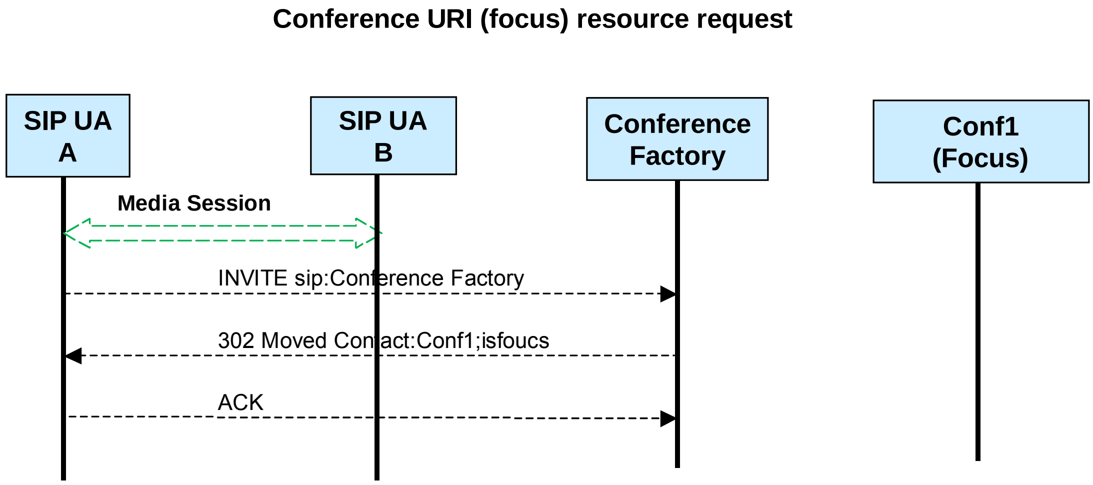


Fig. 10– Conference URI (focus) resource request

3.8.7.3.2 Migration to conference

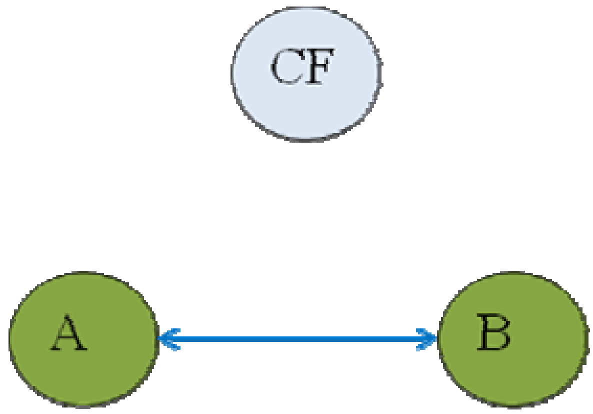


Fig. 11 – Session between UA A and B, CF not involved

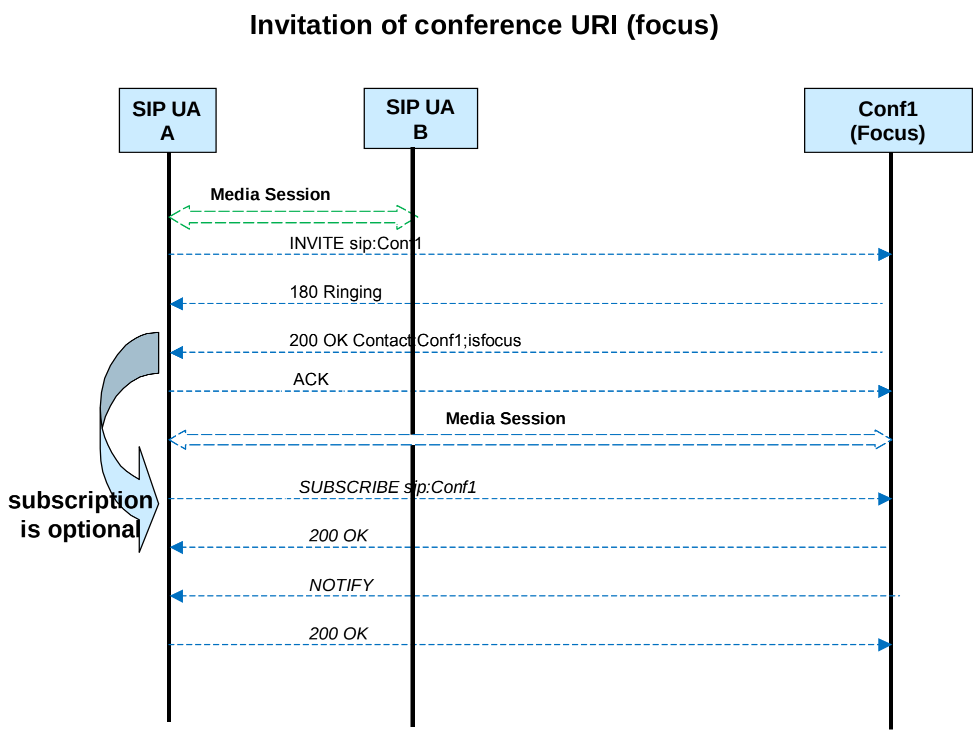


Fig. 12 – Message flow, UA A requests focus

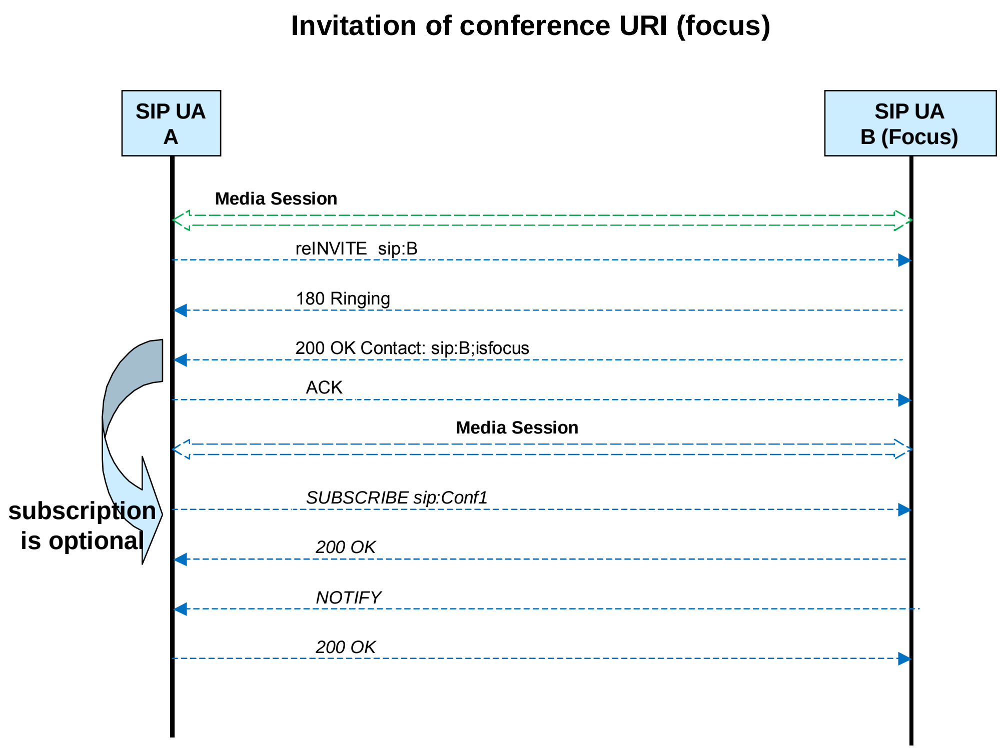


Fig. 13 – Session between UA A and UA B, B is conference focus (CF)

Invitation of conference URI (focus=initiator)

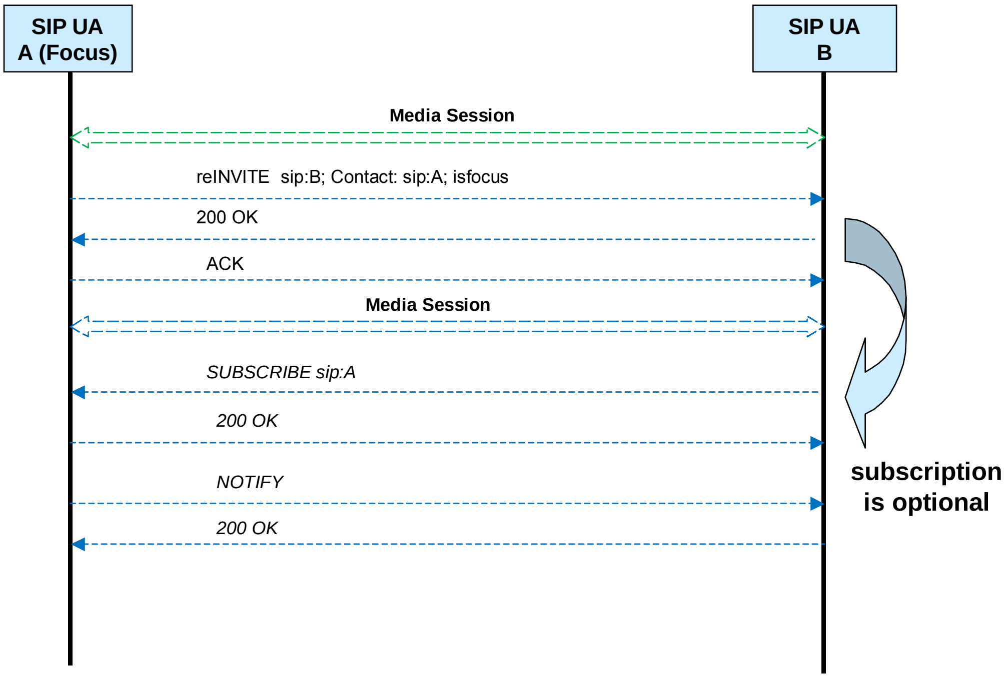


Fig. 14– Message flow, focus hosted by UA B

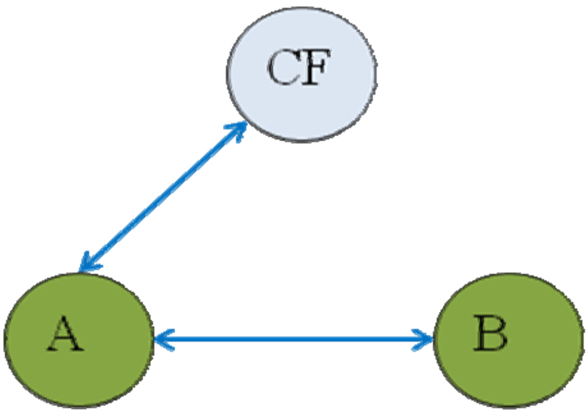


Fig. 15 –UA A has requested focus

Replacement of partner connection

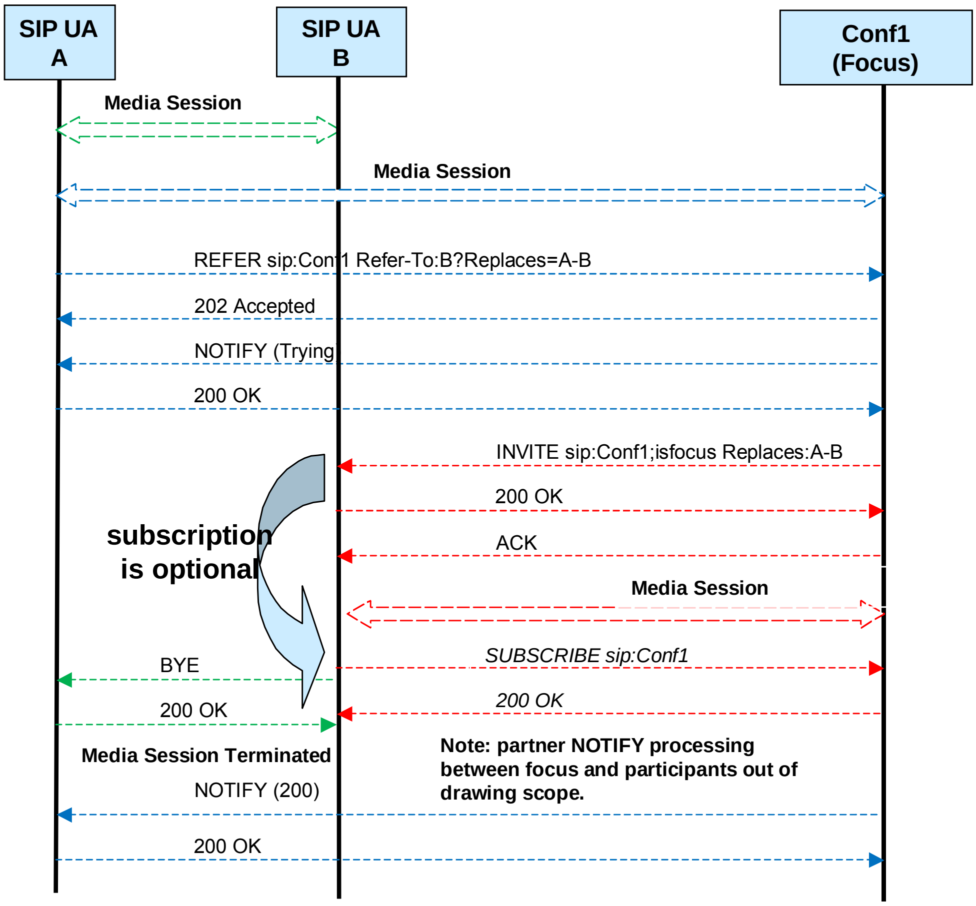


Fig. 16 – Message flow, UA A requests focus to invite B

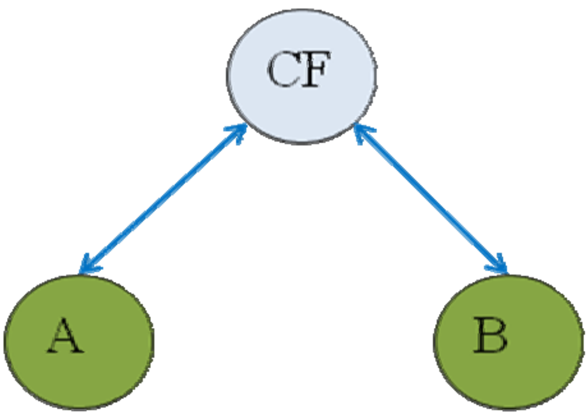


Fig. 17 – UA A and B have conference at focus

3.8.7.3.3 Adding participants

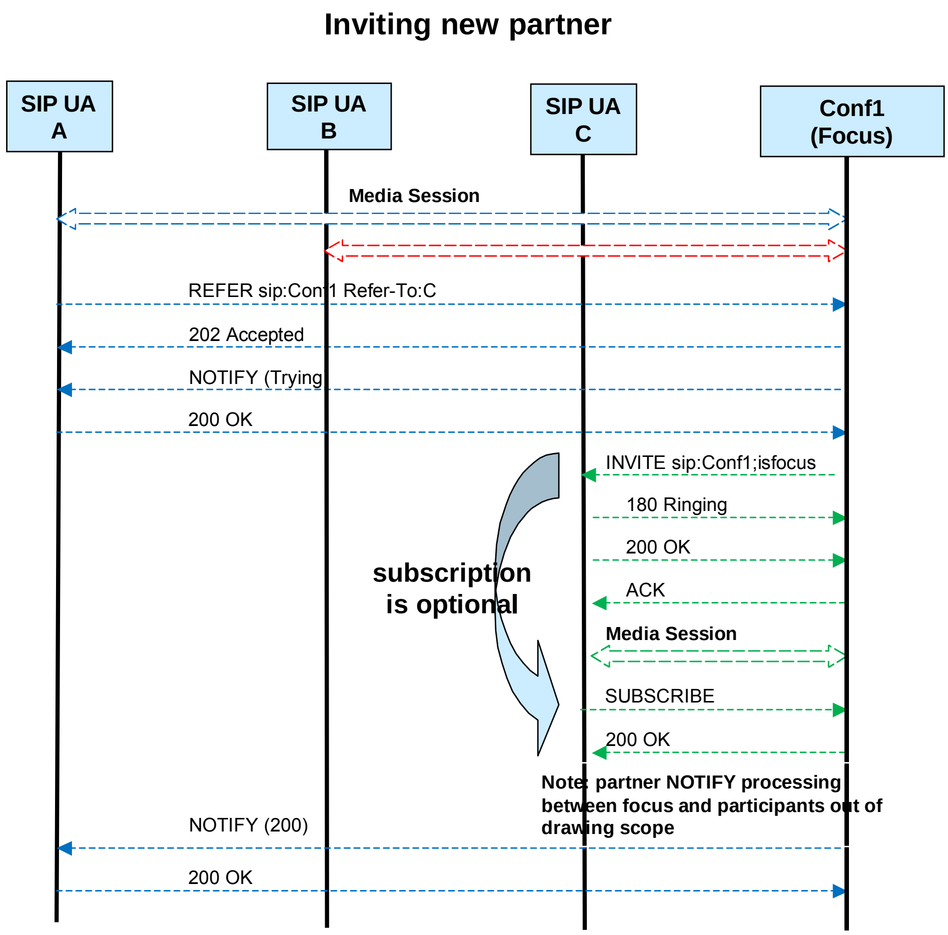


Fig. 18 - Message flow, UA A requests focus to invite C

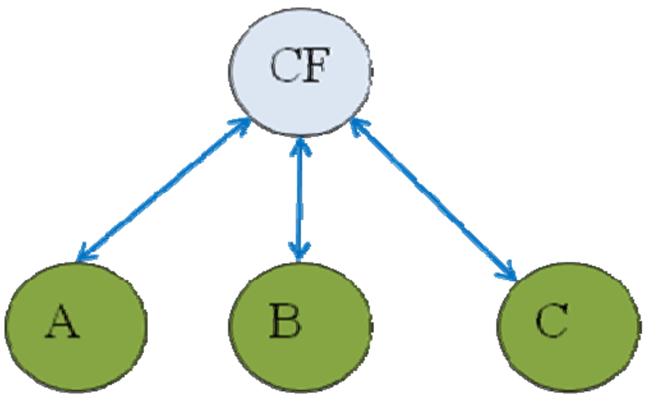


Fig. 19 – Three party conference with focus

Note 10.

To invite another partner (other SIP session, e.g. A-E) Fig. 16 "Replace partner connection" applies.

If the focus role is played by a participant of the conference, e.g. SIP UA X, the same method applies with SIP UA X instead of Focus(Conf1).

3.8.8 Intrusion

The Call Intrusion service is related to Priority DA/IDA call handling, as it is the effect of a Priority call at a Busy Called UA.

In the event that the Calling user has made a Priority call but encounters the Called user busy, Intrusion **SHOULD** take place automatically. If permitted, upon Intrusion all parties **SHALL** be connected together in conference. Before the Intrusion occurs a visual and/or audible intrusion notification **SHALL** be given to users.

Besides, according to ED-136:

a) The Called user applies discretion whether or not to release resources to permit the Priority Call to be answered; the Called user may do this by either clearing a call already in progress or by placing a call on hold.

b) In the event that the Called user releases resources, the Priority Call **SHALL** be either answered automatically or presented as a Priority Call.

c) In the event that the Called user does not release resources within the pre-defined time interval (T1) the Calling user **SHALL** be connected in telephone conference.

It **SHOULD** be possible for any user to be protected against intrusion by other users. This protection **SHALL** be selectable individually on a user-by-user basis. In the case that intrusion is not permitted, the call **SHALL** still be presented at the CWP.

An unwanted user of a call intrusion (the user other than the wanted user in the established call that is to be intruded) with call protection **SHALL** be unable to prevent a call intrusion.

3.8.8.1 Interoperability General Requirements

For the Call Intrusion service the following requirements **SHALL** be fulfilled:

1. Every SIP end **SHOULD** act as focus (IETF RFC 3840 [18]) for a conference and its URI **SHOULD** be a conference URI, therefore:

   - Once a Priority call is connected, the calling party (SIP End User Agent or Gateway) **MAY** send a SUBSCRIBE message (IETF RFC 3265 [10]) to request notification from Called User about the users in the "conference session".

2. Call Intrusion Protection Level (CIPL) of the Unwanted User (the user other than the wanted user in the established call that is to be intruded) **SHALL** be assumed as off, that is, Call Intrusion is allowed. Call intrusion **SHALL** happen unless it is not allowed by the Wanted User or the established active call is a Priority call (Priority header field = "emergency").

3. Dialogue Package (IETF RFC 4235 [23]):

   - All SIP user agents **SHOULD** be able to use a dialogue package allowing them to transmit notification of a change in their state to nominated subscribers;

   - The Allow-Events header field in a SUBSCRIBE method **SHALL** indicate "dialog", the Event header field **SHALL** indicate "dialog", and the Expires header field **SHOULD** be "3600" (1 hour);

   - Dialogue info sent by the User Agent **SHOULD** define: state = "partial", entity = user URI;

   - Dialogue id **SHOULD** also include optional attributes: call-id, local-tag, remote-tag, for each change in state;

   - Possible States include: "Trying", "Proceeding", "Early", "Confirmed" and "Terminated";

- For a call intrusion to occur, the callee **SHALL** be in a “Confirmed” state (Active). The “Confirmed” state and the “Terminated” state only **SHOULD** therefore be indicated in dialogue sent by a SIP User Agent; “Terminated” implies no active call in progress; “Confirmed” implies an active call in progress.

4. Use **SHALL** be made of the INFO method (RFC 2976 [7]) (Content-Type: text/plain) to inform users and gateways of status of the intrusion (“Intrusion in progress”, “Intrusion completed”):

   a) Visual and/or audible intrusion notifications **SHALL** be indicated to Unwanted party/parties upon receipt of a INFO request (RFC2976 [7]) containing a message body (Content-Type: text/plain) with the following content: “Intrusion in progress” or “Intrusion completed” within a SIP dialog.

   b) Visual and/or audible notifications **SHALL** be indicated to Calling (Served) user via the provisional responses 182 (Queued) and 183 (Intrusion in progress), as according to RFC3261 (7.2 Responses) [8], implementations **MAY** choose other text for the Reason-Phrase.

3.8.8.2 Execution of Priority Call Intrusion

When Priority call is invoked by the Calling (Served) user, the Calling UA **SHALL** send an INVITE request with a Priority header field defined as “emergency” to distinguish it from routine DA/IDA call (with Priority header field defined as “urgent”, “normal” or “non-urgent”).

If Intrusion is allowed (i.e. CIPL off) by the Wanted user (i.e. the called user in the intruding call) and the Wanted user is busy, the Wanted UA **SHALL** reply with a 182 (Queued) response and start timer T1 ($0 \le T1 \le$ ANSP dependent value).

> ***Note 11.***
> *The Intrusion warning period ‘T1’ should be configurable to meet requirements in ED-136.*

If the Wanted user releases resources before T1 elapses, the Wanted UA **SHALL** send a 200 (OK) final response to the INVITE (emergency) either automatically or once the Priority Call is manually answered (ANSP dependent).

Should T1 elapse, the Wanted UA **SHALL** send a reINVITE request to the Unwanted UA(s) (UA other than the Wanted UA in the established call that is to be intruded) and a 183 (“Intrusion in progress”) provisional response to the Calling UA.

In case of T1 = 0 (as configured by a specific ANSP), on receiving the INVITE (emergency) request, the Wanted UA **SHALL NOT** send the 182 (Queued) provisional response and proceed directly as specified in the former paragraph.

On receiving the reINVITE request, the Unwanted UA **SHALL** send a 200 (OK) final response to the previous reINVITE.

Upon receipt of the 200 (OK) response, the Wanted UA **SHALL** send the corresponding ACK and an INFO request containing a Text/Plain body indicating “Intrusion in progress” to the Unwanted UA. Besides, the Wanted UA **SHALL** send a 200 (OK) final response to the Calling UA, to allow the Calling user to enter the conference media session. The Wanted UA **SHALL** then act as the focus for the conference (use **SHALL** be made of the “isfocus” feature, defined in IETF RFC 3840 [18], to create a conference media session).

Once the intrusion is effective, Served and Unwanted UAs **MAY** send a dialog subscription (SUBSCRIBE request) to the Wanted UA; then it **SHALL** reply with a notification (NOTIFY request) defining all parties in the conference media session.

If the Unwanted User leaves the conference, the Wanted UA **SHALL** send an INFO request containing a Text/Plain body indicating “Intrusion completed” and a reINVITE request (it’s not focus anymore) to the Served UA.

If the Served user leaves the conference, the Wanted UA **SHALL** send a reINVITE request (it's not focus anymore) to the Unwanted UA.

If intrusion is forbidden by the Wanted UA (i.e. CIPL on), the Wanted UA user **SHALL** reply with a 180 (Ringing) response; Wanted and Unwanted users remain connected. The call is displayed at the user's terminal as a Priority Call and can be manually answered.

3.8.8.3 Message Sequence Charts

The figures below show some typical message sequences that can occur for the intrusion interworking between SIP ends. Each information flow in a Message Sequence Chart (MSC) is named according to the corresponding message sent to or received from a peer UA.

Dashed lines (---) represent signalling messages that are mandatory to the call scenario. The arrow indicates the direction of message flow.

Double dashed lines (===) represent media paths between network elements.

Priority call answered before a predefined time interval

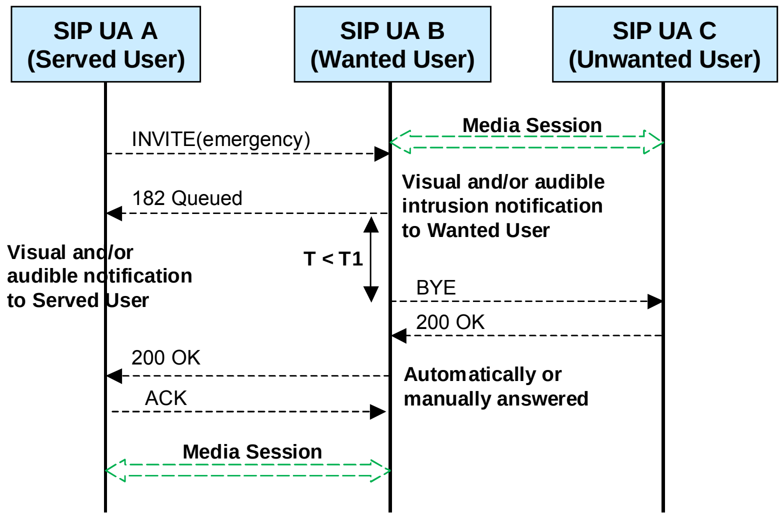


Fig. 20 – Priority call answered after releasing resources

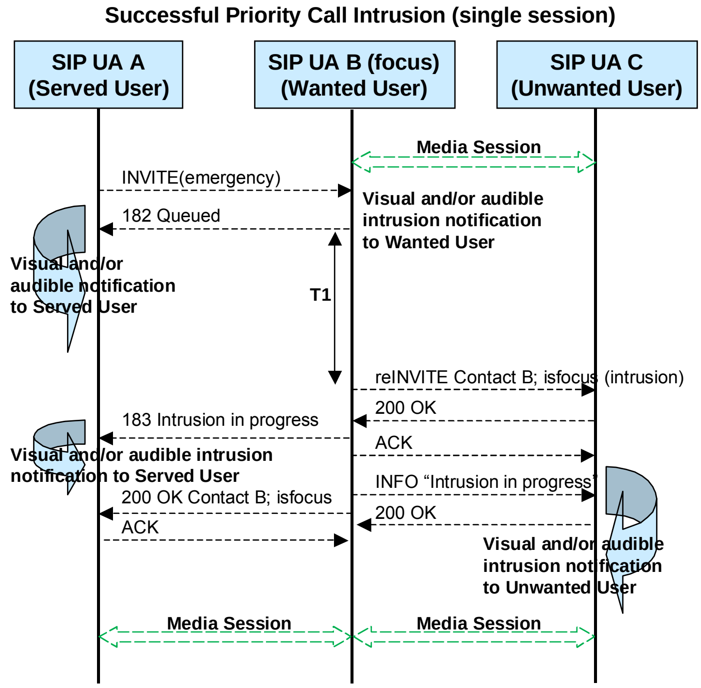


Fig. 21 – Priority call intrusion to a single call

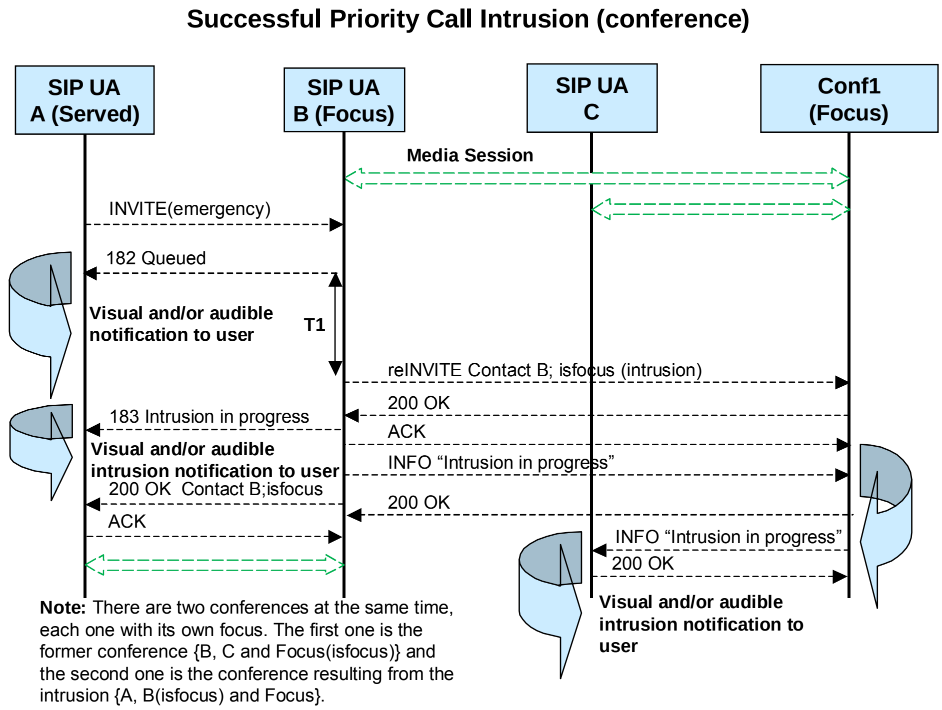


Fig. 22 – Intrusion to Conference

Note 12.
Please, note that there is no difference between single session and conference intrusion processes.

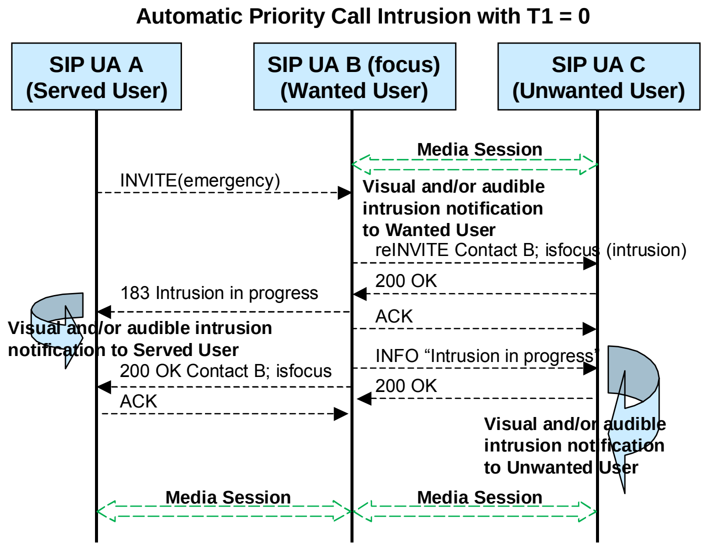


Fig. 23 – Automatic Priority call intrusion with T1 = 0

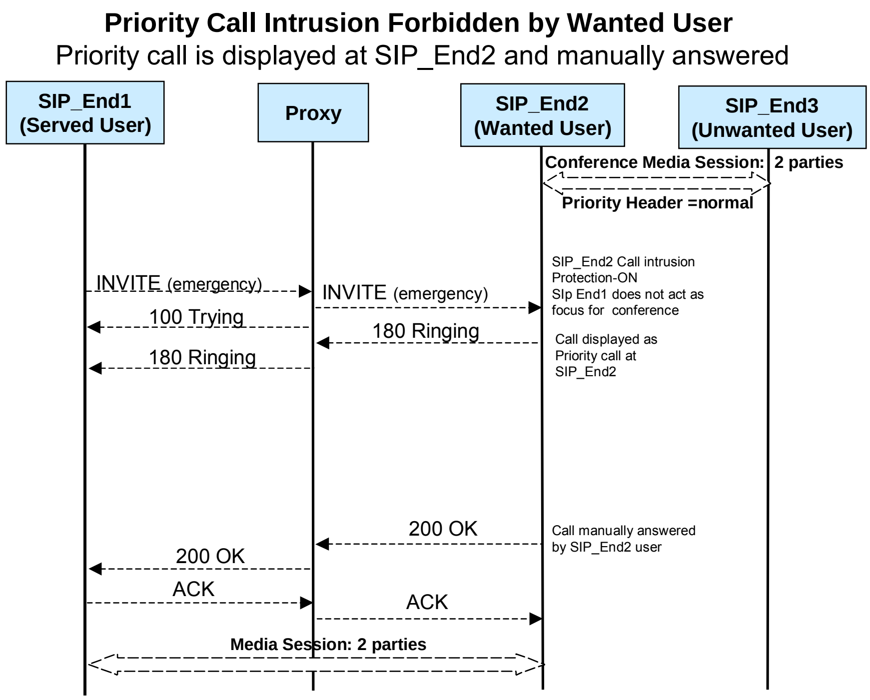


Fig. 24 – Priority Call Intrusion Forbidden by Wanted User

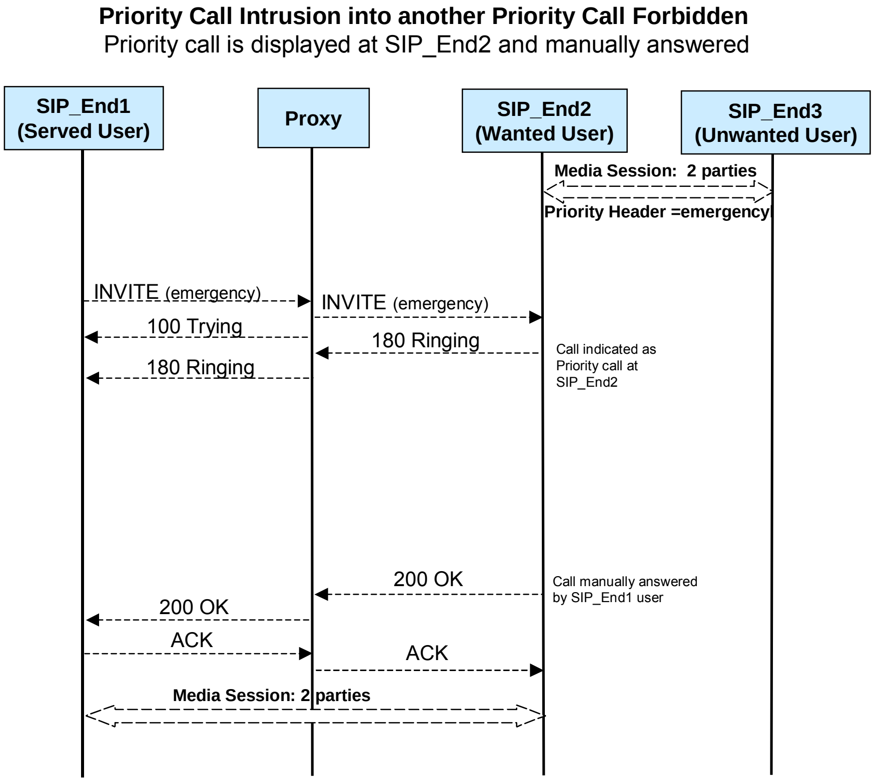


Fig. 25 – Priority Call Intrusion into another Priority Call Forbidden

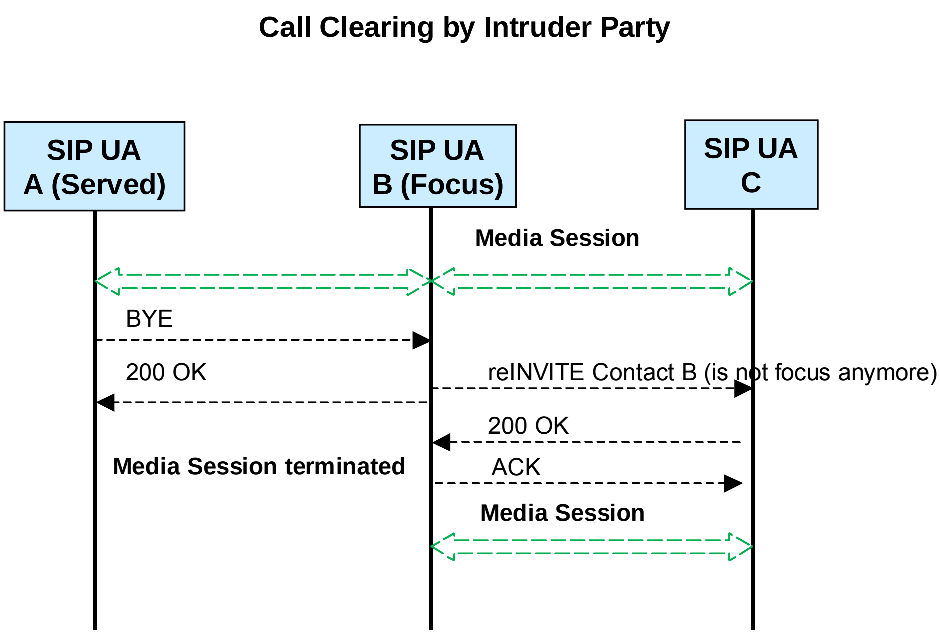


Fig. 26 – Call Clearing by Intruder Party

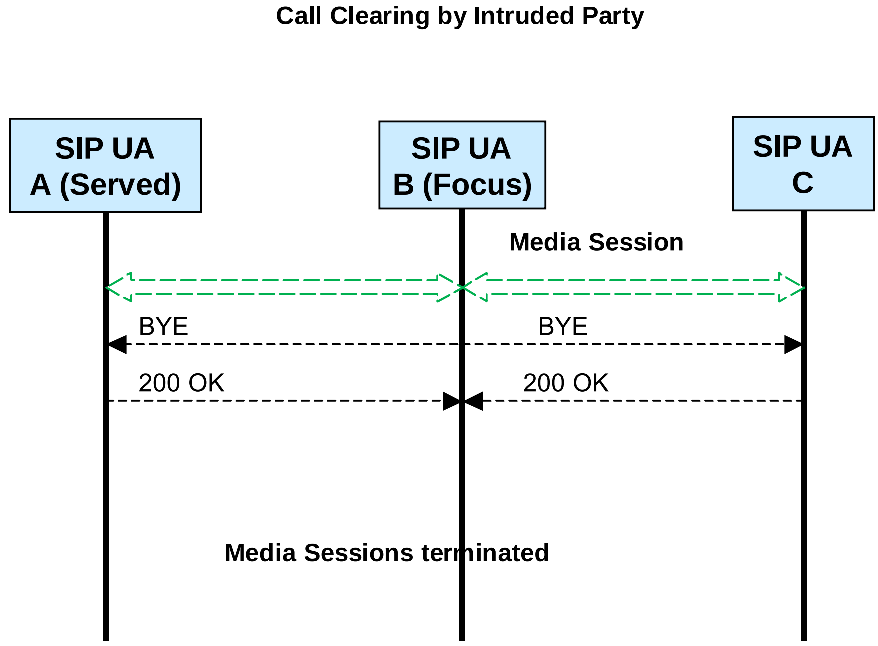


Fig. 27 – Call Clearing by Intruded Party

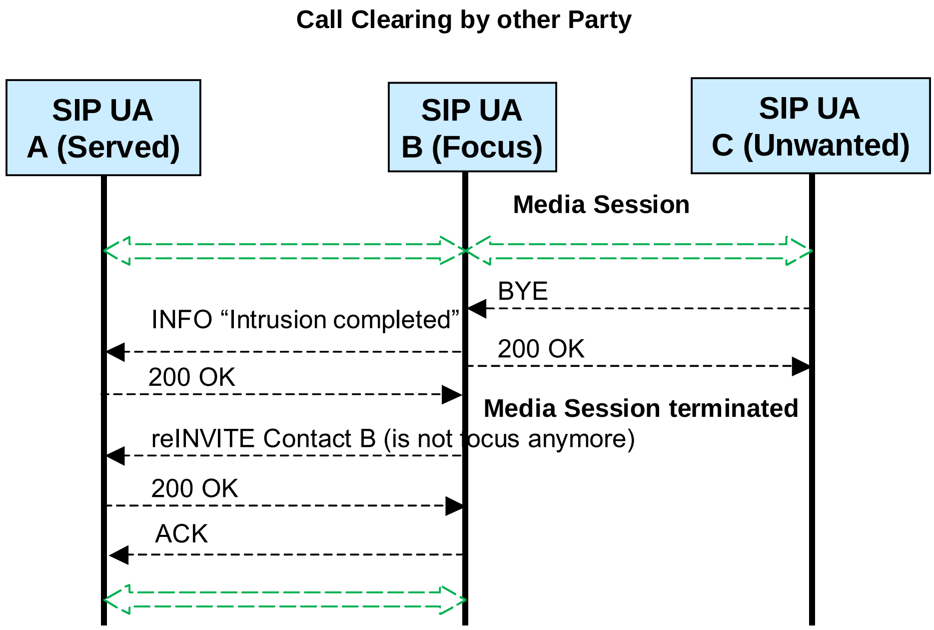


Fig. 28 – Call Clearing by Other ('Unwanted') Party

Call clearing request by a conference focus

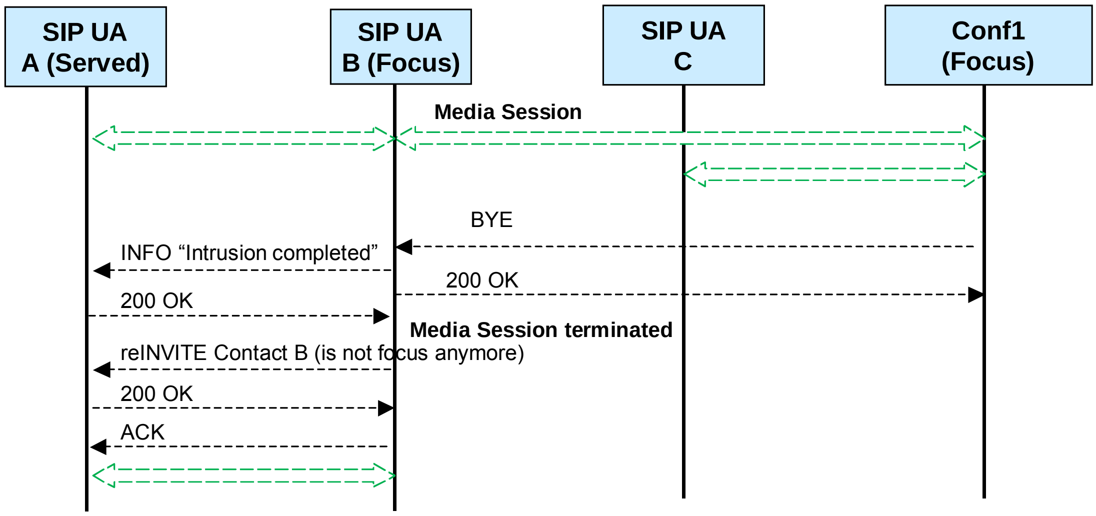


Fig. 29 – Call Clearing Request by a Conference Focus

3.8.9 Position Monitoring

The Position Monitoring service enables a user to hear any active voice call at other user position; the served (monitoring) user hears audio transmitted and received by the monitored position.

> ***Note 13.***
> *This service has to be understood as an independent facility that has nothing to do with the audio monitoring channel present in Instantaneous Access calls.*

Position monitoring **SHOULD** be based on the following principle: According to RFC 3911 [22], a call monitoring is a Join operation; the monitoring UA sends a Join (INVITE with Join header) to the dialog it wants to listen to. It is able to discover the dialog via the dialog state (RFC 4235 [23]) on the monitored UA. The monitoring UA sends SDP in the INVITE which indicates receive only media. Therefore, Position monitoring can be envisaged as simultaneous call monitoring of active calls at the monitored position.

> ***Note 14.***
> *According to this method, the served user may listen to all active calls at the monitored position or freely select the calls to listen to, if required.*

Position monitoring **SHOULD** be handled according to the following requirements:

1. To establish the monitoring session, the monitoring position **SHALL** send a SUBCRIBE request (RFC 3265 [10]) for a dialog package without any dialog identifiers (RFC 4235 [23]) to the position that is going to be monitored.

2. Upon receipt of the corresponding NOTIFY request(s) (RFC 3265 [10], RFC 4235 [23]) from the monitored position, the monitoring position **SHALL** send an INVITE request, including Priority header field with value "non-urgent", Subject header field with value "monitoring" and Join header field (RFC 3911 [22]), for each of the dialogues that were notified; these INVITE requests **SHALL** contain an "a=recvonly" SDP offer attribute.

3. Once the monitoring is established, the monitored audio **SHALL** be dynamically updated through subsequent notifications, corresponding Join operations and/or termination of dialogues.

4. To terminate a Position monitoring session, the monitoring position **SHALL** send a BYE request to

the monitored position for each of the dialogues being listened.

3.8.10 Presence Information

Presence information can be thought of as the state of a user or device at a particular instant. Within the ATS environment, presence information can be used to indicate if a particular remote CWP is operational or not.

Note 15.
Today an ATC supervisor or technical staff may only be informed of a communication problem when either automatic period test calls have failed or a controller can no longer make calls to a remote CWP.

Using Presence information, any user working within an ATS unit could be informed in real time of the operational status of every CWP in their neighbouring ATS units. If a particular CWP was closed down, a message informing them of the fact could be sent.

It would involve the user agent or server (called "watcher") making a subscription (long term relationship) with the Presence Agent located within all neighbouring ATS units.

Note 16.
The Presence Agent is an application running on a Server (i.e. Presence Server or Proxy Server), capable of identifying the presence status of all the registered CWPs in that domain.

The Subscription would request presence information changes to the relevant CWPs in that ATS unit. Once a subscription was made, every time a CWP was closed or opened, notification would be sent to the subscriber ("watcher"). This could lead to improved safety as an ATS unit would know the present status of the CWPs in all its neighbouring ATS units.

Presence information **SHALL** be handled as defined by the Presence Event Package for the Session Initiation Protocol (SIP) (RFC 3856, [19]).

3.8.11 Detection of Link Connection Loss

The system **SHALL** check if telephone call setups are possible to all ATSU locations that have been configured using the OPTIONS method (RFC 3261, [8]) to ping the corresponding VCS servers or gateways.

3.9 AUDIBLE TONES

A SIP UA **SHALL** be capable of generating the tones indicated in Table 9.

Note 17.
Tone generation should however be configurable for the whole system or for specific user terminals, since some ANSPs would prefer to display messages instead of presenting audio tones to the user.

Audible Tone	Locally generated	Sent over the IP Network	Purpose	Tone generated upon receipt of:
Ringing	Yes	No	To indicate successful call establishment and prior to call acceptance by the called user.	• 180 (Ringing) • 182 (Queued) • 183 (Session progress)
Terminal busy	Yes	No	To indicate that all available voice paths to a called user are occupied.	• 480 Temporarily Unavailable • 486 Busy Here • 600 Busy Everywhere • 603 Decline
Congestion	Yes	No	To indicate that a call cannot be completed to a called user due to appropriate inter-VCS links being occupied or otherwise unavailable.	• 503 Service Unavailable

Audible Tone	Locally generated	Sent over the IP Network	Purpose	Tone generated upon receipt of:
Number Unobtainable	Yes	No	To indicate that a terminal is "out of service" or the called user address is unassigned.	• 400 Bad Request • 401 Unauthorized • 403 Forbidden • 404 Not Found • 405 Method Not Allowed • 406 Not Acceptable • 407 Proxy Authentication Required • 408 Request Timeout • 410 Gone • 413 Request Entity Too Large • 414 Request URI Too Long • 415 Unsupported Media Type • 416 Unsupported URI Scheme • 420 Bad Extension • 421 Extension Required • 423 Interval Too Brief • 481 Call Leg/Transaction Does Not Exist • 482 Loop Detected • 483 Too Many Hops • 484 Address Incomplete • 485 Ambiguous • 488 Not Acceptable Here • 489 Bad Event • 491 Request Pending • 493 Undecipherable • 500 Server Internal Error • 501 Not Implemented • 502 Bad Gateway • 504 Server Time-out • 505 Version Not Supported • 513 Message Too Large • 604 Does Not Exist Anywhere • 606 Not Acceptable
Conference notification	No	Yes	To indicate that a new participant is joining the conference.	• N. A. (as it is generated by the conference device that occupies the focus role)
Intrusion warning	Yes	No	Injected into the voice path to warn the Unwanted User of the imminent priority conferencing of an established call.	• A Priority (emergency) call, as long as the intrusion is permitted at the Wanted User CWP
Interrupt warning	Yes	No	Injected into the voice path to warn a user of the imminent priority interruption of an established call.	• A Priority (emergency) call that has to interrupt an established Routine call • INFO (“Intrusion in progress”) request

Table 9 – Audible Tones Generated by SIP Ends

CHAPTER 4

SIGNALLING INTERWORKING BETWEEN SIP AND ATS-R2

4.1 BACKGROUND AND ARCHITECTURE

This chapter specifies signalling interworking between SIP and ATS-R2 in support of basic services as well as ATS supplementary services within an Air Traffic Services Ground Voice Network (AGVN).

ATS-R2 is a Multi-Frequency Compelled inband signalling system adapted for Air Traffic Services networks from the “ITU-T Recommendations Q.400 to Q.490” defining the ITU-T R2 signalling system. It operates in an analogue link between two VCSs and controls call establishment and call clearing on the inter-VCS link. ATS-R2 is specified in the document entitled “ATS R2 and ATS No.5 Signalling Protocol Specifications” [29].

Interworking between ATS-R2 and SIP permits a call originating at a user of a circuit-switched AGVN to terminate at a user of an IP AGVN, or a call originating at a user of an IP AGVN to terminate at a user of a circuit-switched AGVN.

Interworking between a circuit-switched AGVN employing ATS-R2 and a public IP network employing SIP is outside the scope of this specification. However, the functionality specified in this document is in principle applicable to such a scenario when deployed in conjunction with other relevant functionality (e.g., number translation, security functions, etc.).

This specification is applicable to any interworking unit that can act as a gateway between a circuit-switched AGVN employing ATS-R2 and an IP AGVN employing SIP.

ATS-R2 provides a means for establishing and clearing calls that originate and terminate on different VCSs. A call can be routed over a single inter-VCS link connecting the originating and terminating VCS, or over several inter-VCS links in series with switching at intermediate VCSs known as transit VCSs. A call can originate or terminate in another network, in which case it enters or leaves the AGVN environment through a gateway VCS. Parties are identified by numbers, in accordance with a closed numbering plan.

With the aim of exploiting IP to migrate progressively parts of the AGVN network to IP using SIP, SIP equipment in the form of SIP User Agent interfaces, SIP Proxy servers, DNS servers, etc. may be used. The new SIP environment **SHALL** also need to interwork with the ATS-R2-based AGVN in order to support calls originating in one environment and terminating in the other. Interworking is achieved through a gateway.

Another way of migrating is to use an IP network to interconnect two parts of a circuit-switched AGVN and encapsulate ATS-R2 signalling in RTP frames for calls between the two parts of the circuit-switched AGVN. This is outside the scope of this specification.

This document specifies signalling protocol interworking aspects of a gateway between a circuit-switched AGVN employing ATS-R2 signalling and an IP AGVN employing SIP signalling. The gateway appears as a VCS to other VCSs in the circuit-switched network. The gateway appears as a SIP endpoint to other SIP entities in the IP network. Fig. 30 shows the Interconnection Diagram.

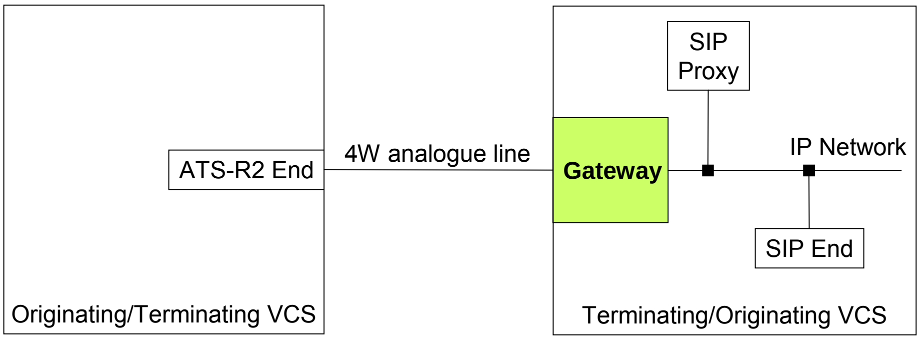


Fig. 30 – ATS-R2 / SIP Interconnection Diagram

In addition to the signalling interworking functionality specified in this document, it is assumed that the gateway also includes the following functionality:

- one or more physical interfaces on the circuit-switched network side supporting one or more inter-VCS links;

- one or more physical interfaces on the IP network side supporting, through layer 1 and layer 2 protocols, IP as the network layer protocol and UDP (RFC 768) [1] and TCP (RFC 793) [3] as transport layer protocols, these being used for the transport of SIP signalling messages and, in the case of UDP, also for media information;

- the support of TLS (RFC 4346 [24]) as additional transport layer secure protocol on the IP network side, this being used for the transport of SIP signalling messages; and

- a means of transferring media information in each direction between the circuit-switched network and the IP network, including as a minimum packetization of media information sent to the IP network and de-packetization of media information received from the IP network.

The protocol model relevant to signalling interworking functionality of a gateway is shown in Fig. 31.

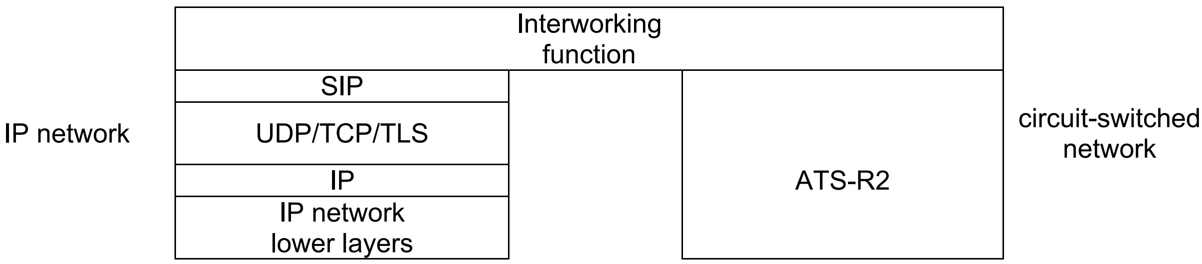


Fig. 31 – SIP / ATS-R2 Protocol Model

In Fig. 31, the SIP box represents SIP syntax and encoding, the SIP transport layer and the SIP transaction layer. The Interworking function includes SIP Transaction User (TU) functionality.

The gateway maps received ATS-R2 signals, where appropriate, to SIP messages and vice versa and maintains an association between an ATS-R2 call and a SIP dialog.

A call from ATS-R2 to SIP is initiated when an ATS-R2 Seizing line signal and a number of ATS-R2 Register signals for the digits sequence comprising called party number, call priority and calling party number arrive at the gateway. The gateway then sends a SIP INVITE request, having translated the ATS-R2 called party number to a URI suitable for inclusion in the Request-URI. The SIP INVITE request and the resulting SIP dialog, if successfully established, are associated with the ATS-R2 call. The SIP 180 (Ringing) response is mapped to an ATS-R2 Status Signal no.6 “Terminal Free”. During establishment, media streams established by SIP and SDP are connected to the bearer channel.

A call from SIP to ATS-R2 is initiated when a SIP INVITE request arrives at the gateway. The gateway sends an ATS-R2 Seizing line signal and a number of ATS-R2 Register signals for the digits sequence comprising called party number, call priority and calling party number to initiate ATS-R2 call establishment, having translated the SIP Request-URI to a number suitable for use as the ATS-R2 called party number. The resulting ATS-R2 call is associated with the SIP INVITE request and with the eventual SIP dialog. The ATS-R2 Status Signal no. 6 “Terminal Free” is mapped to a SIP 200 (OK) response.

4.2 GENERAL REQUIREMENTS

An ATS-R2 / SIP gateway **SHALL** support ATS-R2 in accordance with [29] as a gateway VCS and **SHALL** support SIP in accordance with RFC 3261 [8] as a UA. In particular, the gateway **SHALL** support SIP syntax and encoding, the SIP transport layer and the SIP transaction layer in accordance with RFC 3261. In addition, the gateway **SHALL** support SIP TU behaviour for a UA in accordance with RFC 3261 except where stated otherwise in this specification.

The gateway **SHALL** support SDP in accordance with RFC 4566 [25] and its use in accordance with the offer / answer model in RFC 3264 [9].

The SIP profile specified in CHAPTER 3 **SHALL** apply to the ATS-R2 / SIP gateway.

The gateway **SHALL** support calls from ATS-R2 to SIP and calls from SIP to ATS-R2.

As a result of DNS look-up by the gateway in order to determine where to send a SIP INVITE request, a number of candidate destinations can be attempted in sequence. The way in which this is handled by the gateway is outside the scope of this specification. However, any behaviour specified in this specification on receipt of a SIP final response **SHOULD** apply only when there are no more candidate destinations to try.

4.3 MESSAGE MAPPING REQUIREMENTS

The SIP protocol and ATS-R2 signalling system information flows **SHALL** be mapped to one another as specified in 4.8.

4.3.1 Call Establishment from ATS-R2 to SIP

4.3.1.1 Receipt of an ATS-R2 “Seizing” Line Signal and Addressing and Priority Digits

On receipt of an incoming call (signified by receipt of an ATS R2 “Seizing” line signal and digits) from the ATS-R2 part of the network, containing called party address, call priority and calling party address information that the Gateway determines to be complete, the Gateway **SHALL** map the ATS-R2 call establishment signals to a SIP INVITE request.

The Gateway **SHALL** generate the SIP Request-URI, To and From fields in the SIP INVITE request and **SHALL** also include SDP information as specified in section 4.5.

On receipt of an incoming call containing addressing information that the Gateway determines to be incomplete, or a protocol error during call establishment occurs, the Gateway **SHALL** initiate ATS-R2 call clearing procedures as specified in [29].

4.3.1.2 Receipt of SIP 100 (Trying) Response

A SIP 100 response **SHALL NOT** trigger any ATS-R2 signal. It only serves the purpose of suppressing INVITE request retransmissions.

4.3.1.3 Receipt of 18x Provisional Response

The Gateway **SHALL** map a received SIP 18x response to an ATS-R2 status signal no. 6 "Terminal Free" and supply Ringing tone on the inter-VCS link.

4.3.1.4 Receipt of SIP 2xx Response

If the Gateway receives a SIP 200 (OK) response as the first SIP 200 response to a SIP INVITE request, the Gateway **SHALL** map the SIP 200 (OK) response to an ATS-R2 status signal no. 6 "Terminal Free". The Gateway **SHALL** also send a SIP ACK request to acknowledge the SIP 200 (OK) response. The Gateway **SHALL NOT** include any SDP information in the SIP ACK request. If the Gateway receives further SIP 200 (OK) responses, it **SHALL** respond to each in accordance with RFC 3261 [8] and **SHALL NOT** generate any further ATS-R2 signals.

Media streams will normally have been established in the IP network in each direction. If so, the Gateway **SHALL** connect the media streams to the inter-VCS link if it has not already done so and stop any local Ringing tones.

If the Gateway receives a SIP 2xx response other than SIP 200 (OK), the Gateway **SHALL** send a SIP ACK request and **SHALL NOT** generate any ATS-R2 signal.

4.3.1.5 Receipt of SIP 3xx Response

On receipt of a SIP 3xx response, the Gateway **SHALL** act in accordance with RFC 3261 [8].

No ATS-R2 signal **SHALL** be sent.

4.3.2 Call Establishment from SIP to ATS-R2

4.3.2.1 Receipt of SIP INVITE Request for a New Call

On receipt of a SIP INVITE request for a new call from the IP network, the Gateway **SHALL** attempt to establish a call towards the ATS-R2 network applying the requirements of [29] by sending an ATS-R2 Seizing line signal, and addressing and call priority digits from the received SIP INVITE request. The Gateway **SHALL** also send a SIP 100 (Trying) response towards the IP network.

If no suitable circuit is available the Gateway **SHALL** send a SIP 503 (Service Unavailable) response.

If information in the SIP INVITE request is unsuitable for generating Called party number and Calling party number, the Gateway **SHALL NOT** issue an ATS-R2 Seizing line signal and **SHALL** send a SIP 500 (Server internal error) response.

If the SIP INVITE request does not contain SDP information, the Gateway **SHALL NOT** issue an ATS-R2 Seizing line signal and **SHALL** send a SIP 488 (Not Acceptable Here) response.

4.3.2.2 Receipt of ATS-R2 Status Signal no. 6 "Terminal Free"

The Gateway **SHALL** map an ATS-R2 Status Signal no. 6 "Terminal Free" to a SIP 200 (OK) final response for the SIP INVITE request.

The Gateway **SHALL** connect the media streams to the corresponding inter-VCS link if it has not already done so, provided SDP answer information is included in the transmitted SIP response or has already been sent or received.

4.3.2.3 Receipt of SIP ACK Request

The receipt of a SIP ACK request **SHALL NOT** result in any ATS-R2 signal.

If the SIP ACK contains SDP answer information, the Gateway **SHALL** connect the media streams to the corresponding inter-VCS link if it has not already done so.

4.3.3 Call Clearing and Call Failure

4.3.3.1 Receipt of an ATS-R2 Release Line Signal

On receipt of an ATS-R2 Release line signal, the Gateway behaviour shall depend on the state of call establishment.

1. If the Gateway has sent a SIP 200 (OK) response and received a SIP ACK request or has received a SIP 200 (OK) response and sent a SIP ACK request, the Gateway **SHALL** send a SIP BYE request to clear the call.

2. If the Gateway has sent a SIP 200 (OK) response (indicating that call establishment is complete) but has not received a SIP ACK request, the Gateway **SHALL** wait until a SIP ACK is received and then send a SIP BYE request to clear the call.

3. If the Gateway has sent a SIP INVITE request and received a SIP provisional response but not a SIP final response, the Gateway **SHALL** send a SIP CANCEL request to clear the call.

   Note 18.
   In accordance with RFC 3261 [8], if after sending a SIP CANCEL request a SIP 2xx response is received to the SIP INVITE request, the Gateway shall need to send a SIP BYE request.

4. If the Gateway has sent a SIP INVITE request but received no SIP response, the Gateway **SHALL NOT** send a SIP message. If a SIP final or provisional response is subsequently received, the Gateway **SHALL** then act in accordance with 1, 2 or 3 above respectively.

5. If the Gateway has received a SIP INVITE request but not sent a SIP final response, the Gateway **SHALL** send a SIP final response chosen according to the cause value in the received ATS-R2 Status Signal as specified in Table 10. SIP response 500 (Server internal error) **SHALL** be used as the default for cause values not shown in Table 10.

In all cases the Gateway **SHALL** also disconnect media streams, if established, and allow ATS-R2 and SIP signalling to complete in accordance with [29] and [8] respectively.

ATS-R2 Status Number	ATS-R2 Status Information	SIP Response Code	SIP Response Description
3	Terminal Busy	486	Busy Here
5	Terminal out of service	404	Not Found
8	Trunk congestion	503	Service Unavailable

Table 10 – Mapping of ATS-R2 Status Signal to SIP Error Response

ATS-R2 Line Signal	SIP Response Code	SIP Response Description
Release (Instead of a Status signal)	500	Server Internal Error

Table 11 – Mapping of ATS-R2 Release Line Signal to SIP Error Response

4.3.3.2 Receipt of a SIP BYE Request

On receipt of a SIP BYE request, the Gateway **SHALL** send an ATS-R2 Release line signal. The Gateway **SHALL** also disconnect media streams, if established, and allow SIP signalling to complete in accordance with [8].

Note 19.
When responding to a SIP BYE request, in accordance with RFC 3261 [8] the Gateway is also required to respond to any other outstanding transactions, e.g., with a SIP 487 (Request Terminated) response. This applies in particular if the Gateway has not yet returned a final response to the SIP INVITE request.

4.3.3.3 Receipt of a SIP CANCEL Request

On receipt of a SIP CANCEL request to clear a call for which the Gateway has not sent a SIP final response to the received SIP INVITE request, the Gateway **SHALL** send an ATS-R2 Release line signal. The Gateway **SHALL** also disconnect media streams, if established, and allow SIP signalling to complete in accordance with accordance with [8].

4.3.3.4 Receipt of a SIP 4xx - 6xx Response

On receipt of a SIP final response (4xx-6xx) to a SIP INVITE request, the Gateway **SHALL** transmit an ATS-R2 Status signal or a Release line signal derived from the SIP 4xx-6xx response according to Table 12 and Table 13. The ATS-R2 Release line signal **SHALL** be used as the default for SIP responses not shown in Table 12 and Table 13. The Gateway **SHALL** also disconnect media streams, if established, and allow ATS-R2 and SIP signalling to complete in accordance with [29] and [8] respectively.

	SIP Response Code	SIP Response Description	ATS-R2 Status Number	ATS-R2 Status Information
Client Error	404	Not Found	5	Terminal out of service or Called Number not allocated
	405	Method Not Allowed	5	Terminal out of service
	406	Not Acceptable	5	Terminal out of service
	410	Gone	5	Terminal out of service or Called Number not allocated
	415	Unsupported Media Type	5	Terminal out of service
	480	Temporarily not available	3	Terminal Busy
	484	Address Incomplete	5	Terminal out of service or Called Number not allocated
	485	Ambiguous	5	Terminal out of service or Called Number not allocated
	486	Busy Here	3	Terminal Busy
	488	Not Acceptable Here	5	Terminal out of service
Server Error	501	Not Implemented	5	Terminal out of service
	503	Service Unavailable	8	Trunk congestion
Global Failure	600	Busy Everywhere	3	Terminal Busy
	603	Decline	3	Terminal Busy
	604	Does not exist anywhere	5	Terminal out of service or Called Number not allocated
	606	Not Acceptable	5	Terminal out of service

Table 12 – Mapping of SIP Error Response to ATS-R2 Status Signal

	SIP Response Code	SIP Response Description	ATS-R2 Line Signal
Client Error	400	Bad Request	Release
	401	Unauthorized	Release
	402	Payment Required	Release
	403	Forbidden	Release
	407	Proxy Authentication Required	Release
	408	Request Timeout	Release
	413	Request Entity Too Large	Release
	414	Request-URI Too Large	Release
	416	Unsupported URI Scheme	Release
	420	Bad Extension	Release
	421	Extension Required	Release
	423	Interval Too Brief	Release
	481	Call Leg/Transaction Does Not Exist	Release
	482	Loop Detected	Release
	483	Too Many Hops	Release
	487	Request Terminated	---
	491	Request Pending	Release
	493	Undecipherable	Release
Server Error	500	Server Internal Error	Release
	502	Bad Gateway	Release
	504	Server Time-out	Release
	505	SIP Version not supported	Release
	513	Message Too Large	Release

Table 13 – Mapping of SIP Error Response to ATS-R2 Release Line Signal

4.3.3.5 Gateway-Initiated Call Clearing

If the Gateway initiates clearing of an ATS-R2 call due to ATS-R2 timer expiry or ATS-R2 protocol error in accordance with [29], the Gateway **SHALL** also initiate clearing of the associated SIP call in accordance with subclause 4.3.3.1. If this involves the sending of a final response to a SIP INVITE request, the Gateway **SHALL** use response code 408 (Request timeout) or 500 (Server internal error) as appropriate.

If the Gateway initiates clearing of the SIP call due to SIP timer expiry or SIP protocol error in accordance with RFC 3261 [8], the Gateway **SHALL** also initiate clearing of the associated ATS-R2 call in accordance with [29].

4.3.4 Request to Change Media Characteristics

If after a call has been successfully established the Gateway receives a SIP INVITE request to change the media characteristics of the call in a way that would be incompatible with voice use, the Gateway **SHALL** send back a SIP 503 (Service unavailable) response and **SHALL NOT** change the media characteristics of the existing call.

4.4 NUMBER MAPPING

In ATS-R2, users are identified by numbers according to a closed numbering scheme.

In SIP, users are identified by Universal Resource Identifiers (URIs) conveyed within the Request-URI and various headers, including the From and To headers specified in RFC 3261 [8].

4.4.1 Mapping from ATS-R2 to SIP

4.4.1.1 Using information from the ATS-R2 Called Party Number

When mapping ATS-R2 call setup signals to a SIP INVITE request, the Gateway **SHALL** convert the Called party number to a URI and include that URI in the SIP Request-URI and in the To header fields.

On receipt of an incoming call containing addressing information that the Gateway determines to be complete but no URI is derivable, the Gateway **SHALL** send an ATS-R2 Status Signal no. 5 “Terminal out of service” and **SHALL NOT** send any SIP request.

4.4.1.2 Using information from the ATS-R2 Calling Party Number

When mapping ATS-R2 call setup signals to a SIP INVITE request, the Gateway **SHALL** convert the Calling party number to a URI and include that URI in the SIP From header field.

If no URI is derivable, the Gateway **SHALL** include its own URI in the SIP From header field.

4.4.2 Mapping from SIP to ATS-R2

When mapping a SIP INVITE request to ATS-R2 call setup signals, the Gateway **SHALL** convert the To header field to a Called party number and the From header field to a Calling party number.

If either Called or Calling party numbers are not derivable, the Gateway **SHALL** send a SIP response 500 (Server internal error) and **SHALL NOT** send any ATS-R2 signal.

4.5 MEDIA TYPE IN SDP

The Gateway **SHALL** generate SDP information to include in the SIP INVITE request. The media type included in the SDP information **SHALL** be “audio”.

4.6 REQUIREMENTS FOR SUPPORT OF SUPPLEMENTARY SERVICES

A Gateway **SHALL** support the Priority Call Interruption and the Priority Call Intrusion supplementary services.

4.6.1 Call Priority

4.6.1.1 Mapping at an Incoming Gateway

On receipt of a call from the ATS-R2 network, the Gateway **SHALL** read the ATS-R2 Priority digit of the call and include the relevant Priority and Max-Forwards header fields in the SIP INVITE message with values as specified in Table 14.

Gateway Input: ATS-R2 Priority Digit	Gateway Output		Call Type
	SIP Priority Header Field	SIP Max-Forwards Header Field	
1 or 6	emergency	< 20	Priority call
2 or 7	urgent	< 20	Tactical Routine call
3 or 8	normal	< 20	Strategic Routine call
4 or 9	non-urgent	< 20	General Purpose Routine call

Table 14 – Mapping of ATS-R2 Priority Digit to SIP Header Fields

A VCS **SHOULD** provide a management means of configuring the acceptable (network dependent) Max-Forwards initial value. Nevertheless, it is **RECOMMENDED** that the initial value for the Max-

Forwards header field is less than 20.

4.6.1.2 Mapping at an Outgoing Gateway

On receipt of a SIP INVITE message from the IP network, the Outgoing Gateway **SHALL** map the Priority header field of the SIP INVITE message to the ATS-R2 priority digit as specified in Table 15.

Gateway Input SIP Priority Header Field	Gateway Output ATS-R2 Priority Digit	Call Type
emergency	1 or 6	Priority call
urgent	2 or 7	Tactical Routine call
normal	3 or 8	Strategic Routine call
non-urgent	4 or 9	General Purpose Routine call

Table 15 – Mapping of SIP Priority Header Field to ATS-R2 Priority Digit

Priority Digit values 1-4 **SHALL** be used when the call is routed on a direct route, and values 6-9 **SHALL** be used when detour is necessary since the direct route is occupied.

4.6.2 Priority Call Interruption

Priority Call Interruption is subject to the following restrictions:

1. A Priority call **SHALL NOT** be interrupted from any end;

2. A Routine call **MAY** be interrupted from the ATS-R2 End or by the Gateway when congestion exists.

4.6.2.1 Priority Call Interruption from SIP to ATS-R2

On receipt of an INVITE(emergency) request from the IP network, and all available inter-VCS ATS-R2 links being occupied, the Gateway **SHALL** attempt to establish a priority call towards the ATS-R2 network as specified in [29]. The priority call **SHOULD** interrupt an established routine (non-priority) call (should one exist), thus allowing the priority call to proceed. If interruption is not possible (because all established calls are priority calls), the Gateway **SHALL** attempt to establish the call using a detour route. If interruption is possible, the Gateway **SHALL** select an ATS-R2 call with the lowest Priority level. Before the established routine call is interrupted, all parties engaged in that call **SHALL** receive an interrupt warning tone.

Having selected the call to be interrupted, applying a “Priority Level Interruption Order” principle, the Gateway **SHALL** inform the users that their call is to be released; it **SHALL** inject an audible tone (Interrupt warning tone) into the voice path to each user in the call and start the “Interrupt pending” timer.

If another circuit becomes available prior to the expiry of the “Interrupt pending“ timer, the call interruption **SHALL** be abandoned. The Gateway **SHALL** stop injecting the “Interrupt Warning” tone in the voice path to each user in the call. It **SHALL** then proceed with establishment of the priority call using this available circuit.

On expiry of the Interruption pending timer, the Gateway **SHALL** stop injecting the “Interrupt warning” tone in the voice path to the users and then force release the call to be interrupted by sending an ATS-R2 “Blocking” line signal on the ATS-R2 line.

On termination of the ATS-R2 Blocking line signal, the Gateway **SHALL** continue establishment of the priority call using the newly available ATS-R2 circuit. The Gateway **SHALL** map the SIP emergency call to an ATS-R2 call with the highest priority as specified in Table 15 above.

4.6.2.2 Priority Call Interruption from ATS-R2 to SIP

On receipt of an ATS-R2 Blocking line signal, after a previous Interrupt warning tone injected by the ATS-R2 End into the voice path, the Gateway **SHALL** send a SIP BYE request containing a Text/Plain body indicating “Emergency - Forced Release” to the SIP End.

4.6.3 Priority Call Intrusion

4.6.3.1 Priority Call Intrusion from ATS-R2 to SIP

An ATS-R2 priority call (having Priority level 1 or 6) made towards the Gateway **SHALL** cause Gateway SIP User Agent to send a SIP INVITE request with a Priority header defined as “emergency” to distinguish it from a routine call (with Priority header defined as “urgent, “normal”, non-urgent).

The Wanted SIP_End User Agent, receiving an INVITE(emergency), acts as the focus for a conference (use is made of the “isfocus” feature defined in RFC 3840 [16] to create a conference media session).

On receiving a SIP 182 (Queued) provisional response, the Gateway **SHALL NOT** send any ATS-R2 signals towards the ATS-R2 End.

On receiving a SIP 183 (Intrusion in progress) provisional response, the Gateway **SHALL** send an ATS-R2 status signal no. 6 “Terminal Free” to the ATS-R2 End. Should no early media be present, Ringing tone **SHALL** be injected by the Gateway. Upon receipt of a 200 (OK) final response, the Gateway **SHALL** stop injection of the Ringing tone if no early media was present.

On receiving a SIP 200 (OK) final response without having received a previous SIP 183 (Intrusion in progress) provisional response, the Gateway **SHALL** send an ATS-R2 status signal no. 6 “Terminal Free” to the ATS-R2 End.

> ***Note 20.***
> *SIP 183 (Intrusion in progress) and/or 200 (OK) responses have to be received within P22 (12s), in accordance with [29]. This implies T1 (see paragraph 3.8.8) < P22.*

If intrusion is forbidden (i.e. CIPL on), the Wanted user reply is SIP 180 (Ringing). Wanted and Unwanted users remain connected; Call from ATS-R2 user is displayed at user’s terminal as Priority Call and can be manually answered. On receiving a SIP 180 (Ringing) response, the Gateway **SHALL** send an ATS-R2 status signal no. 6 “Terminal Free” and supply Ringing tone to the ATS-R2 End.

4.6.3.2 Priority Call Intrusion from SIP to ATS-R2

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. The Gateway **SHALL** be configured to operate with T1 = 0 and automatic Priority call answer.

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, 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.

4.7 AUDIBLE TONES

A gateway **SHALL** be capable of generating the audible tones indicated in Table 16 and sending them over the ATS-R2 line.

Audible Tone	Purpose	Tone generated upon receipt of:

Audible Tone	Purpose	Tone generated upon receipt of:
Ringing	Sent by the gateway towards the ATS-R2 network after successful call establishment and prior to call acceptance by the called user.	• 180 (Ringing) response
Interrupt warning	Injected into the voice path to warn a user of the imminent priority interruption of an established call. This signal is sent by the gateway that is handling the call interruption over the inter-VCS link.	• A Priority (emergency) call that has to interrupt an established Routine call

Table 16 – ATS-R2 Audible Tones Transmitted on Line

4.8 MESSAGE SEQUENCE CHARTS

The paragraphs below show some typical message sequences that can occur for an interworking between ATS-R2 and SIP.

The Message Sequence Charts (MSC) in figures below show the information flows between the Call Control entity (labelled "Gateway") and respective Protocol Control entities for each signalling system (labelled "ATS-R2_End" and "SIP_End") within a Gateway VCS. Each information flow is named according to the corresponding message or signal sent to or received from a peer VCS.

Dashed lines (---) represent signalling messages that are mandatory to the call scenario. These messages can be SIP or ATS-R2 signalling. The arrow indicates the direction of message flow.

Double dashed lines (===) represent media paths between network elements.

4.8.1 Successful ATS-R2 to SIP Routine Call

The MSC shown below is a typical message sequence for a successful call setup of an incoming routine call (to a gateway) from a route employing the ATS-R2 signalling system which is routed on an IP network to the called user employing SIP.

```
 ATS-R2_End        Gateway            Proxy            SIP_End
      |                |                 |                 |
      | SEIZING signal |                 |                 |
      |--------------->|                 |                 |
      |    MF Digit    |                 |                 |
      |--------------->|                 |                 |
      |  MF Digit Ack  |                 |                 |
      |<---------------|                 |                 |
      |       .        |                 |                 |
      |       .        |                 |                 |
      |Priority(2,3,4,7,8 or 9)          |                 |
      |--------------->|                 |                 |
      |Prior.Digit ACK |                 |                 |
      |<---------------|                 |                 |
      |       .        |                 |                 |
      |       .        |                 |                 |
      |    MF Digit    |                 |                 |
      |--------------->|                 |                 |
      |  MF Digit Ack  |                 |                 |
      |<---------------|                 |                 |
    A |                |     INVITE      |                 |
    | |                |---------------->|     INVITE      |
    | |                |   100 Trying    |---------------->|
  P22 |                |<----------------|   180 Ringing   |
    | |                |   180 Ringing   |<----------------|
    | |"Terminal Free" |<----------------|                 |
    V |<---------------|                 |                 |
      |   Status Ack   |                 |                 |
      |--------------->|                 |                 |
      |  Ringing Tone  |                 |                 |
      |<===============|                 |                 |
      |                |                 |                 |
      |                |                 |     200 OK      |
      |                |     200 OK      |<----------------|
      | (Ringing Tone  |<----------------|                 |
      |  Stopped)      |     ACK         |                 |
      |                |---------------->|     ACK         |
      |                |                 |---------------->|
      |(Both Way Voice)|      (Both Way RTP Media)         |
      |<==============>|<=================================>|
      |                |                 |                 |
```

Fig. 32 – Successful ATS-R2 to SIP Routine Call

Note 21.
For a routine call, the value of the "Priority" header field in the INVITE method shall be "urgent", "normal" or "non-urgent", as indicated in Table 14.

Note 22.
As defined in [29], "P22" is the time interval between the end of the 13th Acknowledge signal and the start of a register type Status signal for general transit interworking; its time-out is 12s.

4.8.2 Successful SIP to ATS-R2 Routine Call

This is a typical message sequence for a successful call setup of an incoming routine call (to a gateway) from an IP network employing SIP which is routed on a route employing the ATS-R2 signalling system.

```
ATS-R2_End        Gateway          Proxy           SIP_End
     |                |                |                |
     |                |                |     INVITE     |
     |                |     INVITE     |<---------------|
     | SEIZING signal |<---------------|   100 Trying   |
     |<---------------|   100 Trying   |--------------->|
     |    MF Digit    |--------------->|                |
     |<---------------|                |                |
     |  MF Digit Ack  |                |                |
     |--------------->|                |                |
     |       .        |                |                |
     |       .        |                |                |
     |Priority(2,3,4,7,8 or 9)         |                |
     |<---------------|                |                |
     |Prior.Digit ACK |                |                |
     |--------------->|                |                |
     |       .        |                |                |
     |       .        |                |                |
     |    MF Digit    |                |                |
     |<---------------|                |                |
     |  MF Digit Ack  |                |                |
     |--------------->|                |                |
   A |                |                |                |
   | |                |                |                |
P6/P22                |                |                |
   | | “Terminal Free”|                |                |
   V |--------------->|     200 OK     |                |
     |   Status Ack   |--------------->|     200 OK     |
     |<---------------|                |--------------->|
     |                |                |      ACK       |
     |  Ringing Tone  |      ACK       |<---------------|
     |===============>|<---------------|                |
     |                |                |                |
     |                |                |                |
     | (Ringing Tone  |                |                |
     |  Stopped)      |                |                |
     |(Both Way Voice)|      (Both Way RTP Media)       |
     |<==============>|<===============================>|
     |                |                |                |
```

Fig. 33 – Successful SIP to ATS-R2 Routine Call

Note 23.
For a routine call, the ATS-R2 Priority Digit value shall be 2, 3, 4, 7, 8 or 9, as indicated in Table 15.

Note 24.
As defined in [29], “P6” is the time interval between the end of the 13th Acknowledge signal and the start of a register type Status signal; its time-out is 1 s. “P6” is equivalent to “P22” for general transit interworking; their time-out in this case is 12 s.

4.8.3 Normal Call Clearing from ATS-R2 End

Typical message sequence for Call Clearing from ATS-R2 to SIP subsequent to call establishment.

```
ATS-R2_End         Gateway            Proxy            SIP_End
    |                 |                 |                 |
    |(Both Way Voice)|       (Both Way RTP Media)        |
    |<===============>|<=================================>|
    |                 |                 |                 |
    | RELEASE signal  |                 |                 |
    |---------------->|      BYE        |                 |
    |                 |---------------->|      BYE        |
    |                 |                 |---------------->|
    |                 |                 |     200 OK      |
    |                 |     200 OK      |<----------------|
    |                 |<----------------|                 |
    |                 |                 |                 |
```

Fig. 34 – Normal Call Clearing from ATS-R2 End

4.8.4 Normal Call Clearing from SIP End

Typical message sequence for Call Clearing from SIP to ATS-R2 subsequent to call establishment.

```
ATS-R2_End         Gateway            Proxy            SIP_End
    |                 |                 |                 |
    |(Both Way Voice)|       (Both Way RTP Media)        |
    |<===============>|<=================================>|
    |                 |                 |                 |
    |                 |                 |       BYE       |
    |                 |      BYE        |<----------------|
    | RELEASE signal  |<----------------|                 |
    |<----------------|     200 OK      |                 |
    |                 |---------------->|     200 OK      |
    |                 |                 |---------------->|
    |                 |                 |                 |
```

Fig. 35 – Normal Call Clearing from SIP End

4.8.5 Unsuccessful ATS-R2 to SIP Call

4.8.5.1 Call Clearing from ATS-R2 End

This is a typical message sequence for Call Clearing from ATS-R2 to SIP during establishment of a call from ATS-R2 to SIP, in which the Gateway has received a provisional response (1xx) to the INVITE request but not a final response (2xx, 3xx, 4xx, 5xx, 6xx).

```
ATS-R2_End        Gateway          Proxy           SIP_End
    |                |                |                |
    | SEIZING signal |                |                |
    |--------------->|                |                |
    |    MF Digit    |                |                |
    |--------------->|                |                |
    |  MF Digit ACK  |                |                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
    |    MF Digit    |                |                |
    |--------------->|                |                |
    |  MF Digit Ack  |                |                |
    |<---------------|     INVITE     |                |
    |                |--------------->|     INVITE     |
    |                |   100 Trying   |--------------->|
    |                |<---------------|   180 Ringing  |
    |                |   180 Ringing  |<---------------|
    |”Terminal Free” |<---------------|                |
    |<---------------|                |                |
    |   Status ACK   |                |                |
    |--------------->|                |                |
    |                |                |                |
    | RELEASE signal |                |                |
    |--------------->|     CANCEL     |                |
    |                |--------------->|     CANCEL     |
    |                |     200 OK     |--------------->|
    |                |<---------------|     200 OK     |
    |                |                |<---------------|
    |                |                |487 Req. Termin.|
    |                |487 Req. Termin.|<---------------|
    |                |<---------------|       ACK      |
    |                |       ACK      |--------------->|
    |                |--------------->|                |
    |                |                |                |
```

**Fig. 36 – Unsuccessful ATS-R2 to SIP Call.
Call Clearing from ATS-R2 End**

4.8.5.2 Call Clearing from SIP Network

4.8.5.2.1 Mapping of SIP Response to ATS-R2 Status Signal

This is a typical message sequence for Call Clearing from SIP to ATS-R2 during establishment of a call from ATS-R2 to SIP, in which the Gateway has not previously received a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and the SIP response can be mapped to one of these ATS-R2 Status Signals: “Terminal Busy”, “Terminal Out of Service”, or “Trunk Congestion”.

```
ATS-R2_End         Gateway            Proxy           SIP_End
    |                 |                 |                 |
    | SEIZING signal  |                 |                 |
    |---------------->|                 |                 |
    |    MF Digit     |                 |                 |
    |---------------->|                 |                 |
    |  MF Digit Ack   |                 |                 |
    |<----------------|                 |                 |
    |        .        |                 |                 |
    |        .        |                 |                 |
    |        .        |                 |                 |
    |                 |                 |                 |
    |    MF Digit     |                 |                 |
    |---------------->|                 |                 |
    |  MF Digit Ack   |                 |                 |
    |<----------------|                 |                 |
    |                 |     INVITE      |                 |
    |                 |---------------->|     INVITE      |
    |                 |   100 Trying    |---------------->|
    |                 |<----------------|  486 Busy Here  |
    |                 |                 |<----------------|
    |                 |                 |       ACK       |
    |                 |  486 Busy Here  |---------------->|
    |                 |<----------------|                 |
    |                 |       ACK       |                 |
    |”Terminal Busy”  |---------------->|                 |
    |<----------------|                 |                 |
    |  Status ACK     |                 |                 |
    |---------------->|                 |                 |
    | RELEASE signal  |                 |                 |
    |---------------->|                 |                 |
    |                 |                 |                 |
```

**Fig. 37 – Unsuccessful ATS-R2 to SIP Call.
Mapping of SIP Response to ATS-R2 Status Signal**

4.8.5.2.2 Mapping of SIP Error Response to ATS-R2 Release Line Signal

This is a typical message sequence for Call Clearing from SIP to ATS-R2 during establishment of a call from ATS-R2 to SIP, in which the Gateway has not previously received a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request and the SIP response cannot be mapped to one of the ATS-R2 Status Signals.

```
ATS-R2_End        Gateway           Proxy           SIP_End
    |                |                |                |
    | SEIZING signal |                |                |
    |--------------->|                |                |
    |    MF Digit    |                |                |
    |--------------->|                |                |
    |  MF Digit Ack  |                |                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
    |    MF Digit    |                |                |
    |--------------->|                |                |
    |  MF Digit Ack  |                |                |
    |<---------------|     INVITE     |                |
    |                |--------------->|     INVITE     |
    |                |   100 Trying   |--------------->|
    |                |<---------------|     4xx-6xx    |
    |                |     4xx-6xx    |<---------------|
    | RELEASE signal |<---------------|       ACK      |
    |<---------------|       ACK      |--------------->|
    |                |--------------->|                |
    |                |                |                |
```

Fig. 38 – Unsuccessful ATS-R2 to SIP Call.
Mapping of SIP Error Response to ATS-R2 Release Line Signal

4.8.6 Unsuccessful SIP to ATS-R2 Call

4.8.6.1 Call Clearing from SIP Network

This is a typical message sequence for Call Clearing from SIP to ATS-R2 during establishment of a call from SIP to ATS-R2, in which the Gateway has sent a provisional response (1xx) to the INVITE request but not a final response (2xx, 3xx, 4xx, 5xx, 6xx).

```
ATS-R2_End         Gateway            Proxy           SIP_End
    |                 |                 |                 |
    |                 |                 |     INVITE      |
    |                 |     INVITE      |<----------------|
    | SEIZING signal  |<----------------|   100 Trying    |
    |<----------------|   100 Trying    |---------------->|
    |    MF Digit     |---------------->|                 |
    |<----------------|                 |                 |
    |  MF Digit Ack   |                 |                 |
    |---------------->|                 |                 |
    |        .        |                 |                 |
    |        .        |                 |                 |
    |        .        |                 |     CANCEL      |
    |                 |     CANCEL      |<----------------|
    | RELEASE signal  |<----------------|     200 OK      |
    |<----------------|     200 OK      |---------------->|
    |                 |---------------->|                 |
    |                 |487 Req. Termin. |                 |
    |                 |---------------->|487 Req. Termin.|
    |                 |      ACK        |---------------->|
    |                 |<----------------|      ACK        |
    |                 |                 |<----------------|
    |                 |                 |                 |
```

Fig. 39 – Unsuccessful SIP to ATS-R2 Call.
Call Clearing from SIP Network

4.8.6.2 Call Clearing from ATS-R2 End

4.8.6.2.1 Receipt of Status Signal from ATS-R2 End

This is a typical message sequence for Call Clearing from ATS-R2 to SIP during establishment of a call from SIP to ATS-R2, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and one of these ATS-R2 Status Signals is received: “Terminal Busy”, “Terminal Out of Service”, or “Trunk Congestion”.

```
ATS-R2_End        Gateway           Proxy           SIP_End
    |                |                |                |
    |                |                |     INVITE     |
    |                |    INVITE      |<---------------|
    | SEIZING signal |<---------------|   100 Trying   |
    |<---------------|   100 Trying   |--------------->|
    |    MF Digit    |--------------->|                |
    |<---------------|                |                |
    |  MF Digit Ack  |                |                |
    |--------------->|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
    |    MF Digit    |                |                |
    |<---------------|                |                |
    |  MF Digit Ack  |                |                |
    |--------------->|                |                |
    | “Terminal Busy”|                |                |
    |--------------->|  486 Busy Here |                |
    |   Status ACK   |--------------->|                |
    |<---------------|     ACK        |                |
    | RELEASE signal |<---------------|  486 Busy Here |
    |<---------------|                |--------------->|
    |                |                |      ACK       |
    |                |                |<---------------|
    |                |                |                |
```

Fig. 40 – Unsuccessful SIP to ATS-R2 Call.
Receipt of Status Signal from ATS-R2 End

4.8.6.2.2 Receipt of Release Line Signal from ATS-R2 End

This is a typical message sequence for Call Clearing from ATS-R2 to SIP during establishment of a call from SIP to ATS-R2, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and ATS-R2 Release line signal is received.

```
ATS-R2_End          Gateway            Proxy            SIP_End
    |                  |                 |                 |
    |                  |                 |      INVITE     |
    |                  |     INVITE      |<----------------|
    | SEIZING signal   |<----------------|   100 Trying    |
    |<-----------------|   100 Trying    |---------------->|
    |     MF Digit     |---------------->|                 |
    |<-----------------|                 |                 |
    |        .         |                 |                 |
    |        .         |                 |                 |
    |        .         |                 |                 |
    |                  |                 |                 |
    | RELEASE signal   |                 |                 |
    |----------------->|500 Svr.Int.Err. |                 |
    |                  |---------------->|500 Svr.Int.Err. |
    |                  |       ACK       |---------------->|
    |                  |<----------------|       ACK       |
    |                  |                 |<----------------|
    |                  |                 |                 |
```

Fig. 41 – Unsuccessful SIP to ATS-R2 Call.
Receipt of Release Line Signal from ATS-R2 End

Note 25.
This scenario can arise where a protocol error is encountered by the ATS-R2 End during call set-up.

4.8.6.3 Call Clearing from Gateway

This is a typical message sequence for Call Clearing from Gateway during establishment of a call from SIP to ATS-R2, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and ATS-R2 protocol error is encountered by the Gateway during call set-up.

```
ATS-R2_End        Gateway           Proxy          SIP_End
    |                |                |                |
    |                |                |     INVITE     |
    |                |     INVITE     |<---------------|
    | SEIZING signal |<---------------|   100 Trying   |
    |<---------------|   100 Trying   |--------------->|
    |    MF Digit    |--------------->|                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
    | RELEASE signal |408 Req.timeout/|                |
    |<---------------|500 Svr.Int.Err.|408 Req.timeout/|
    |                |--------------->|500 Svr.Int.Err.|
    |                |      ACK       |--------------->|
    |                |<---------------|      ACK       |
    |                |                |<---------------|
    |                |                |                |
```

Fig. 42 – Unsuccessful SIP to ATS-R2 Call.
Call Clearing from Gateway

4.8.7 Interworking of Supplementary Services

4.8.7.1 Priority Call Interruption

4.8.7.1.1 ATS-R2 to SIP Priority Call Interruption

The message sequence shown below corresponds to a scenario in which a routine (non-priority) call, established through a gateway, is interrupted by a priority call from the ATS-R2_End.

```
ATS-R2_End        Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|       (Both Way RTP Media)      |
    |<==============>|<===============================>|
    |                |                |                |
    |Interrupt Warning Tone (from ATS-R2 End)          |
  A |===============>|================================>|
  | |                |                |                |
  | |                |                |                |
  | |                |                |                |
 P13|                |                |                |
  | |                |                |                |
  | |                |                |                |
  | |BLOCKING signal |                |                |
  V |--------------->|      BYE       |                |
    |                |--------------->|      BYE       |
    |                |                |--------------->|
    |                |                |     200 OK     |
    |                |     200 OK     |<---------------|
    |                |<---------------|                |
    | SEIZING signal |                |                |
    |--------------->|                |                |
    |    MF Digit    |                |                |
    |--------------->|                |                |
    |  MF Digit Ack  |                |                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |Priority= 1 or 6|                |                |
    |--------------->|                |                |
    |Prior.Digit ACK |                |                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
```

Fig. 43 – ATS-R2 to SIP Priority Call Interruption

4.8.7.1.2 ATS-R2 to SIP Priority Call Interruption Abandon

The message sequences shown below correspond to a scenario in which during the interrupt warning period another line becomes available and has been seized for the priority call.

```
ATS-R2_End        Gateway           Proxy          SIP_End
    |                |                 |                 |
    |(Both Way Voice)|        (Both Way RTP Media)       |
    |<==============>|<=================================>|
    |                |                 |                 |
    |Interrupt Warning Tone (from ATS-R2 End)            |
    |===============>|==================================>|
    |                |                 |                 |
    |(Interrupt Warning Tone Stopped)  |                 |
    |                |                 |                 |
    |             (Users remain connected)               |
    |                |                 |                 |
```

Fig. 44 – ATS-R2 to SIP Priority Call Interruption Abandon before the Blocking Line Signal is sent

```
ATS-R2_End        Gateway           Proxy          SIP_End
    |                |                 |                 |
    |(Both Way Voice)|        (Both Way RTP Media)       |
    |<==============>|<=================================>|
    |                |                 |                 |
    |Interrupt Warning Tone (from ATS-R2 End)            |
    |===============>|==================================>|
    |                |                 |                 |
    |                |                 |                 |
    |                |                 |                 |
    |                |                 |                 |
    |                |                 |                 |
    |                |                 |                 |
    |BLOCKING signal |                 |                 |
    |--------------->|       BYE       |                 |
    |                |---------------->|       BYE       |
    |                |                 |---------------->|
    |                |                 |      200 OK     |
    |                |      200 OK     |<----------------|
    |                |<----------------|                 |
    | RELEASE signal |                 |                 |
    |--------------->|                 |                 |
    |                |                 |                 |
```

Fig. 45 – ATS-R2 to SIP Priority Call Interruption Abandon while the Blocking Line Signal is being sent

4.8.7.1.3 SIP to ATS-R2 Priority Call Interruption

The message sequence shown below corresponds to a scenario in which a routine (non-priority) call, established through a gateway, is interrupted by a priority call from the SIP Network.

```
ATS-R2_End        Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|                |                |
    |<==============>|                |                |
    |                |                |INVITE(emergency)
    |   Interrupt    |INVITE(emergency)<---------------|
    |  Warning Tone  |<---------------|   100 Trying   |
    |<===============|   100 Trying   |--------------->|
    |                |--------------->|                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    | BLOCKING signal|                |                |
    |<---------------|                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    | SEIZING signal |                |                |
    |<---------------|                |                |
    |    MF Digit    |                |                |
    |<---------------|                |                |
    |  MF Digit Ack  |                |                |
    |--------------->|                |                |
    |       .        |        .       |        .       |
    |       .        |        .       |        .       |
    |       .        |        .       |        .       |
    |                |                |                |
```

Fig. 46 – SIP to ATS-R2 Priority Call Interruption

4.8.7.1.4 SIP to ATS-R2 Priority Call Interruption Abandon

The message sequences shown below correspond to a scenario in which during the interrupt warning period another line becomes available and has been seized for the priority call.

```
ATS-R2_End        Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|                |                |
    |<==============>|                |                |
    |                |                |INVITE(emergency)
    |   Interrupt    |INVITE(emergency)<---------------|
    |  Warning Tone  |<---------------|  100 Trying    |
    |<===============|   100 Trying   |--------------->|
    |                |--------------->|                |
    |                |                |                |
    |(Interr. Warning|                |                |
    | Tone Stopped)  |                |                |
    |                |                |                |
    |(Users remain   |                |                |
    | connected)     |                |                |
    |                |                |                |
```

Fig. 47 – SIP to ATS-R2 Priority Call Interruption Abandon before the Blocking Line Signal is sent

```
ATS-R2_End        Gateway           Proxy           SIP_End1
    |                |                |                |
    |(Both Way Voice)|                |                |
    |<==============>|                |                |
    |                |                |INVITE(emergency)
    |   Interrupt    |INVITE(emergency)<---------------|
    |  Warning Tone  |<---------------|  100 Trying    |
    |<===============|   100 Trying   |--------------->|
    |                |--------------->|                |
    |                |                |        SIP_End2|
    |                |                |           |    |
    |                |                |           |    |
    |                |                |           |    |
    | BLOCKING signal|                |           |    |
    |<---------------|                |           |    |
    | RELEASE signal |      BYE       |           |    |
    |<---------------|--------------->|    BYE    |    |
    |                |                |---------->|    |
    |                |     200 OK     |<--200 OK--|    |
    |                |<---------------|           |    |
    |                |                |           |    |
```

Fig. 48 – SIP to ATS-R2 Priority Call Interruption Abandon while the Blocking Line Signal is sent

Note 26.
Once the ATS-R2 Blocking line signal is being sent, the Gateway shall continue to send the Blocking signal for its full duration and then clear the call by sending the corresponding ATS-R2 Release line signal and SIP BYE message.

4.8.7.2 Priority Call Intrusion

4.8.7.2.1 Priority Call at an Incoming Gateway

Priority call answered within a predetermined time interval T1

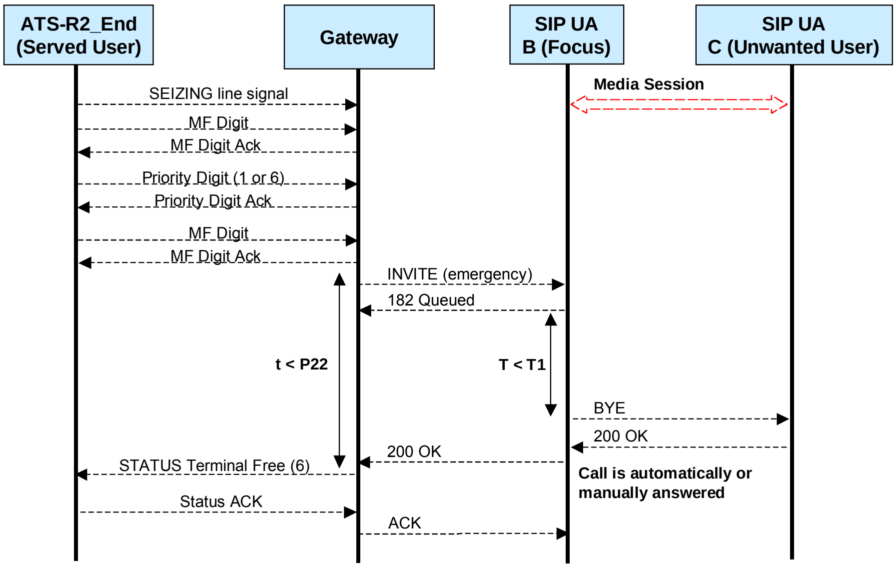


Fig. 49 – ATS-R2 to SIP Priority call answered after releasing previous call

Successful ATS-R2 End Intrusion to SIP UA

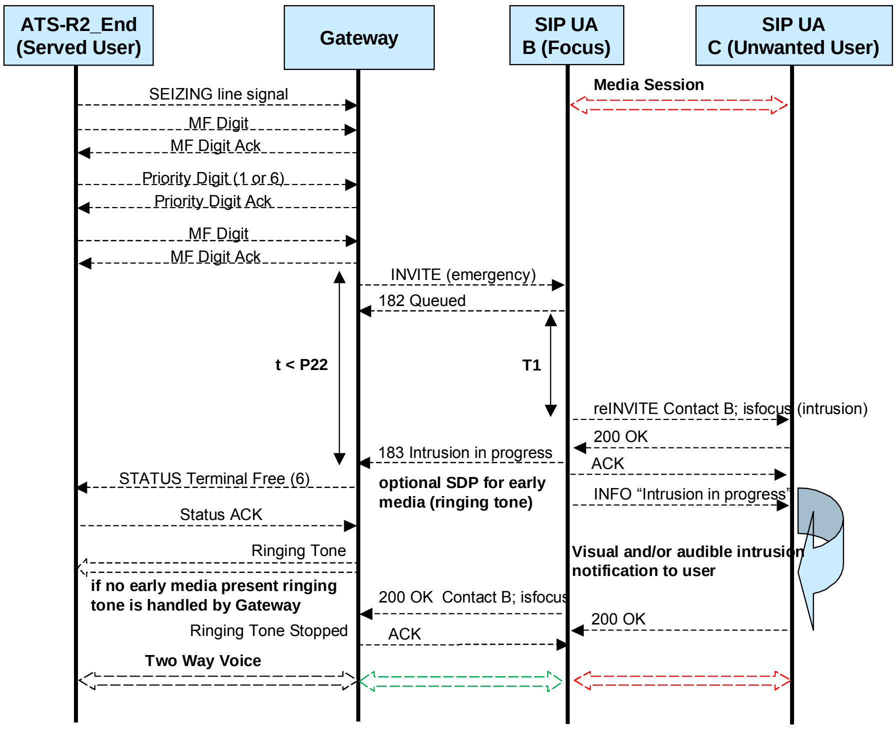


Fig. 50 – ATS-R2 to SIP Successful Priority Call Intrusion

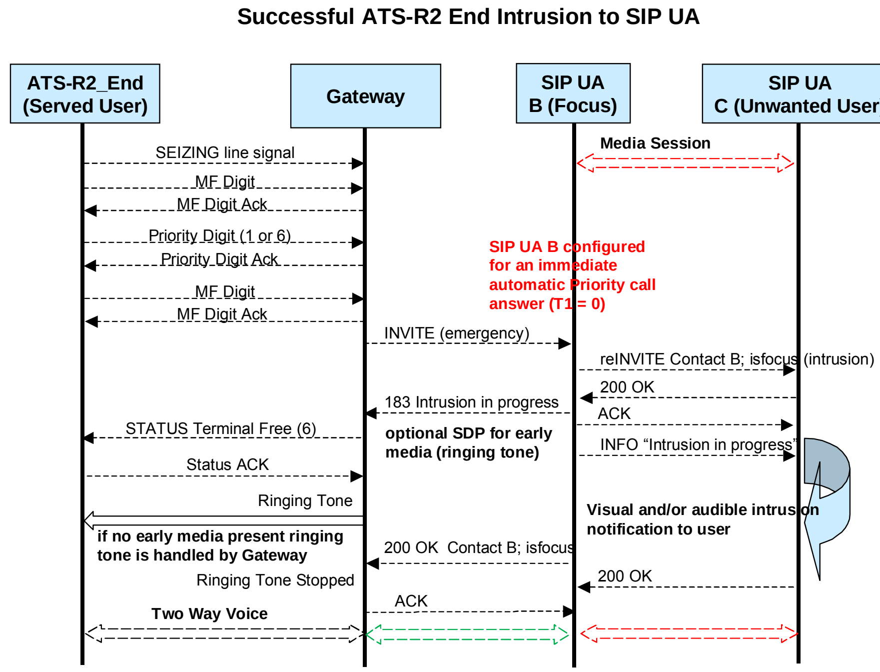


Fig. 51 – ATS-R2 to SIP Successful Priority Call Intrusion with T1 = 0

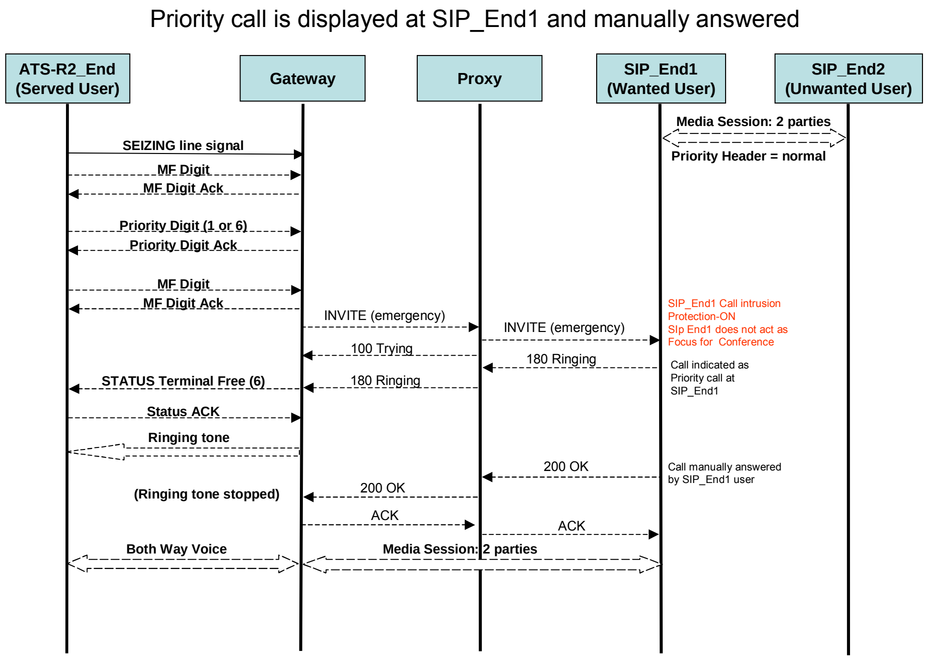


Fig. 52 – ATS-R2 to SIP Priority Call Intrusion Forbidden by Wanted User

- The Call Intrusion Protection Level of the Unwanted User shall be assumed as OFF, that is call intrusion permitted, or not, determined exclusively by the Wanted User; there is no SIP signalling for SIP_End2 (Unwanted User) to forbid call intrusion.

- Call intrusion will happen unless the established active call is a Priority call (Priority Header Field = “emergency”).

Fig. 53 – ATS-R2 to SIP Priority Call Intrusion Cannot Be Forbidden by Unwanted User

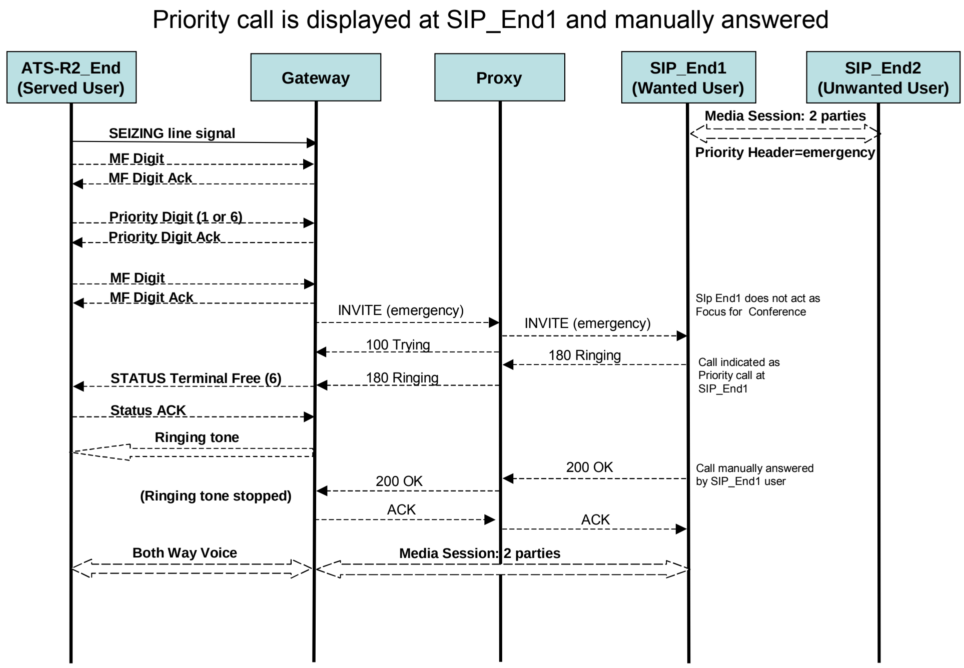


Fig. 54 – ATS-R2 to SIP Priority Call Intrusion into another Priority Call Forbidden

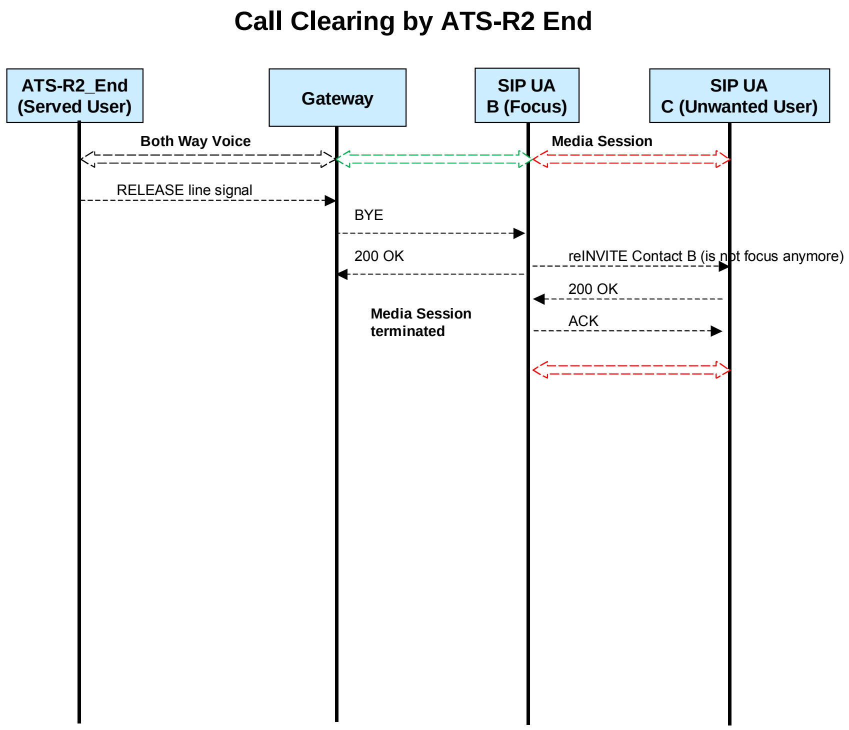


Fig. 55 – Call Clearing by ATS-R2 End

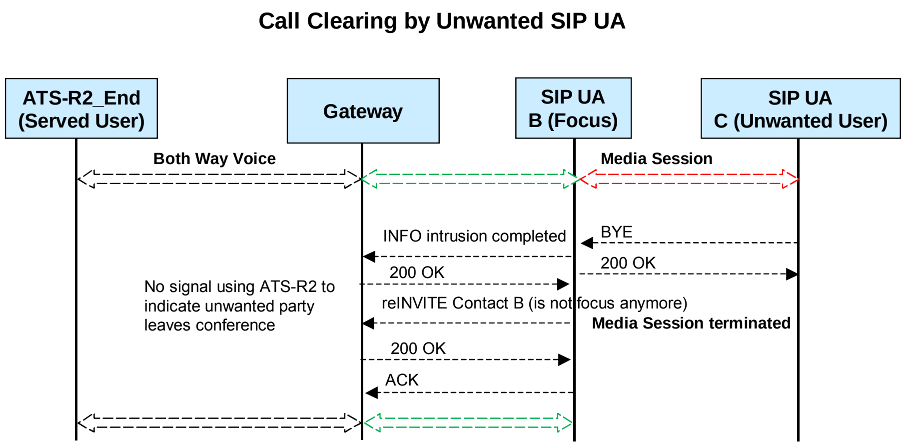


Fig. 56 – Call Clearing by 'Unwanted' SIP UA

4.8.7.2.2 Priority Call at an Outgoing Gateway

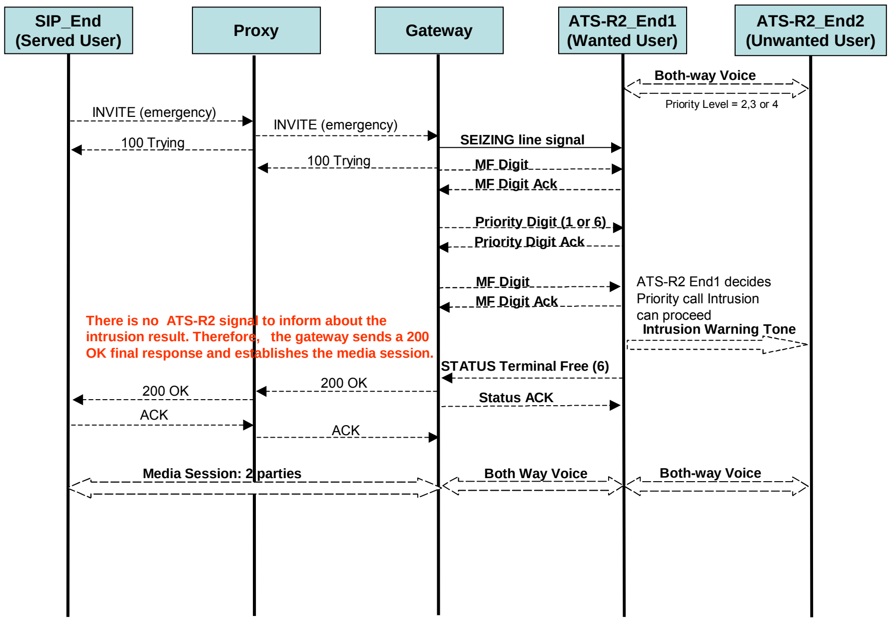


Fig. 57 – SIP to ATS-R2 Successful Priority Call Intrusion

Successful SIP UA Intrusion to SIP - ATS-R2 Call

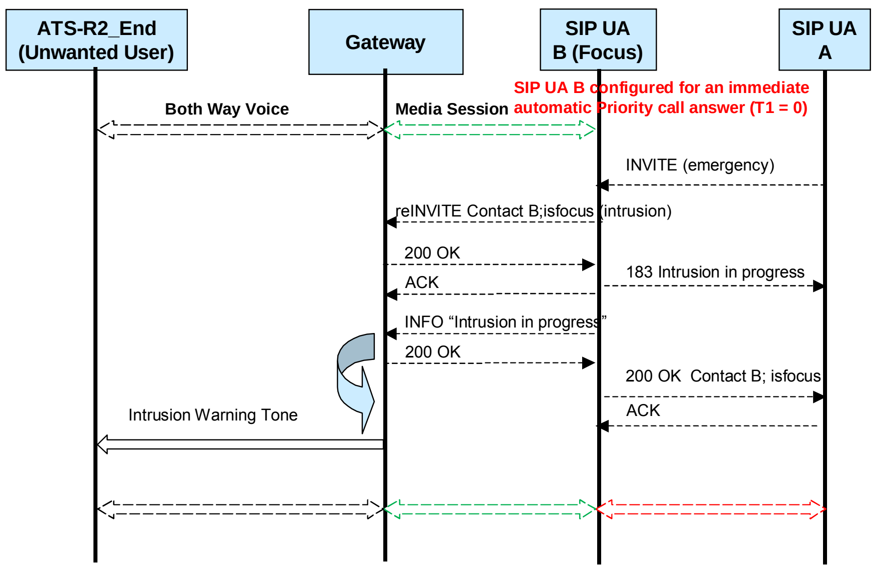


Fig. 58 – Intrusion to a SIP - ATS-R2 Call

Priority call is displayed at ATS-R2 End1 and manually answered

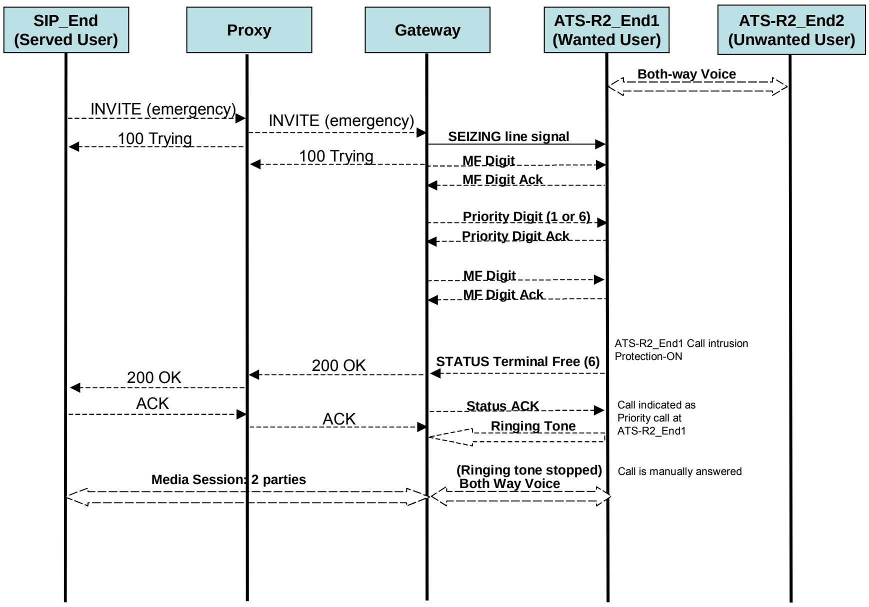


Fig. 59 – SIP to ATS-R2 Priority Call Intrusion Forbidden by Wanted User

- The Call Intrusion Protection Level of the Unwanted User shall be assumed as OFF, that is call intrusion permitted, or not, determined exclusively by the Wanted User; there is no ATS-R2 signalling for ATS-R2_End2 (Unwanted User) to forbid call intrusion.

- Call intrusion will happen unless the established active call is a Priority call (Priority digit = 1 or 6).

Fig. 60 – SIP to ATS-R2 Priority Call Intrusion Cannot Be Forbidden by Unwanted User

Priority call is displayed at ATS-R2 End1 and manually answered

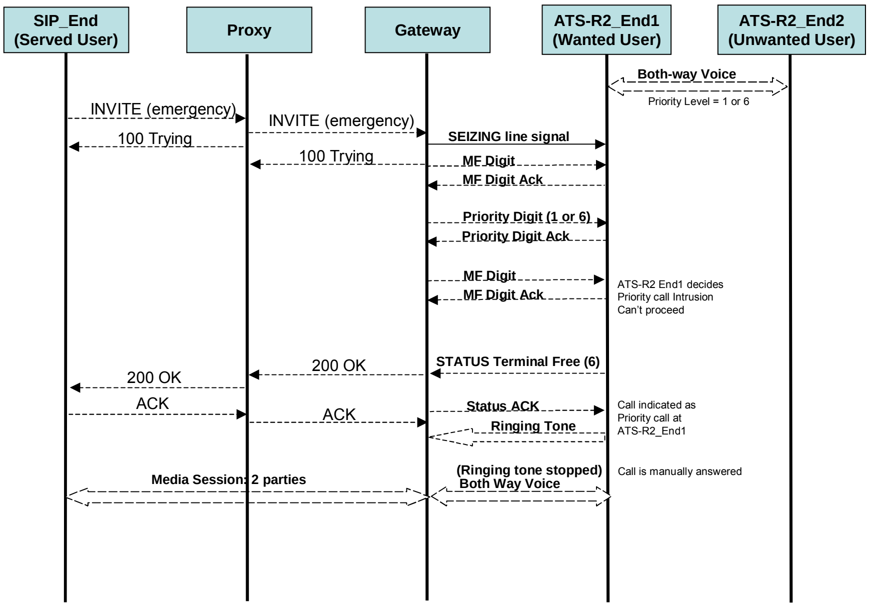


Fig. 61 – SIP to ATS-R2 Priority Call Intrusion into another Priority Call Forbidden

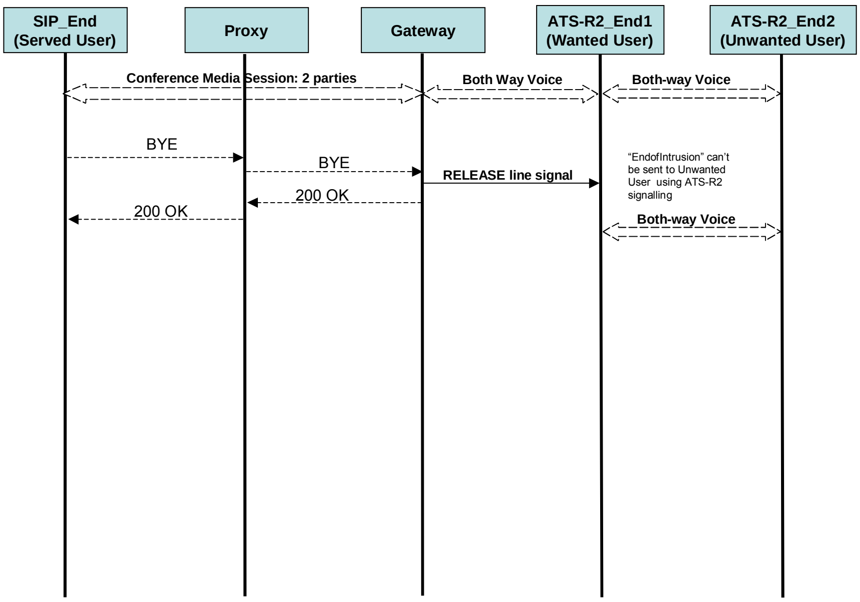


Fig. 62 –Call Clearing by SIP UA

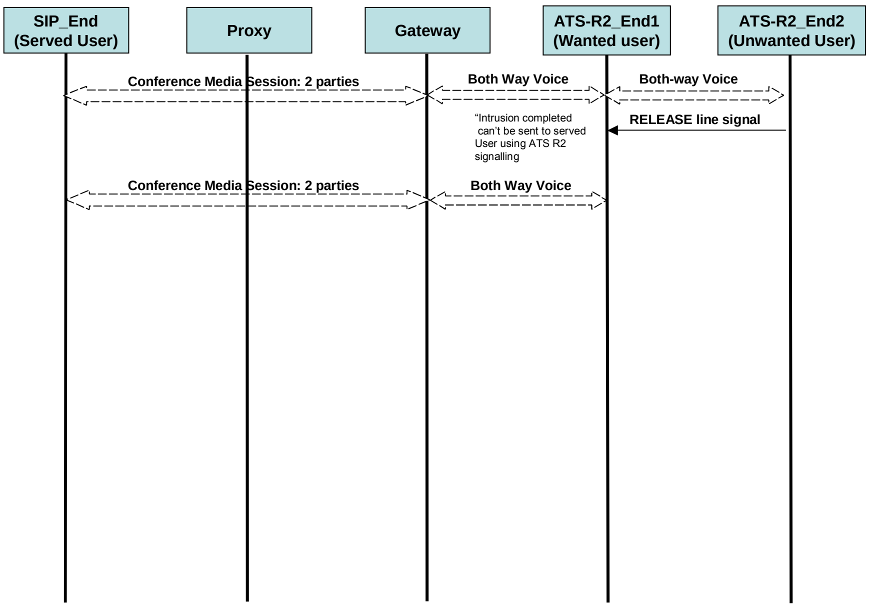


Fig. 63 – Call Clearing by ATS-R2 Unwanted User

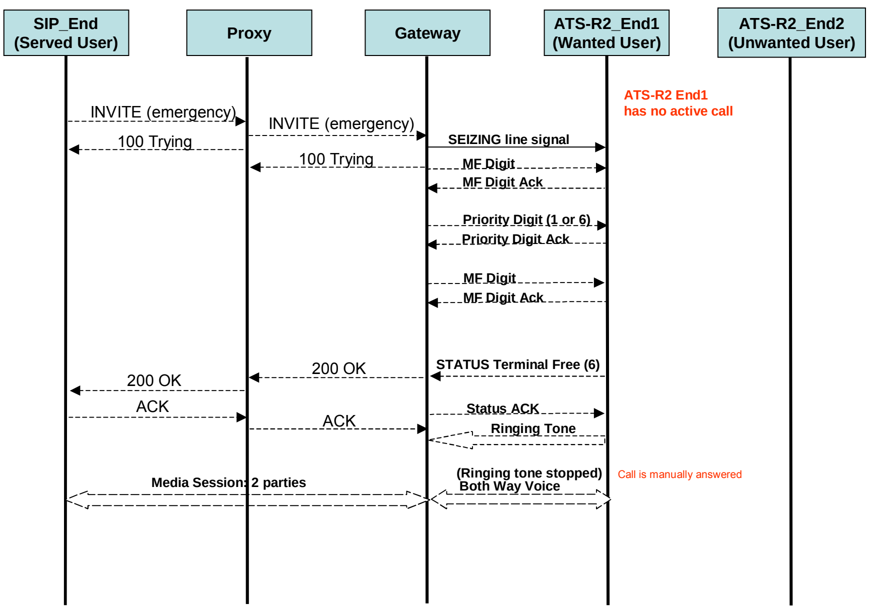
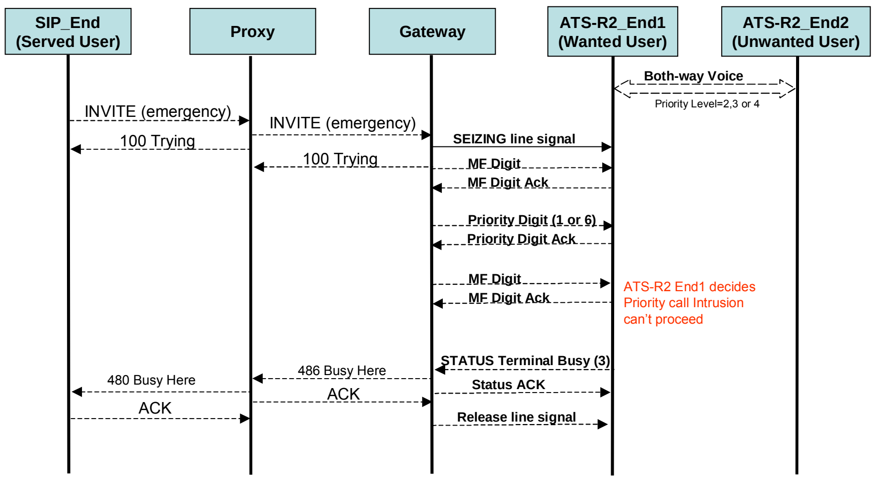
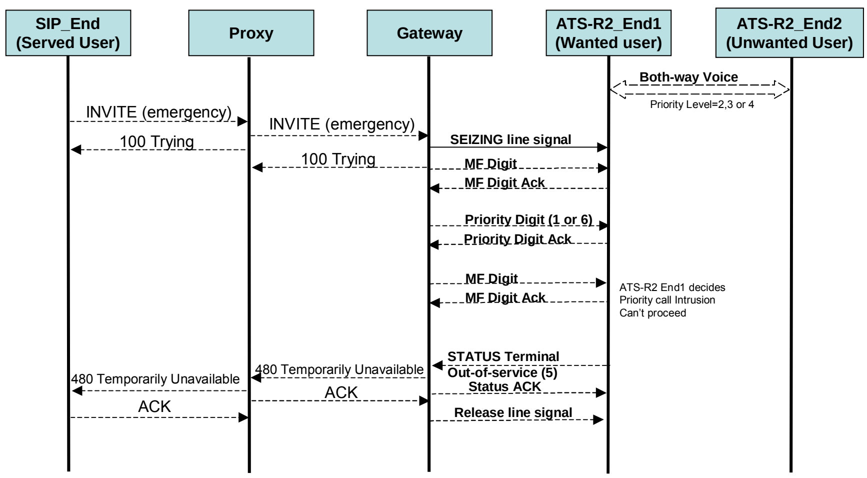


Fig. 64 – SIP to ATS-R2 Priority Call Intrusion to a Non-busy Wanted User



This scenario is unusual as it results in Priority call not being connected

Fig. 65 – SIP to ATS-R2 Priority Call Intrusion Receiving Status Signal "Terminal Busy"



This scenario is unusual as it results in Priority call not being connected

Fig. 66 – SIP to ATS-R2 Priority Call Intrusion Receiving Status Signal "Terminal Out-of-Service"

CHAPTER 5

SIGNALLING INTERWORKING BETWEEN SIP AND ATS-No.5

5.1 BACKGROUND AND ARCHITECTURE

This chapter specifies signalling interworking between SIP and ATS-No.5 in support of basic services as well as ATS supplementary services within an Air Traffic Services Ground Voice Network (AGVN).

ATS-No.5 is a Multi-Frequency inband signalling system adapted for Air Traffic Services networks from the “ITU-T Recommendations Q.140 to Q.164” defining the ITU-T No.5 signalling system. It operates in an analogue link between two VCSs and controls call establishment and call clearing on the inter-VCS link. ATS-No.5 is specified in the document entitled “ATS R2 and ATS No.5 Signalling Protocol Specifications” [29].

Interworking between ATS-No.5 and SIP permits a call originating at a user of a circuit-switched AGVN to terminate at a user of an IP AGVN, or a call originating at a user of an IP AGVN to terminate at a user of a circuit-switched AGVN.

Interworking between a circuit-switched AGVN employing ATS-No.5 and a public IP network employing SIP is outside the scope of this specification. However, the functionality specified in this document is in principle applicable to such a scenario when deployed in conjunction with other relevant functionality (e.g., number translation, security functions, etc.).

This specification is applicable to any interworking unit that can act as a gateway between a circuit-switched AGVN employing ATS-No.5 and an IP AGVN employing SIP.

ATS-No.5 provides a means for establishing and clearing calls that originate and terminate on different VCSs. A call can be routed over a single inter-VCS link connecting the originating and terminating VCS, or over several inter-VCS links in series with switching at intermediate VCSs known as transit VCSs. A call can originate or terminate in another network, in which case it enters or leaves the AGVN environment through a gateway VCS. Parties are identified by numbers, in accordance with a closed numbering plan.

With the aim of exploiting IP to migrate progressively parts of the AGVN network to IP using SIP, SIP equipment in the form of SIP User Agent interfaces, SIP Proxy servers, DNS servers, etc. may be used. The new SIP environment **SHOULD** also need to interwork with the ATS-No.5-based AGVN in order to support calls originating in one environment and terminating in the other. Interworking is achieved through a gateway.

Another way of migrating is to use an IP network to interconnect two parts of a circuit-switched AGVN and encapsulate ATS-No.5 signalling in RTP frames for calls between the two parts of the circuit-switched AGVN. This is outside the scope of this specification.

This document specifies signalling protocol interworking aspects of a gateway between a circuit-switched AGVN employing ATS-No.5 signalling and an IP AGVN employing SIP signalling. The gateway appears as a VCS to other VCSs in the circuit-switched network. The gateway appears as a SIP endpoint to other SIP entities in the IP network. Fig. 67 shows the Interconnection Diagram.

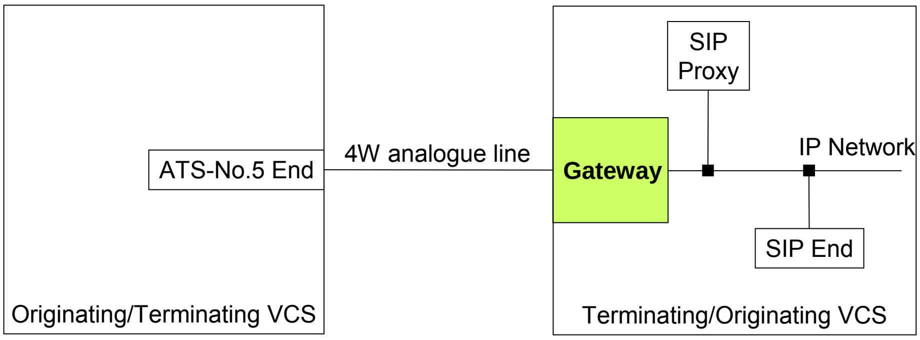


Fig. 67 – ATS-No.5 / SIP Interconnection Diagram

In addition to the signalling interworking functionality specified in this document, it is assumed that the gateway also includes the following functionality:

- one or more physical interfaces on the circuit-switched network side supporting one or more inter-VCS links;
- one or more physical interfaces on the IP network side supporting, through layer 1 and layer 2 protocols, IP as the network layer protocol and UDP (RFC 768) [1] and TCP (RFC 793) [3] as transport layer protocols, these being used for the transport of SIP signalling messages and, in the case of UDP, also for media information;
- the support of TLS (RFC 4346 [24]) as additional transport layer secure protocol on the IP network side, this being used for the transport of SIP signalling messages; and
- a means of transferring media information in each direction between the circuit-switched network and the IP network, including as a minimum packetization of media information sent to the IP network and de-packetization of media information received from the IP network.

The protocol model relevant to signalling interworking functionality of a gateway is shown in Fig. 68.

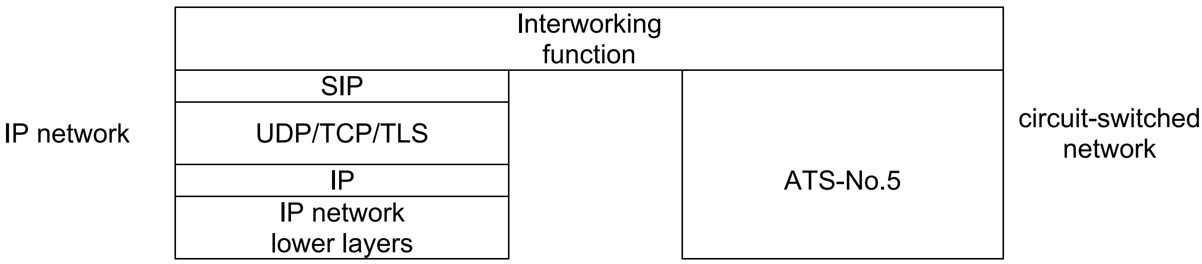


Fig. 68 – SIP / ATS-No.5 Protocol Model

In Fig. 68, the SIP box represents SIP syntax and encoding, the SIP transport layer and the SIP transaction layer. The Interworking function includes SIP Transaction User (TU) functionality.

The gateway maps received ATS-No.5 signals, where appropriate, to SIP messages and vice versa and maintains an association between an ATS-No.5 call and a SIP dialog.

A call from ATS-No.5 to SIP is initiated when an ATS-No.5 Seizing line signal and a number of ATS-No.5 Register signals for the digits sequence comprising called party number, call priority and calling party number arrive at the gateway. The gateway then sends a SIP INVITE request, having translated the ATS-No.5 called party number to a URI suitable for inclusion in the Request-URI. The SIP INVITE request and the resulting SIP dialog, if successfully established, are associated with the ATS-No.5 call. The SIP 180 (Ringing) response is mapped to an ATS-No.5 Status Signal no.6 “Terminal Free”. During establishment, media streams established by SIP and SDP are connected to the bearer channel.

A call from SIP to ATS-No.5 is initiated when a SIP INVITE request arrives at the gateway. The gateway sends an ATS-No.5 Seizing line signal and a number of ATS-No.5 Register signals for the digits sequence comprising called party number, call priority and calling party number to initiate ATS-No.5 call establishment, having translated the SIP Request-URI to a number suitable for use as the ATS-No.5 called party number. The resulting ATS-No.5 call is associated with the SIP INVITE request and with the eventual SIP dialog. The ATS-No.5 Status Signal no. 6 “Terminal Free” is mapped to a SIP 183 (Session Progress) response.

5.2 GENERAL REQUIREMENTS

An ATS-No.5 / SIP gateway **SHALL** support ATS-No.5 in accordance with [29] as a gateway VCS and **SHALL** support SIP in accordance with RFC 3261 [8] as a UA. In particular, the gateway **SHALL** support SIP syntax and encoding, the SIP transport layer and the SIP transaction layer in accordance with RFC 3261. In addition, the gateway **SHALL** support SIP TU behaviour for a UA in accordance with RFC 3261 except where stated otherwise in this specification.

The gateway **SHALL** support SDP in accordance with RFC 4566 [25] and its use in accordance with the offer / answer model in RFC 3264 [9].

The SIP profile specified in CHAPTER 3 **SHALL** apply to the ATS-No.5 / SIP gateway.

The gateway **SHALL** support calls from ATS-No.5 to SIP and calls from SIP to ATS-No.5.

As a result of DNS look-up by the gateway in order to determine where to send a SIP INVITE request, a number of candidate destinations can be attempted in sequence. The way in which this is handled by the gateway is outside the scope of this specification. However, any behaviour specified in this specification on receipt of a SIP final response **SHOULD** apply only when there are no more candidate destinations to try.

5.3 MESSAGE MAPPING REQUIREMENTS

The SIP protocol and ATS-No.5 signalling system information flows **SHALL** be mapped to one another as specified in 5.8.

5.3.1 Call Establishment from ATS-No.5 to SIP

5.3.1.1 Receipt of an ATS-No.5 “Seizing” Line Signal and Register Signalling

On receipt of an incoming call (signified by receipt of an ATS No.5 “Seizing” line signal and Register signalling) from the ATS-No.5 part of the network, containing called party address, call priority and calling party address information that the Gateway determines to be complete, the Gateway **SHALL** map the ATS-No.5 call establishment signals to a SIP INVITE request.

The Gateway **SHALL** generate the SIP Request-URI, To and From fields in the SIP INVITE request and **SHALL** also include SDP information as specified in section 5.5.

On receipt of an incoming call containing addressing information that the Gateway determines to be incomplete, or a protocol error during call establishment occurs, the Gateway **SHALL** initiate ATS-No.5 call clearing procedures as specified in [29].

5.3.1.2 Receipt of SIP 100 (Trying) Response

A SIP 100 response **SHALL NOT** trigger any ATS-No.5 signal. It only serves the purpose of suppressing INVITE request retransmissions.

5.3.1.3 Receipt of 18x Provisional Response

The Gateway **SHALL** map a received SIP 18x response to an ATS-No.5 status signal no. 6 “Terminal Free” and supply Ringing tone on the inter-VCS link.

5.3.1.4 Receipt of SIP 2xx Response

If the Gateway receives a SIP 200 (OK) response as the first SIP 200 response to a SIP INVITE request, the Gateway **SHALL** map the SIP 200 (OK) response to an ATS-No.5 “Answer” line signal. The Gateway **SHALL** also send a SIP ACK request to acknowledge the SIP 200 (OK) response. The Gateway **SHALL NOT** include any SDP information in the SIP ACK request. If the Gateway receives further SIP 200 (OK) responses, it **SHALL** respond to each in accordance with RFC 3261 [8] and **SHALL NOT** generate any further ATS-No.5 signals.

Media streams will normally have been established in the IP network in each direction. If so, the Gateway **SHALL** connect the media streams to the inter-VCS link if it has not already done so and stop any local Ringing tones.

If the Gateway receives a SIP 2xx response other than SIP 200 (OK), the Gateway **SHALL** send a SIP ACK request and **SHALL NOT** generate any ATS-No.5 signal.

5.3.1.5 Receipt of SIP 3xx Response

On receipt of a SIP 3xx response, the Gateway **SHALL** act in accordance with RFC 3261 [8].

No ATS-No.5 signal **SHALL** be sent.

5.3.2 Call Establishment from SIP to ATS-No.5

5.3.2.1 Receipt of SIP INVITE Request for a New Call

On receipt of a SIP INVITE request for a new call from the IP network, the Gateway **SHALL** attempt to establish a call towards the ATS-No.5 network applying the requirements of [29] by sending an ATS-No.5 Seizing line signal and Register signalling from the received SIP INVITE request. The Gateway **SHALL** also send a SIP 100 (Trying) response.

If no suitable circuit is available the Gateway **SHALL** send a SIP 503 (Service Unavailable) response.

If information in the SIP INVITE request is unsuitable for generating Called party number and Calling party number, the Gateway **SHALL NOT** issue an ATS-No.5 Seizing line signal and **SHALL** send a SIP 500 (Server internal error) response.

If the SIP INVITE request does not contain SDP information, the Gateway **SHALL NOT** issue an ATS-No.5 Seizing line signal and **SHALL** send a SIP 488 (Not Acceptable Here) response.

5.3.2.2 Receipt of ATS-No.5 Status Signal no. 6 “Terminal Free”

The Gateway **SHALL** map an ATS-No.5 Status Signal no. 6 “Terminal Free” to a SIP 180 (Ringing) provisional response for the SIP INVITE request.

5.3.2.3 Receipt of ATS-No.5 Answer Line Signal

The Gateway **SHALL** map an ATS-No.5 Answer line signal to a SIP 200 (OK) final response for the SIP INVITE request.

5.3.2.4 Receipt of SIP ACK Request

The receipt of a SIP ACK request **SHALL** map an ATS-No.5 Answer Ack line signal.

If the SIP ACK contains SDP answer information, the Gateway **SHALL** connect the media streams to the corresponding inter-VCS link if it has not already done so.

5.3.3 Call Clearing and Call Failure

5.3.3.1 Receipt of an ATS-No.5 Clear-Forward Line Signal

On receipt of an ATS-No.5 Clear-Forward line signal, the Gateway behaviour shall depend on the state of call establishment.

1. If the Gateway has received a SIP 200 (OK) response and sent a SIP ACK request, the Gateway **SHALL** send a SIP BYE request and an ATS-No.5 Release Guard line signal to clear the call.

2. If the Gateway has sent a SIP INVITE request and received a SIP provisional response but not a SIP final response, the Gateway **SHALL** send a SIP CANCEL request and an ATS-No.5 Release Guard line signal to clear the call.

   Note 27.
   In accordance with RFC 3261 [8], if after sending a SIP CANCEL request a SIP 2xx response is received to the SIP INVITE request, the Gateway shall need to send a SIP BYE request.

3. If the Gateway has sent a SIP INVITE request but received no SIP response, the Gateway **SHALL** send an ATS-No.5 Release Guard line signal but **SHALL NOT** send a SIP message. If a SIP final or provisional response is subsequently received, the Gateway **SHALL** then act in accordance with 1 or 2 above respectively.

5.3.3.2 Receipt of an ATS-No.5 Clear-Back Line Signal

On receipt of an ATS-No.5 Clear-Back line signal, the Gateway behaviour shall depend on the state of call establishment.

1. If the Gateway has sent a SIP 200 (OK) response and received a SIP ACK request, the Gateway **SHALL** send a SIP BYE request and ATS-No.5 Clear-Back-Ack and Clear-Forward line signals to clear the call.

2. If the Gateway has sent a SIP 200 (OK) response (indicating that call establishment is complete) but has not received a SIP ACK request, the Gateway **SHALL** wait until a SIP ACK is received and then send a SIP BYE request and ATS-No.5 Clear-Back-Ack and Clear-Forward line signals to clear the call.

3. If the Gateway has received a SIP INVITE request but not sent a SIP final response, the Gateway **SHALL** send a SIP final response chosen according to the cause value in the received ATS-No.5 Status Signal as specified in Table 17. SIP response 500 (Server internal error) **SHALL** be used as the default for cause values not shown in Table 17.

In all cases the Gateway **SHALL** also disconnect media streams, if established, and allow ATS-No.5 and SIP signalling to complete in accordance with [29] and [8] respectively.

ATS-No.5 Status Number	ATS-No.5 Status Information	SIP Response Code	SIP Response Description
3	Terminal Busy	486	Busy Here
5	Terminal out of service	404	Not Found
8	Trunk congestion	503	Service Unavailable

Table 17 – Mapping of ATS-No.5 Status Signal to SIP Error Response

ATS-No.5 Line Signal	SIP Response Code	SIP Response Description
Clear Back (instead of a Status Signal)	500	Server Internal Error

Table 18 – Mapping of ATS-No.5 Clear-Back Line Signal to SIP Error Response

5.3.3.3 Receipt of a SIP BYE Request

On receipt of a SIP BYE request, the Gateway **SHALL** send an ATS-No.5 Clear Forward or Clear Back line signal, depending upon the call establishment scenario (outgoing gateway call or incoming gateway call, respectively). The Gateway **SHALL** also disconnect media streams, if established, and allow ATS-No.5 and SIP signalling to complete in accordance with [29] and [8], respectively.

> ***Note 28.***
> *When responding to a SIP BYE request, in accordance with RFC 3261 [8] the Gateway is also required to respond to any other outstanding transactions, e.g., with a SIP 487 (Request Terminated) response. This applies in particular if the Gateway has not yet returned a final response to the SIP INVITE request.*

5.3.3.4 Receipt of a SIP CANCEL Request

On receipt of a SIP CANCEL request to clear a call for which the Gateway has not sent a SIP final response to the received SIP INVITE request, the Gateway **SHALL** send an ATS-No.5 Clear Forward line signal. The Gateway **SHALL** also disconnect media streams, if established, and allow ATS-No.5 and SIP signalling to complete in accordance with [29] and [8], respectively.

5.3.3.5 Receipt of a SIP 4xx - 6xx Response

On receipt of a SIP final response (4xx-6xx) to a SIP INVITE request, the Gateway **SHALL** transmit an ATS-No.5 Status signal or a Clear Back line signal derived from the SIP 4xx-6xx response according to Table 19 and Table 20. The ATS-No.5 Clear Back line signal **SHALL** be used as the default for SIP responses not shown in Table 19 and Table 20. The Gateway **SHALL** also disconnect media streams, if established, and allow ATS-No.5 and SIP signalling to complete in accordance with [29] and [8], respectively.

	SIP Response Code	SIP Response Description	ATS-No.5 Status Number	ATS-No.5 Status Information
Client Error	404	Not Found	5	Terminal out of service or Called Number not allocated
	405	Method Not Allowed	5	Terminal out of service
	406	Not Acceptable	5	Terminal out of service
	410	Gone	5	Terminal out of service or Called Number not allocated
	415	Unsupported Media Type	5	Terminal out of service
	480	Temporarily not available	3	Terminal Busy
	484	Address Incomplete	5	Terminal out of service or Called Number not allocated
	485	Ambiguous	5	Terminal out of service or Called Number not allocated
	486	Busy Here	3	Terminal Busy
	488	Not Acceptable Here	5	Terminal out of service
Server Error	501	Not Implemented	5	Terminal out of service
	503	Service Unavailable	8	Trunk congestion
Global	600	Busy Everywhere	3	Terminal Busy

	SIP Response Code	SIP Response Description	ATS-No.5 Status Number	ATS-No.5 Status Information
Failure	603	Decline	3	Terminal Busy
	604	Does not exist anywhere	5	Terminal out of service or Called Number not allocated
	606	Not Acceptable	5	Terminal out of service

Table 19 – Mapping of SIP Error Response to ATS-No.5 Status Signal

	SIP Response Code	SIP Response Description	ATS-No.5 Line Signal
Client Error	400	Bad Request	Clear Back
	401	Unauthorized	Clear Back
	402	Payment Required	Clear Back
	403	Forbidden	Clear Back
	407	Proxy Authentication Required	Clear Back
	408	Request Timeout	Clear Back
	413	Request Entity Too Large	Clear Back
	414	Request-URI Too Large	Clear Back
	416	Unsupported URI Scheme	Clear Back
	420	Bad Extension	Clear Back
	421	Extension Required	Clear Back
	423	Interval Too Brief	Clear Back
	481	Call Leg/Transaction Does Not Exist	Clear Back
	482	Loop Detected	Clear Back
	483	Too Many Hops	Clear Back
	487	Request Terminated	---
	491	Request Pending	Clear Back
	493	Undecipherable	Clear Back
Server Error	500	Server Internal Error	Clear Back
	502	Bad Gateway	Clear Back
	504	Server Time-out	Clear Back
	505	SIP Version not supported	Clear Back
	513	Message Too Large	Clear Back

Table 20 – Mapping of SIP Error Response to ATS-No.5 Clear Back Line Signal

5.3.3.6 Gateway-Initiated Call Clearing

If the Gateway initiates clearing of an ATS-No.5 call due to ATS-No.5 timer expiry or ATS-No.5 protocol error in accordance with [29], the Gateway **SHALL** also initiate clearing of the associated SIP call in accordance with subclause 5.3.3.1. If this involves the sending of a final response to a SIP INVITE request, the Gateway **SHALL** use response code 408 (Request timeout) or 500 (Server internal error) as appropriate.

If the Gateway initiates clearing of a SIP call owing to SIP timer expiry or SIP protocol error in accordance with RFC 3261 [8], the Gateway **SHALL** also initiate clearing of the associated ATS-No.5 call in accordance with [29].

5.3.4 Request to Change Media Characteristics

If after a call has been successfully established the Gateway receives a SIP INVITE request to change the media characteristics of the call in a way that would be incompatible with voice use, the Gateway **SHALL** send back a SIP 503 (Service unavailable) response and **SHALL NOT** change the media characteristics of the existing call.

5.4 NUMBER MAPPING

In ATS-No.5, users are identified by numbers according to a closed numbering scheme.

In SIP, users are identified by Universal Resource Identifiers (URIs) conveyed within the Request-URI and various headers, including the From and To headers specified in RFC 3261 [8].

5.4.1 Mapping from ATS-No.5 to SIP

5.4.1.1 Using information from the ATS-No.5 Called Party Number

When mapping ATS-No.5 call setup signals to a SIP INVITE request, the Gateway **SHALL** convert the Called party number to a URI and include that URI in the SIP Request-URI and in the To header fields.

On receipt of an incoming call containing addressing information that the Gateway determines to be complete but no URI is derivable, the Gateway **SHALL** send an ATS-No.5 Status Signal no. 5 “Terminal out of service” and **SHALL NOT** send any SIP request.

5.4.1.2 Using information from the ATS-No.5 Calling Party Number

When mapping ATS-No.5 call setup signals to a SIP INVITE request, the Gateway **SHALL** convert the Calling party number to a URI and include that URI in the SIP From header field.

If no URI is derivable, the Gateway **SHALL** include its own URI in the SIP From header field.

5.4.2 Mapping from SIP to ATS-No.5

When mapping a SIP INVITE request to ATS-No.5 call setup signals, the Gateway **SHALL** convert the To header field to a Called party number and the From header field to a Calling party number.

If either Called or Calling party numbers are not derivable, the Gateway **SHALL** send a SIP response 500 (Server internal error) and **SHALL NOT** send any ATS-No.5 signal.

5.5 MEDIA TYPE IN SDP

The Gateway **SHALL** generate SDP information to include in the SIP INVITE request. The media type included in the SDP information **SHALL** be “audio”.

5.6 REQUIREMENTS FOR SUPPORT OF SUPPLEMENTARY SERVICES

A Gateway **SHALL** support the Priority Call Interruption and the Priority Call Intrusion supplementary services.

5.6.1 Call Priority

5.6.1.1 Mapping at an Incoming Gateway

On receipt of a call from the ATS-No.5 network, the Gateway **SHALL** read the ATS-No.5 Priority digit of the call and include the relevant Priority and Max-Forwards header fields in the SIP INVITE message with values as specified in Table 21.

Gateway Input: ATS-No.5 Priority Digit	Gateway Output		Call Type
	SIP Priority Header Field	SIP Max-Forwards Header Field	
1 or 6	emergency	< 20	Priority call
2 or 7	urgent	< 20	Tactical Routine call
3 or 8	normal	< 20	Strategic Routine call
4 or 9	non-urgent	< 20	General Purpose Routine call

Table 21 – Mapping of ATS-No.5 Priority Digit to SIP Header Fields

A VCS **SHOULD** provide a management means of configuring the acceptable (network dependent) Max-Forwards initial value. Nevertheless, it is **RECOMMENDED** that the initial value for the Max-Forwards header field is less than 20.

5.6.1.2 Mapping at an Outgoing Gateway

On receipt of a SIP INVITE message from the IP network, the Outgoing Gateway **SHALL** map the Priority header field of the SIP INVITE message to the ATS-No.5 priority digit as specified in Table 22.

Gateway Input SIP Priority Header Field	Gateway Output ATS-No.5 Priority Digit	Call Type
emergency	1 or 6	Priority call
urgent	2 or 7	Tactical Routine call
normal	3 or 8	Strategic Routine call
non-urgent	4 or 9	General Purpose Routine call

Table 22 – Mapping of SIP Priority Header Field to ATS-No.5 Priority Digit

Priority Digit values 1-4 **SHALL** be used when the call is routed on a direct route, and values 6-9 **SHALL** be used when detour is necessary since the direct route is occupied.

5.6.2 Priority Call Interruption

Priority Call Interruption is subject to the following restrictions:

1. A Priority call **SHALL NOT** be interrupted from any end;

2. A Routine call **MAY** be interrupted from the ATS-No.5 End or by the Gateway when congestion exists.

5.6.2.1 Priority Call Interruption from SIP to ATS-No.5

On receipt of an INVITE(emergency) request from the IP network, and all available inter-VCS ATS-No.5 links being occupied, the Gateway **SHALL** attempt to establish a priority call towards the ATS-No.5 network as specified in [29]. The priority call **SHOULD** interrupt an established routine call (should one exist), thus allowing the priority call to proceed. If interruption is not possible (because all established calls are priority calls), the Gateway **SHALL** attempt to establish the call using a detour route. If interruption is possible, the Gateway **SHALL** select an ATS-No.5 call with the lowest Priority level. Before the established routine call is interrupted, all parties engaged in that call **SHALL** receive an interrupt warning tone.

Having selected the call to be interrupted, applying a “Priority Level Interruption Order” principle, the Gateway **SHALL** inform the users that their call is to be released; it **SHALL** inject an audible tone (Interrupt warning tone) into the voice path to each user in the call and start the “Interrupt pending” timer.

If another circuit becomes available prior to the expiry of the “Interrupt pending“ timer, the call interruption **SHALL** be abandoned. The Gateway **SHALL** stop injecting the “Interrupt Warning” tone in the voice path to each user in the call. It **SHALL** then proceed with establishment of the priority call using this available circuit.

On expiry of the Interruption pending timer, the Gateway **SHALL** stop injecting the “Interrupt warning” tone in the voice path to the users and then force release the call to be interrupted by sending an ATS-No.5 “Blocking” line signal on the ATS-No.5 line.

On termination of the ATS-No.5 Blocking line signal, the Gateway **SHALL** continue establishment of the priority call using the newly available ATS-No.5 circuit. The Gateway **SHALL** map the SIP(emergency) call to an ATS-No.5 call with the highest priority as specified in Table 22 above.

5.6.2.2 Priority Call Interruption from ATS-No.5 to SIP

On receipt of an ATS-No.5 Blocking line signal, after a previous Interrupt warning tone injected by the ATS-No.5 End into the voice path, the Gateway **SHALL** send a SIP BYE request containing a Text/Plain body indicating “Emergency - Forced Release” to the SIP End and a Release Guard line signal to the ATS-No.5 End.

5.6.3 Priority Call Intrusion

5.6.3.1 Priority Call Intrusion from ATS-No.5 to SIP

An ATS-No.5 priority call (having Priority level 1 or 6) made towards the Gateway **SHALL** cause Gateway SIP User Agent to send a SIP INVITE request with a Priority header defined as “emergency” to distinguish it from a routine call (with Priority header defined as “urgent”, “normal” or “non-urgent”).

The Wanted SIP_End User Agent, receiving an INVITE(emergency), acts as the focus for a conference (use is made of the “isfocus” feature defined in RFC 3840 [16] to create a conference media session).

On receiving a SIP 182 (Queued) provisional response, the Gateway **SHALL NOT** send any ATS-No.5 signals towards the ATS-No.5 End.

On receiving a SIP 180 (Ringing) or 183 (Intrusion progress) provisional response, the Gateway **SHALL** send an ATS-No.5 status signal no. 6 “Terminal Free” to the ATS-No.5 End. Should no early media be present, Ringing tone **SHALL** be injected by the Gateway. Upon receipt of a 200 (OK) final response, the Gateway **SHALL** stop injection of the Ringing tone, if no early media was present, an send the ATS-No.5 Answer line signal to the ATS-No.5 End.

On receiving a SIP 200 (OK) final response without having received a previous SIP 183 (Intrusion in progress) provisional response, the Gateway **SHALL** send an ATS-No.5 status signal no. 6 “Terminal Free” and an ATS-No.5 Answer line signal to the ATS-No.5 End.

> ***Note 29.***
> *SIP 180 (Ringing), SIP 183 (Intrusion in progress) and/or 200 (OK) responses have to be received within P22 (12s), in accordance with [29]. This implies T1 (see paragraph 3.8.8) < P22.*

If intrusion is forbidden (i.e. CIPL on), the Wanted user reply is SIP 180 (Ringing). Wanted and Unwanted users remain connected; Call from ATS-No.5 user is displayed at user’s terminal as Priority Call and can be manually answered. On receiving a SIP 180 (Ringing) response, the Gateway **SHALL** send an ATS-No.5 status signal no. 6 “Terminal Free” and supply Ringing tone to the ATS-No.5 End.

5.6.3.2 Priority Call Intrusion from SIP to ATS-No.5

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-No.5 network. The Gateway **SHALL** be configured to operate with T1 = 0 and automatic Priority call answer.

On receipt of ATS-No.5 status signal no. 6 “Terminal Free”, the Gateway **SHALL** send a SIP 180 (Ringing) provisional response towards the IP-network.

On receipt of an ATS-No.5 Answer line signal, the Gateway **SHALL** send a SIP 200 (OK) response towards the IP-network and an ATS-No.5 Answer Ack line signal toward the ATS-No-5 network. Other possible ATS-No.5 responses **SHALL** be handled in accordance with subclause 5.3.3.

Once a priority call is answered, 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.

5.7 AUDIBLE TONES

A gateway **SHALL** be capable of generating the audible tones indicated in Table 23 and sending them over the ATS-No.5 line.

Audible Tone	Purpose	Tone generated upon receipt of:
Ringing	Sent by the gateway towards the ATS-No.5 network after successful call establishment and prior to call acceptance by the called user.	• 180 (Ringing) response
Interrupt warning	Injected into the voice path to warn a user of the imminent priority interruption of an established call. This signal is sent by the gateway that is handling the call interruption over the inter-VCS link.	• A Priority (emergency) call that has to interrupt an established Routine call

Table 23 – ATS-No.5 Audible Tones Transmitted on Line

5.8 MESSAGE SEQUENCE CHARTS

The paragraphs below show some typical message sequences that can occur for an interworking between ATS-No.5 and SIP.

The Message Sequence Charts (MSC) in figures below show the information flows between the Call Control entity (labelled "Gateway") and respective Protocol Control entities for each signalling system (labelled "ATS-No.5_End" and "SIP_End") within a Gateway VCS. Each information flow is named according to the corresponding message or signal sent to or received from a peer VCS.

Dashed lines (---) represent signalling messages that are mandatory to the call scenario. These messages can be SIP or ATS-No.5 signalling. The arrow indicates the direction of message flow.

Double dashed lines (===) represent media paths between network elements.

5.8.1 Successful ATS-No.5 to SIP Routine Call

The MSC shown below is a typical message sequence for a successful call setup of an incoming routine call (to a gateway) from a route employing the ATS-No.5 signalling system which is routed on an IP network to the called user employing SIP.

```
ATS-No.5_End       Gateway          Proxy          SIP_End
     |                |                |                |
     | SEIZING signal |                |                |
     |--------------->|                |                |
     |PROCEED-TO-SEND |                |                |
     |<---------------|                |                |
     |       KP       |                |                |
     |--------------->|                |                |
     |    MF Digit    |                |                |
     |--------------->|                |                |
     |       .        |                |                |
     |       .        |                |                |
     |Priority(2,3,4,7,8 or 9)         |                |
     |--------------->|                |                |
     |       .        |                |                |
     |       .        |                |                |
     |    MF Digit    |                |                |
     |--------------->|                |                |
     |       ST       |                |                |
   A |--------------->|     INVITE     |                |
   | |                |--------------->|     INVITE     |
   | |                |   100 Trying   |--------------->|
  P22|                |<---------------|  180 Ringing   |
   | |                |  180 Ringing   |<---------------|
   | |"Terminal Free" |<---------------|                |
   V |<---------------|                |                |
     | (One Way Voice)|                |                |
     |<===============|                |                |
     |  Ringing Tone  |                |                |
     |                |                |     200 OK     |
     |                |     200 OK     |<---------------|
     |     ANSWER     |<---------------|                |
     |<---------------|      ACK       |                |
     |   ANSWER ACK   |--------------->|      ACK       |
     |--------------->|                |--------------->|
     |(Both Way Voice)|       (Both Way RTP Media)      |
     |<==============>|<===============================>|
     |                |                |                |
```

Fig. 69 – Successful ATS-No.5 to SIP Routine Call

Note 30.
For a routine call, the value of the “Priority” header field in the INVITE method shall be “urgent”, “normal” or “non-urgent”, as indicated in Table 21.

Note 31.
As defined in [29], “P22” is the time interval between the end of the ST signal and the start of a register type Status signal for general transit interworking; its time-out is 12 s.

5.8.2 Successful SIP to ATS-No.5 Routine Call

This is a typical message sequence for a successful call setup of an incoming routine call (to a gateway) from an IP network employing SIP which is routed on a route employing the ATS-No.5 signalling system.

```
ATS-No.5_End        Gateway           Proxy           SIP_End
      |                |                |                |
      |                |                |     INVITE     |
      |                |     INVITE     |<---------------|
      | SEIZING signal |<---------------|   100 Trying   |
      |<---------------|   100 Trying   |--------------->|
      |PROCEED-TO-SEND |--------------->|                |
      |--------------->|                |                |
      |       KP       |                |                |
      |<---------------|                |                |
      |    MF Digit    |                |                |
      |<---------------|                |                |
      |       .        |                |                |
      |       .        |                |                |
      |Priority(2,3,4,7,8 or 9)         |                |
      |<---------------|                |                |
      |       .        |                |                |
      |       .        |                |                |
      |   MF Digit     |                |                |
      |<---------------|                |                |
      |       ST       |                |                |
    A |<---------------|                |                |
    | |                |                |                |
T9/P22                 |                |                |
    | | “Terminal Free”|                |                |
    V |--------------->|  180 Ringing   |                |
      |                |--------------->|  180 Ringing   |
      |                |                |--------------->|
      |                |                |                |
      |                |                |                |
      |     ANSWER     |                |                |
      |--------------->|     200 OK     |                |
      |   ANSWER ACK   |--------------->|     200 OK     |
      |<---------------|                |--------------->|
      |                |                |      ACK       |
      |                |      ACK       |<---------------|
      |                |<---------------|                |
      |(Both Way Voice)|       (Both Way RTP Media)      |
      |<==============>|<===============================>|
      |                |                |                |
```

Fig. 70 – Successful SIP to ATS-No.5 Routine Call

Note 32.
For a routine call, the ATS-No.5 Priority Digit value shall be 2, 3, 4, 7, 8 or 9, as indicated in Table 22.

Note 33.
As defined in [29], “T9” is the time interval between the end of the ST signal and the start of a register type Status signal; its time-out is 2 s. “T9” is equivalent to “P22” for general transit interworking; their time-out in this case is 12 s.

5.8.3 Normal Call Clearing from ATS-No.5 End

The MSCs below represent two typical message sequences for Call Clearing from ATS-No.5 to SIP subsequent to call establishment; the first one is for an Incoming gateway call scenario (call was made from ATS-No.5 to SIP), and the second one is for an Outgoing gateway call scenario (call was made from SIP to ATS-No.5).

```
ATS-No.5_End      Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|       (Both Way RTP Media)      |
    |<==============>|<===============================>|
    |                |                |                |
    | CLEAR FORWARD  |                |                |
    |--------------->|      BYE       |                |
    |  RELEASE GUARD |--------------->|      BYE       |
    |<---------------|                |--------------->|
    |                |                |     200 OK     |
    |                |     200 OK     |<---------------|
    |                |<---------------|                |
    |                |                |                |
```

Fig. 71 – Normal Call Clearing from ATS-No.5 End (Incoming Gateway Call)

```
ATS-No.5_End      Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|       (Both Way RTP Media)      |
    |<==============>|<===============================>|
    |                |                |                |
    |   CLEAR BACK   |                |                |
    |--------------->|                |                |
    | CLEAR BACK ACK |                |                |
    |<---------------|                |                |
    | CLEAR FORWARD  |                |                |
    |<---------------|      BYE       |                |
    |  RELEASE GUARD |--------------->|      BYE       |
    |--------------->|                |--------------->|
    |                |                |     200 OK     |
    |                |     200 OK     |<---------------|
    |                |<---------------|                |
    |                |                |                |
```

Fig. 72 – Normal Call Clearing from ATS-No.5 End (Outgoing Gateway Call)

5.8.4 Normal Call Clearing from SIP End

The MSCs below represent two typical message sequences for Call Clearing from SIP to ATS-No.5 subsequent to call establishment; the first one is for an Incoming gateway call scenario (call was made from ATS-No.5 to SIP), and the second one is for an Outgoing gateway call scenario (call was made from SIP to ATS-No.5).

```
ATS-No.5_End       Gateway           Proxy            SIP_End
     |                 |                 |                 |
     |(Both Way Voice) |       (Both Way RTP Media)        |
     |<===============>|<=================================>|
     |                 |                 |                 |
     |                 |                 |       BYE       |
     |                 |       BYE       |<----------------|
     |   CLEAR BACK    |<----------------|                 |
     |<----------------|     200 OK      |                 |
     | CLEAR BACK ACK  |---------------->|     200 OK      |
     |---------------->|                 |---------------->|
     | CLEAR FORWARD   |                 |                 |
     |---------------->|                 |                 |
     |  RELEASE GUARD  |                 |                 |
     |<----------------|                 |                 |
     |                 |                 |                 |
```

Fig. 73 – Normal Call Clearing from SIP End (Incoming Gateway Call)

```
ATS-No.5_End       Gateway           Proxy            SIP_End
     |                 |                 |                 |
     |(Both Way Voice) |       (Both Way RTP Media)        |
     |<===============>|<=================================>|
     |                 |                 |                 |
     |                 |                 |       BYE       |
     |                 |       BYE       |<----------------|
     | CLEAR FORWARD   |<----------------|                 |
     |<----------------|     200 OK      |                 |
     |  RELEASE GUARD  |---------------->|     200 OK      |
     |---------------->|                 |---------------->|
     |                 |                 |                 |
```

Fig. 74 – Normal Call Clearing from SIP End (Outgoing Gateway Call)

5.8.5 Unsuccessful ATS-No.5 to SIP Call

5.8.5.1 Call Clearing from ATS-No.5 End

This is a typical message sequence for Call Clearing from ATS-No.5 to SIP during establishment of a call from ATS-No.5 to SIP, in which the Gateway has received a provisional response (1xx) to the INVITE request but not a final response (2xx, 3xx, 4xx, 5xx, 6xx).

```
ATS-No.5_End       Gateway            Proxy           SIP_End
    |                 |                 |                 |
    | SEIZING signal  |                 |                 |
    |---------------->|                 |                 |
    |PROCEED-TO-SEND  |                 |                 |
    |<----------------|                 |                 |
    |       KP        |                 |                 |
    |---------------->|                 |                 |
    |    MF Digit     |                 |                 |
    |---------------->|                 |                 |
    |       .         |                 |                 |
    |       .         |                 |                 |
    |       .         |                 |                 |
    |    MF Digit     |                 |                 |
    |---------------->|                 |                 |
    |       ST        |                 |                 |
    |---------------->|     INVITE      |                 |
    |                 |---------------->|     INVITE      |
    |                 |   100 Trying    |---------------->|
    |                 |<----------------|   180 Ringing   |
    |                 |   180 Ringing   |<----------------|
    |"Terminal Free"  |<----------------|                 |
    |<----------------|                 |                 |
    |                 |                 |                 |
    | CLEAR FORWARD   |                 |                 |
    |---------------->|     CANCEL      |                 |
    |  RELEASE GUARD  |---------------->|     CANCEL      |
    |<----------------|     200 OK      |---------------->|
    |                 |<----------------|     200 OK      |
    |                 |                 |<----------------|
    |                 |                 |487 Req. Termin. |
    |                 |487 Req. Termin. |<----------------|
    |                 |<----------------|       ACK       |
    |                 |       ACK       |---------------->|
    |                 |---------------->|                 |
    |                 |                 |                 |
```

Fig. 75 – Unsuccessful ATS-No.5 to SIP Call. Call Clearing from ATS-No.5 End

5.8.5.2 Call Clearing from SIP Network

5.8.5.2.1 Mapping of SIP Response to ATS-No.5 Status Signal

This is a typical message sequence for Call Clearing from SIP to ATS-No.5 during establishment of a call from ATS-No.5 to SIP, in which the Gateway has not previously received a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and the SIP response can be mapped to one of these ATS-No.5 Status Signals: “Terminal Busy”, “Terminal Out of Service”, or “Trunk Congestion”.

```
ATS-No.5_End      Gateway           Proxy           SIP_End
    |                 |                 |                 |
    | SEIZING signal  |                 |                 |
    |---------------->|                 |                 |
    |PROCEED-TO-SEND  |                 |                 |
    |<----------------|                 |                 |
    |       KP        |                 |                 |
    |---------------->|                 |                 |
    |    MF Digit     |                 |                 |
    |---------------->|                 |                 |
    |       .         |                 |                 |
    |       .         |                 |                 |
    |       .         |                 |                 |
    |    MF Digit     |                 |                 |
    |---------------->|                 |                 |
    |       ST        |                 |                 |
    |---------------->|     INVITE      |                 |
    |                 |---------------->|     INVITE      |
    |                 |   100 Trying    |---------------->|
    |                 |<----------------|  486 Busy Here  |
    |                 |                 |<----------------|
    |                 |                 |       ACK       |
    |                 |  486 Busy Here  |---------------->|
    |                 |<----------------|                 |
    |                 |       ACK       |                 |
    |”Terminal Busy”  |---------------->|                 |
    |<----------------|                 |                 |
    | CLEAR FORWARD   |                 |                 |
    |---------------->|                 |                 |
    |  RELEASE GUARD  |                 |                 |
    |<----------------|                 |                 |
    |                 |                 |                 |
```

Fig. 76 – Unsuccessful ATS-No.5 to SIP Call.
Mapping of SIP Response to ATS-No.5 Status Signal

5.8.5.2.2 Mapping of SIP Error Response to ATS-No.5 Clear Back Line Signal

This is a typical message sequence for Call Clearing from SIP to ATS-No.5 during establishment of a call from ATS-No.5 to SIP, in which the Gateway has not previously received a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request and the SIP response cannot be mapped to one of the ATS-No.5 Status Signals.

```
ATS-No.5_End      Gateway            Proxy            SIP_End
    |                |                 |                 |
    | SEIZING signal |                 |                 |
    |--------------->|                 |                 |
    |PROCEED-TO-SEND |                 |                 |
    |<---------------|                 |                 |
    |       KP       |                 |                 |
    |--------------->|                 |                 |
    |    MF Digit    |                 |                 |
    |--------------->|                 |                 |
    |       .        |                 |                 |
    |       .        |                 |                 |
    |       .        |                 |                 |
    |    MF Digit    |                 |                 |
    |--------------->|                 |                 |
    |       ST       |                 |                 |
    |--------------->|     INVITE      |                 |
    |                |---------------->|     INVITE      |
    |                |   100 Trying    |---------------->|
    |                |<----------------|     4xx-6xx     |
    |                |     4xx-6xx     |<----------------|
    |   CLEAR BACK   |<----------------|       ACK       |
    |<---------------|       ACK       |---------------->|
    | CLEAR BACK ACK |---------------->|                 |
    |--------------->|                 |                 |
    | CLEAR FORWARD  |                 |                 |
    |--------------->|                 |                 |
    |  RELEASE GUARD |                 |                 |
    |<---------------|                 |                 |
    |                |                 |                 |
```

Fig. 77 – Unsuccessful ATS-No.5 to SIP Call.
Mapping of SIP Error Response to ATS-No.5 Clear Back Line Signal

5.8.6 Unsuccessful SIP to ATS-No.5 Call

5.8.6.1 Call Clearing from SIP Network

This is a typical message sequence for Call Clearing from SIP to ATS-No.5 during establishment of a call from SIP to ATS-No.5, in which the Gateway has sent a provisional response (1xx) to the INVITE request but not a final response (2xx, 3xx, 4xx, 5xx, 6xx).

```
ATS-No.5_End       Gateway           Proxy           SIP_End
    |                 |                 |                 |
    |                 |                 |     INVITE      |
    |                 |     INVITE      |<----------------|
    | SEIZING signal  |<----------------|   100 Trying    |
    |<----------------|   100 Trying    |---------------->|
    |PROCEED-TO-SEND  |---------------->|                 |
    |---------------->|                 |                 |
    |       KP        |                 |                 |
    |<----------------|                 |                 |
    |    MF Digit     |                 |                 |
    |<----------------|                 |                 |
    |    MF Digit     |                 |                 |
    |<----------------|                 |                 |
    |        .        |                 |                 |
    |        .        |                 |                 |
    |        .        |                 |     CANCEL      |
    |                 |     CANCEL      |<----------------|
    |  CLEAR FORWARD  |<----------------|     200 OK      |
    |<----------------|     200 OK      |---------------->|
    |  RELEASE GUARD  |---------------->|                 |
    |---------------->|487 Req. Termin.|                 |
    |                 |---------------->|487 Req. Termin.|
    |                 |       ACK       |---------------->|
    |                 |<----------------|       ACK       |
    |                 |                 |<----------------|
    |                 |                 |                 |
```

**Fig. 78 – Unsuccessful SIP to ATS-No.5 Call.
Call Clearing from SIP Network**

5.8.6.2 Call Clearing from ATS-No.5 End

5.8.6.2.1 Receipt of Status Signal from ATS-No.5 End

This is a typical message sequence for Call Clearing from ATS-No.5 to SIP during establishment of a call from SIP to ATS-No.5, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and one of these ATS-No.5 Status Signals is received: “Terminal Busy”, “Terminal Out of Service”, or “Trunk Congestion”.

```
ATS-No.5_End       Gateway            Proxy           SIP_End
    |                |                |                |
    |                |                |     INVITE     |
    |                |     INVITE     |<---------------|
    | SEIZING signal |<---------------|   100 Trying   |
    |<---------------|   100 Trying   |--------------->|
    |PROCEED-TO-SEND |--------------->|                |
    |--------------->|                |                |
    |       KP       |                |                |
    |<---------------|                |                |
    |    MF Digit    |                |                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |   MF Digit     |                |                |
    |<---------------|                |                |
    |       ST       |                |                |
    |<---------------|                |                |
    |                |                |                |
    |                |                |                |
    | “Terminal Busy”|                |                |
    |--------------->| 486 Busy Here  |                |
    |  CLEAR FORWARD |--------------->|                |
    |<---------------|      ACK       |                |
    |  RELEASE GUARD |<---------------| 486 Busy Here  |
    |--------------->|                |--------------->|
    |                |                |      ACK       |
    |                |                |<---------------|
    |                |                |                |
```

Fig. 79 – Unsuccessful SIP to ATS-No.5 Call. Receipt of Status Signal from ATS-No.5 End

5.8.6.2.2 Receipt of Clear Back Line Signal from ATS-No.5 End

This is a typical message sequence for Call Clearing from ATS-No.5 to SIP during establishment of a call from SIP to ATS-No.5, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and the ATS-No.5 Clear Back line signal is received.

```
ATS-No.5_End      Gateway          Proxy           SIP_End
    |                |                |                |
    |                |                |     INVITE     |
    |                |     INVITE     |<---------------|
    | SEIZING signal |<---------------|   100 Trying   |
    |<---------------|   100 Trying   |--------------->|
    |PROCEED-TO-SEND |--------------->|                |
    |--------------->|                |                |
    |       KP       |                |                |
    |<---------------|                |                |
    |    MF Digit    |                |                |
    |<---------------|                |                |
    |    MF Digit    |                |                |
    |<---------------|                |                |
    |       .        |                |                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
    |   CLEAR BACK   |                |                |
    |--------------->|500 Svr.Int.Err.|                |
    | CLEAR BACK ACK |--------------->|500 Svr.Int.Err.|
    |<---------------|      ACK       |--------------->|
    | CLEAR FORWARD  |<---------------|      ACK       |
    |<---------------|                |<---------------|
    |  RELEASE GUARD |                |                |
    |--------------->|                |                |
    |                |                |                |
```

Fig. 80 – Unsuccessful SIP to ATS-No.5 Call.
Receipt of Release Line Signal from ATS-No.5 End

Note 34.
This scenario can arise where a protocol error is encountered by the ATS-No.5 End during call set-up.

5.8.6.3 Call Clearing from Gateway

This is a typical message sequence for Call Clearing from Gateway during establishment of a call from SIP to ATS-No.5, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and ATS-No.5 protocol error is encountered by the Gateway during call set-up.

```
ATS-No.5_End      Gateway           Proxy           SIP_End
    |                |                |                |
    |                |                |    INVITE      |
    |                |    INVITE      |<---------------|
    | SEIZING signal |<---------------|   100 Trying   |
    |<---------------|   100 Trying   |--------------->|
    |       .        |--------------->|                |
    |       .        |                |                |
    |       .        |                |                |
    |                |                |                |
    |  CLEAR FORWARD |408 Req.timeout/|                |
    |<---------------|500 Svr.Int.Err.|408 Req.timeout/|
    |  RELEASE GUARD |--------------->|500 Svr.Int.Err.|
    |--------------->|      ACK       |--------------->|
    |                |<---------------|      ACK       |
    |                |                |<---------------|
    |                |                |                |
```

**Fig. 81 – Unsuccessful SIP to ATS-No.5 Call.
Call Clearing from Gateway**

5.8.7 Interworking of Supplementary Services

5.8.7.1 Priority Call Interruption

5.8.7.1.1 ATS-No.5 to SIP Priority Call Interruption

The message sequence shown below corresponds to a scenario in which a routine (non-priority) call, established through a gateway, is interrupted by a priority call from the ATS-No.5_End.

```
ATS-No.5_End         Gateway             Proxy            SIP_End
       |                 |                 |                 |
       |(Both Way Voice)|        (Both Way RTP Media)        |
       |<===============>|<=================================>|
       |                 |                 |                 |
       | Interrupt Warning Tone (from ATS-No.5 End)          |
     A |================>|==================================>|
     | |                 |                 |                 |
     | |                 |                 |                 |
     | |                 |                 |                 |
    T19|                 |                 |                 |
     | |                 |                 |                 |
     | |                 |                 |                 |
     | |BLOCKING signal  |                 |                 |
     V |---------------->|       BYE       |                 |
       |  RELEASE GUARD  |---------------->|       BYE       |
       |<----------------|                 |---------------->|
       |                 |                 |      200 OK     |
       |                 |      200 OK     |<----------------|
       |                 |<----------------|                 |
       | SEIZING signal  |                 |                 |
       |---------------->|                 |                 |
       |PROCEED-TO-SEND  |                 |                 |
       |<----------------|                 |                 |
       |        KP       |                 |                 |
       |---------------->|                 |                 |
       |     MF Digit    |                 |                 |
       |---------------->|                 |                 |
       |        .        |                 |                 |
       |        .        |                 |                 |
       |        .        |                 |                 |
       |Priority= 1 or 6|                 |                 |
       |---------------->|                 |                 |
       |        .        |                 |                 |
       |        .        |                 |                 |
       |        .        |                 |                 |
       |                 |                 |                 |
```

Fig. 82 – ATS-No.5 to SIP Priority Call Interruption

5.8.7.1.2 ATS-No.5 to SIP Priority Call Interruption Abandon

The message sequences shown below correspond to a scenario in which during the interrupt warning period another line becomes available and has been seized for the priority call.

```
ATS-No.5_End        Gateway           Proxy           SIP_End
       |                |                |                |
       |(Both Way Voice)|       (Both Way RTP Media)      |
       |<==============>|<===============================>|
       |                |                |                |
       | Interrupt Warning Tone (from ATS-No.5 End)       |
     A |===============>|================================>|
     | |                |                |                |
     | |                |                |                |
    T19|(Interrupt Warning Tone Stopped) |                |
     | |                |                |                |
       |            (Users remain connected)              |
       |                |                |                |
```

Fig. 83 – ATS-No.5 to SIP Priority Call Interruption Abandon before the Blocking Line Signal is sent

```
   ATS-No.5_End       Gateway            Proxy           SIP_End
       |                |                |                |
       |(Both Way Voice)|       (Both Way RTP Media)      |
       |<==============>|<===============================>|
       |                |                |                |
       | Interrupt Warning Tone (from ATS-No.5 End)       |
     A |===============>|================================>|
     | |                |                |                |
     | |                |                |                |
     | |                |                |                |
    T19|                |                |                |
     | |                |                |                |
     | |                |                |                |
     | |BLOCKING signal |                |                |
     V |--------------->|      BYE       |                |
       |  RELEASE GUARD |--------------->|      BYE       |
       |<---------------|                |--------------->|
       |                |                |     200 OK     |
       |                |     200 OK     |<---------------|
       |                |<---------------|                |
       |CLEAR FORW./BACK|                |                |
       |--------------->|                |                |
       |        .       |                |                |
       |        .       |                |                |
       |        .       |                |                |
```

Fig. 84 – ATS-No.5 to SIP Priority Call Interruption Abandon while the Blocking Line Signal is being sent

Note 35.
The Clear Forward line signal is sent after the Blocking signal if the interrupted call was made from the ATS-No.5 End; otherwise, the Clear back signal is used.

5.8.7.1.3 SIP to ATS-No.5 Priority Call Interruption

The message sequence shown below corresponds to a scenario in which a routine (non-priority) call, established through a gateway, is interrupted by a priority call from the SIP Network.

```
ATS-No.5_End      Gateway          Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|                |                |
    |<==============>|                |                |
    |                |                |INVITE(emergency)
    |   Interrupt    |INVITE(emergency)<---------------|
    |  Warning Tone  |<---------------|   100 Trying   |
    |<===============|   100 Trying   |--------------->|
    |                |--------------->|                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    | BLOCKING signal|                |                |
    |<---------------|                |                |
    |  RELEASE GUARD |                |                |
    |--------------->|                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    | SEIZING signal |                |                |
    |<---------------|                |                |
    |PROCEED-TO-SEND |                |                |
    |--------------->|                |                |
    |       KP       |                |                |
    |<---------------|                |                |
    |    MF Digit    |                |                |
    |<---------------|                |                |
    |       .        |       .        |       .        |
    |       .        |       .        |       .        |
    |       .        |       .        |       .        |
    |                |                |                |
```

Fig. 85 – SIP to ATS-No.5 Priority Call Interruption

5.8.7.1.4 SIP to ATS-No.5 Priority Call Interruption Abandon

The message sequences shown below correspond to a scenario in which during the interrupt warning period another line becomes available and has been seized for the priority call.

```
ATS-No.5_End       Gateway           Proxy           SIP_End
      |                |                |                |
      |(Both Way Voice)|                |                |
      |<==============>|                |                |
      |                |                |INVITE(emergency)
      |   Interrupt    |INVITE(emergency)<---------------|
      |  Warning Tone  |<---------------|   100 Trying   |
      |<===============|   100 Trying   |--------------->|
      |                |--------------->|                |
      |                |                |                |
      |(Interr. Warning|                |                |
      | Tone Stopped)  |                |                |
      |                |                |                |
      |(Users remain   |                |                |
      | connected)     |                |                |
      |                |                |                |
```

Fig. 86 – SIP to ATS-No.5 Priority Call Interruption Abandon before the Blocking Line Signal is sent

```
ATS-No.5_End         Gateway           Proxy           SIP_End1
      |                |                |                |
      |(Both Way Voice)|                |                |
      |<==============>|                |                |
      |                |                |INVITE(emergency)
      |   Interrupt    |INVITE(emergency)<---------------|
      |  Warning Tone  |<---------------|   100 Trying   |
      |<===============|   100 Trying   |--------------->|
      |                |--------------->|                |
      |                |                |       SIP_End2 |
      |                |                |           |    |
      |                |                |           |    |
      |                |                |           |    |
      | BLOCKING signal|                |           |    |
      |<---------------|                |           |    |
      |  RELEASE GUARD |                |           |    |
      |--------------->|                |           |    |
      |CLEAR FORW./BACK|      BYE       |           |    |
      |<---------------|--------------->|    BYE    |    |
      |        .       |                |---------->|    |
      |        .       |     200 OK     |<--200 OK--|    |
      |        .       |<---------------|           |    |
      |                |                |           |    |
```

Fig. 87 – SIP to ATS-No.5 Priority Call Interruption Abandon while the Blocking Line Signal is sent

Note 36.

Once the ATS-No.5 Blocking line signal is being sent, the Gateway shall continue to send the Blocking signal for its full duration and then clear the call by sending the corresponding ATS-No.5 Clear Forward or Clear Back line signal (depending upon the originator of the interrupted call) and SIP BYE message.

5.8.7.2 Priority Call Intrusion

5.8.7.2.1 Priority Call at an Incoming Gateway

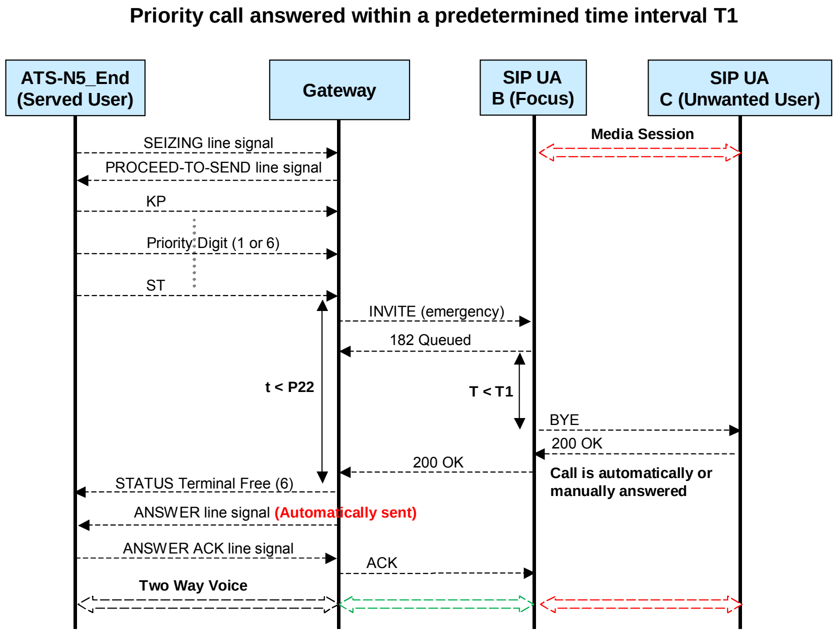


Fig. 88 – ATS-No.5 to SIP Priority call answered after releasing previous call

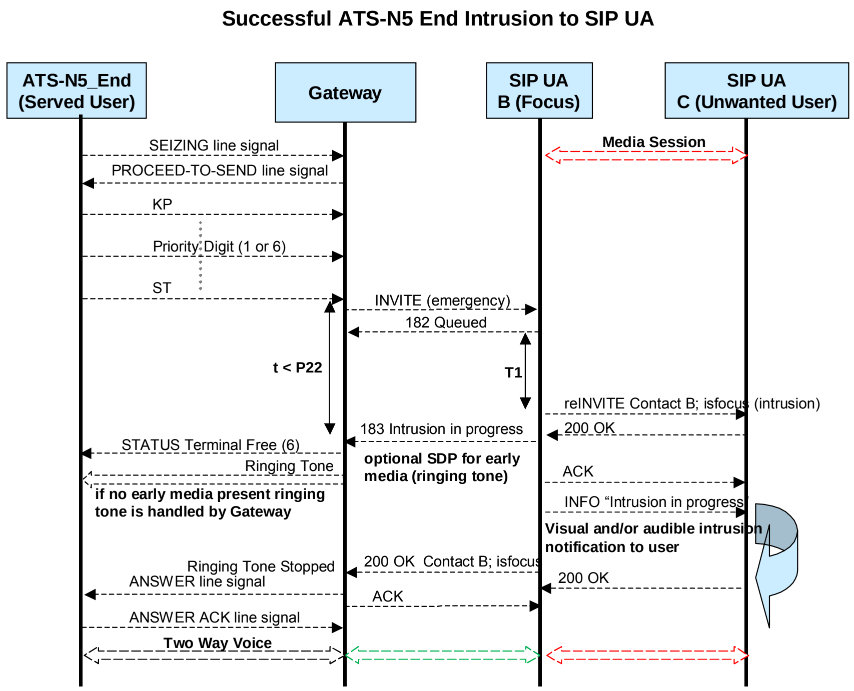


Fig. 89 – ATS-No.5 to SIP Successful Priority Call Intrusion

Successful ATS-N5 End Intrusion to SIP UA

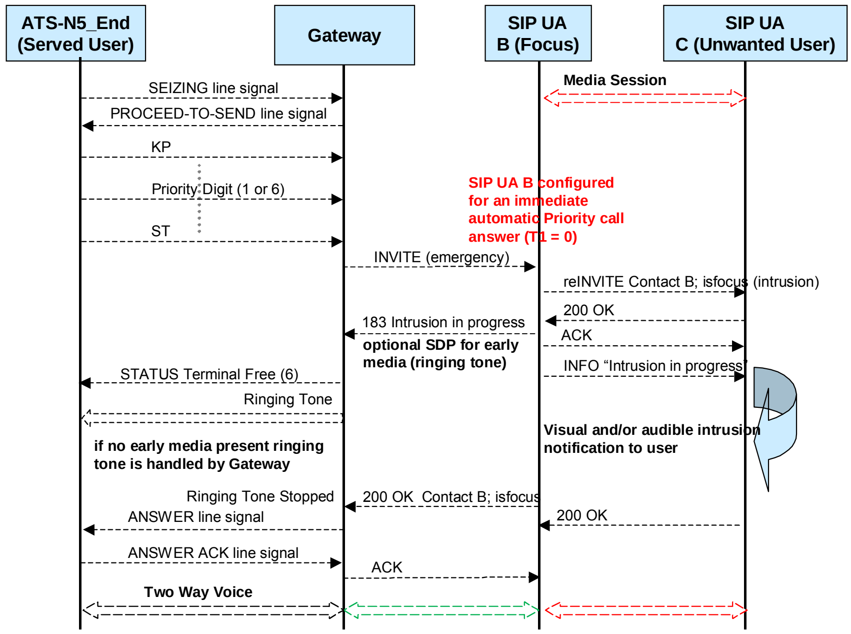


Fig. 90 – ATS-No.5 to SIP Successful Priority Call Intrusion with T1 = 0

Priority call is displayed at SIP_End1 and manually answered

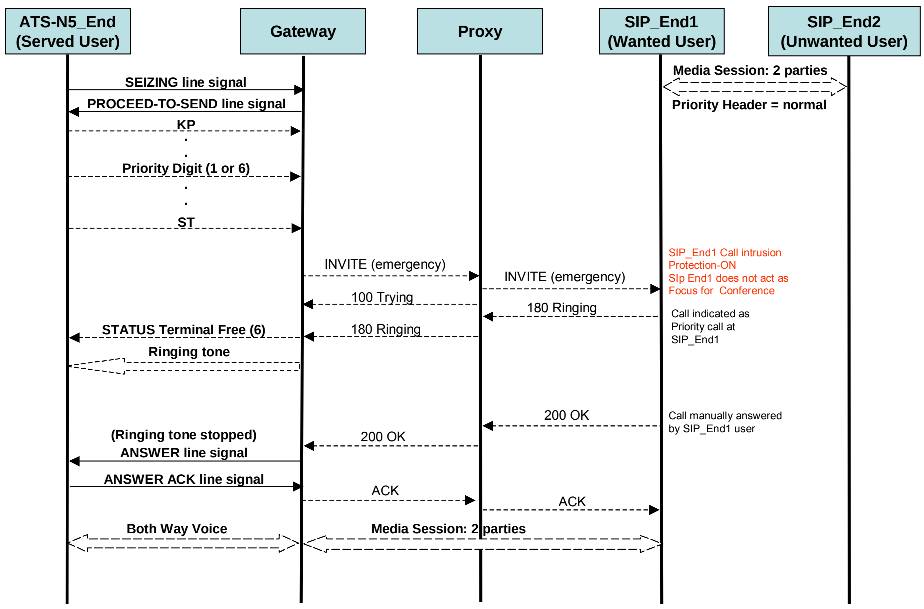


Fig. 91 – ATS-No.5 to SIP Priority Call Intrusion Forbidden by Wanted User

- The Call Intrusion Protection Level of the Unwanted User shall be assumed as OFF, that is call intrusion permitted, or not, determined exclusively by the Wanted User; there is no SIP signalling for SIP_End2 (Unwanted User) to forbid call intrusion.

- Call intrusion will happen unless the established active call is a Priority call (Priority Header Field = “emergency”).

Fig. 92 – ATS-No.5 to SIP Priority Call Intrusion Cannot Be Forbidden by Unwanted User

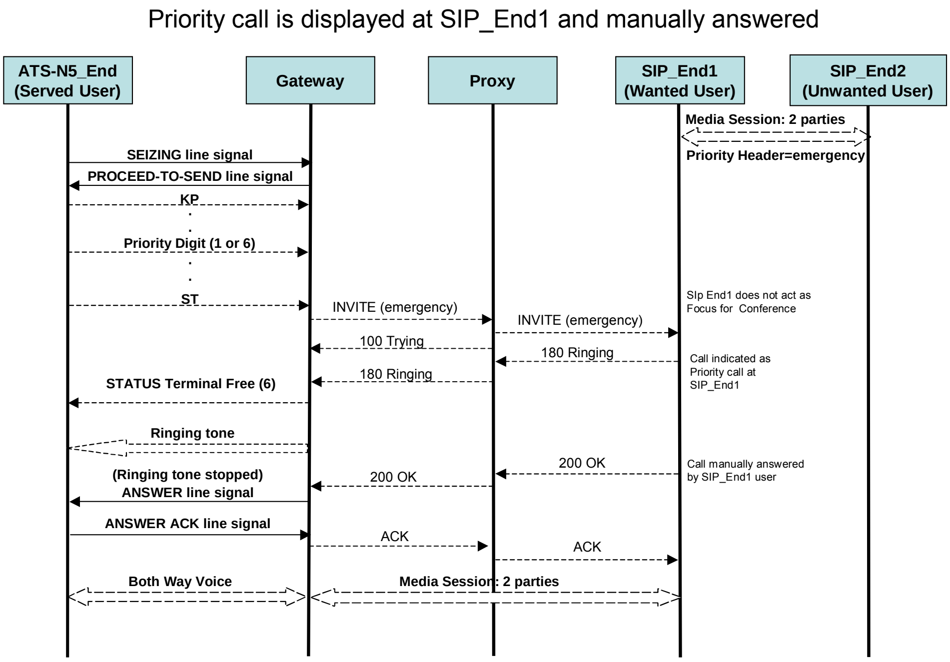


Fig. 93 – ATS-No.5 to SIP Priority Call Intrusion into another Priority Call Forbidden

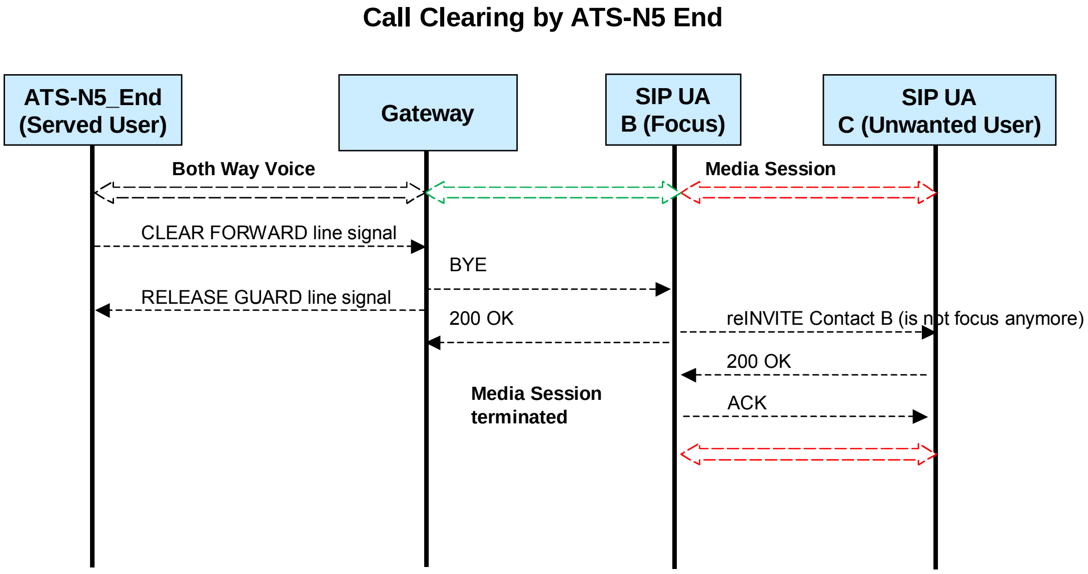


Fig. 94 – Call Clearing by ATS-No.5 End

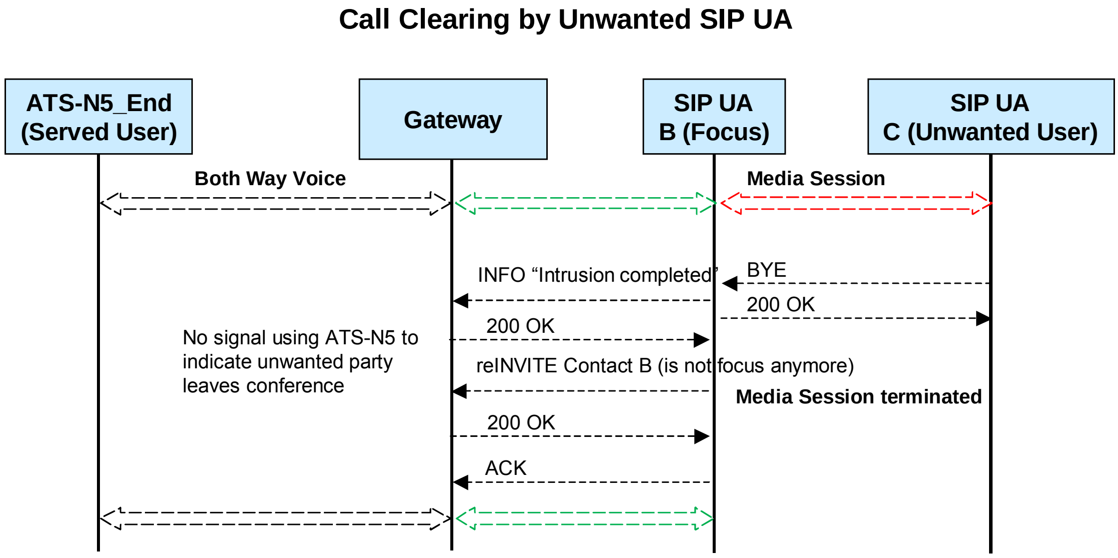


Fig. 95 – Call Clearing by Unwanted SIP UA

5.8.7.2.2 Priority Call at an Outgoing Gateway

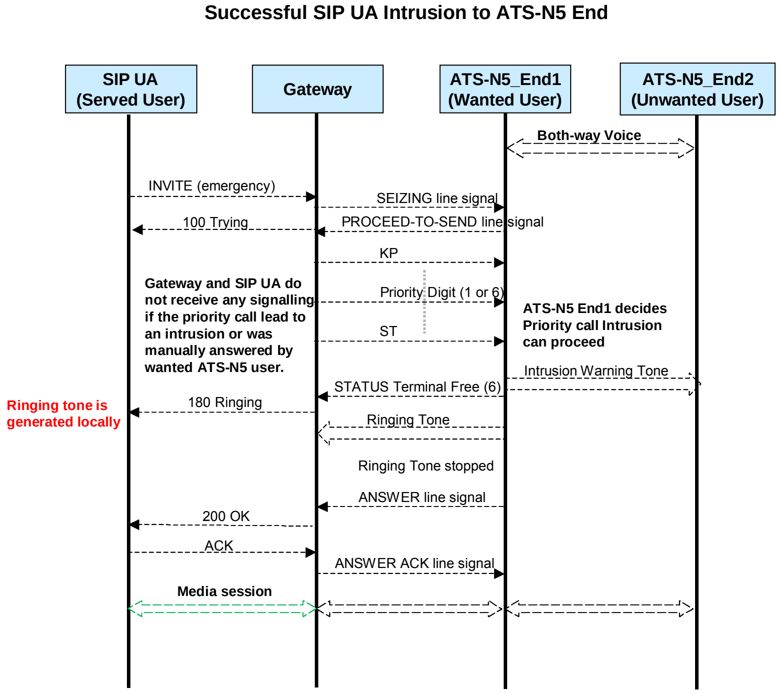


Fig. 96 – SIP to ATS-No.5 Successful Priority Call Intrusion

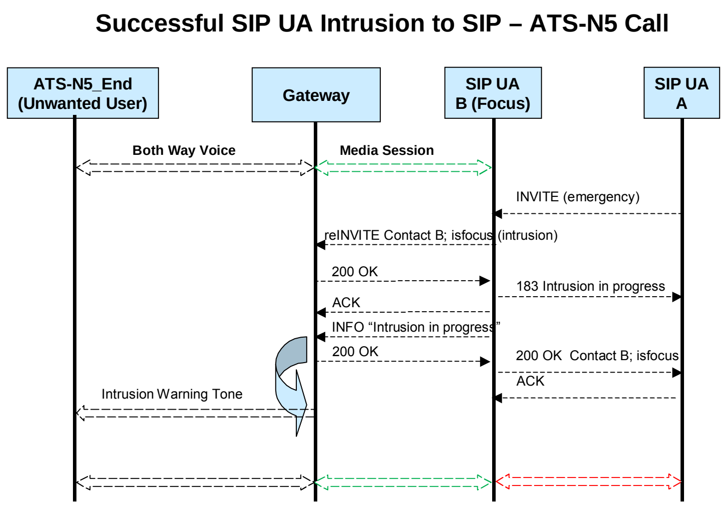


Fig. 97 – Intrusion to a SIP - ATS-No.5 Call

Priority call is displayed at ATS-N5 End1 and manually answered

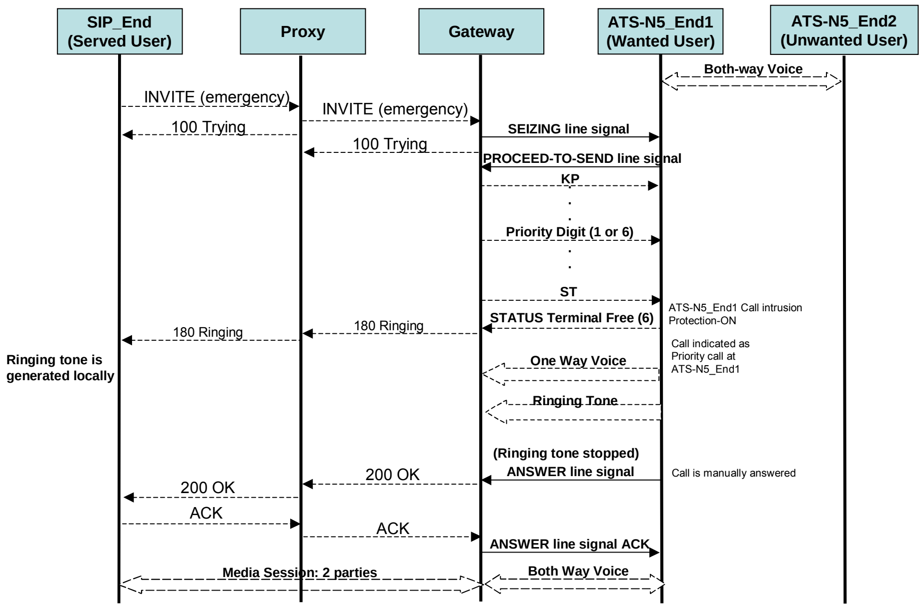


Fig. 98 – SIP to ATS-No.5 Priority Call Intrusion Forbidden by Wanted User

- The Call Intrusion Protection Level of the Unwanted User shall be assumed as OFF, that is call intrusion permitted, or not, determined exclusively by the Wanted User; there is no ATS-N5 signalling for ATS-N5_End2 (Unwanted User) to forbid call intrusion.

- Call intrusion will happen unless the established active call is a Priority call (Priority digit = 1 or 6).

Fig. 99 – SIP to ATS-No.5 Priority Call Intrusion Cannot Be Forbidden by Unwanted User

Priority call is displayed at ATS-N5 End1 and manually answered

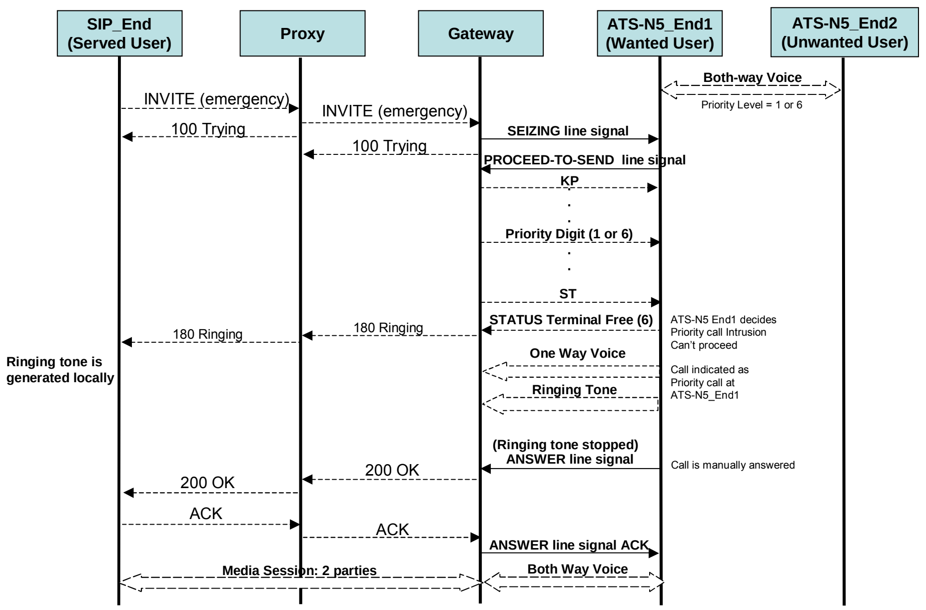


Fig. 100 – SIP to ATS-No.5 Priority Call Intrusion into another Priority Call Forbidden

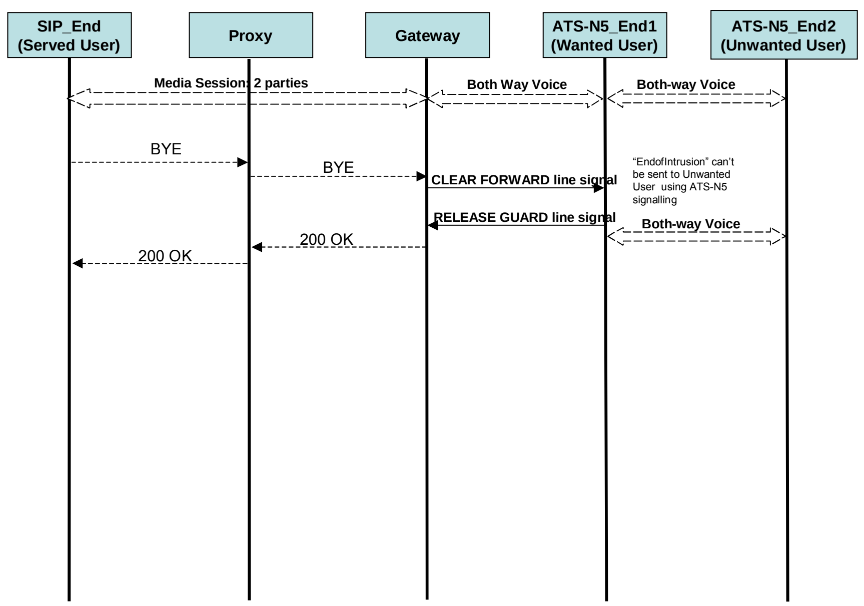


Fig. 101 – Call Clearing by SIP UA

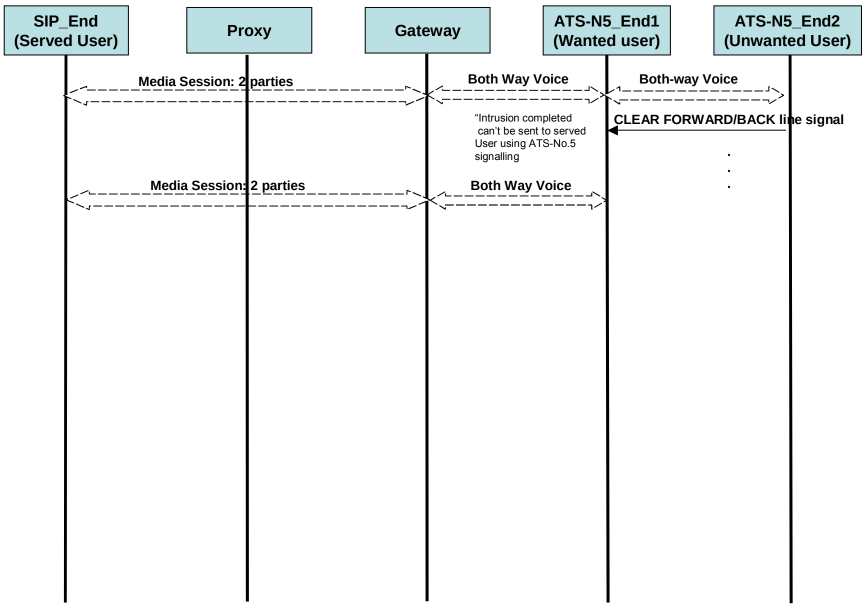


Fig. 102 – Call Clearing by ATS-No.5 Unwanted User

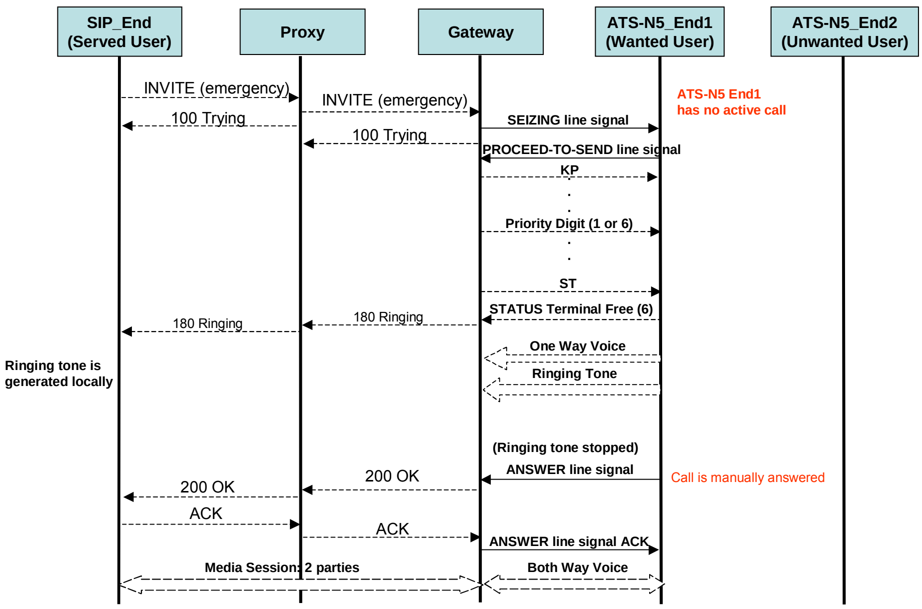
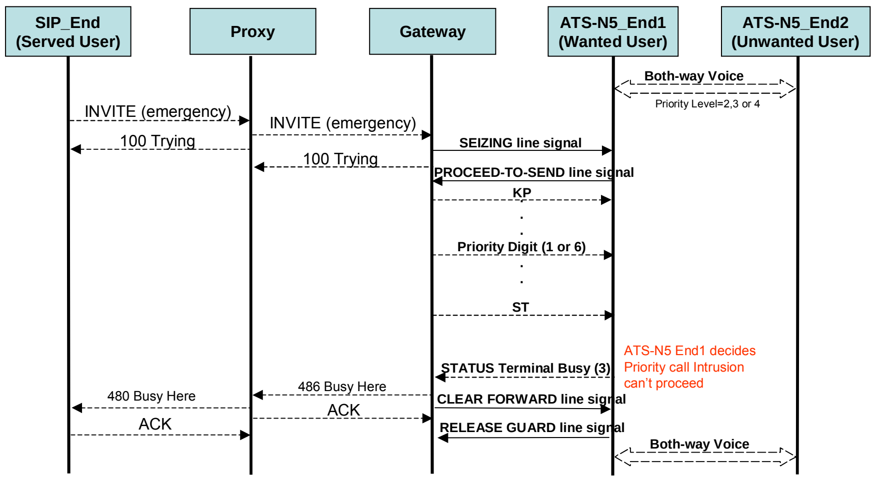
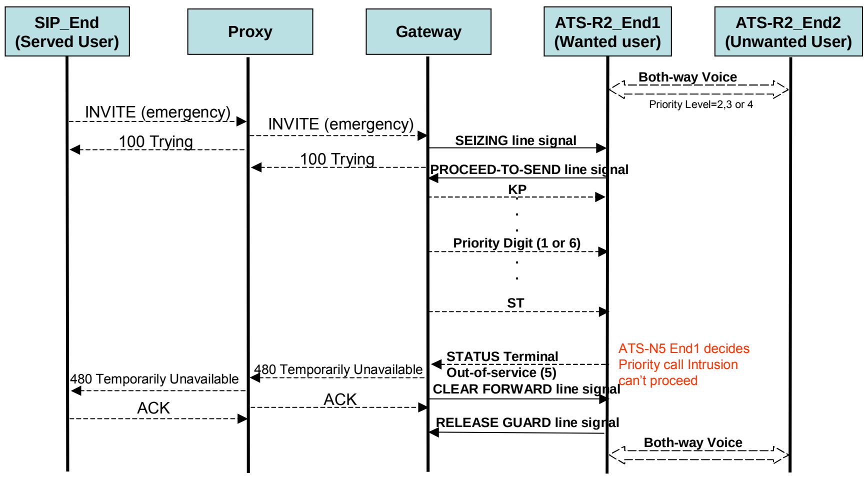


Fig. 103 – SIP to ATS-No.5 Priority Call Intrusion to a Non-busy Wanted User



This scenario is unusual as it results in Priority call not being connected

Fig. 104 – SIP to ATS-No.5 Priority Call Intrusion Receiving Status Signal "Terminal Busy"



This scenario is unusual as it results in Priority call not being connected

Fig. 105 – SIP to ATS-No.5 Priority Call Intrusion Receiving Status Signal "Terminal Out-of-Service"

CHAPTER 6

SIGNALLING INTERWORKING BETWEEN SIP AND ATS-QSIG

6.1 BACKGROUND AND ARCHITECTURE

This chapter specifies signalling interworking between SIP and ATS-QSIG in support of basic services as well as ATS supplementary services within an Air Traffic Services Ground Voice Network (AGVN).

ATS-QSIG is a signalling protocol that operates between Voice Communication Systems (VCS) within an Air Traffic Services Ground Voice Network (AGVN). An AGVN provides basic services and supplementary services to its users. ATS-QSIG is specified in ECMA-312 ("Profile Standard for the Use of PSS1 (QSIG) in Air Traffic Services Networks") [33].

Interworking between ATS-QSIG and SIP permits a call originating at a user of a circuit-switched AGVN to terminate at a user of an IP AGVN, or a call originating at a user of an IP AGVN to terminate at a user of a circuit-switched AGVN.

Interworking between a circuit-switched AGVN employing ATS-QSIG and a public IP network employing SIP is outside the scope of this specification. However, the functionality specified in this document is in principle applicable to such a scenario when deployed in conjunction with other relevant functionality (e.g., number translation, security functions, etc.).

This specification is applicable to any interworking unit that can act as a gateway between a circuit-switched AGVN employing ATS-QSIG and an IP AGVN employing SIP.

ATS-QSIG provides a means for establishing and clearing calls that originate and terminate on different VCSs. A call can be routed over a single inter-VCS link connecting the originating and terminating VCS, or over several inter-VCS links in series with switching at intermediate VCSs known as transit VCSs. A call can originate or terminate in another network, in which case it enters or leaves the AGVN environment through a gateway VCS. Parties are identified by numbers, in accordance with a closed numbering plan. This basic call capability is specified in ECMA-312 [33]. In addition to basic call capability, ATS-QSIG specifies a number of further capabilities supporting the use of ATS supplementary services in VCSs.

With the aim of exploiting IP to progressively migrate parts of the AGVN network to IP using SIP, SIP equipment in the form of SIP User Agent interfaces, SIP Proxy servers, DNS servers, etc. may be used. The new SIP environment **SHALL** also need to interwork with the ATS-QSIG-based AGVN in order to support calls originating in one environment and terminating in the other. Interworking is achieved through a gateway.

Another way of migrating is to use a SIP network to interconnect two parts of a circuit-switched AGVN and encapsulate ATS-QSIG signalling in SIP messages for calls between the two parts of the circuit-switched AGVN. This is outside the scope of this specification.

This document specifies signalling protocol interworking aspects of a gateway between a circuit-switched AGVN employing ATS-QSIG signalling and an IP AGVN employing SIP signalling. The gateway appears as a PINX to other PINXs in the circuit-switched network. The gateway appears as a SIP endpoint to other SIP entities in the IP network. Fig. 106 shows the Interconnection Diagram.

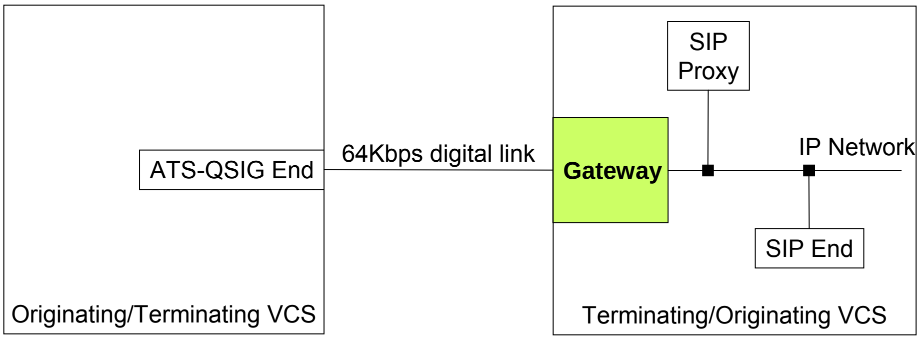


Fig. 106 – ATS-QSIG / SIP Interconnection Diagram

In addition to the signalling interworking functionality specified in this document, it is assumed that the gateway also includes the following functionality:

- one or more physical interfaces on the circuit-switched network side supporting one or more inter-VCS links, each link providing one or more constant bit rate channels for media information and a reliable layer 2 connection (e.g., over a fixed rate physical channel) for transporting ATS-QSIG signalling messages;

- one or more physical interfaces on the IP network side supporting, through layer 1 and layer 2 protocols, IP as the network layer protocol and UDP (RFC 768 [1]) and TCP (RFC 793 [3]) as transport layer protocols, these being used for the transport of SIP signalling messages and, in the case of UDP, also for media information;

- the support of TLS (RFC 4346 [24]) as additional transport layer secure protocol on the IP network side, this being used for the transport of SIP signalling messages; and

- a means of transferring media information in each direction between the circuit-switched network and the IP network, including as a minimum packetization of media information sent to the IP network and de-packetization of media information received from the IP network.

The protocol model relevant to signalling interworking functionality of a gateway is shown in Fig. 107.

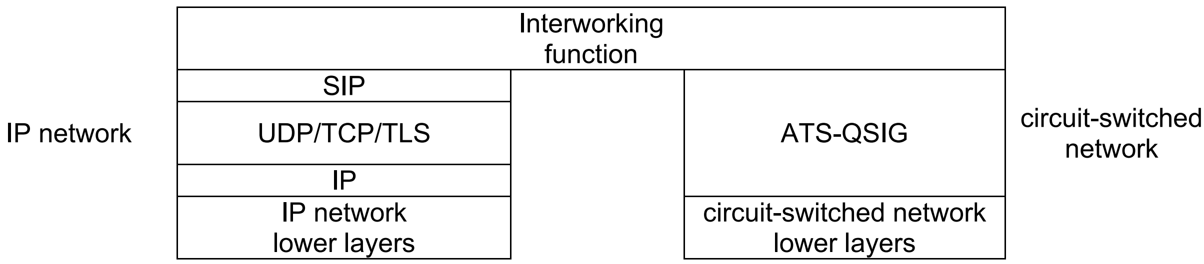


Fig. 107 – SIP / ATS-QSIG Protocol Model

In Fig. 107, the SIP box represents SIP syntax and encoding, the SIP transport layer and the SIP transaction layer. The Interworking function includes SIP Transaction User (TU) functionality.

The gateway maps received ATS-QSIG messages, where appropriate, to SIP messages and vice versa and maintains an association between a QSIG call and a SIP dialog.

A call from ATS-QSIG to SIP is initiated when an ATS-QSIG SETUP message arrives at the gateway. The ATS-QSIG SETUP message initiates ATS-QSIG call establishment and an initial response message completes negotiation of the bearer channel to be used for that call. The gateway then sends a SIP INVITE request, having translated the ATS-QSIG called party number to a URI suitable for inclusion in the Request-URI. The SIP INVITE request and the resulting SIP dialog, if successfully established, are associated with the ATS-QSIG call. The SIP 200 (OK) response is mapped to an ATS-QSIG CONNECT message, signifying answer of the call. During establishment, media streams established by SIP and SDP are connected to the bearer channel.

A call from SIP to ATS-QSIG is initiated when a SIP INVITE request arrives at the gateway. The gateway sends an ATS-QSIG SETUP message to initiate ATS-QSIG call establishment, having translated the SIP Request-URI to a number suitable for use as the ATS-QSIG called party number. The resulting ATS-QSIG call is associated with the SIP INVITE request and with the eventual SIP dialog. Receipt of an initial ATS-QSIG response message completes negotiation of the bearer channel to be used, allowing media streams established by SIP and SDP to be connected to that bearer channel. The ATS-QSIG CONNECT message is mapped to a SIP 200 (OK) response.

6.2 GENERAL REQUIREMENTS

A gateway **SHALL** support ATS-QSIG in accordance with ECMA-312 [33] as a gateway PINX and **SHALL** support SIP in accordance with RFC 3261 [8] as a UA. In particular the gateway **SHALL** support SIP syntax and encoding, the SIP transport layer and the SIP transaction layer in accordance with RFC 3261 [8]. In addition, the gateway **SHALL** support SIP TU behaviour for a UA in accordance with RFC 3261 except where stated otherwise in this specification.

The gateway **SHALL** support SDP in accordance with RFC 4566 [25] and its use in accordance with the offer / answer model in RFC 3264 [9].

The SIP profile specified in CHAPTER 3 **SHALL** apply to the ATS-QSIG / SIP gateway.

The gateway **SHALL** support calls from ATS-QSIG to SIP and calls from SIP to ATS-QSIG.

As a result of DNS look-up by the gateway in order to determine where to send a SIP INVITE request, a number of candidate destinations can be attempted in sequence. The way in which this is handled by the gateway is outside the scope of this specification. However, any behaviour specified in this specification on receipt of a SIP final response **SHOULD** apply only when there are no more candidate destinations to try.

6.3 MESSAGE MAPPING REQUIREMENTS

Message mapping requirements of ECMA-339 [34] **SHALL** apply, with modifications as specified in the following sections.

6.3.1 Message Validation and Handling of Protocol Errors

The gateway **SHALL** validate received ATS-QSIG messages in accordance with the requirements of ECMA-312 [33] and **SHALL** act in accordance with ECMA-312 on detection of an ATS-QSIG protocol error.

Message segmentation and re-assembly of ECMA-143 [32] **SHALL NOT** apply.

The gateway **SHALL** run ATS-QSIG protocol timers as specified in ECMA-312 and **SHALL** act in accordance with ECMA-312 if an ATS-QSIG protocol timer expires.

6.3.2 Call Establishment from ATS-QSIG to SIP

6.3.2.1 Overlap Procedures

An Incoming Gateway need not support the Overlap Receiving state specified in ECMA-143 [32].

6.3.3 Call Establishment from SIP to ATS-QSIG

6.3.3.1 Receipt of SIP INVITE Request for a New Call

On receipt of an INVITE request from the IP network, the Outgoing Gateway **SHALL** attempt to establish a call towards the ATS-QSIG network applying the requirements of ECMA-143 [32] with modifications as specified in ECMA-312 [33].

6.3.3.2 Use of Overlap Procedures

An Outgoing Gateway **SHALL NOT** initiate calls using states and procedures for overlap sending.

6.3.3.3 Receipt of ATS-QSIG CONNECT Message

On receipt of an ATS-QSIG CONNECT message, the Outgoing Gateway **SHALL NOT** send a QSIG CONNECT ACKNOWLEDGE message.

6.3.3.4 Receipt of a SIP INVITE Request for a Call Already Being Established

A gateway **SHALL NOT** support the behaviour derived from overlap procedures; therefore, a gateway should not receive multiple SIP INVITE requests belonging to the same call, but having different Request-URI and To fields.

6.4 NUMBER MAPPING

In ATS-QSIG, users are identified by numbers according to a closed numbering scheme. Numbers are conveyed within the Called party number and Calling party number information elements.

In SIP, users are identified by Universal Resource Identifiers (URIs) conveyed within the Request-URI and various headers, including the From and To headers specified in RFC 3261 [8].

Number mapping requirements of ECMA-339 [34] **SHALL** apply, with modifications as specified in the following sections.

6.4.1 Mapping from ATS-QSIG to SIP

6.4.1.1 Using Information from the ATS-QSIG Calling Party Number Information Element

An incoming gateway receiving an ATS-QSIG message containing the Presentation Indicator octet in the Calling Party Number information element **SHALL** ignore the contents of this octet. Therefore, sections 9.1.2.1, 9.1.2.2, 9.1.2.3 and 9.1.2.4 of ECMA-339 **SHALL** be replaced by the following.

6.4.1.1.1 No URI derived

The gateway **SHALL NOT** generate a P-Asserted-Identity header, **SHALL NOT** generate a Privacy header and **SHALL** include a URI identifying the gateway in the From header.

6.4.1.1.2 URI derived

The gateway **SHALL** generate a P-Asserted-Identity header containing the derived URI, if the gateway supports this header. The gateway **SHALL NOT** generate a Privacy header and **SHALL** include the derived URI in the From header.

6.4.1.2 Using Information from the QSIG Connected Number Information Element

An incoming gateway receiving an ATS-QSIG message containing the Connected Number information element **SHALL** ignore the contents of this information element when converting the calling party number to a URI. Therefore, those requirements contained in sections 9.1.3.1, 9.1.3.2, 9.1.3.3 and 9.1.3.4 of ECMA-339 **SHALL NOT** be applicable.

6.4.2 Mapping from SIP to ATS-QSIG

6.4.2.1 Generating the ATS-QSIG Calling party number information element

When mapping a SIP INVITE request to an ATS-QSIG SETUP message, the gateway **SHALL** generate a Calling Party Number information element as follows.

If the SIP INVITE request contains a P-Asserted-Identity header and the gateway supports that header, the gateway **SHALL** attempt to derive a number from the URI in that header. If no number is derivable from a P-Asserted-Identity header (including the case where there is no P-Asserted-Identity header), the gateway **SHALL** attempt to derive a number from the URI in the From header. If a number is derived either from the P-Asserted-Identity header, or from the From header, the gateway **SHALL** include it in the Calling Party Number information element that **SHALL** be encoded as specified in section 10.16.5 of ECMA-312.

If no number is derivable, the gateway **SHALL** reject the call by sending a SIP response 500 (Server internal error).

6.4.2.2 Generating QSIG Connected Number Information Element

A gateway **SHALL NOT** generate the QSIG Connected Number information element for an outgoing message.

6.5 MAPPING BETWEEN ATS-QSIG Transit Counter INFORMATION ELEMENT AND SIP Max-forwards HEADER FIELD

In accordance with 3.4.5, the gateway **SHOULD** map between ATS-QSIG Transit Counter information element and SIP Max-forwards header field according to Table 24 and Table 25.

ATS QSIG Transit Counter Information Element	SIP Max-Forwards Header Field
0	19
1	15
2	11
3	7
≥ 4	3

Table 24 – Mapping of ATS-QSIG Transit Counter Information Element to SIP Max-Forwards Header Field at an Incoming Gateway

SIP Max-Forwards Header Field	ATS QSIG Transit Counter Information Element
≥ 19	0
16 – 18	1
11 – 15	2
7 – 10	3
< 7	4

Table 25 – Mapping of SIP Max-Forwards Header Field to ATS-QSIG Transit Counter Information Element at an Outgoing Gateway

6.6 REQUIREMENTS FOR SUPPORT OF BASIC SERVICES

6.6.1 Derivation of ATS-QSIG Bearer Capability Information Element

The outgoing gateway **SHALL** generate the Bearer capability information element in the ATS-QSIG SETUP message based on SDP information received along with the SIP INVITE request. If the SIP INVITE request does not contain SDP information or the media type in the SDP information is only 'audio' then the Bearer Capability information element **SHALL** be generated according to Table 26. Coding of the Bearer Capability information element for 'data' transfer **SHALL** be encoded according to Table 27.

Field	Value
Coding standard	"ISO/IEC std." (01)
Information transfer capability	"Speech" (00000)
Transfer mode	"Circuit-mode" (00)
Information transfer rate	"16 Kbit/s" (10010)
Multiplier	Octet omitted
User information layer 1 protocol	"G.728 LD-CELP" (01010)

Table 26 – Bearer Capability Encoding for 'audio' Transfer

Field	Value
Coding standard	"ISO/IEC std." (01)
Information transfer capability	"Unrestricted digital information" (01000)
Transfer mode	"Circuit-mode" (00)
Information transfer rate	"16 Kbit/s" (10010)

Table 27 – Bearer Capability Encoding for 'data' Transfer

6.6.2 Derivation of Media Type in SDP

The incoming gateway **SHALL** generate SDP information to include in the SIP INVITE request based on the Bearer Capability information element received in the ATS-QSIG SETUP message. The media type included in the SDP information **SHALL** be according to Table 28.

ATS-QSIG	SDP		
Information Transfer Capability in Bearer Capability information element	**Media parameter**	**Transport protocol parameter**	**Format parameter**
"Speech" (00000)	audio	RTP/AVP	15
"Unrestricted digital information" (01000)	data	TCP	

Table 28 – Media Descriptions ("m=") Setting in SDP Based on Bearer Capability Information Element

6.6.3 Derivation of Attributes Type in SDP

The incoming gateway **SHALL** generate SDP information to include in the SIP INVITE request based on the Bearer Capability information element received in the ATS-QSIG SETUP message. The attribute type included in the SDP information **SHALL** be according to Table 29.

ATS-QSIG	SDP
Information Transfer Capability in Bearer Capability information element	**Attributes of media session**
"Speech" (00000)	rtpmap:15 G728/8000

Table 29 – Media Attributes (“a=”) Setting in SDP Based on Bearer Capability Information Element

6.7 REQUIREMENTS FOR SUPPORT OF SUPPLEMENTARY SERVICES

A gateway **SHALL** support the Call Priority Interruption and the Call Intrusion supplementary services.

6.7.1 Call Priority

6.7.1.1 Mapping of ATS-QSIG Priority Levels to SIP Priority Header Field

On receipt of a call from the ATS-QSIG network, the Gateway **SHALL** initiate the call towards the IP network. The Gateway **SHALL** read the ATS-QSIG Facility information elements defining the CPICL, CPIPL and CICL of the call and include the relevant Priority header field in the SIP INVITE request with values as specified in Table 30.

Gateway Input ATS-QSIG CPICL, CPIPL and CICL	Gateway Output SIP Priority Header Field	Call Type
CPICL = 3 CPIPL = 3 CICL = 3	emergency	Priority call
CPICL = 0 CPIPL = 2 or 3 CICL = 0	urgent	Tactical Routine call
CPICL = 0 CPIPL = 1 CICL = 0	normal	Strategic Routine call
CPICL = 0 CPIPL = 0 CICL = 0	non-urgent	General Purpose Routine call

Table 30 – Mapping of ATS-QSIG CPICL, CPIPL and CICL to SIP Priority Header Field

In the case that Facility information elements defining the CPICL, CPIPL and CICL are not present, then it **SHALL** be assumed that their values are zero.

In those exceptional cases when Facility information elements defining the CICL and/or CPICL with values of 1 or 2, then values of zero **SHALL** also be assumed.

6.7.1.2 Mapping of SIP Priority Header Field to ATS-QSIG Priority Levels

On receipt of a SIP INVITE request from the IP network, the Outgoing Gateway **SHALL** attempt to establish a call towards the ATS-QSIG network. The Outgoing Gateway **SHALL** also map the Priority header field of the SIP INVITE request to the equivalent ATS-QSIG priority levels as specified in Table 31.

Gateway Input SIP Priority Header Field	Gateway Output ATS-QSIG CPICL, CPIPL and CICL	Call Type
emergency	CPICL = 3 CPIPL = 3 CICL = 3	Priority call
urgent	CPICL = 0 CPIPL = 2 CICL = 0	Tactical Routine call (Medium Interrupt Protection)
normal	CPICL = 0 CPIPL = 1 CICL = 0	Strategic Routine call (Low Interrupt Protection)
non-urgent	CPICL = 0 CPIPL = 0 CICL = 0	General Purpose Routine call (No Interrupt Protection)

Table 31 – Mapping of SIP Priority Header Field to ATS-QSIG Priority Levels

In the case that Facility information elements defining the CPICL, CPIPL and CICL have the value zero, it is not mandatory to include them in the ATS-QSIG SETUP message.

If the resulting Call Priority Interrupt Protection Level CPIPL of the calling party < 3 at the gateway output, the CPIPL negotiation **SHALL** take place only between endpoints in the ATS-QSIG network.

6.7.2 Priority Call Interruption

Every call (Priority or Non-priority) **SHALL** be associated with a CPIPL value, however:

- A Priority call **SHALL NOT** be interrupted from any end.
- A Routine call with CPIPL = 3 (highest protection) **SHALL NOT** be interrupted from any end.
- A Routine call with a CPIPL < 3 **MAY** be interrupted from the ATS-QSIG End or by the gateway when congestion exists.
- When a non-priority call was originated from the ATS-QSIG_End, the CPIPL of the established call **SHALL** be associated with the CPIPL value in the ATS-QSIG SETUP message.
- When a non-priority call was originated from the SIP_End, the CPIPL of the established call **SHALL** be associated with the higher of the CPIPL values in the ATS-QSIG SETUP message and corresponding ATS-QSIG CONNECT message.

6.7.2.1 Priority Call Interruption from SIP to ATS-QSIG

On receipt of a SIP INVITE(emergency) request from the IP network and all B-channels in the preferred route of the circuit-switched network being occupied, the gateway **SHALL** attempt to establish a Priority call towards the ATS-QSIG network as specified in section 10.20.4 of ECMA-312 [33].

Having selected the call to be interrupted, the Gateway **SHALL** inform the parties involved that the call is to be released; an ATS-QSIG NOTIFY message containing the notification value "InterruptionIsImpending" **SHALL** be sent to the ATS-QSIG End (party in the circuit-switched network) and a SIP INFO message including a Text/Plain body indicating "Interruption is impending" **SHALL** also be sent to the SIP End (party in the IP-network). The Gateway **SHALL** also start the "Interruption pending" timer.

On expiry of the Interruption pending timer, the Gateway **SHALL** send an ATS-QSIG DISCONNECT message containing the notification value "InterruptionForcedRelease" to the ATS-QSIG_End and a SIP BYE request containing a Text/Plain body indicating "Emergency - Forced Release" to the SIP

End, after having stopped the Interrupt Warning Tone.

When clearing of the interrupted call has been completed, the Gateway **SHALL** continue the establishment of the priority call using the newly available B-channel.

6.7.2.2 Priority Call Interruption from ATS-QSIG to SIP

6.7.2.2.1 Receipt of a NOTIFY message containing notification value “InterruptionIsImpending”

On receipt of an ATS-QSIG NOTIFY message containing Notification value “InterruptionIsImpending” from the ATS-QSIG network, the Gateway **SHALL** send a SIP INFO message including a Text/Plain body indicating “Interruption is impending” to the SIP End.

6.7.2.2.2 Receipt of a DISCONNECT message containing notification value “InterruptionForcedRelease”

On receipt of an ATS-QSIG DISCONNECT message containing Notification value “InterruptionForcedRelease”, the Gateway **SHALL** send a SIP BYE request containing a Text/Plain body indicating “Emergency - Forced Release” to the SIP End.

6.7.2.2.3 Receipt of a NOTIFY message containing notification value “InterruptionTerminated”

On receipt of an ATS-QSIG NOTIFY message containing Notification Value “InterruptionTerminated” from the ATS-QSIG network, the Gateway **SHALL** send a SIP INFO message including a Text/Plain body indicating “Interruption Terminated” to the SIP End.

6.7.3 Priority Call Intrusion

6.7.3.1 Priority Call Intrusion from ATS-QSIG to SIP

An ATS-QSIG Priority call (having CPICL = 3, CPIPL = 3 and CICL = 3) made towards an Incoming Gateway **SHALL** cause Gateway SIP User Agent to send SIP INVITE with a Priority header defined as “emergency” to distinguish it from a routine call (with Priority header defined as “urgent”, “normal”, “non-urgent”).

If Intrusion is allowed (i.e. CIPL=0) by the Wanted SIP End User, the Wanted SIP End User will then act as the focus for the conference (use is made of the “isfocus” feature, defined in RFC 3840 [16] to create a conference media session).

On receiving SIP 182 (Queued) provisional response, the Gateway **SHALL** send an ATS-QSIG ALERTING message toward the ATS-QSIG End.

On receiving SIP 183 (Intrusion in progress) provisional response, the Gateway **SHALL** send an ATS-QSIG NOTIFY message indicating “intrusionIsImpending” to the ATS-QSIG End.

If after a SIP 182 (Queued) provisional response, a SIP 200 (OK) response is received, the Gateway **SHALL** send an ATS-QSIG CONNECT message containing a Facility information element of type ciError indicating “NotAuthorised” to the ATS-QSIG End.

If after a SIP 183 (Intrusion in progress) provisional response, a SIP 200 (OK) response is received, the Gateway **SHALL** send an ATS-QSIG CONNECT message containing a Facility information element of type ciRequest.res indicating “unwantedUserIntruded” to the ATS-QSIG End.

If intrusion is forbidden (i.e. CIPL=3), the Wanted user will reply with a SIP 180 (Ringing) response. Wanted and Unwanted users remain connected; The call from the ATS-QSIG user is displayed at the user’s terminal as a Priority Call and can be manually answered. On receiving a SIP 180 (Ringing) response, the Gateway **SHALL** send an ATS-QSIG ALERTING message containing a Facility information element of type ciError indicating “notAuthorised” to the ATS-QSIG End.

Once an intrusion is effective, if the Unwanted User leaves the conference, the Gateway will receive a SIP INFO request including a Text/Plain body indicating “Intrusion completed”. On receipt of the SIP

INFO request (“Intrusion completed”), the Gateway **SHALL** send an ATS-QSIG FACILITY message containing a facility information element “ciCompleted.inv” to the ATS-QSIG End (Served User).

6.7.3.2 Priority Call Intrusion from SIP to ATS-QSIG

On receipt of a SIP INVITE(emergency) request from the IP network, the Gateway **SHALL** attempt to establish a Priority call (ATS-QSIG SETUP message having CPICL = 3, CPIPL = 3 and CICL = 3) towards the ATS-QSIG network as specified in ECMA-312. The Gateway **SHALL** be configured to operate with T1 = 0 and automatic Priority call answer.

If Intrusion is allowed (i.e. CIPL =0), the Wanted User (ATS-QSIG End) reply will be an ATS-QSIG NOTIFY message containing Notification Value “InterruptionIsImpending” and then an ATS-QSIG CONNECT message containing a Facility information element of type ciRequest.res indicating “unwantedUserIntruded”. On receipt of the ATS-QSIG NOTIFY message, the Gateway **SHALL** send a SIP 183 (Intrusion in progress) provisional response to the SIP End Served User. On receipt of the ATS-QSIG CONNECT message, the Gateway **SHALL** send a SIP 200 (OK) final response to the SIP End Served User. Once the intrusion is effective, the Gateway User Agent **SHALL** be ready to receive a dialog subscription from its SIP end-caller (Served User); it **SHALL** reply with a notification defining all parties in the conference media session.

If intrusion is forbidden (i.e. CIPL=3), the Wanted User reply will be an ATS-QSIG ALERTING message containing a Facility information element of type ciError indicating “notAuthorised”. On receipt of the ATS-QSIG ALERTING message, the Gateway **SHALL** send a SIP 180 (Ringing) response towards the IP-network.

Once an intrusion is effective, if the Unwanted User leaves the conference, the Gateway will receive an ATS-QSIG FACILITY message containing a Facility information element of type “ciCompleted.inv”. On receipt of the ATS-QSIG FACILITY message containing a Facility information element “ciCompleted.inv”, the Gateway **SHALL** send a SIP INFO request including a Text/Plain body indicating “Intrusion completed”.

6.8 MESSAGE SEQUENCE CHARTS

The following paragraphs show some typical message sequences that can occur for an interworking between ATS-QSIG and SIP.

The Message Sequence Charts (MSC) in the figures below show the information flows between the Call Control entity (labelled "Gateway") and respective Protocol Control entities for each signalling system (labelled "ATS-QIG_End" and "SIP_End") within a Gateway VCS. Each information flow is named according to the corresponding message or signal sent to or received from a peer VCS.

Dashed lines (---) represent signalling messages that are mandatory to the call scenario. These messages can be SIP or ATS-QSIG signalling. The arrow indicates the direction of message flow.

Double dashed lines (===) represent media paths between network elements.

6.8.1 Successful ATS-QSIG to SIP Routine Call

This is a typical message sequence for a successful call setup of an incoming routine call (to a gateway) from a route employing the ATS-QSIG signalling system which is routed on an IP network to the called user employing SIP.

```
ATS-QSIG_End       Gateway          Proxy          SIP_End
    |                |                |                |
    |     SETUP      |                |                |
    |--------------->|     INVITE     |                |
    | CALL PROCEEDING|--------------->|     INVITE     |
    |<---------------|   100 Trying   |--------------->|
    |                |<---------------|                |
    |                |                |  180 Ringing   |
    |                |  180 Ringing   |<---------------|
    |    ALERTING    |<---------------|                |
    |<---------------|                |                |
    |                |                |                |
    |                |                |                |
    |                |                |                |
    |                |                |     200 OK     |
    |                |     200 OK     |<---------------|
    |    CONNECT     |<---------------|                |
    |<---------------|      ACK       |                |
    |                |--------------->|      ACK       |
    |                |                |--------------->|
    |(Both Way Voice)|      (Both Way RTP Media)       |
    |<==============>|<===============================>|
    |                |                |                |
```

Fig. 108 – Successful ATS-QSIG to SIP Routine Call

Note 37.
For a routine call, CPIPL = 0, 1, 2 or 3, CPICL = 0 and CICL = 0, as indicated in Table 30.

Note 38.
In accordance with section 10.9 of ECMA-312, on sending the CONNECT message a CONNECT ACKNOWLEDGE message will not be received.

6.8.2 Successful SIP to ATS-QSIG Routine Call

This is a typical message sequence for a successful call setup of an incoming routine call (to a gateway) from an IP network employing SIP which is routed on a route employing the ATS-QSIG signalling system.

```
ATS-QSIG_End       Gateway           Proxy            SIP_End
     |                |                |                |
     |                |                |     INVITE     |
     |                |     INVITE     |<---------------|
     |      SETUP     |<---------------|   100 Trying   |
     |<---------------|   100 Trying   |--------------->|
     | CALL PROCEEDING|--------------->|                |
     |--------------->|                |                |
     |    ALERTING    |                |                |
     |--------------->|  180 Ringing   |                |
     |                |--------------->|  180 Ringing   |
     |                |                |--------------->|
     |     CONNECT    |                |                |
     |--------------->|     200 OK     |                |
     |                |--------------->|     200 OK     |
     |                |                |--------------->|
     |                |                |      ACK       |
     |                |      ACK       |<---------------|
     |                |<---------------|                |
     |(Both Way Voice)|       (Both Way RTP Media)      |
     |<==============>|<===============================>|
```

Fig. 109 – Successful SIP to ATS-QSIG Routine Call

Note 39.
For a routine call, the value of the "Priority" header field in the INVITE method shall be "urgent", "normal" or "non-urgent", as indicated in Table 31.

Note 40.
In accordance with section 10.9 of ECMA-312, on receipt of the CONNECT message a CONNECT ACKNOWLEDGE message shall not be sent.

6.8.3 Normal Call Clearing from ATS-QSIG End

Typical message sequence for Call Clearing from ATS-QSIG to SIP subsequent to call establishment.

```
ATS-QSIG_End       Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|       (Both Way RTP Media)      |
    |<==============>|<===============================>|
    |                |                |                |
    |   DISCONNECT   |                |                |
    |--------------->|      BYE       |                |
    |    RELEASE     |--------------->|      BYE       |
    |<---------------|                |--------------->|
    |RELEASE COMPLETE|                |     200 OK     |
    |--------------->|     200 OK     |<---------------|
    |                |<---------------|                |
    |                |                |                |
```

Fig. 110 – Normal Call Clearing from ATS-QSIG End

6.8.4 Normal Call Clearing from SIP End

Typical message sequence for Call Clearing from SIP to ATS-QSIG subsequent to call establishment.

```
ATS-QSIG_End       Gateway           Proxy           SIP_End
    |                |                |                |
    |(Both Way Voice)|       (Both Way RTP Media)      |
    |<==============>|<===============================>|
    |                |                |                |
    |                |                |       BYE      |
    |                |      BYE       |<---------------|
    |   DISCONNECT   |<---------------|                |
    |<---------------|     200 OK     |                |
    |    RELEASE     |--------------->|     200 OK     |
    |--------------->|                |--------------->|
    |RELEASE COMPLETE|                |                |
    |<---------------|                |                |
    |                |                |                |
```

Fig. 111 – Normal Call Clearing from SIP End

6.8.5 Unsuccessful ATS-QSIG to SIP Call

6.8.5.1 Call Clearing from ATS-QSIG End

This is a typical message sequence for Call Clearing from ATS-QSIG to SIP during establishment of a call from ATS-QSIG to SIP, in which the Gateway has received a provisional response (1xx) to the INVITE request but not a final response (2xx, 3xx, 4xx, 5xx, 6xx).

```
 ATS-QSIG_End        Gateway            Proxy           SIP_End
      |                 |                 |                 |
      |      SETUP      |                 |                 |
      |---------------->|      INVITE     |                 |
      | CALL PROCEEDING|---------------->|      INVITE     |
      |<----------------|   100 Trying    |---------------->|
      |                 |<----------------|   180 Ringing   |
      |                 |   180 Ringing   |<----------------|
      |     ALERTING    |<----------------|                 |
      |<----------------|                 |                 |
      |    DISCONNECT   |                 |                 |
      |---------------->|      CANCEL     |                 |
      |     RELEASE     |---------------->|      CANCEL     |
      |<----------------|      200 OK     |---------------->|
      |RELEASE COMPLETE|<----------------|      200 OK     |
      |---------------->|                 |<----------------|
      |                 |                 |487 Req. Termin.|
      |                 |487 Req. Termin.|<----------------|
      |                 |<----------------|       ACK       |
      |                 |       ACK       |---------------->|
      |                 |---------------->|                 |
      |                 |                 |                 |
```

Fig. 112 – Unsuccessful ATS-QSIG to SIP Call.
Call Clearing from ATS-QSIG End

6.8.5.2 Call Clearing from SIP Network

This is a typical message sequence for Call Clearing from SIP to ATS-QSIG during establishment of a call from ATS-QSIG to SIP, in which the Gateway has not previously received a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request.

```
ATS-QSIG_End       Gateway            Proxy            SIP_End
     |                 |                 |                 |
     |     SETUP       |                 |                 |
     |---------------->|     INVITE      |                 |
     | CALL PROCEEDING |---------------->|     INVITE      |
     |<----------------|   100 Trying    |---------------->|
     |                 |<----------------|     4xx-6xx     |
     |                 |                 |<----------------|
     |                 |                 |       ACK       |
     |                 |     4xx-6xx     |---------------->|
     |                 |<----------------|                 |
     |                 |       ACK       |                 |
     |   DISCONNECT    |---------------->|                 |
     |<----------------|                 |                 |
     |    RELEASE      |                 |                 |
     |---------------->|                 |                 |
     |RELEASE COMPLETE |                 |                 |
     |<----------------|                 |                 |
     |                 |                 |                 |
```

Fig. 113 – Unsuccessful ATS-QSIG to SIP Call. Call Clearing from SIP Network

The mapping of the SIP responses to ATS-QSIG Cause values **SHALL** be as specified in ECMA-339.

6.8.6 Unsuccessful SIP to ATS-QSIG Call

6.8.6.1 Call Clearing from SIP Network

This is a typical message sequence for Call Clearing from SIP to ATS-QSIG during establishment of a call from SIP to ATS-QSIG, in which the Gateway has sent a provisional response (1xx) to the INVITE request but not a final response (2xx, 3xx, 4xx, 5xx, 6xx).

```
ATS-QSIG_End       Gateway           Proxy           SIP_End
     |                |                |                |
     |                |                |     INVITE     |
     |                |     INVITE     |<---------------|
     |     SETUP      |<---------------|   100 Trying   |
     |<---------------|   100 Trying   |--------------->|
     |CALL PROCEEDING |--------------->|     CANCEL     |
     |--------------->|     CANCEL     |<---------------|
     |   DISCONNECT   |<---------------|     200 OK     |
     |<---------------|     200 OK     |--------------->|
     |    RELEASE     |--------------->|                |
     |--------------->|487 Req. Termin.|                |
     |RELEASE COMPLETE|--------------->|                |
     |<---------------|      ACK       |                |
     |                |<---------------|487 Req. Termin.|
     |                |                |--------------->|
     |                |                |      ACK       |
     |                |                |<---------------|
     |                |                |                |
```

**Fig. 114 – Unsuccessful SIP to ATS-QSIG Call.
Call Clearing from SIP Network**

6.8.6.2 Call Clearing from ATS-QSIG End

This is a typical message sequence for Call Clearing from ATS-QSIG to SIP during establishment of a call from SIP to ATS-QSIG, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request.

```
ATS-QSIG_End       Gateway           Proxy            SIP_End
    |                 |                 |                 |
    |                 |                 |     INVITE      |
    |                 |     INVITE      |<----------------|
    |     SETUP       |<----------------|   100 Trying    |
    |<----------------|   100 Trying    |---------------->|
    |CALL PROCEEDING  |---------------->|                 |
    |---------------->|                 |                 |
    |   DISCONNECT    |                 |                 |
    |---------------->|     4xx-6xx     |                 |
    |     RELEASE     |---------------->|                 |
    |<----------------|       ACK       |                 |
    |RELEASE COMPLETE |<----------------|     4xx-6xx     |
    |---------------->|                 |---------------->|
    |                 |                 |       ACK       |
    |                 |                 |<----------------|
    |                 |                 |                 |
```

**Fig. 115 – Unsuccessful SIP to ATS-QSIG Call.
Call Clearing from ATS-QSIG End**

The mapping of the SIP responses to ATS-QSIG Cause values **SHALL** be as specified in ECMA-339.

6.8.6.3 Call Clearing from Gateway

This is a typical message sequence for Call Clearing from Gateway during establishment of a call from SIP to ATS-QSIG, in which the Gateway has not sent a final response (2xx, 3xx, 4xx, 5xx, 6xx) to the INVITE request, and ATS-QSIG protocol error is encountered by the Gateway during call set-up.

```
ATS-QSIG_End       Gateway           Proxy            SIP_End
    |                 |                 |                 |
    |                 |                 |     INVITE      |
    |                 |     INVITE      |<----------------|
    |     SETUP       |<----------------|   100 Trying    |
    |<----------------|   100 Trying    |---------------->|
    |CALL PROCEEDING  |---------------->|                 |
    |---------------->|                 |                 |
    |   DISCONNECT    |                 |                 |
    |<----------------|500 Svr.Int.Err. |                 |
    |     RELEASE     |---------------->|500 Svr.Int.Err. |
    |---------------->|       ACK       |---------------->|
    |RELEASE COMPLETE |<----------------|       ACK       |
    |<----------------|                 |<----------------|
    |                 |                 |                 |
```

**Fig. 116 – Unsuccessful SIP to ATS-QSIG Call.
Call Clearing from Gateway**

6.8.7 Interworking of Supplementary Services

6.8.7.1 Priority Call Interruption

6.8.7.1.1 ATS-QSIG to SIP Priority Call Interruption

The message sequence shown below corresponds to a scenario in which a routine call, established through a gateway, is interrupted by a priority call from the ATS-QSIG_End.

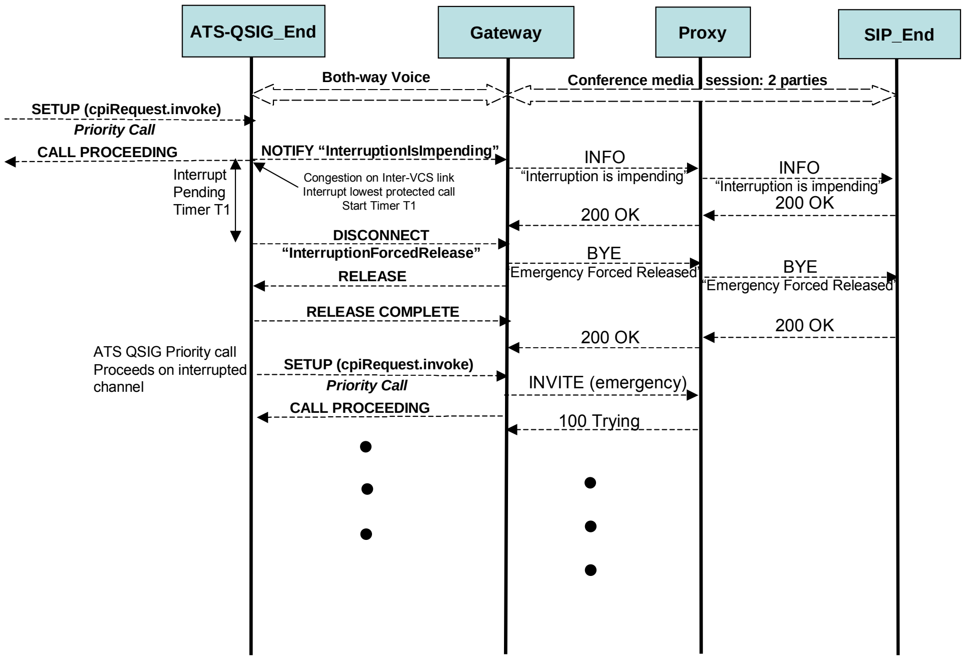


Fig. 117 – ATS-QSIG to SIP Priority Call Interruption

6.8.7.1.2 ATS-QSIG to SIP Priority Call Interruption Abandon

The message sequence shown below correspond to a scenario in which during the interrupt warning period another channel becomes available and it is used to establish the priority call.

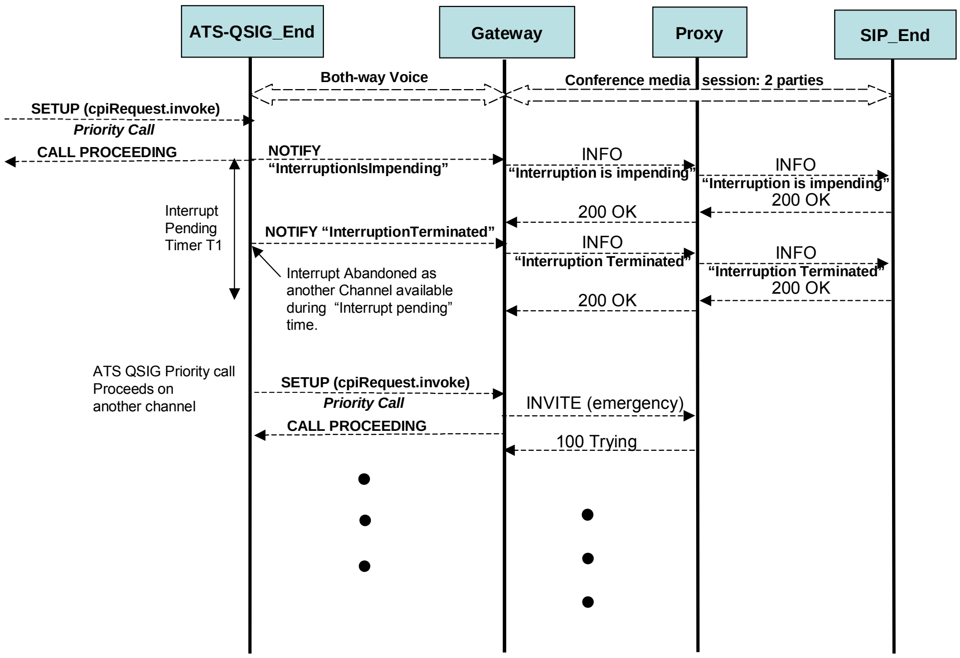


Fig. 118 – ATS-QSIG to SIP Priority Call Interruption Abandon

6.8.7.1.3 SIP to ATS-QSIG Priority Call Interruption

The message sequence shown below corresponds to a scenario in which a routine call, established through a gateway, is interrupted by a priority call from the SIP Network. An established routine call between SIP_End1 and ATS-QSIG_End1 across a gateway could be interrupted if an emergency call from SIP_End2 to ATS-QSIG_End2 finds congestion on the Gateway to ATS-QSIG_End1 link and decides to interrupt.

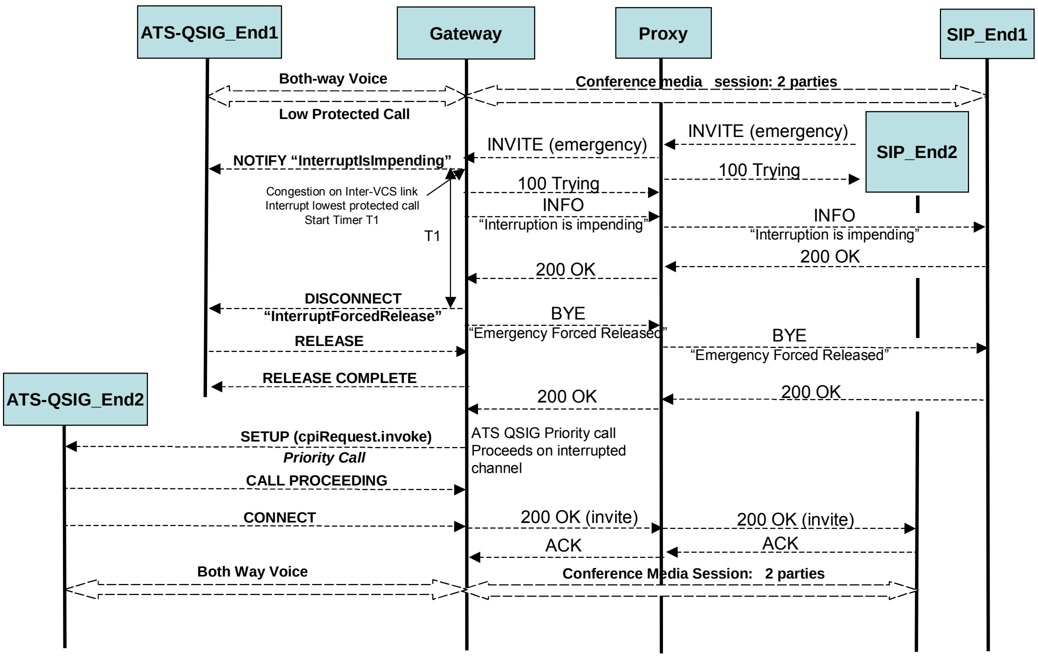


Fig. 119 – SIP to ATS-QSIG Priority Call Interruption

6.8.7.1.4 SIP to ATS-QSIG Priority Call Interruption Abandon

The message sequence shown below correspond to a scenario in which during the interrupt warning period another channel becomes available and it is used for the priority call.

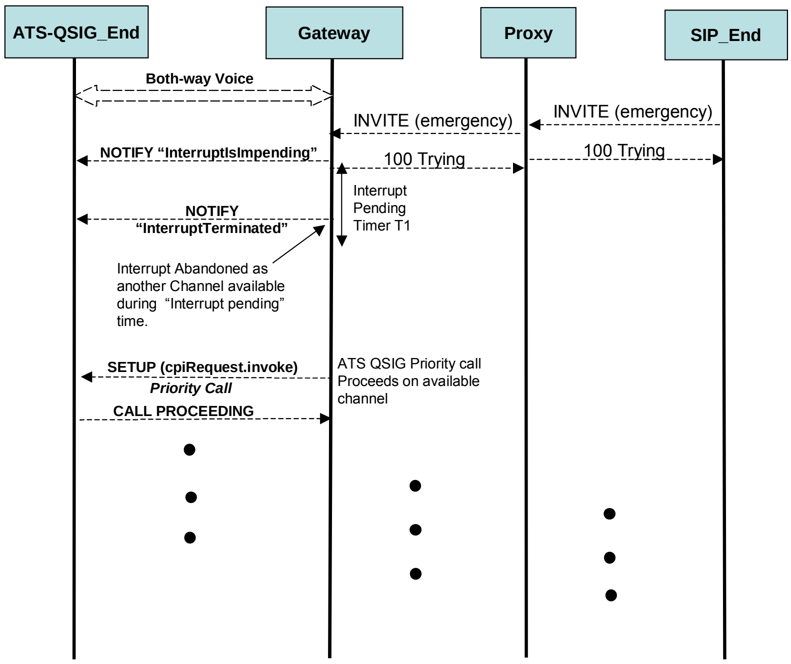


Fig. 120 – SIP to ATS-QSIG Priority Call Interruption Abandon

6.8.7.2 Priority Call Intrusion

6.8.7.2.1 Priority Call at an Incoming Gateway

Priority call answered within a predetermined time interval T1

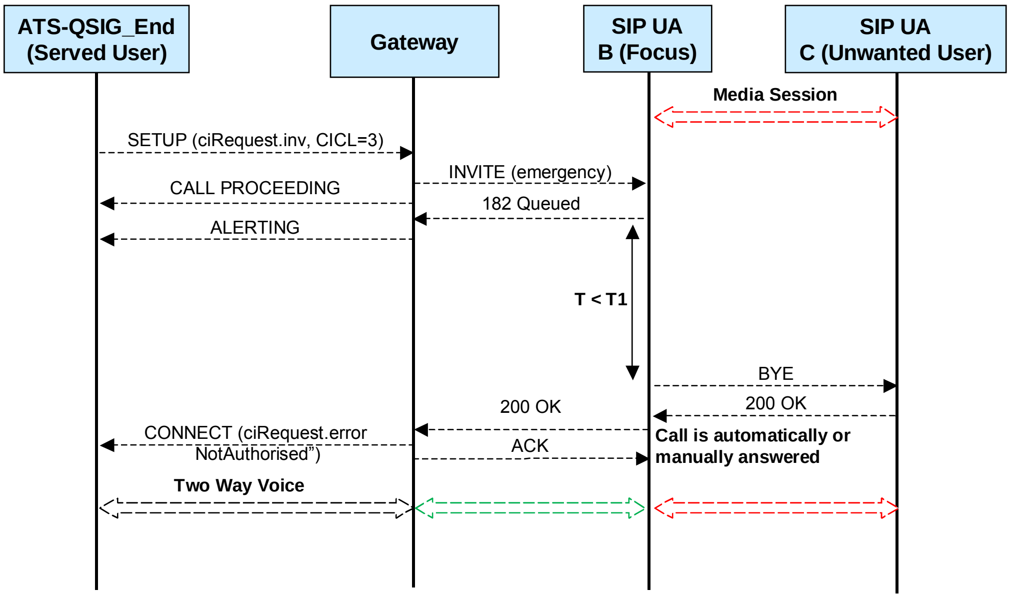


Fig. 121 – ATS-QSIG to SIP Priority call answered after releasing previous call

Successful ATS-QSIG End Intrusion to SIP UA

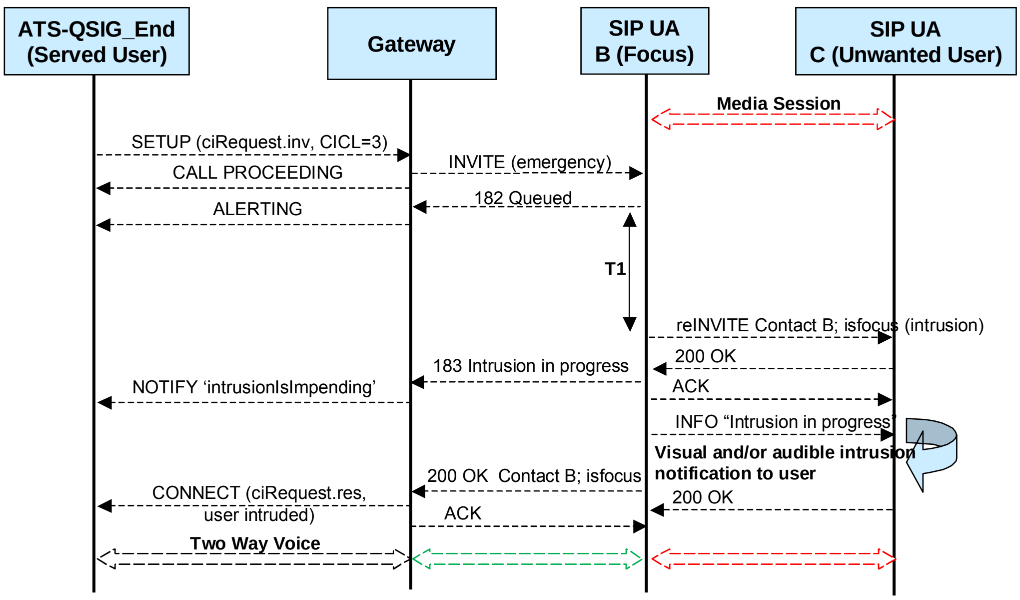


Fig. 122 – ATS-QSIG to SIP Successful Priority Call Intrusion

Successful ATS-QSIG End Intrusion to SIP UA

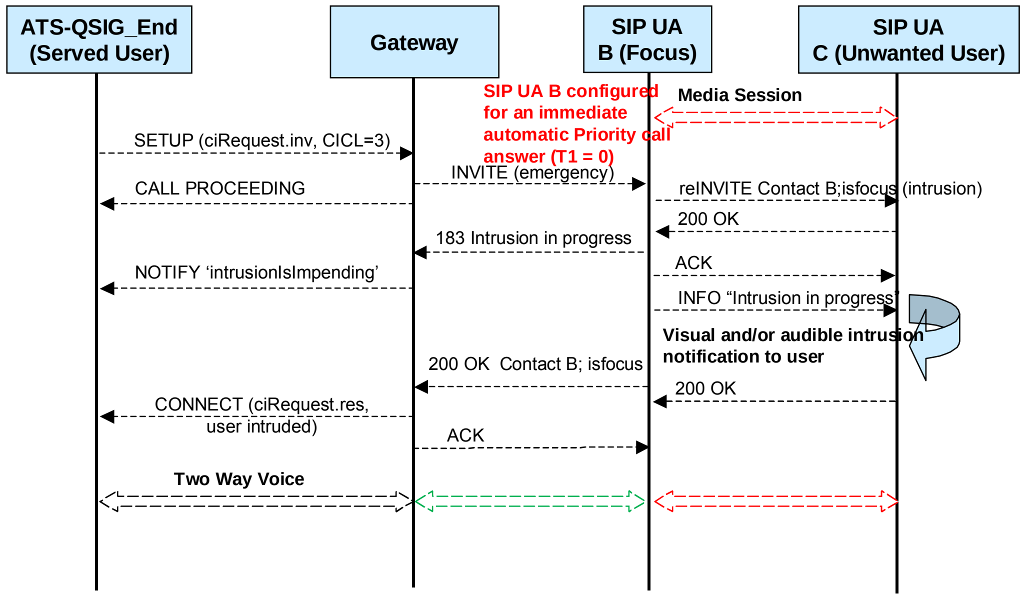


Fig. 123 – ATS-QSIG to SIP Successful Priority Call Intrusion with T1 = 0

Priority call is displayed at SIP_End1 and manually answered

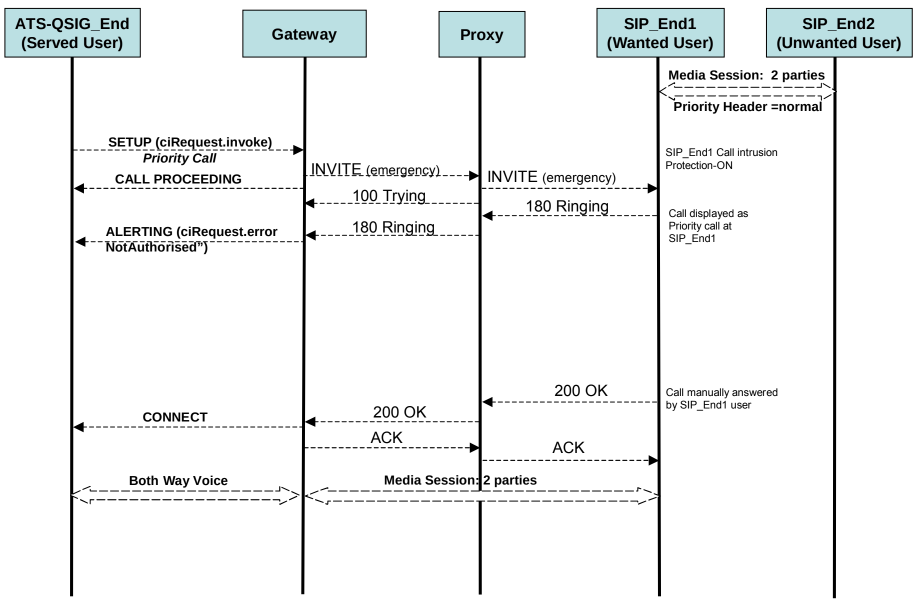


Fig. 124 – ATS-QSIG to SIP Priority Call Intrusion Forbidden by Wanted User

- The Call Intrusion Protection Level of the Unwanted User shall be assumed as 0 (OFF), that is call intrusion permitted, or not, determined exclusively by the Wanted User; there is no SIP signalling for SIP_End2 (Unwanted User) to forbid call intrusion.

- Call intrusion will happen unless the established active call is a Priority call (Priority header field = “emergency”).

Fig. 125 – ATS-QSIG to SIP Priority Call Intrusion Cannot Be Forbidden by Unwanted User

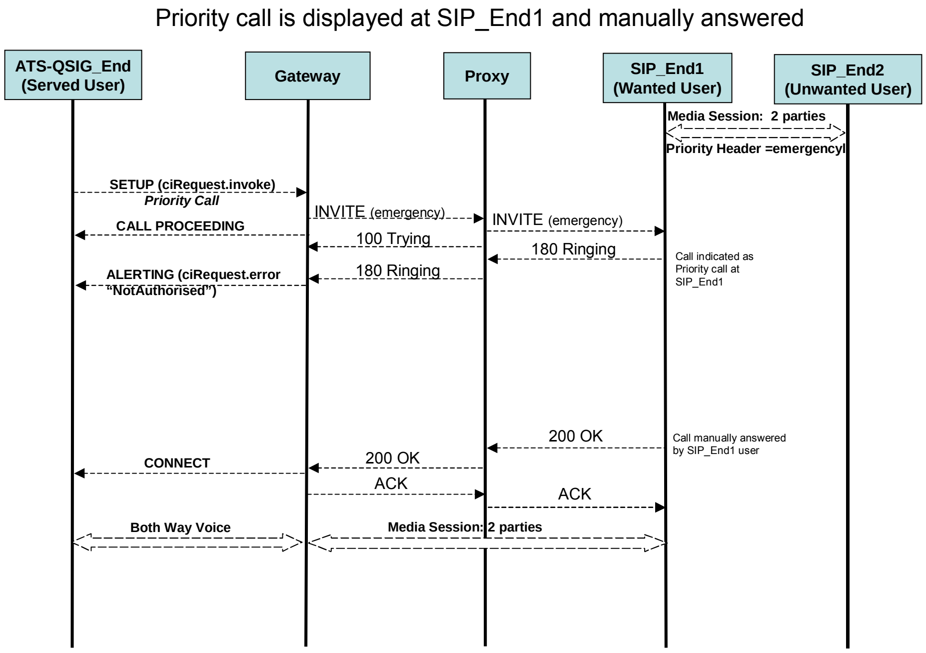


Fig. 126 – ATS-QSIG to SIP Priority Call Intrusion into another Priority Call Forbidden

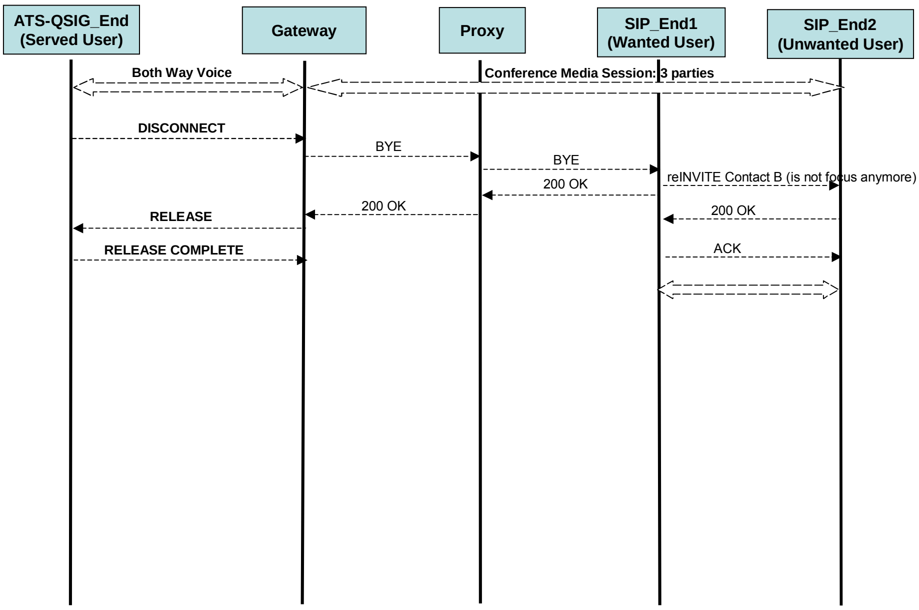


Fig. 127 – Call Clearing by ATS-QSIG End

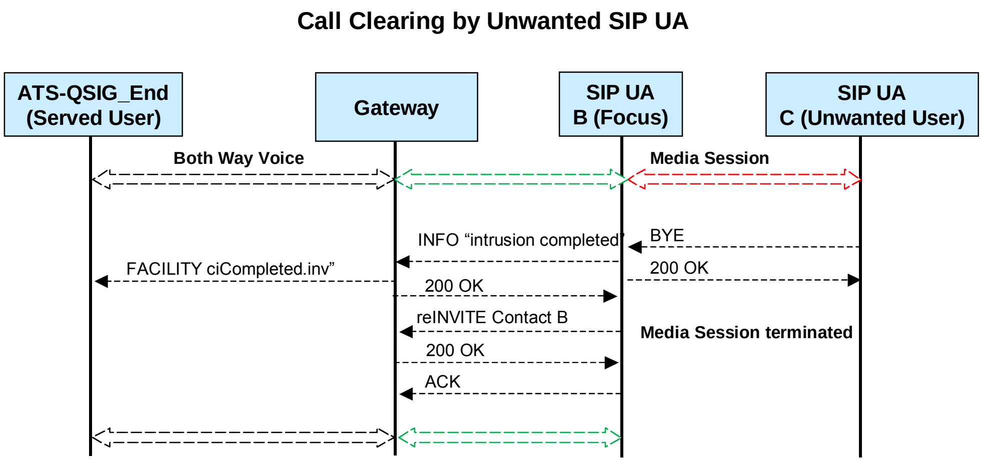


Fig. 128 – Call Clearing by Unwanted User

6.8.7.2.2 Priority Call at an Outgoing Gateway

Successful SIP UA Intrusion to ATS-QSIG End

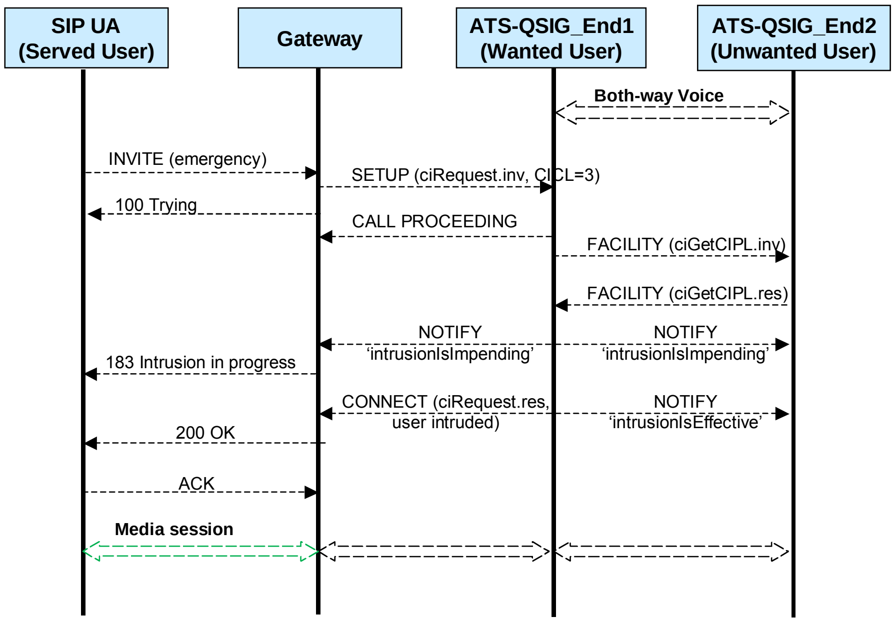


Fig. 129 – SIP to ATS-QSIG Successful Priority Call Intrusion

Priority call is displayed at ATS-QSIG End1 and manually answered

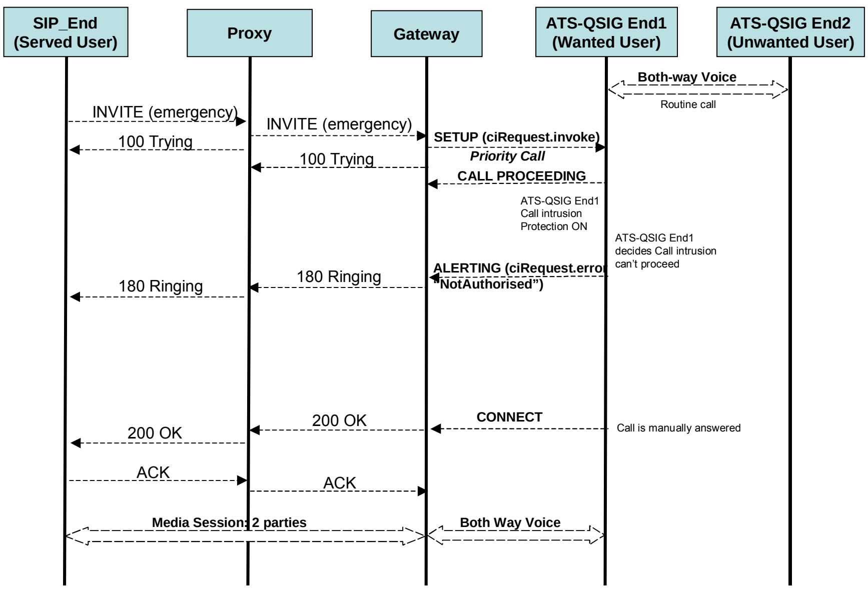


Fig. 130 – SIP to ATS-QSIG Priority Call Intrusion Forbidden by Wanted User

Priority call is displayed at ATS-QSIG End1 and manually answered

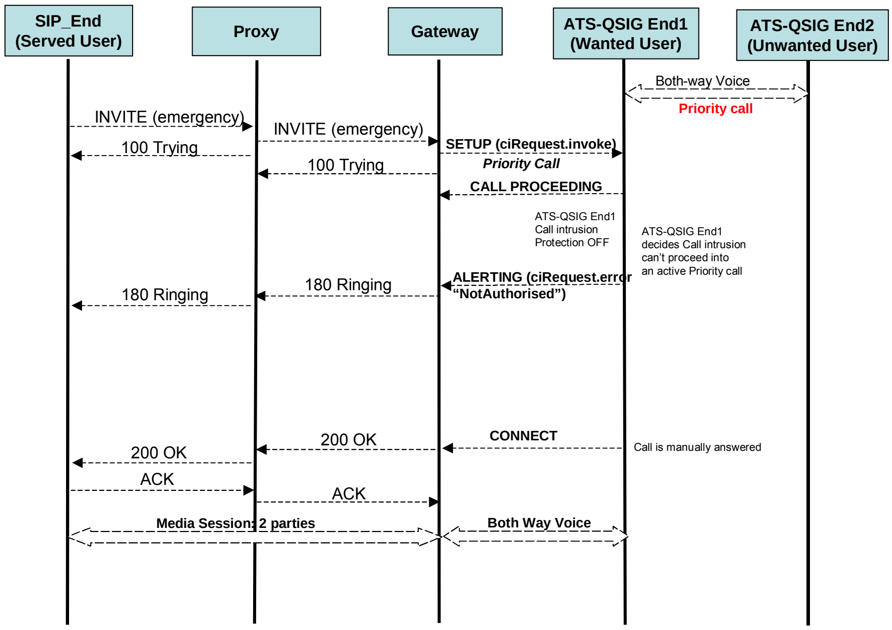


Fig. 131 – SIP to ATS-QSIG Priority Call Intrusion into another Priority Call Forbidden

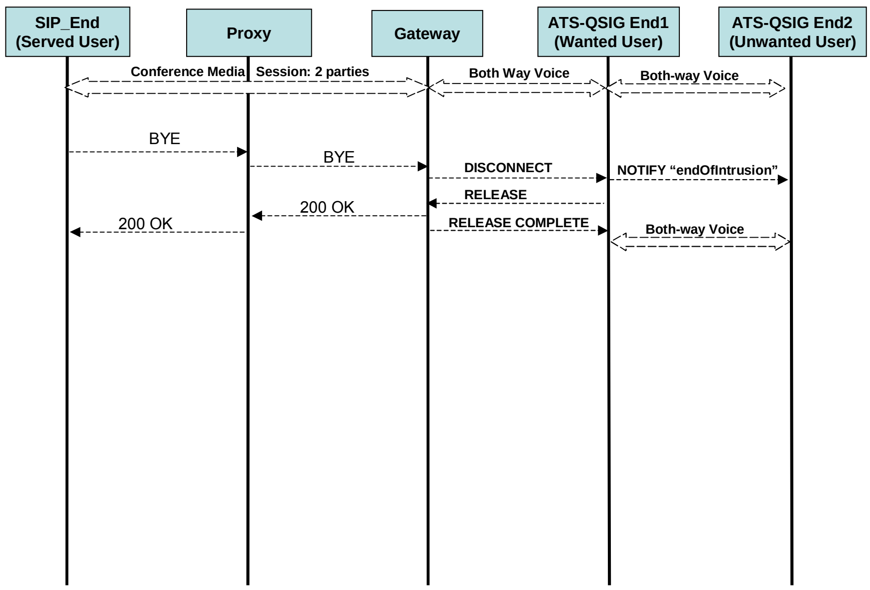


Fig. 132 – Call Clearing by SIP UA

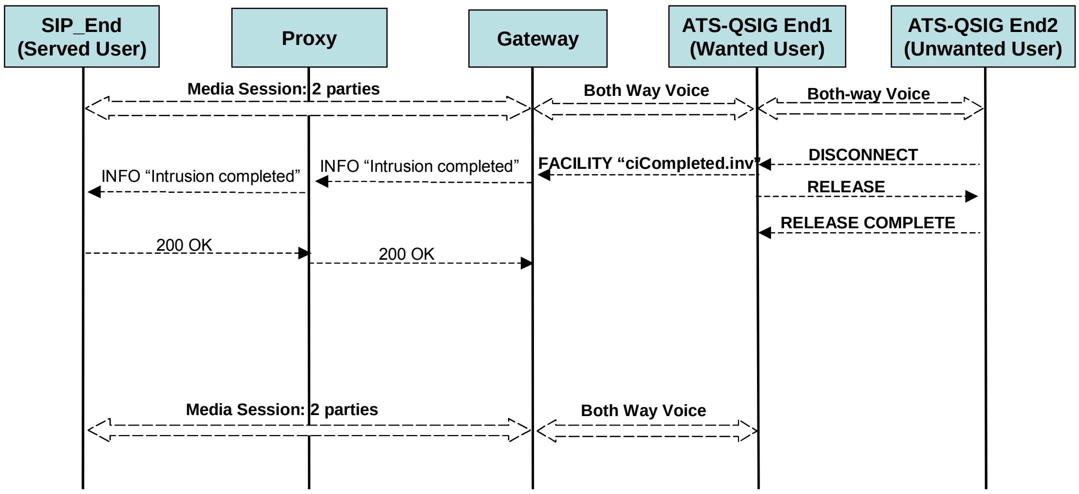


Fig. 133 – Call Clearing by Unwanted ATS-QSIG End

Automatic connection of Priority Call is not allowed

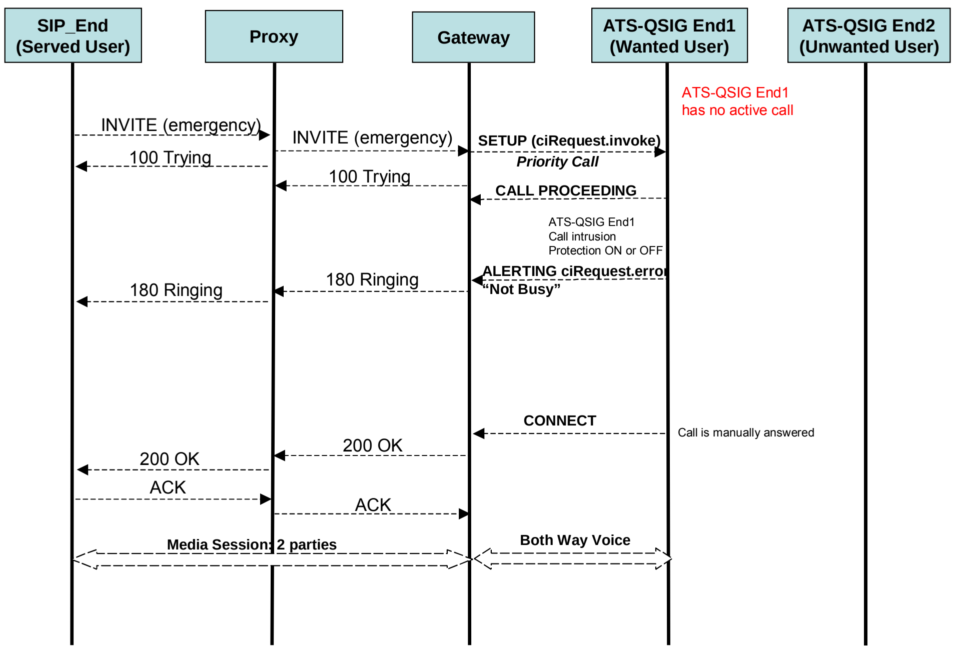


Fig. 134 – SIP to ATS-QSIG Priority Call Intrusion to a Non-busy Wanted User

ANNEX A

REFERENCES

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ANNEX B

ACRONYMS

Ack	Acknowledge
AGVN	Air Traffic Services Ground Voice communications Network
A/G	Air/Ground
AM	Amplitude Modulation
ANSP	Air Navigation Service Provider
ATA	Analogue Telephone Adapter
ATC	Air Traffic Control
ATM	Air Traffic Management
ATS	Air Traffic Services
ATS-No.5	Air Traffic Services – No.5 signalling system
ATS-QSIG	Air Traffic Services – Q reference point SIGnalling system
ATS-R2	Air Traffic Services – R2 signalling system
AVP	Audio/Video Profile
CICL	Call Intrusion Capability Level
CIPL	Call Intrusion Protection Level
CPICL	Call Priority Interruption Capability Level
CPIPL	Call Priority Interruption Protection Level
CWP	Controller Working Position
DA	Direct Access
DNS	Domain Name Service
ECMA	European Computer Manufacturers Association
G/G	Ground/Ground
HMI	Human Machine Interface
HTTP	HyperText Transfer Protocol
IA	Instantaneous Access
IANA	Internet Assigned Numbers Authority
ICCVC	Instantaneous Controller-Controller Voice Communication
IDA	InDirect Access
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication standardization sector
LAN	Local Area Network
LD-CELP	Low Delay - Code Excited Linear Prediction
MF	Multi-Frequency
MFC	Multi-Frequency Code
MSC	Message Sequence Chart
N. A.	Not Applicable
PABX	Private Automatic Branch eXchange
PCM	Pulse Code Modulation
PINX	Private Integrated services Network eXchange
PISN	Private Integrated Services Network
PSS1	Private Signalling System no. 1
PSTN	Public Switched Telephone Network
QoS	Quality of Service
Rec.	Recommendation
RFC	Request For Comments
RTCP	Real-time Control Protocol
RTP	Real-time Transport Protocol
Rx	Reception
S/MIME	Secure / Multipurpose Internet Mail Extensions
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SS-IA	Instantaneous Access Supplementary Service
TCP	Transmission Control Protocol

TDM	Time Division Multiplexing
TLS	Transport Layer Secure protocol
TU	Transaction User
Tx	Transmission
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
URI	Universal Resource Identifier
UHF	Ultra-High Frequency
VCS	Voice Communications System
VHF	Very High Frequency
VoIP	Voice over the Internet Protocol
WAN	Wide Area Network

ANNEX C

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