



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

INTEROPERABILITY STANDARDS FOR VOIP ATM COMPONENTS

VOLUME 3: EUROPEAN LEGACY TELEPHONE INTERWORKING

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ED-137/3B

January 2012

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FOREWORD

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

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

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

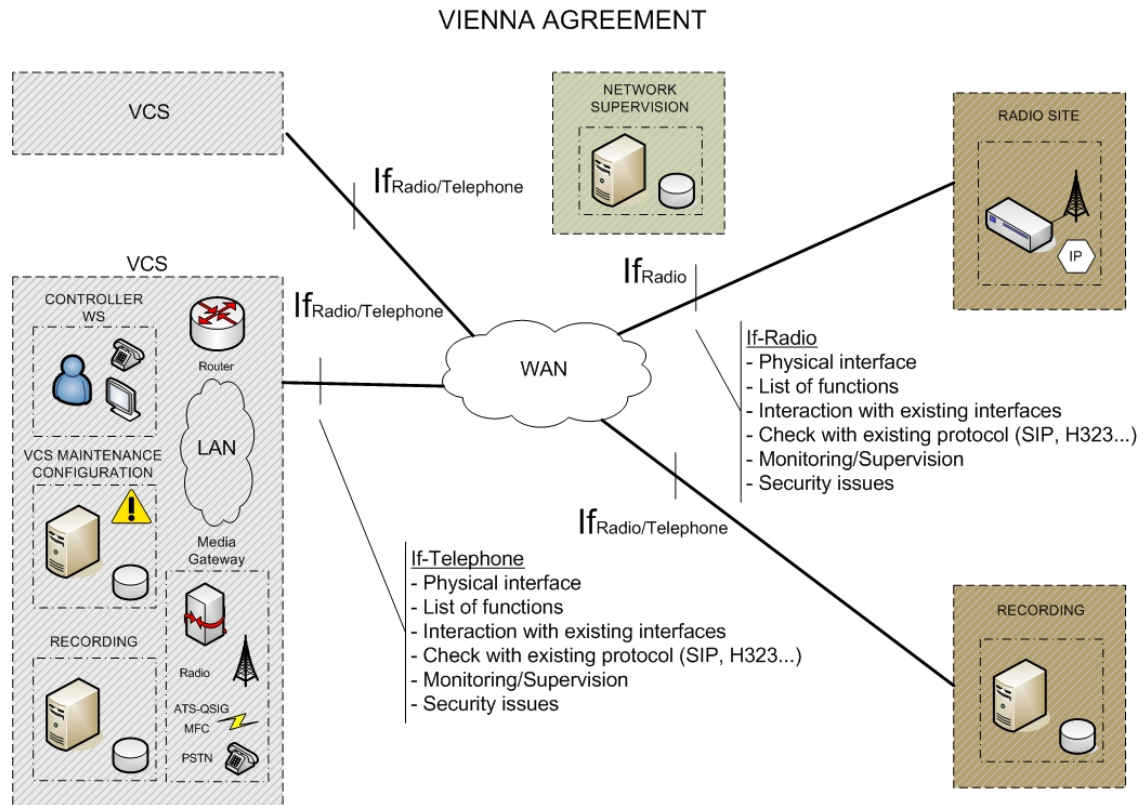
Consequently the following three documents were delivered:

ED-136 - VoIP ATM System Operational and Technical Requirements

ED-137 - Interoperability Standards for VoIP ATM Components

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

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



VoIP components are interconnected through an IP network and suppliers are free to define their internal architecture (IP/Ethernet, TDM/PCM - Time Division Multiplexing/Pulsed Code Modulation, etc.) 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, etc.)
- **VoIP Ground Radio Station Components** performing DSB-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

1.2 ED-137 VOLUME 3 PRESENTATION

The scope of the WG67 ED-137 Documents is to define the rules for VoIP implementations to support ATM communications:

- Radio for the Volume 1
- Telephone for the Volume 2
- European Legacy Telephone Interworking for the Volume 3
- Recording for the Volume 4
- Supervision for the Volume 5

This includes the performances requested for the related communications.

The present document, ED-137 Volume 3, proposes a profile standard for the use of gateway between SIP and legacy standards to establish, terminate and modify speech media sessions of the Ground Telephone Service in an Air Traffic Services Ground Voice Network (AGVN).

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

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

Detailed description of terminology:

1. **SHALL:** This word has the same meaning as the phrase "REQUIRED" and means that the definition is an absolute requirement of the specification.
2. **SHALL NOT:** This phrase means that the definition is an absolute prohibition of the specification.
3. **SHOULD:** This word, or the adjective "RECOMMENDED", means that there may exist valid reasons in particular circumstances to ignore a particular item, but the full implications must be understood and carefully weighed before choosing a different course.
4. **SHOULD NOT:** This phrase, or the phrase "NOT RECOMMENDED" mean that there may exist valid reasons in particular circumstances when the particular behaviour is acceptable or even useful, but the full implications should be understood and the case carefully weighed before implementing any behaviour described with this label.
5. **MAY:** This word, or the adjective „OPTIONAL“, means that an item is truly optional.

1.4 VERSION TRACKING

The present document, ED-137 Volume 3, **SHALL** be referenced as **legacy-eu.01** in the WG67-Version SIP header field as described in [35].

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 legacy standard 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 legacy 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 legacy 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 among others by corresponding IETF RFCs, ITU recommendations, ECMA standards and Eurocontrol specifications and recommendations, as indicated in each case.

2.2.1 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 signalling and control functionality for multimedia communications. SIP is an essential component in the context of other protocols to enable communication architectures.

In addition to the signalling functionality specified in this document, it is assumed that the IP AGVN components also include 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.

2.2.2 Audio Specifications

The audio transport **SHALL** be supported by the real-time transport protocol (RTP) (RFC 3550 [14]), and **SHOULD** be 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. G.711 A-law (in Europe) or μ -law (in USA/Japan) coding. If compression is needed ITU-T Rec. G.728 **SHALL** be used without Packet Loss Concealment. ITU-T Rec.

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

The session setup between ATS-QSIG and SIP **SHALL** include an fmtp attribute complementing the G.728 definition with the parameter sendsync within the media section of the session description, as specified in [35], in order to tell the remote party that it does insert the G.728 inband synchronization bit into the outbound RTP stream according to the bit stealing method specified in ECMA-312 [34].

An ATS-QSIG Gateway that receives this parameter from a remote party **SHALL NOT** insert the G.728 inband synchronization bit into the audio stream towards the ATS-QSIG end. It **SHALL** forward frames as they are received.

In the case, where the sendsync parameter is missing, the ATS-QSIG Gateway **SHALL** insert the G.728 inband synchronization bit into the audio stream towards the ATS-QSIG end. It **SHALL** only forward such modified frames and **SHALL** ensure that each frame starts with a G.728 codevector.

ITU-T Rec. G.729 audio coding **MAY** be used.

2.2.3 Interworking with ATS Legacy Systems

The SIP environment **SHALL** need to interwork with legacy (e.g. ATS-R2, ATS-No.5 or ATS-QSIG)-based AGVN in order to support calls originating in one environment and terminating in the other. 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.

2.2.4 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]

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 and in EUROCAE ED-137 "Interoperability Standards for VoIP ATM Components, Volume 2: Telephone [35].

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. Figure 2 shows the Interconnection Diagram.

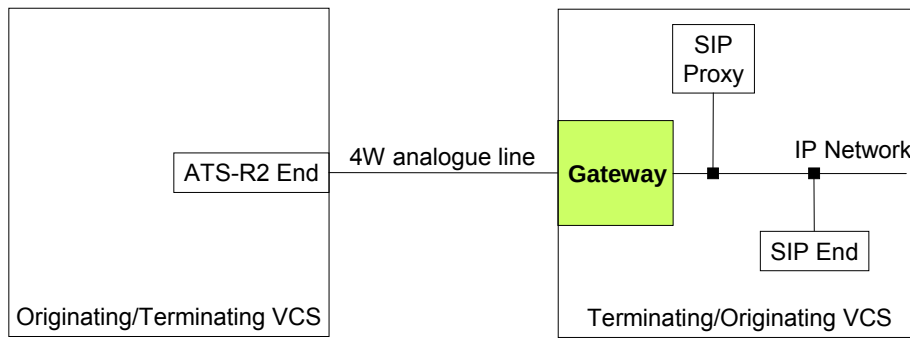


FIGURE 2: ATS-R2 / SIP INTERCONNECTION DIAGRAM

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 Figure 3.

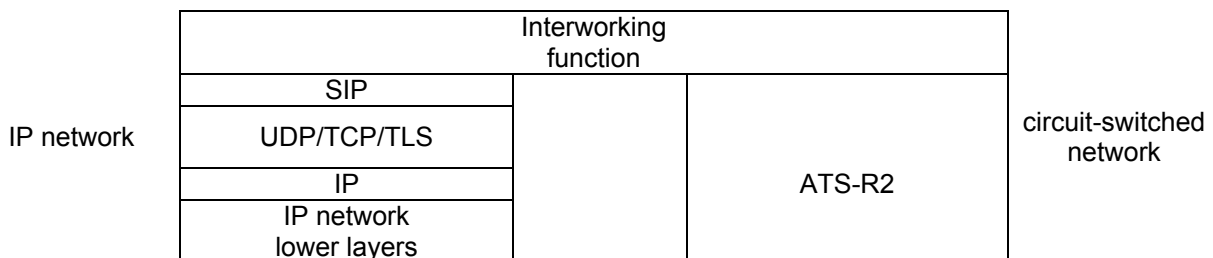


FIGURE 3: SIP / ATS-R2 PROTOCOL MODEL

In Figure 3, 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. The gateway **SHALL** ignore the SIP intermediate responses 182 or 183 and **SHALL NOT** generate any ATS-R2 signal towards the ATS-R2 end.

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 1. *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 1. SIP response 500 (Server internal error) **SHALL** be used as the default for cause values not shown in Table 1.

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 1 – 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 2 – 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 2. *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 [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 3 and Table 4. The ATS-R2 Release line signal **SHALL** be used as the default for SIP responses not shown in Table 3 and Table 4. 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 3 – 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 4 – 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 5.

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 5 – 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. The initial value for the Max-Forwards header field **SHOULD** be 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 6.

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 6 – 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 ATS-R2 parties engaged in that call **SHALL** receive an interrupt warning tone and the SIP provisional response 183 (Interruption in progress) shall be sent to the requesting SIP UA.

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 the ATS-R2 End, and send a INFO request containing a message body (Content-Type: text/plain) with the following content: "Interruption is impending" to the SIP UA (see Chapter 4.8.1 in [35]). It **SHALL** then 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 the ATS-R2 End and send a INFO request containing a message body (Content-Type: text/plain) with the following content: "Interruption terminated" to the SIP UA (see Chapter 4.8.1 in [35]). 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 ATS-R2 End. It **SHALL** then force release the call to be interrupted by sending an ATS-R2 "Blocking" line signal on the ATS-R2 line.

The Gateway **SHALL** send a SIP BYE request to the SIP UA being released containing a Reason header as defined in [35] with the following content:

Reason: WG-67; cause=1001; text="Emergency - Forced Release"

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 6 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 Reason header as defined in [35] with the following content:

Reason: WG-67; cause=1001; text="Emergency - Forced Release"

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 3. *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 4.8 in [35]) < P22 – 4s, with 4s guard time.*

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. 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 0.

Once a priority call is answered at the ATS-R2 end, the Gateway User Agent **SHOULD** 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 **SHOULD** send a notification about the intruded party, refer to Chapter 4.10 in [35].

4.7 AUDIBLE TONES

A gateway **SHALL** be capable of generating the audible tones indicated in Table 7 and sending them over the ATS-R2 line.

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 7 – 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.

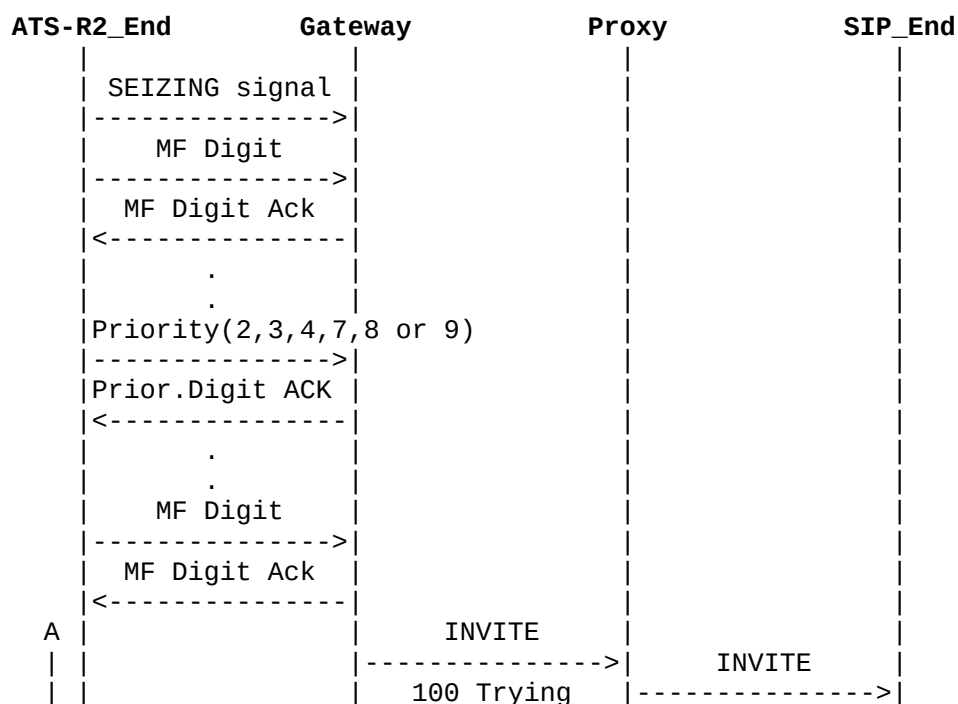
The Message Sequence Charts (MSC) in figures below show the information flows between the Call Control entity (labelled "Gateway") and respective Protocol Control entities for each signalling system (labelled "ATS-R2_End" and "SIP_End") within a Gateway VCS. Each information flow is named according to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



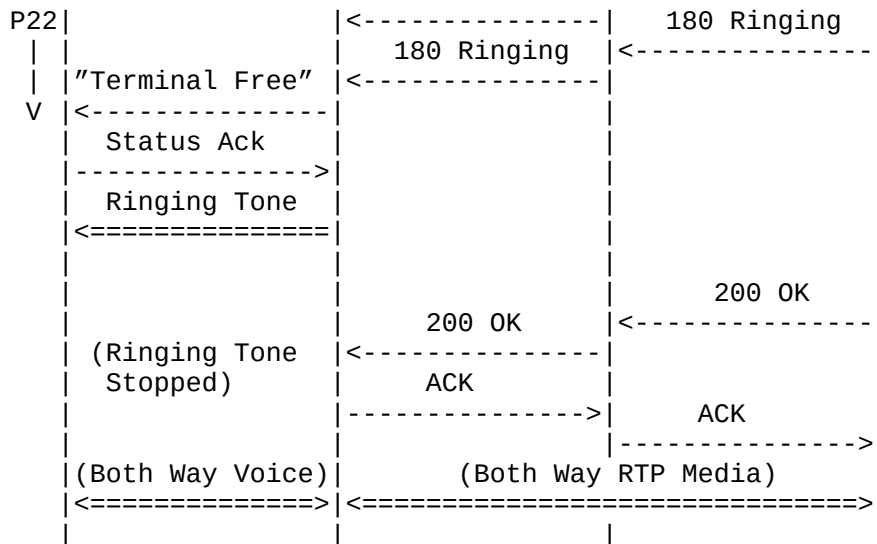


FIGURE 4: SUCCESSFUL ATS-R2 TO SIP ROUTINE CALL

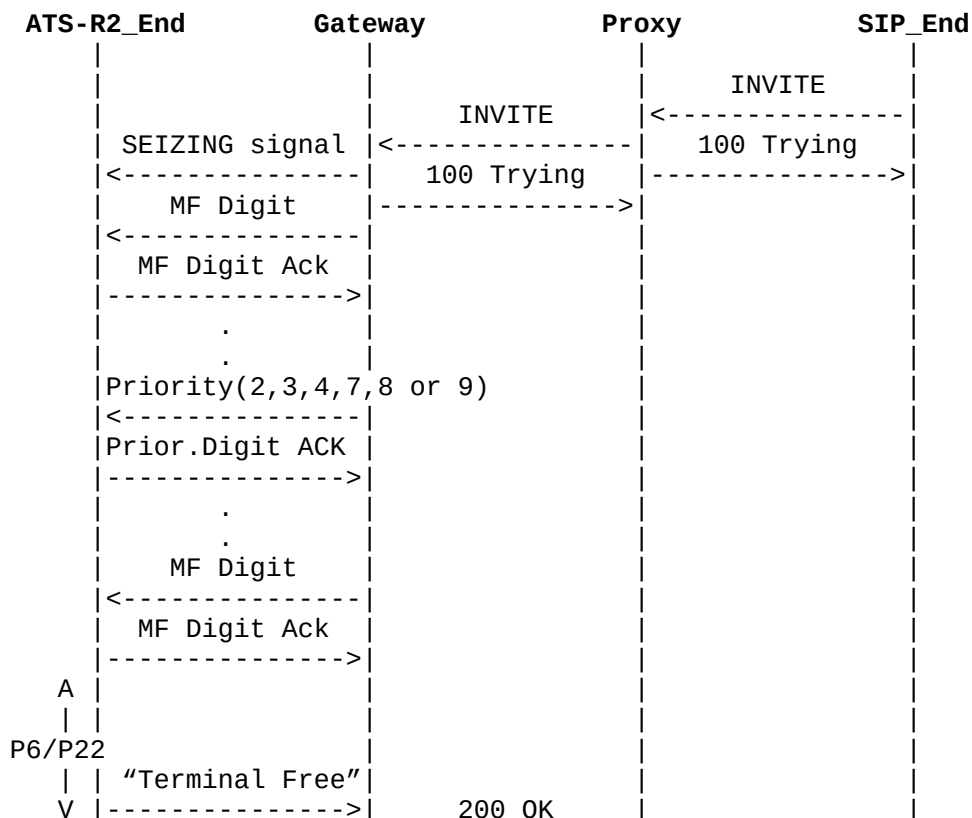
NOTE 4. For a routine call, the value of the "Priority" header field in the INVITE method shall be "urgent", "normal" or "non-urgent", as indicated in OTable 5..

NOTE 5. 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.



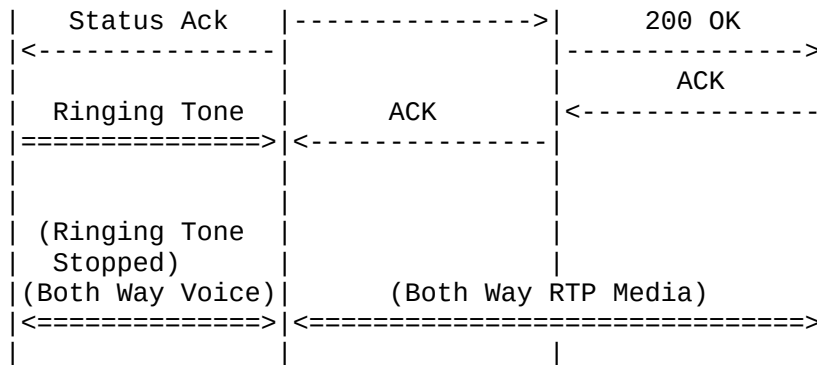


FIGURE 5: SUCCESSFUL SIP TO ATS-R2 ROUTINE CALL

NOTE 6. For a routine call, the ATS-R2 Priority Digit value shall be 2, 3, 4, 7, 8 or 9, as indicated in Table 6.

NOTE 7. 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.

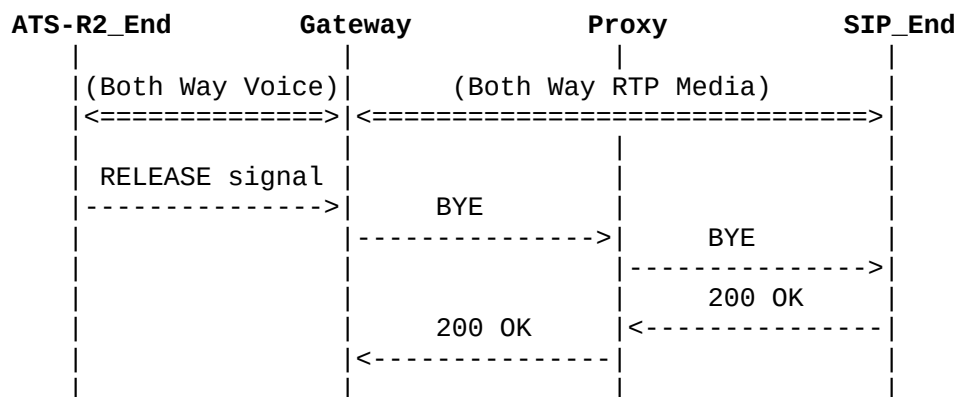


FIGURE 6: NORMAL CALL CLEARING FROM ATS-R2 END

4.8.4 Normal Call Clearing from SIP End

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

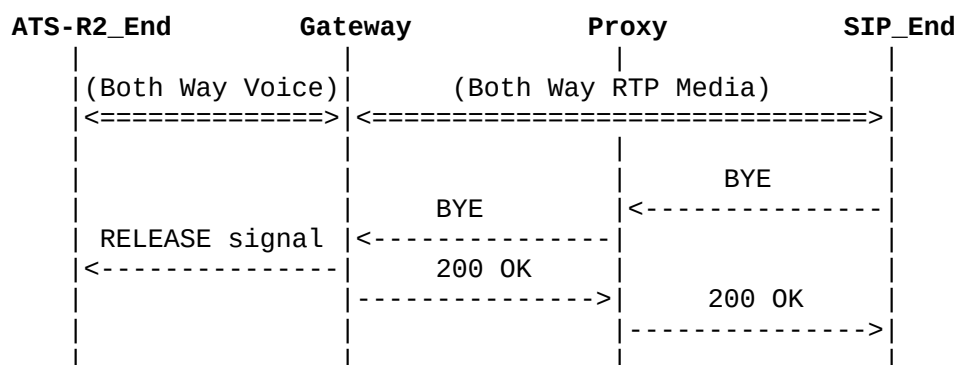
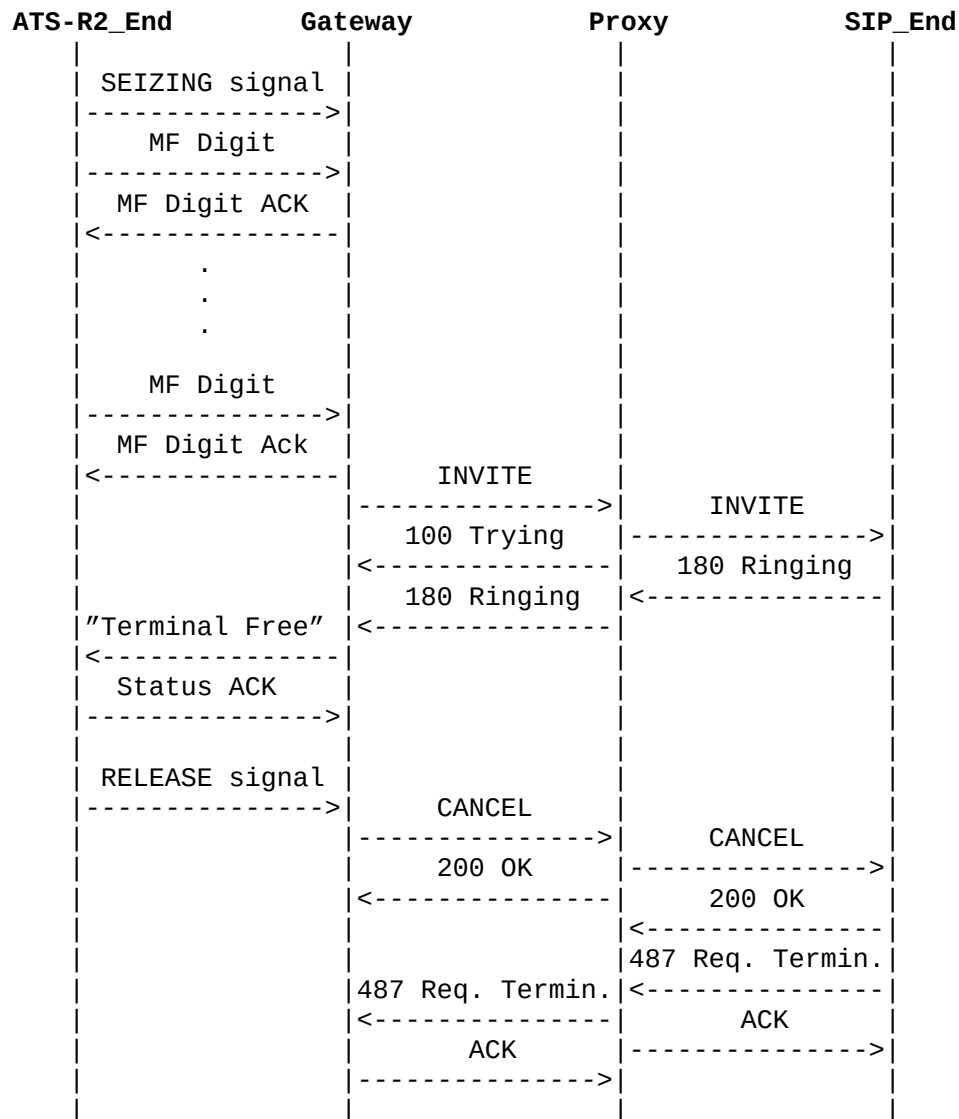


FIGURE 7: NORMAL CALL CLEARING FROM SIP END

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).

FIGURE 8: 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”.

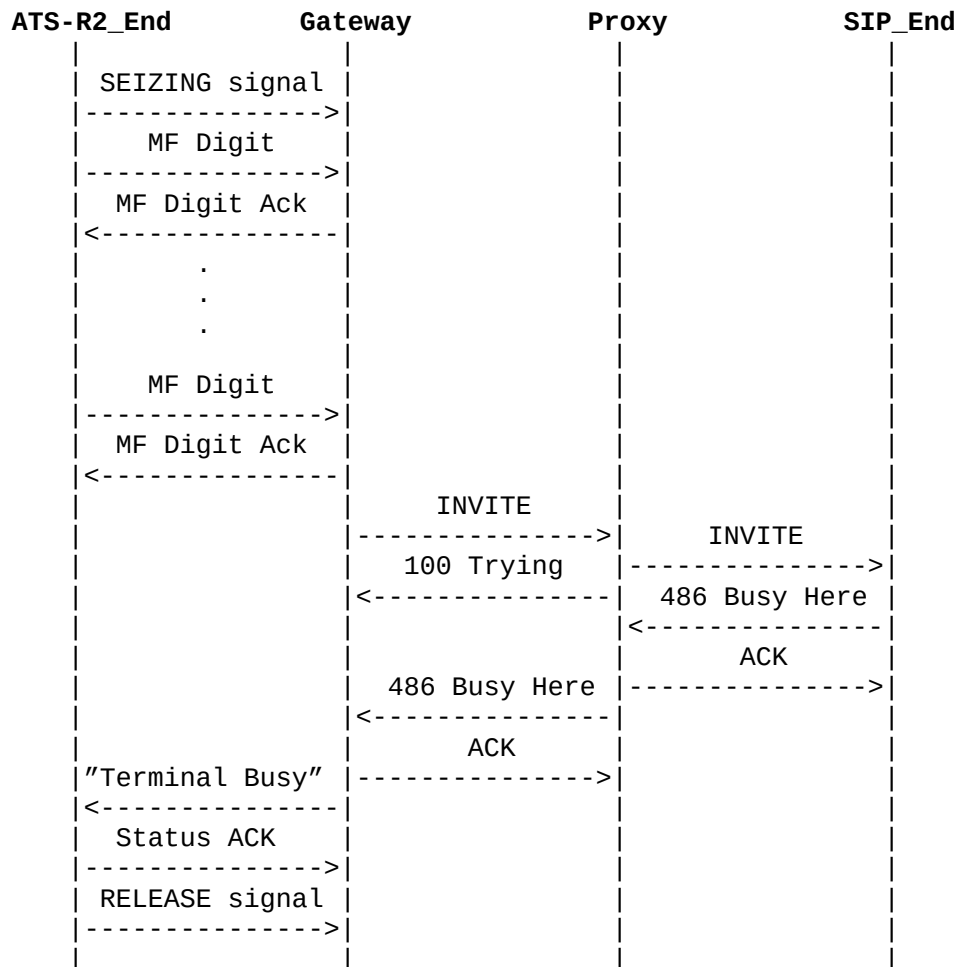


**FIGURE 9: UNSUCCESSFUL ATS-R2 TO SIP CALL.
MAPPING OF SIP RESPONSE TO ATS-R2 STATUS SIGNAL**

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.

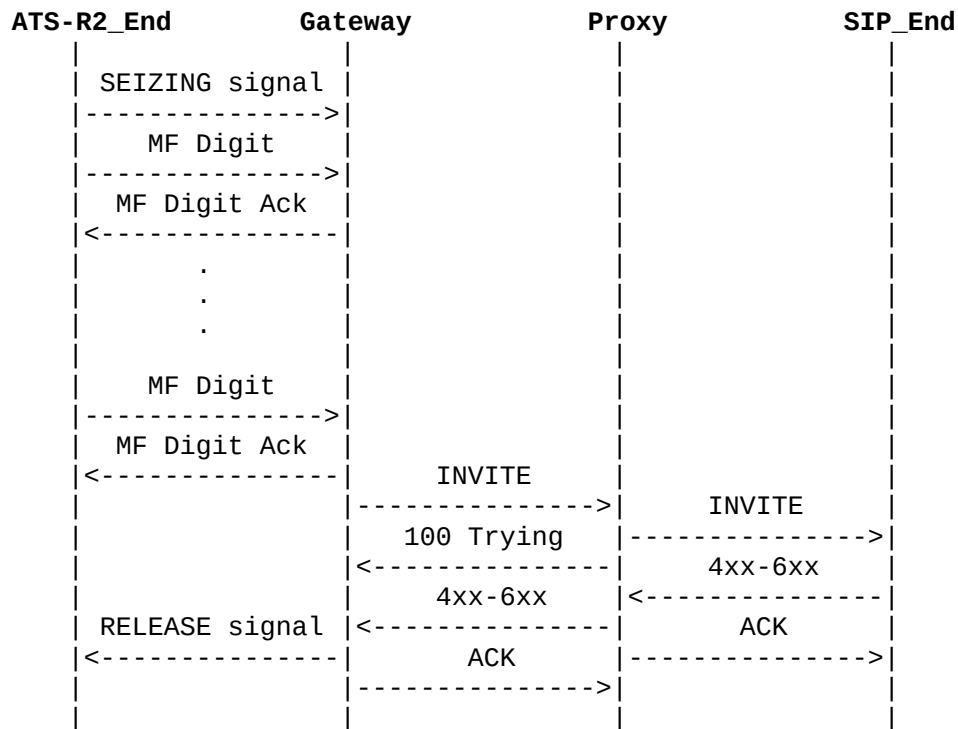


**FIGURE 10: UNSUCCESSFUL ATS-R2 TO SIP CALL.
MAPPING OF SIP ERROR RESPONSE TO ATS-R2 RELEASE LINE SIGNAL**

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).

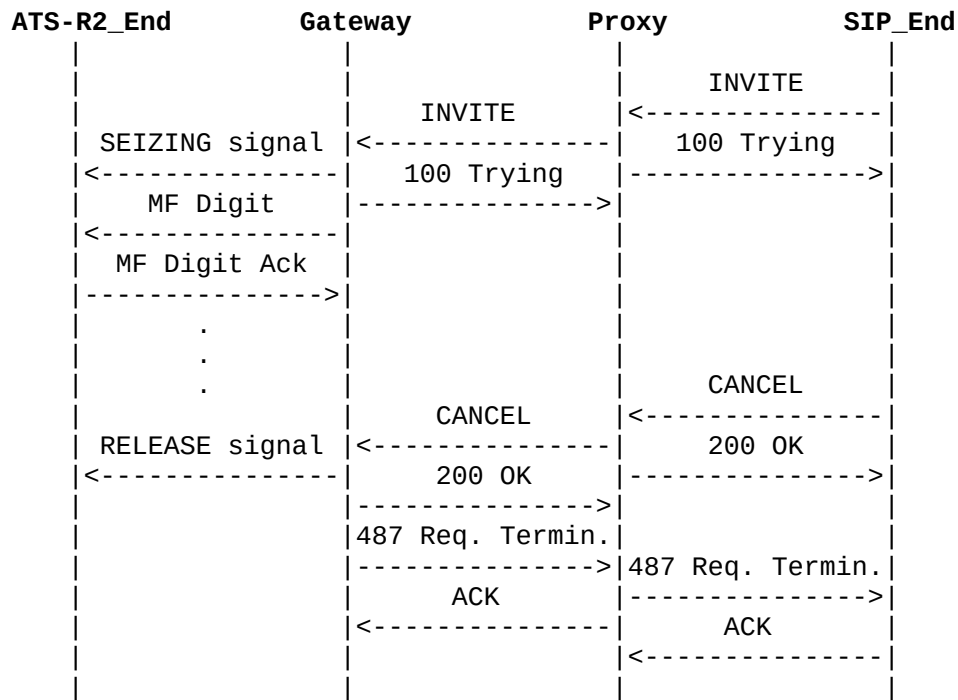


**FIGURE 11: UNSUCCESSFUL SIP TO ATS-R2 CALL.
CALL CLEARING FROM SIP NETWORK**

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".

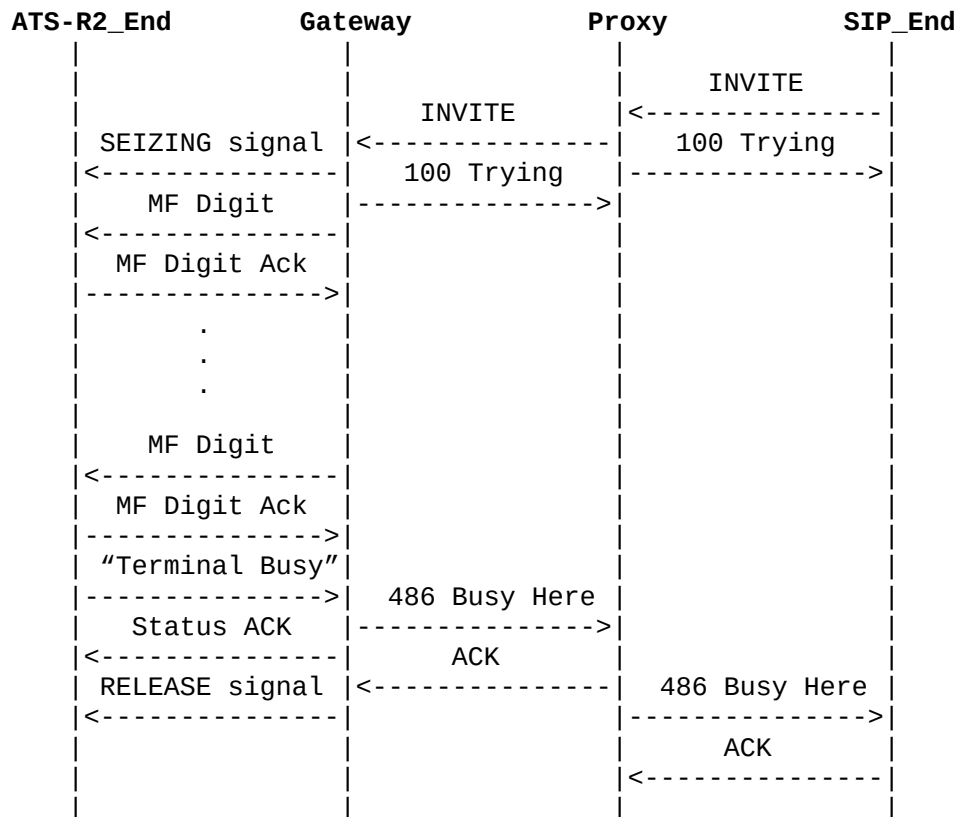


**FIGURE 12: UNSUCCESSFUL SIP TO ATS-R2 CALL.
RECEIPT OF STATUS SIGNAL FROM ATS-R2 END**