SIPForum Video Relay Service (VRS)

US VRS Provider Interoperability Profile

SIP Forum Document Number: VRS US Providers Profile TWG-6-1.0

1 Abstract

The US SIP Video Relay Service (VRS) Interoperability Profile is a profile of the Session Initiation Protocol (SIP) and related media aspects that enables inter-provider call handling for United States (US) Video Relay Service (VRS) calls. It specifies the minimal set of call flows, IETF and ITU-T standards that must be supported, provides guidance where the standards leave multiple implementation options, and specifies minimal and extended capabilities for US VRS calls.

2 Table of Contents

3 Introduction

Video Relay Services (VRS) are an important form of communications for the Deaf, Deaf-Blind, and Hard-of-Hearing. While working to improve and differentiate VRS services, providers also know the importance of maintaining interoperability for inter-provider calling. US providers have cooperated and published this SIP profile as the benchmark for implementing, testing and maintaining SIP interoperability. And as necessary, from time-to-time this document will receive updates to further advance this goal.

4 Scope

The scope of this document is intentionally limited. It covers only interoperation among US VRS providers subject to FCC regulations. The intent is to assist the quick migration of those providers from H.323 to a SIP-based infrastructure. It does not attempt to provide any features not present previously. Of note, while the security is weak, it is an incremental improvement on current VRS practice.

This document specifies only the interface between providers, and that between the providers and the iTRS database. It intentionally does not specify the interface between a RUE and a provider, allowing providers the most freedom to reuse their existing infrastructure.

This document is expected to be only a first step. Subsequent, more expansive, documents are anticipated, as discussed in section 13 (Future Plans).

5 Conventions and Terminology

5.1 Terminology from Requirements

Call. A SIP dialog or conference between two or more RUEs, or PSTN UEs.

Communication Modality (Modality). A particular form of communication that may be employed by two users. For example: English voice, Spanish voice, American Sign Language, English lip reading, French real-time-text, English

MSRP instant messaging. Here one Communication Modality is assumed to encompass both the language and the manner in which that language is exchanged. E.g., English voice and French voice are two different communication modalities.

Communication Relaying. the act of translating or interpreting between different Communication Modalities.

Communications Assistant (CA). An individual person who performs the function of Communication Relaying, using a CAUE. For example, with Video Relay, this is a Sign Language Interpreter.

Communications Assistant Identifier (CAID). An alphanumeric string that uniquely identifies a CA within a specific provider's service. Relay Service users can use this ID to provide feedback on service and law enforcement can use the ID to contact a CA involved in an emergency conversation.

Communications Assistant User Equipment (CAUE). A UE used by a CA to facilitate the relay of a call.

Default Relay Service: The Relay Service operated by a user's Default Relay Service Provider.

Default Relay Service Provider (Default Provider). The Relay Service Provider that registers and assigns a telephone number to a Relay User. A Relay User's Default Provider provides the relay service that handles incoming Relay Calls to the user. It also handles outgoing Relay Calls by default.

Dial-around Call: A Relay Call where the Relay User specified the use of a Relay Service Provider other than the Default Provider when initiating the call.

Full Intra Request (FIR). A request to a media sender, requiring that sender, to send a Decoder Refresh Point at the earliest opportunity. FIR is sometimes known as "instantaneous decoder refresh request"; "video fast update request"; or "fast update request".

Hearing Carry Over (HCO). A form of VRS where a Relay User is able to listen to the other party and in reply the CA uses their voice to interpret the Sign Language of the Relay User.

Interactive Media Response (IMR). A device that interacts with a caller via Communication Modalities supported by the caller. Typically used while awaiting the availability of a CA or the callee.

Media Description. An SDP description of a proposed media stream. This starts with an "m=" line and includes all following SDP lines up to but not including the next "m=" line.

Point-to-Point Call (P2P Call). A call directly between two RUEs.

PSTN UE. Equipment that interfaces with a human being via the PSTN, and mediates communication via voice.

PSTN User. An individual using a PSTN UE.

Relay Call. A call that allows persons with hearing or speech disabilities to use a RUE to talk to users of traditional voice services with the aid of a Communication Assistant (CA) to relay the communication. See [FCC-VRS-GUIDE].

Relay-to-Relay Call. A call between two Relay Users each using different forms (Video Relay, IP Relay, TTY) of Relay and associated Communication Assistants to assist in relaying the conversation.

Relay Numbering Administrator. The administrator of a Relay Numbering Directory.

Relay Numbering Directory (RND). A database administered by the Relay Numbering Administrator, the purpose of which is to map each registered Relay User's Relay Telephone Number to a URI at which the Relay User's RUE may be reached.

Relay Service (RS). A family of services that allow a registered Relay User to use an RUE to make and receive Relay Calls and Point-to-Point Calls. The functions provided by the Relay Service Platform include the provision of media links supporting the Communication Modalities used by the caller and callee, user registration and validation, authentication, authorization, ACD platform functions, routing (including emergency call routing), call setup, mapping, call features (such as call forwarding and video mail), and assignment of CAs to Relay Calls.

Relay Service Provider. An organization that operates a Relay Service. A Relay User selects a Relay Service Provider to assign and register a telephone number for their use, to register with for receipt of incoming calls, and as the default service for outgoing calls.

Relay Telephone Number. Telephone number assigned to a Relay User in the format defined by E.164.

Relay User. An individual that has registered with a Relay Service Provider, and who obtains service by using Relay User Equipment. Relay Users may be subject to regional requirements for using the service.

Relay User Address of Record (User AoR). The SIP address of record for the RUE.

Relay User E164 Number (User E164). The telephone number assigned to the RUE, in E.164 format.

Relay User Equipment (RUE). An SIP User Agent enhanced with extra features to support a Relay User in requesting and using Relay Calls. A RUE may take many forms. For example:

- a stand-alone device;
- an application running in standard device like a smart phone or tablet; or
- proprietary equipment connected to a server that provides the RUE interface.

Sign language. A language which uses hand gestures and body language to convey meaning, including but not limited to American Sign Language (ASL).

Telecommunications Relay Services (TRS). (from the FCC): "Telephone transmission services that provide the ability for an individual who has a hearing or speech disability to engage in communication by wire or radio with a hearing individual in a manner that is functionally equivalent to the ability of an individual who does not have a hearing or speech disability to communicate using voice communication services by wire or radio. Such term includes services that enable two-way communication between an individual who uses a text telephone or other nonvoice terminal device and an individual who does not use such a device, speech-to-speech services, video relay services and non-English relay services."

Two-stage dial-around: the Relay User first calls the dial-around provider. Then he/she gives the number of the callee to the CA, and the CA connects to the callee.

Video Interpreter (VI). A CA who can relay between sign language and speech.

Video Relay Service (VRS). A Relay Service for people with hearing or speech disabilities who use sign language to communicate using video equipment (Video RUE) with other people in real time. The video link allows the CA to view and interpret the Relay User's signed conversation and relay the conversation back and forth with the other party.

Voice Carry Over (VCO). A form of VRS where a person with a hearing disability is able to speak directly to the other party and in reply the Video Interpreter listens to the other party and uses Sign Language to communicate back to the person with the hearing disability.

5.2 Normative Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

6 Reference Architecture

The interfaces to be defined are labeled in drawing 1:

U2: The SIP signaling and media interface between RUEs and Relay Services.
R1: The SIP signaling and media interface between Relay Services.
D1: The interface between Relay Services and Relay Numbering Directories.

Components in drawing 1, further defined in the Terminology section 5.1 :

CAUE: Communications Assistant User Equipment

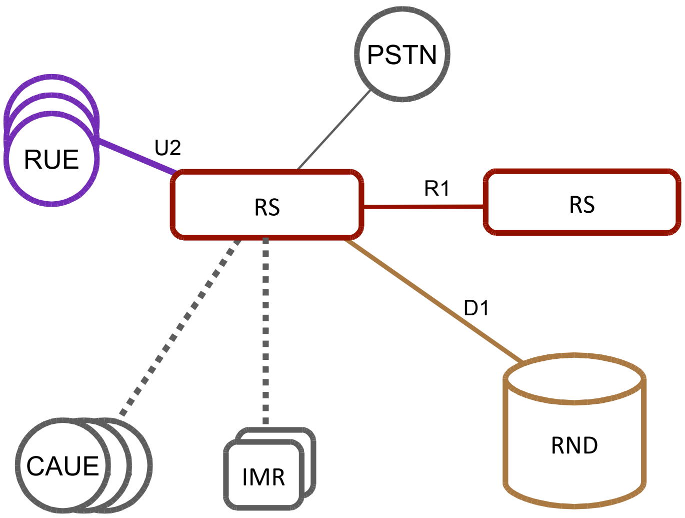
IMR: Interactive Media Response device

PSTN: PSTN UE

RND: Relay Numbering Directory.

RS: Relay Service.

RUE: Relay User Equipment



Drawing 1: Relay Service Interfaces

7 Key Assumptions and Limitations of Scope

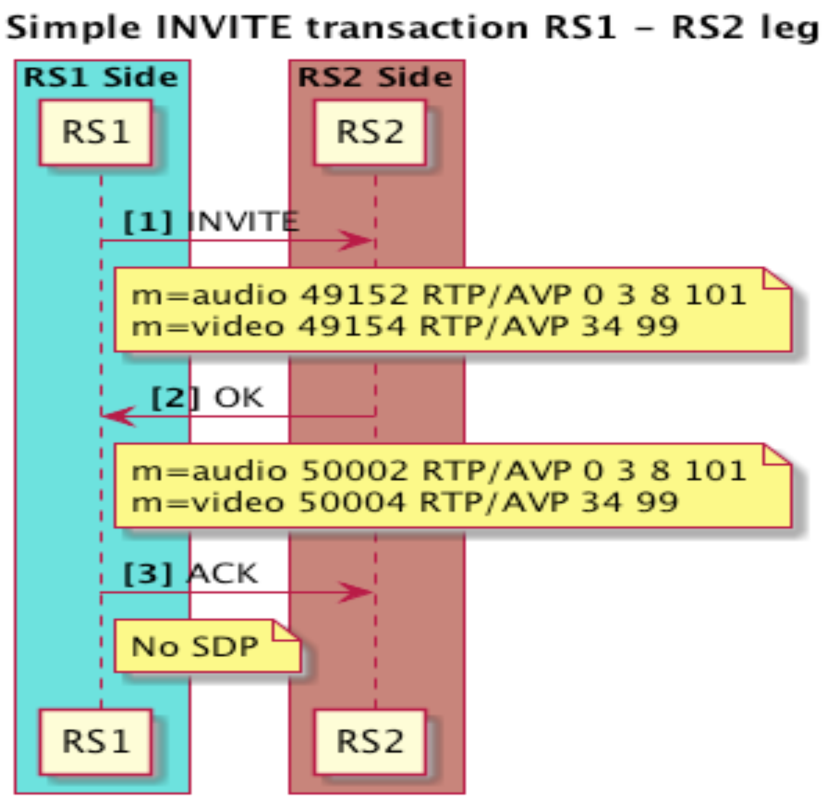
This profile is explicitly limited in scope to US providers operating under US FCC regulations. Addressing the needs of other jurisdictions is deferred.

Configuration of RUEs, and re-association of RUEs to a new default provider is not addressed by this document.

8 Use Cases

The two main variants of INVITE transactions that this document refers to are: INVITE with media; and INVITE without media (sometimes referred to as delayed media negotiation). These are referenced in the call flows starting at section 8.1 by use of the term 'basic INVITE'.

The first of these is the most common case, where the caller sends SDP in the INVITE. The callee responds with SDP in the final response. In the diagram, only the 'm' lines of the SDP are shown and non-final responses such as TRYING/RINGING are not shown. 'm' lines are examples only and real calls may have different media.



In the second variant, the INVITE contains no SDP and so the media negotiation is delayed until the final 200 OK response to the INVITE is sent. The ACK to the final response carries the caller's SDP answer. Again, only the 'm' lines of the SDP are shown and non-final responses such as TRYING/RINGING are not shown.

8.1 PSTN to RUE: two stage manual dial around

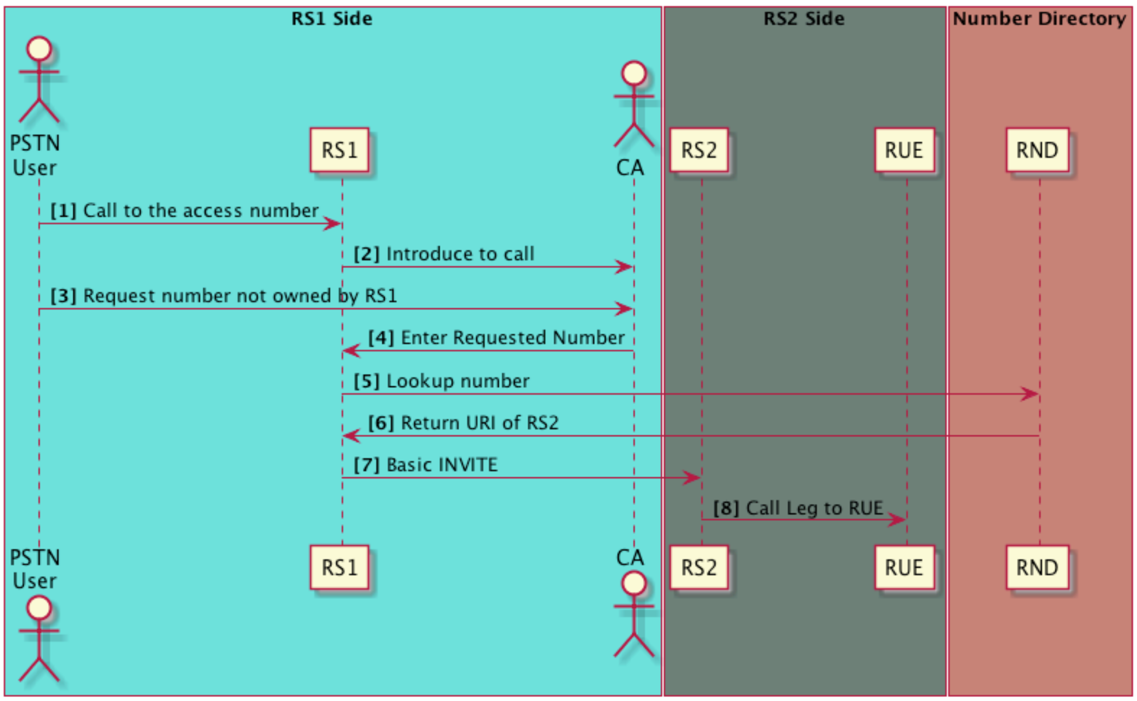
A PSTN user makes a call, via Relay Service 1 (RS1), to a user of Relay Service 2 (RS2). Relaying is performed by RS1.

8.1.1 End to End Overview

1. A PSTN user makes a call to the access number of a relay service (RS1);
2. RS1 answers the call and connects a Communications Assistant (CA);
3. The PSTN user requests a number that does not belong to RS1;
4. CA Enters Requested Number;

5. RS1 looks up the requested number in the Relay Number Directory (RND) to find the relay service (RS2) that owns the requested number;
6. RS1 connects to the edge proxy of RS2;
7. RS2 connects the call to the destination RUE; then
8. The call continues with RS1's CA .

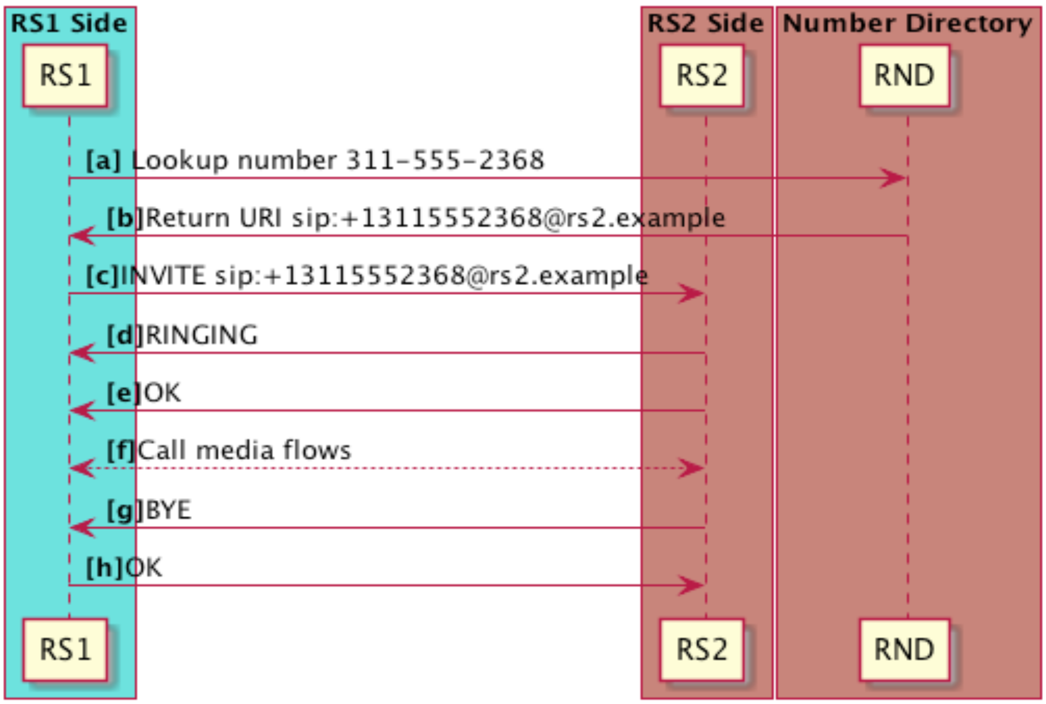
PSTN to RUE: End to End Call Flow



8.1.2 Detail on the RS1 – RS2 leg

a) RS1 looks up the requested number in the RND to find the relay service (RS2) that owns the requested number;
b) RND returns SIP URI referencing the called number at RS2;
c) RS1 sends an INVITE to RS2;
d) RS2 sends call progress information e.g. 180 RINGING;
e) 200 OK;
f) Call media flows;
g) BYE; then
h) 200 OK.

PSTN to RUE: Detail on the RS1 – RS2 leg



8.2 RUE – PSTN: two stage manual dial around

A RUE user makes a call, via Relay Service 1 (RS1), to a different Relay Service 2 (RS2). The RUE user uses the interpreting services of RS2 to contact a PSTN user. Relaying is performed by RS2.

8.2.1 End to End Overview

1. A RUE user makes a call to the access number of a non-default relay service (RS2);
2. RS1 looks up the requested number in the Relay Number Directory (RND) to find the relay service (RS2) that owns the requested number;
3. RND provides RS1 with the URI for RS2;
4. RS1 sends a Basic INVITE to RS2;
5. RS2 answers the call and connects a Communications Assistant (CA);
6. The RUE user requests a number associated with a PSTN user;
7. Call leg for CA setup with RS2 PSTN interface/gateway;
8. Call continues to the PSTN user.

8.2.2 Detail on the RS1 – RS2 leg

The call flow between RS1 and RS2 follows the basic SIP signaling specified in Section 8 .

8.3 RUE to RUE Point-to-Point call between users of different providers

An RUE user dials the Telephone Number(TN) of a user with a different default provider. The RUE user's default provider routes the call through destination provider and then the call goes on to the destination RUE. The caller's provider (RS1) discovers the destination provider (RS2) by looking up the destination TN in the RND.

1. A RUE user makes a call to RUE2 number of a non-default relay service (RS2);
2. RS1 looks up the requested number in the Relay Number Directory (RND);
3. RND provides RS1 with the URI for RUE2;
4. RS1 routes the call to RS2;
5. RS2 calls RUE2.

8.4 Video Mail

Video Mail service is provided to a VRS User by that user's default provider. Video mail may be recorded by the default provider for incoming calls to the VRS User in the following circumstances:

- the user has no RUE connected to the default provider;
- a call is delivered to the user's RUE but the RUE does not answer the call in a reasonable time;
- a call is delivered to the user's RUE but the user, via the RUE, indicates that the call should be rejected;
- a call is delivered to the user's RUE but the RUE is busy with another call.

In such cases the default provider may choose to divert the call to a video mail recording service.

In the following cases the call may reach the default provider via the R1 interface from another VRS provider before going to video mail recording:

- A RUE to RUE Point-to-Point call between users of different providers (section 8.3);
- PSTN to RUE: two stage manual dial around (section 8.1).

In both cases the profiled flow (a Basic Call) between RS1 and RS2 does not change. Only the unprofiled behavior between RS2, the called user's RUE, and the video mail server differs.

To allow RS2 time to timeout an unanswered call and direct it to a video mail server the call originator (RS1) must not impose a time limit less than the default SIP Invite transaction timeout of 3 minutes. (However the originating user may manually terminate the call before this timeout.)

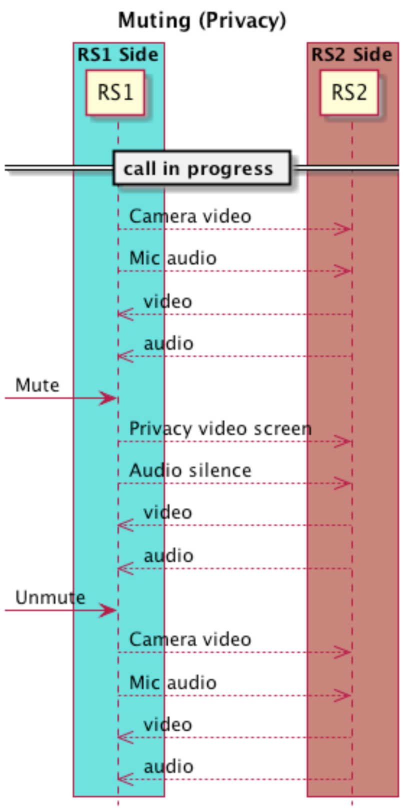
8.5 Muting (Privacy)

During a call, a video user wants to stop sending media. This can be achieved by sending a privacy screen and comfort noise, music, or other audio, or nothing. It is essential that the RUE's camera video and microphone audio NOT be sent while muted.

When not sending media on an RTP session it is important to periodically send something to prevent NAT bindings from being dropped.

It may also be possible to selectively mute video and/or audio. This is not shown in the diagram.

Muting is not signaled in SIP.



9 Relay Service

9.1 Interface D1

For this profile the function of the Relay Numbering Directory is provided by the iTRS database. The D1 interface is specified in the iTRS Provisioning Interface [iTRS-PI] and iTRS Query Interface [iTRS-QI].

The database accessed via the provisioning and query interfaces is based on ENUM [RFC3761] – a usage of DNS [RFC6116]. Queries typically take a telephone number as input and return DNS NAPTR records that may be evaluated, yielding a URI identifying the server that is responsible for calls to that telephone number. The provisioning interface allows a Relay Service to change the NAPTR records for selected telephone numbers.

9.1.1 Relay Numbering Directory entry format for SIP devices

For use with this profile:

- a NAPTR record MUST have a service-field of "E2U+sip";
- the hostname portion of the SIP URI generated by evaluating the NAPTR MUST contain a DNS FQDN, not an IP address;
- the generated URI MUST NOT contain a “user=phone” parameter;
- the generated URI MAY optionally include a port number.
- The *order* value for SIP NAPTR records MUST be lower than any H323 NAPTR records for the same phone number

For example:

```
2.1.2.1.5.5.5.1.0.8.1.itrs.us. 5 IN NAPTR 10 11 "u"
"E2U+sip" "!^(.*)$!sip:\1@providerXYZ.example.com!"

2.1.2.1.5.5.5.1.0.8.1.itrs.us. 5 IN NAPTR 20 11 "u"
"E2U+h323" "!^(.*)$!h323:\1@192.0.2.111:54321!"
```

(The h323 record is shown only to illustrate the relative *order* values.)

If a provider needs to stop SIP calls to an endpoint it MAY remove the SIP NAPTR record.

9.2 Interface R1

9.2.1 Properties

The following are attributes of a Relay Service:

- Each Relay Service MUST publish their source peering IP addresses to all other providers using an out-of-band mechanism.
- Each Relay Service has a set of User Address of Records (AoRs) that it manages.
  - For each User AoR there MUST be a unique Relay User E164 Number.
  - Each User AoR MUST be constructed using the procedure in section 10 (URI Representation of Telephone Numbers) using the Relay User E164 Number as the dial string.
  - The Relay Service serves as the Default Relay Service for each User AoR that it manages.

- Relay Service URIs and User AoRs MUST resolve (in accord with [RFC3263]) to globally routable IPv4 addresses. The AoRs MAY also resolve to IPv6 addresses.

9.2.2 Authentication and Authorization

9.2.2.1 Connection Authentication and Authorization

Authentication of the peer provider is achieved by checking the source IP address of SIP signaling traffic received on R1 to ensure that the sender's address is one of the known peering addresses for the peer provider.

9.2.2.2 RUE Location Information

In order to satisfy FCC requirements, the IP address of the RUE must be available to the Relay Service that provides relaying for a call. To meet this need in dial-around cases, the default provider MUST supply the IP address of the RUE (the public IP address of the RUE as observed by the default provider) to the dial-around provider. There are three cases:

- for calls originated by a RUE, the IP address of the RUE must be supplied in initial INVITE requests forwarded over the R1 interface;
- for INVITE requests received on the R1 interface and forwarded to a RUE, the IP address of the RUE must be supplied in all non-100 provisional responses and successful final responses to the INVITE;
- for INVITE requests received on the R1 interface and forwarded to a video mail server, the IP address of the video mail server must be supplied on all non-100 provisional responses and successful final responses to the INVITE.

The RUE IP address MUST be included in a SIP Call-Info header field containing a 'purpose' parameter with the value *trs-user-ip*.[1]

The URI element of this header field MUST contain the IP address of the RUE that originated the call, in the form of a SIP URI containing only an IP address. E.g.,

```
Call-Info: <sip:203.0.113.1>; purpose=trs-user-ip
```

The receiving provider MAY use this IP address to infer the geographic location of the RUE user.

If a dial-around provider does not receive the IP address of the originating RUE from the default provider, or is unable to verify that this address meets regulator

1 This is a non-standard purpose value, defined only by this specification. This approach has been adopted due to inability to obtain a standardized mechanism in a timely way. In the future this approach may be replaced with a standardized mechanism.

requirements for call reimbursement, it MAY refuse to provide relay service for the call. It then MAY signal this refusal to the default provider by returning a '403 Forbidden' response to the default provider.

If the request is refused in this way, the reason phrase of the response can be used to indicate that the cause of the rejection is lack of the RUE IP address. (E.g., the reason phrase could be: "No trs-user-ip provided".)

9.2.2.3 Request Authentication and Authorization

When a request is received on an authorized connection it MUST trust the validity of the From-URI, P-Asserted-Identity URI(s), and RUE location information.

9.2.3 General Handling of Incoming requests (from another provider)

If the source IP address of the request is not in the list of peers then the request must not be processed.

9.2.4 General Handling of Outgoing Requests (to another provider)

The sending provider MUST look up the AoR for the destination number in the RND. The request MUST be sent to the address obtained by resolving the SIP URI of that AoR according to standard SIP rules, as specified in [RFC3261] and [RFC3263].

	Called Provider Protocol Support		
Calling Provider Protocol Support	***SIP & H.323***	***SIP***	***H.323***
SIP & H.323	SIP	SIP	H.323
SIP	SIP	SIP	(Fail)
H.323	H.323	(Fail)	H.323

Table 1

Support for H.323 is not specified by this document. However the RND MAY also contain NAPTR entries for H.323. For providers that also support H.323 Table 1 shows which protocol to use based on the capabilities of calling and called provider.

The From-URI MUST satisfy one of the following criteria:

- SIP:Anonymous@anonymous.invalid (in the case of CallerId blocking)
- the User AoR.

If the original sender of the request is one of the sending provider's own users, or can otherwise be verified by the sending provider, then a P-Asserted-Identity header containing the SIP AoR of the sender MUST be included (see 12.2 , Use of P-Asserted-Identity in requests).

> NOTE: The above includes the case where the sending provider is forwarding a call from the PSTN. The provider needs to decide if caller identity received from the PSTN interface is trusted. If and only if that is so it will include that identity in P-Asserted-Identity.

When P-Asserted-Identity is included, the From-URI MUST match it or else be the anonymous URI.

9.2.5 Outgoing REFER requests

Transfer processing must be handled within a single relay service. REFER requests [RFC3515] MUST NOT be sent over the R1 interface, and REFER MUST NOT be listed in the Allow header.

9.2.6 Incoming Calls (from another provider)

An incoming INVITE request is processed according to section 9.2.3 (General Handling of Incoming requests (from another provider)) and section 9.2.9 (Offer/Answer Procedures for all calls).

When the incoming INVITE request is addressed to a RUE connected to this provider, then the IP address of the RUE MUST be included in the 2xx response to the INVITE, in accord with section 9.2.2.2 (RUE Location Information).

9.2.7 Outgoing Calls (to another provider)

An outgoing INVITE request is processed according to section 9.2.4 (General Handling of Outgoing Requests (to another provider)) and section 9.2.9 (Offer/Answer Procedures for all calls).

When the outgoing call is from a RUE connected to this provider, the IP address of the RUE MUST be included in the INVITE, in accord with section 9.2.2.2 (RUE Location Information).

The originating Relay Service MUST NOT impose a time limit less than the default SIP Invite transaction timeout. (The reason is to allow time for terminating provider to decide if the called user has not answered, and transfer the call to video mail.)

9.2.8 Transfers

Transfers are not supported on this interface.

9.2.9 Offer/Answer Procedures for all calls

Unless otherwise specified here, offer/answer behavior MUST comply with basic offer/answer rules specified in RFC3261, RFC3264, and clarified by RFC6337.

9.2.9.1 All Offers and Answers

The following conditions apply to all offers and answers:

- Media addresses (in “c=” lines) MUST be globally routable IPv4 addresses. IPv6 MAY be used in ICE candidates when agreed upon between providers. NAT traversal procedures MUST NOT be required to send/receive.
- Media addresses MAY use a different IP address than the one used for SIP signaling.
- Media offers MAY be asymmetrical, meaning that sending capabilities may differ from receiving capabilities. If multiple codecs are negotiated for an “m=” line, usage of codecs may switch without a re-INVITE.
- All RTP m-lines SHOULD specify RTP/AVP as the declaration of the supported RTP profile, even when the RTP/AVPF profile is supported.
- RTP/AVPF attributes MAY be declared even though RTP/AVP was declared for the profile.
- Supported RTP/AVPF attributes MUST be declared.

9.2.9.2 Initial Offer in Call

The initial offer in a call may appear in the initial INVITE, or in a response to the initial INVITE. The following conditions apply to the initial offer:

- It MUST include a single video m-line.
- It MUST include a single audio m-line.
- It MUST include all the mandatory to implement (MTI) codecs listed in section 11.1 (Media Attributes per Component) for each media type included, and SHOULD include all other supported codecs for those media.
- A text media stream MAY be included.

9.2.9.3 Subsequent Offers

During a call, INVITE transactions MAY be used to put all media on hold and to retrieve all media from hold. Offers SHOULD use the SDP attribute 'sendonly' to signal that a stream is on holding if hold media is to be sent, and SHOULD use 'inactive' when holding media is not to be sent.

9.2.9.4 Answers

This profile does not specify behavior if a received offer has more than one m-line of the same media type.

When answering a received offer including a media description that contains a media type and Relay Service MTI codec listed in section 11.1 , the receiving Relay Service MUST NOT reject that media description in the answer; it MAY however accept any matching codec, not necessarily a MTI codec.

9.2.9.5 Media Direction Attributes

The SDP media direction attributes (sendrecv, sendonly, recvonly, inactive) can be used at the discretion of the two parties in a call to negotiate the flow of media – audio, video, and text. These attributes can be used as part of implementation of features such as call hold/resume. (The receipt of an offer containing one of these attributes does not provide enough information for the recipient to infer what feature is intended.)

The media direction may be changed concurrently and consistently for all the media streams, or the media direction may be changed independently for each media stream.

An answerer SHOULD NOT reject one media stream simply because its direction differs from that of another media stream.

10 URI Representation of Telephone Numbers

URIs derived from non-URI sources (dial strings) MUST be represented as follows:

- A dial string that represents an E.164 number MUST be represented as a SIP URI with a URI "user=phone" parameter. The user part of the URI MUST be in conformance with 'global-number' defined in RFC 3966. The user part MUST NOT contain any 'visual-separator' characters.
- The 'hostport' (RFC 3261) to use in the URI is dependent upon the context in which the URI is to be used.

11 Media / Codecs

11.1 Media Attributes per Component

	R1 Interface
Video Codecs	

MTI *(before January 1, 2016)*	*H.263 [H263] [RFC4629]*
MTI *(after January 1, 2016)*	H.264 [H264] [RFC6184] Constrained Baseline Profile, Level 1.3, packetization mode 1
recommended *(before January 1, 2016)*	H.264 [H264] [RFC6184] Constrained Baseline Profile, Level 1.3, packetization mode 1
Audio Codecs	
MTI	G.711, telephone-events [RFC4733]
recommended	G.722.2
Text Codecs	
MTI	T.140 [RFC4103] (if offered)
recommended	T.140

11.2 RTP & RTCP

All media streams MUST be exchanged using the real-time transport protocol (RTP) as described in [RFC 3550].

All RTP and RTCP traffic over UDP MUST use symmetric RTP [RFC4961].

Receivers of RTP traffic MUST be capable of processing RTP packets with a different packetization rate than the rate used for sending.

11.3 Bandwidth Negotiation and Flow Control

During a call, codec control messages SHOULD be used as described in RFC 5104, to negotiate maximum bitrate. Specifically Temporary Maximum Media Stream Bit Rate Request TMMBR SHOULD be used where RUEs have detected the need to decrease or increase the bit rate.

Where either side of a session doesn't support CCM TMMBR, INVITE messages MAY be used during a call to renegotiate the use of bandwidth.

11.4 RTP/AVPF Profile

Implementations MUST support the RTP/AVPF profile per RFC 4585 for video RTP sessions, but SHOULD signal “RTP/AVP” in the SDP m-line as specified in section 9.2.9.1 .[2] Supporting the RTP/AVPF profile allows implementations to use advanced RTCP mechanisms, like indicating packet loss, requesting intra frame and temporary bitrate change indication, which are essential for video streams.

Supported AVPF messages MUST be declared by RTCP Feedback attributes. Since implementations convey media streams from RUEs of varying background, there may be situations when no AVPF attributes are supported in a session.

11.5 Intraframe Request

Use of call control messages for signaling FIR SHOULD be used as described in RFC 5104. Where CCM FIR has not been negotiated because either side of the call cannot support it, SIP INFO messages MAY be used to send XML encoded FIR messages according to RFC 5168.

12 Asserted Identity

This profile makes use of the mechanisms defined in [RFC3325], with a single trust domain spanning all of the US VRS Relay Providers.

12.1 Spec(T)

The Spec(T) applicable to the profile is defined as follows:

1. Protocol requirements

   This document defines the protocol requirements for all members of Spec(T). (This includes RFC3261, RFC3323 and RFC3325.)

2. Authentication Requirements

   Authentication requirements for participants in the R1 interface are specified in 9.2.2.1 (Connection Authentication and Authorization).

   RUEs authenticate to their Default Provider in accord with provider-defined mechanisms.

   Internal nodes with a provider authenticate to one another in accord with provider-defined mechanisms.

3. Security Requirements

2 This use of “RTP/AVP” for RTP/AVPF profile is documented in IMTC SIP Video Profile Best Practices [IMTC1013].

SIP connections between providers, over the R1 interface use TCP by default but MAY use TLS by private agreement. The TCP connections are not secured.

4. Scope of Trust Domain

   The Trust Domain specified in this agreement consists of :

   - a set of Relay Service Providers that have established a full mesh of peering relationships and IP peering addresses with one another;
   - a set of Relay Service peering nodes, each operated by one of the Relay Service Providers, reachable at the IP peering addresses;
   - other Relay Service nodes, operated by Relay Service Providers, that exchange SIP messages with the Relay Service peering nodes of the same Relay Service Provider.

   The following are explicitly *excluded* from the Trust Domain:

   - Devices (including RUEs) operated by Relay Users;
   - PSTN interfaces and gateways.

5. Implicit handling when no Privacy header is present

   If no Privacy header is present, all P-Asserted-Identity header fields MUST be removed from messages leaving the trust domain.

12.2 Use of P-Asserted-Identity in requests

The P-Asserted-Identity header is used across the R1 interface to convey the authenticated identity of the sender (when known to the sending provider) to the receiving provider.

The P-Asserted-Identity header can also be passed among other nodes operated by a provider that fall within the trust domain.

12.3 Use of P-Asserted-Identity in responses

This profile intentionally does not require, or define (or forbid) the use of P-Asserted-Identity in responses.

12.4 Use of P-Preferred-Identity

While [RFC3325] defines the P-Preferred-Identity header, this profile does not specify its use. Nodes receiving messages containing P-Preferred-Identity MAY ignore and/or remove it.

13 Future Plans

As noted in the Scope section, this document describes a first step for US VRS providers to begin interoperating via SIP. It is anticipated that this document will be followed by one or more others with broader, international, scope encompassing more of the primary goals and other features already described in [VRS-Charter]. Methods for securing the signaling and media will be in scope for those efforts.

14 Contributors

The following people have contributed substantial text and/or review comments to this document:

Ruben Alanis, Grant Beckmann, Scot Brooksby, Leopold Claussen, David Estrella, Josh Evans, Jared Evans, James Hamlin, Gunnar Hellström, Jeremy Jack, George Lee, John Martin, Bruce Mattie, Jose Periera, Kristofer Pettersson, Isaac Roach, Lydia Runnels, Steven Saunders, Ed Sayers, Jeremy Shaffner, Joshua Shaffner, Richard Shields, Roger Slusser.

15 References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

[RFC3263] Rosenberg, J., "Session Initiation Protocol (SIP): Locating SIP Servers", RFC 3263, June 2002.

[RFC3264] Rosenberg, J. and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)", RFC 3264, June 2002.

[RFC3323] Peterson, J., "A Privacy Mechanism for the Session Initiation Protocol (SIP)", RFC 3323, November 2002.

[RFC3325] Jennings, C., Peterson, J., and M. Watson, "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks", RFC 3325, November 2002.

[RFC3515] Sparks, R., "The Session Initiation Protocol (SIP) Refer Method", RFC 3515, April 2003.

[RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.

[RFC3761] Faltstrom, P., “The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)”, RFC 3761, April 2004.

[RFC4103] Hellstrom, G., “RTP Payload for Text Conversation”, RFC 4103, June 2005.

[RFC4961] Wing, D., “Symmetric RTP / RTP Control Protocol (RTCP)”, July 2007.

[RFC4629] Ott, J., Borman, C., Sullivan, G., Wenger, S., and R. Even, Ed., "RTP Payload Format for ITU-T Rec. H.263 Video", RFC 4629, November 2006.

[RFC4733] Schulzrinne, H., “RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals”, RFC 4733, December 2006.

[RFC5104] Wenger, S., “Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)”, RFC 5104, February 2008.

[RFC5168] Levin, O., “XML Schema for Media Control”, RFC 5168, March 2008.

[RFC6116] Bradner, S., “The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)”, RFC 6116, March 2011.

[RFC6184] Wang, Y., Even, R., Kristensen, T., and R. Jesup, "RTP Payload Format for H.264 Video", RFC 6184, May 2011.

[RFC6337] Okumura, S., Sawada, T., Kyzivat, P., “Session Initiation Protocol (SIP) Usage of the Offer/Answer Model”, August 2011.

[H263] International Telecommunications Union - Telecommunication Standardization Sector, "Video coding for low bit rate communication", ITU-T Recommendation H.263, January 2005.

[H264] ITU-T Recommendation H.264, "Advanced video coding for generic audiovisual services", March 2010.

[FCC-VRS-GUIDE] “Guide: Telecommunications Relay Service (TRS)”, June 2014, http://www.fcc.gov/guides/telecommunications-relay-service-trs

[IMTC1013] SIP Video Profile Best Practices, IMTC, October 2014. http://portal.imtc.org/DesktopModules/Inventures_Document/FileDownload.aspx?ContentID=21794

[iTRS-PI] Neustar,”iTRS Provisioning Interface Guide”,iTRS_PI_Guide_4.1_v1.0, June 2012

[iTRS-QI] Neustar,”iTRS Query Interface Guide”,iTRS_QI_Guide_4.0_v1.0, February 2012

[VRS-Charter] “Video Relay Service (VRS) Task Group Charter “, http://www.sipforum.org/content/view/404/291/

16 Source for Call Flow Diagrams

16.1 Simple INVITE transaction RS1 - RS2 leg

```
@startuml

title Simple INVITE transaction RS1 - RS2 leg

box "RS1 Side" #5fddd6
participant RS1
end box


box "RS2 Side" #bb7068
participant RS2
end box


autonumber "<b>[0]"

RS1 -> RS2: INVITE
note right of RS1
    m=audio 49152 RTP/AVP 0 3 8 101
    m=video 49154 RTP/AVP 34 99
end note
RS2 -> RS1: OK
note right of RS1
    m=audio 50002 RTP/AVP 0 3 8 101
    m=video 50004 RTP/AVP 34 99
end note
RS1 -> RS2: ACK
note right of RS1
    No SDP
end note

@enduml
```

16.2 PSTN to RUE: End to End Call Flow

```
@startuml

title PSTN to RUE: End to End Call Flow

box "RS1 Side" #5fddd6
actor "PSTN\nUser" as PSTNUSER
participant "RS1" as RS1
actor CA
end box


box "RS2 Side" #5c7068
participant RS2
```

```
participant RUE
end box


box "Number Directory" #bb7068
participant RND
end box

autonumber "<b>[0]"

PSTNUSER -> RS1: Call to the access number
RS1 -> CA: Introduce to call
PSTNUSER -> CA: Request number not owned by RS1
CA -> RS1: Enter Requested Number
RS1 -> RND: Lookup number
RND -> RS1: Return URI of RS2
RS1 -> RS2: Basic INVITE
RS2 -> RUE: Call Leg to RUE


@enduml
```

16.3 PSTN to RUE: Detail on the RS1 – RS2 leg

```
@startuml

title PSTN to RUE: Detail on the RS1 - RS2 leg

box "RS1 Side" #5fddd6
participant RS1
end box


box "RS2 Side" #bb7068
participant RS2
end box


box "Number Directory" #bb7068
participant RND
end box


RS1 -> RND: <b>[a]</b> Lookup number 311-555-2368
RND -> RS1: <b>[b]</b>Return URI sip:+13115552368@rs2.example
RS1 -> RS2: <b>[c]</b>INVITE sip:+13115552368@rs2.example
RS2 -> RS1: <b>[d]</b>RINGING
RS2 -> RS1: <b>[e]</b>OK
RS2 <--> RS1: <b>[f]</b>Call media flows
RS2 -> RS1: <b>[g]</b>BYE
RS2 <- RS1: <b>[h]</b>OK

@enduml
```

16.4 Muting (Privacy)

```
@startuml

hide footbox
title Muting (Privacy)

box "RS1 Side" #5fddd6
participant RS1
end box

box "RS2 Side" #bb7068
participant RS2
end box

== call in progress ==
RS1 -->> RS2: Camera video
RS1 -->> RS2: Mic audio
RS2 -->> RS1: video
RS2 -->> RS1: audio
[-> RS1: Mute
RS1 -->> RS2: Privacy video screen
RS1 -->> RS2: Audio silence
RS2 -->> RS1: video
RS2 -->> RS1: audio
[-> RS1: Unmute
RS1 -->> RS2: Camera video
RS1 -->> RS2: Mic audio
RS2 -->> RS1: video
RS2 -->> RS1: audio

@enduml
```

16.5 RUE to PSTN: End to End Call Flow

```
@startuml

title RUE to PSTN: End to End Call Flow

box "RS1 Side" #5fddd6
actor "RUE User" as RUEUSER
participant "RS1" as RS1
end box

box "RS2 Side" #5c7068
participant "RS2 Proxy" as RS2PROXY
participant "RS2 PSTN" as RS2PSTN
actor "CA"
end box

box "Number Directory" #bb7068
participant RND
end box
```

```
box "PSTN Side" #6fc65f
actor "PSTN"
end box

autonumber "<b>[0]"
RUEUSER -> RS1: Call RS2 Access Number
RS1 -> RND: Lookup number
RND -> RS1: Return URI of RS2
RS1 -> RS2: Basic INVITE
RS2 -> CA: Introduce to call
RUEUSER -> CA: Request PSTN number
CA -> RS2PSTN: Call leg to PSTN
RS2PSTN -> PSTN: Call

@enduml
```

16.6 RUE to non-homed RUE Call Flow

```
@startuml

title RUE to non-homed RUE Call Flow

box "RS1 Side" #5fddd6
actor "RUE User" as RUEUSER
participant "RS1" as RS1
end box

box "RS2 Side" #5c7068
participant "RS2" as RS2
actor “RUE2”
end box

box "Number Directory" #bb7068
participant RND
end box

autonumber "<b>[0]"
RUEUSER -> RS1: Call RUE2 Number
RS1 -> RND: Lookup number
RND -> RS1: Return URI of RUE2
RS1 -> RS2: Route
RS2 -> RUE2: Call

@enduml
```

16.7 Refer

```
@startuml

title REFER

box "RS1 Side" #5fddd6
participant RUE1
participant "RS1" as RS1
```

```
actor CA
end box


box "RS2 Side" #5c7068
participant “RS2 EDGE PROXY 1” as RS2EP1
participant “RS2 EDGE PROXY 2” as RS2EP2
participant “RS2 User Access Proxy” as RS2UAP
participant RUE2
end box


box “Number Directory” #bb7068
participant RND
end box

autonumber "<b>[0]"

RUE1 -> RS1: Call to number for RUE2
RS1 -> RND: Lookup number
RND -> RS1: Return URI of RS2
RS1 -> CA: Introduce to call
RS1 -> RS2EP1: Basic INVITE
RS2EP1 -> RS2UAP: Basic INVITE
RS2UAP -> RUE2: Basic INVITE
RS2EP1 -> RS1: Refer-to RS2-EP2
RS1 -> RS2EP2: Call (replacing original call)
RS2EP2 -> RS2UAP: Call (replacing original call)

@enduml
```

16.8 Refer: Detail on the RS1 – RS2 leg

```
@startuml

title Refer: Detail on the RS1 - RS2 leg

box "RS1 Side" #5fddd6
participant RS1
end box


box "RS2 Side" #bb7068
participant “RS2 Edge Proxy 1” as RS2EP1
participant “RS2 Edge Proxy 1” as RS2EP2
end box


box “Number Directory” #bb7068
participant RND
end box


RS1 -> RND: <b>[a]</b> Lookup number 311-555-2368
RND -> RS1: <b>[b]</b>Return URI 311-555-2368@rs2-ep1.example
```

```
RS1 -> RS2EP1: <b>[c]</b>INVITE 311-555-2368@rs2-ep1.example
RS2EP1 -> RS1: <b>[d]</b>OK (INVITE)
note right of RS1
    Audio, video and text
    media established between
    RS2 Edge Proxy 1 and RS1
end note
RS2EP1 -> RS1: <b>[e]</b>REFER Refer-To 311-555-2368@rs2-ep2.example
RS1 -> RS2EP1: <b>[f]</b>OK (REFER)
RS1 -> RS2EP2: <b>[g]</b>INVITE 311-555-2368@rs2-ep2.example replacing
call via \
                         Edge Proxy 1
RS2EP2 -> RS1: <b>[h]</b>OK (INVITE)
note right of RS1
    Audio, video and text
    media established between
    RS2 Edge Proxy 2 and RS1
end note
note right of RS1
    RS1 stops sending media
    to RS2 Edge Proxy 1
end note
RS1 -> RS2EP1: <b>[i]</b>BYE (for original call)
note right of RS1
    RS2 Edge Proxy 1 stops
    sending media to RS1
end note
RS2EP1 -> RS1: <b>[j]</b>OK (BYE)
RS2EP2 -> RS1: <b>[k]</b>BYE
RS1 -> RS2EP2: <b>[l]</b>OK (BYE)

@enduml
```