

rum
Internet-Draft
Intended status: Standards Track
Expires: 26 February 2021

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25 August 2020

Interoperability Profile for Relay User Equipment
draft-ietf-rum-rue-03

Abstract

Video Relay Service (VRS) is a term used to describe a method by which a hearing persons can communicate with deaf/Hard of Hearing user using an interpreter ("Communications Assistant") connected via a videophone to the deaf/HoH user and an audio telephone call to the hearing user. The CA interprets using sign language on the videophone link and voice on the telephone link. Often the interpreters may be supplied by a company or agency termed a "provider" in this document. The provider also provides a video service that allows users to connect video devices to their service, and subsequently to CAs and other deaf/HoH users. It is desirable that the videophones used by the deaf/HoH/H-I user conform to a standard so that any device may be used with any provider and that video calls direct between deaf/HoH users work. This document describes the interface between a videophone and a provider.

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Table of Contents

1. Introduction	3
2. Terminology	3
3. Requirements Language	5
4. General Requirements	6
5. SIP Signaling	6
5.1. Registration	6
5.2. Session Establishment	8
5.2.1. Normal Call Origination	8
5.2.2. One-Stage Dial-Around Origination	9
5.2.3. RUE Contact Information	10
5.2.4. Incoming Calls	10
5.2.5. Emergency Calls	11
5.3. Mid Call Signaling	11
5.4. URI Representation of Phone Numbers	12
5.5. Transport	12
6. Media	12
6.1. SRTP and SRTCP	13
6.2. Text-Based Communication	13
6.3. Video	13
6.4. Audio	13
6.5. DTMF Digits	13
6.6. Session Description Protocol	13
6.7. Privacy	14
6.8. Negative Acknowledgment, Packet Loss Indicator, and Full Intraframe Request Features	14
7. Contacts	14
7.1. CardDAV Login and Synchronization	14
7.2. Contacts Import/Export Service	15
8. Mail Waiting Indicator (MWI)	15
9. Provisioning and Provider Selection	15
9.1. RUE Provider Selection	16
9.2. RUE Configuration Service	17
9.3. Schemas	20
10. Acknowledgements	26
11. IANA Considerations	26
12. Security Considerations	26
13. Normative References	26

Author's Address 32

1. Introduction

Video Relay Service (VRS) is a form of Telecommunications Relay Service (TRS) that enables persons with hearing disabilities who use sign language, such as American Sign Language (ASL), to communicate with voice telephone users through video equipment. These services also enable communication between such individuals directly in suitable modalities, including any combination of sign language via video, real-time text (RTT), and speech.

This Interoperability Profile for Relay User Equipment (RUE) is a profile of the Session Initiation Protocol (SIP) and related media protocols that enables end-user equipment registration and calling for VRS calls. It specifies the minimal set of call flows, Internet Engineering Task Force (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 RUE calls.

Both deaf/HoH to provider (interpreted) and direct deaf/HoH to deaf/HoH calls are supported on this interface. While there are some accommodations in this document to maximize backwards compatibility with devices and services that conform to this document, backwards compatibility is not a requirement, and some interwork may be required to allow direct video calls to older devices. This document only describes the interface between the device and the provider, and not any other interface the provider may have.

2. Terminology

Communication Assistant (CA): The ASL interpreter stationed in a TRS-registered call center working for a VRS Provider, acting as part of the wire of a call to provide functionally equivalent phone service.

Communication modality (modality): A specific form of communication that may be employed by two users, e.g., English voice, Spanish voice, American Sign Language, English lip-reading, or French real-time-text. Here, one communication modality is assumed to encompass both the language and the way that language is exchanged. For example, English voice and French voice are two different communication modalities.

Default video relay service: The video relay service operated by a subscriber's default VRS provider.

Default video relay service Provider (default Provider): The VRS provider that registers, and assigns a telephone number to, a specific subscriber. A subscriber's default Provider provides the VRS that handles incoming relay calls to the user. The default Provider also handles outgoing relay calls by default.

Dial-around call: A relay call where the subscriber specifies the use of a VRS provider other than one of the Providers with whom the subscriber is registered. This can be accomplished by the user dialing a "front-door" number for a VRS provider and signing or texting a phone number to call ("two-stage"). Alternatively, this can be accomplished by the user's RUE software instructing the server of its default VRS provider to automatically route the call through the alternate Provider to the desired public switched telephone network (PSTN) directory number ("one-stage").

Full Intra Request (FIR): A request to a media sender, requiring that media 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". Point-to-Point Call (P2P Call): A call between two RUEs, without including a CA.

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. Please refer to FCC-VRS-GUIDE.

Relay-to-relay call: A call between two subscribers each using different forms of relay (video relay, IP relay, TTY), each with a separate CA to assist in relaying the conversation.

Relay service (RS): A service that allow a registered subscriber to use a RUE to make and receive relay calls, point-to-point calls, and relay-to-relay calls. The functions provided by the relay service include the provision of media links supporting the communication modalities used by the caller and callee, and user registration and validation, authentication, authorization, automatic call distributor (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 (Provider): An organization that operates a relay service. A subscriber selects a relay service Provider to assign and register a telephone number for their use, to register with for receipt of incoming calls, and to provide the default service for outgoing calls.

Relay user: Please refer to "subscriber".

Relay user E.164 Number (user E.164): The telephone number assigned to the RUE in ITU-T E.164 format.

Relay user equipment (RUE): A SIP user agent (UA) enhanced with extra features to support a subscriber in requesting and using relay calls. A RUE may take many forms, including a stand-alone device; an application running on a general-purpose computing device such as a laptop, tablet or smart phone; or proprietary equipment connected to a server that provides the RUE interface.

RUE Interface: the SIP interface between a RUE and the provider who supports it

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

Subscriber: An individual who has registered with a Provider and who obtains service by using relay user equipment. This is the traditional telecom term for an end-user customer, which in our case is a relay user.

Telecommunications relay services (TRS): Telephone transmission services that provide the ability for an individual who has a hearing impairment or speech impairment 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 impairment or speech impairment to communicate using voice communication services by wire or radio. TRS includes services that enable two-way communication between an individual who uses a Telecommunications Device for the Deaf (TDD) or other non-voice terminal device and an individual who does not use such a device.

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 subscriber's signed conversation and relay the conversation back and forth with the other party.

3. Requirements Language

The keywords "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]

4. General Requirements

All HTTP/HTTPS connections specified throughout this document MUST use HTTPS. Both HTTPS and all SIP connections MUST use TLS conforming to at least [RFC7525] and must support [RFC8446]

All text data payloads not otherwise constrained by a specification in another standards document MUST be encoded as Unicode UTF/8.

5. SIP Signaling

Implementations of the RUE Interface MUST conform to the following core SIP standards [RFC3261] (Base SIP) [RFC3263] (Locating SIP Servers), [RFC3264] (Offer/Answer), [RFC3840] (User Agent Capabilities), [RFC5626] (Outbound), [RFC4566] (Session Description Protocol), [RFC3323] (Privacy), [RFC3605] (RTCP Attribute in SDP), [RFC6665] (SIP Events), [RFC3311] (UPDATE Method), [RFC5393] (Loop-Fix), [RFC5658] (Record Route fix), [RFC5954] (ABNF fix), [RFC3960] (Early Media), and [RFC6442] (Geolocation Header).

In the above documents the RUE device conforms to the requirements of a SIP user Agent, and the provider conforms to the requirements of Registrar and Proxy Server where the document specifies different behavior for different roles. The only requirement on providers for RFC6665 (Events) is support for the Message Waiting Indicator (See Section Section 8), which is optional and providers not supporting MWI need not support RFC6665.

In addition, implementation MUST conform to [RFC3327] (Path), [RFC5245] (ICE), [RFC3326] (Reason header), [RFC3515] (REFER Method), [RFC3891] (Replaces Header), [RFC3892] (Referred-By).

Implementations MUST include a "User-Agent" header field uniquely identifying the RUE application, platform, and version in all SIP requests, and MUST include a "Server" header field with the same content in SIP responses.

5.1. Registration

The RUE MUST register with a SIP registrar, following [RFC3261] and [RFC5626] at a provider it has an account with. If the configuration (please refer to Section 11) contains multiple "outbound-proxies", then the RUE MUST use them as specified in [RFC5626] to establish multiple flows.

The request-URI for the REGISTER request MUST contain the "provider-domain" from the configuration. The To-URI and From-URI MUST be identical URIs, formatted as specified in Section 13, using the "phone-number" and "provider-domain" from the configuration.

The RUE determines the URI to resolve by initially determining if an outbound proxy is configured. If it is, the URI will be that of the outbound proxy. If no outbound proxy is configured, the URI will be the Request-URI from the REGISTER request. The RUE extracts the domain from that URI and consults the DNS record for that domain. The DNS entry MUST contain NAPTR records conforming to RFC3263. One of those NAPTR records MUST specify TLS as the preferred transport for SIP. For example, a DNS NAPTR query for "sip:pl.red.example.netv" could return:

```
IN NAPTR 50 50 "s" "SIPS+D2T" "" _sips._tcp.pl.red.example.net
IN NAPTR 90 50 "s" "SIP+D2T" "" _sip._tcp.pl.red.example.net
```

If the RUE receives a 439 (First Hop Lacks Outbound Support) response to a REGISTER request, it MUST re-attempt registration without using the outbound mechanism.

The registrar MAY authenticate using SIP MD5 digest authentication. The credentials to be used (username and password) MUST be supplied within the credentials section of the configuration and identified by the realm the registrar uses in a digest challenge. This username/password combination SHOULD NOT be the same as that used for other purposes, such as retrieving the RUE configuration or logging into the Provider's customer service portal. Because MD5 is considered insecure, [I-D.yusef-sipcore-digest-scheme] SHOULD be implemented by all implementations and SHA-based digest algorithms SHOULD be used for digest authentication.

If the registration request fails with an indication that credentials from the configuration are invalid, then the RUE SHOULD retrieve a fresh version of the configuration. If credentials from a freshly retrieved configuration are found to be invalid, then the RUE MUST cease attempts to register and SHOULD inform the RUE User of the problem.

Support for multiple simultaneous registrations is OPTIONAL.

Multiple simultaneous RUE SIP registrations from different RUE devices with the same SIP URI SHOULD be permitted by the Provider. The Provider MAY limit the total number of simultaneous registrations. When a new registration request is received that results in exceeding the limit on simultaneous registrations, the Provider MAY then prematurely terminate another registration; however, it SHOULD NOT do this if it would disconnect an active call.

If a Provider prematurely terminates a registration to reduce the total number of concurrent registrations with the same URI, it SHOULD take some action to prevent the affected RUE from automatically re-registering and re-triggering the condition.

5.2. Session Establishment

5.2.1. Normal Call Origination

After initial SIP registration, the RUE adheres to SIP [RFC3261] basic call flows, as documented in [RFC3665].

A RUE device MUST routes all outbound calls through an outbound proxy if configured.

INVITE requests used to initiate calls SHOULD NOT contain Route headers. Route headers MAY be included in one-stage dial-around calls and emergency calls. The SIP URIs in the To field and the Request-URI MUST be formatted as specified in subsection 6.4 using the destination phone number. The domain field of the URIs SHOULD be the "provider-domain" from the configuration (e.g., sip:+13115552368@red.example.com;user=phone). The same exceptions apply, including anonymous calls.

Anonymous calls MUST be supported by all implementations. An anonymous call is signaled per [RFC3323].

The From-URI MUST be formatted as specified in Section 5.4, using the phone-number and "provider-domain" from the configuration. It SHOULD also contain the display-name from the configuration when present. (Please refer to Section 9.2.)

Negotiated media MUST follow the guidelines specified in Section 6 of this document.

To allow time to timeout an unanswered call and direct it to a videomail server, the User Agent Client MUST NOT impose a time limit less than the default SIP Invite transaction timeout of 3 minutes.

5.2.2. One-Stage Dial-Around Origination

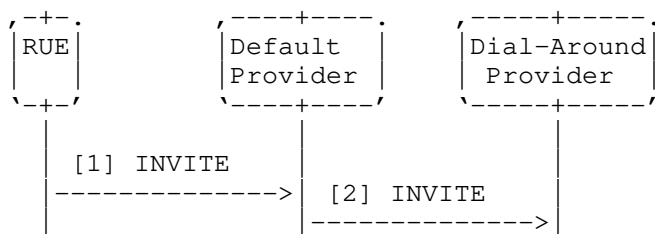
Outbound dial-around calls allow a RUE user to select any Provider to provide interpreting services for any call. "Two-stage" dial-around calls involve the RUE calling a telephone number that reaches the dial-around Provider and using signing or DTMF to provide the called party telephone number. In two-stage dial-around, the To URI is the URI of the dial-around Provider and the domain of the URI is the Provider domain from the configuration.

One-stage dial-around is a method where the called party telephone number is provided in the To URI and the Request-URI, using the domain of the dial-around Provider.

For one-stage dial-around, the RUE MUST follow the procedures in Section 5.2.1 with the following exception: the domain part of the SIP URIs in the To field and the Request-URI MUST be the domain of the dial-around Provider, discovered according to Section 9.1.

The following is a partial example of a one-stage dial-around call from VRS user +1-555-222-0001 hosted by red.example.com to a hearing user +1-555-123-4567 using dial-around to green.example.com for the relay service. Only important details of the messages are shown and many header fields have been omitted:

One Stage Dial-Around



Message Details:

[1] INVITE Rue -> Default Provider

```

INVITE sip:+15551234567@green.example.net;user=phone SIP/2.0
To: <sip:+15551234567@green.example.net;user=phone>
From: "Bob Smith" <sip:+18135551212@red.example.net;user=phone>
Route: sip:green.example.net
  
```

[2] INVITE Default Provider -> Dial-Around Provider

```

INVITE sip:+15551234567@green.example.net;user=phone SIP/2.0
To: <sip:+15551234567@green.example.net;user=phone>
From: "Bob Smith" sip:+18135551212@red.example.net;user=phone
P-Asserted-Identity: sip:+18135551212@red.example.net
  
```

Figure 1

5.2.3. RUE Contact Information

To identify the owner of a RUE, the initial INVITE for a call from a RUE, or the 200 OK accepting a call by a RUE, identifies the owner by sending a Call-Info header with a purpose parameter of "rue-owner". The URI MAY be an HTTPS URI or Content-Indirect URL. The latter is defined by [RFC2392] to locate message body parts. This URI type is present in a SIP message to convey the RUE ownership information as a MIME body. The form of the RUE ownership information is a jCard [RFC7095]. Please refer to [RFC6442] for an example of using Content-Indirect URLs in SIP messages. Note that use of the Content-Indirect URL usually implies multiple message bodies ("mime/multipart").

5.2.4. Incoming Calls

The RUE MUST accept inbound calls sent to it by the proxy mentioned in the configuration.

If Multiple simultaneous RUE SIP registrations from different RUE devices with the same SIP URI exist, the Provider MUST parallel fork the call to all registered RUEs so that they ring at the same time. The first RUE to reply with a 200 OK answers the call and the Provider MUST CANCEL other call branches.

5.2.5. Emergency Calls

Implementations MUST conform to [RFC6881] for handling of emergency calls, except that if the device is unable to determine its own location, it MAY send the emergency call without a Geolocation header and without a Route header (since it would be unable to query the LoST server for a route per RFC6881). If an emergency call arrives at the provider without a Geolocation header, the provider MUST supply location by adding the Geolocation header, and MUST supply the route by querying the LoST server with that location.

If the emergency call is to be handled using existing country specific procedures, the Provider is responsible for modifying the INVITE to conform to the country-specific requirements. In this case, location MAY be extracted from the RFC6881 conformant INVITE and used to propagate it to the appropriate country-specific entities. Because the RUE may have a more accurate and timely location of the device than the a manual entry location for nomadic RUE devices, but country-specific procedures require the location to be pre-loaded in some entity prior to placing an emergency call, implementations of a RUE device MAY send a Geolocation header containing its location in the REGISTER request if the configuration specifies it. That information MAY be used to populate the location to appropriate country-specific entities.

Implementations MUST implement Additional Data, [RFC7852]. RUE devices MUST implement Data Provider, Device Implementation and Owner/Subscriber Information blocks. Providers MUST implement Data Provider and Service Information blocks as the call is forwarded to the PSAP.

5.3. Mid Call Signaling

Implementations MUST support re-INVITE to renegotiate media session parameters (among other uses). Per Section 6.1, implementations MUST, be able to support an INFO request for full frame refresh for devices that do not support RTCP mechanisms (please refer to Section 6.8). Implementations MUST support an in-dialog REFER ([RFC3515] updated by [RFC7647] and including support for norefersub per [RFC4488]) with the Replaces header [RFC3891] to enable call transfer.

5.4. URI Representation of Phone Numbers

SIP URIs constructed from non-URI sources (dial strings) and sent to SIP proxies by the RUE MUST be represented as follows, depending on whether they can be represented as an E.164 number.

A dial string that can be written as an E.164 formatted phone number MUST be represented as a SIP URI with a URI ";user=phone" tag. The user part of the URI MUST be in conformance with 'global-number' defined in [RFC3966]. The user part MUST NOT contain any 'visual-separator' characters.

Dial strings that cannot be written as E.164 numbers MUST be represented as dialstring URIs, as specified by [RFC4967], e.g., sip:411@red.example.net;user=dialstring.

The domain part of Relay Service URIs and User Address of Records (AoR) MUST (using resolve (in accord with [RFC3263])) to globally routable IPv4 addresses. The AoRs MAY also resolve to IPv6 addresses.

5.5. Transport

Implementations MUST conform to [I-D.ietf-rtcweb-transports] except that that this specification does not use the WebRTC data channel. See Section 6.2 for how RUE supports real time text without the data channel.

Implementations MUST support SIP outbound [RFC5626] (please also refer to Section 5.1).

6. Media

This specification adopts the media specifications for WebRTC ([I-D.ietf-rtcweb-overview]). Where WebRTC defines how interactive media communications may be established using a browser as a client, this specification assumes a normal SIP call. The RTP, RTCP, SDP and specific media requirements specified for WebRTC are adopted for this document. The RUE is a WebRTC non-browser endpoint, except as noted expressly below.

The following sections specify the WebRTC documents to which conformance is required. "Mandatory to Implement" means a conforming implementation must implement the specified capability. It does not mean that the capability must be used in every session. For example, OPUS is a mandatory to implement audio codec, and all conforming implementations must support OPUS. However, implementation presenting a call across the RUE Interface where the call originates

in the Public Switched Telephone Network, or an older, non-RUE-compatible device, which only offers G.711 audio, does not need to include the OPUS codec in the offer, since it cannot be used with that call.

6.1. SRTP and SRTCP

Implementations MUST support [I-D.ietf-rtcweb-rtp-usage] except that `MediaStreamTracks` are not used. Implementations MUST conform to Section 6.4 of [I-D.ietf-rtcweb-security-arch].

6.2. Text-Based Communication

Implementations MUST support real-time text ([RFC4102] and [RFC4103]) via T.140 media. One original and two redundant generations MUST be transmitted and supported, with a 300 ms transmission interval. Note that this is not how real time text is transmitted in WebRTC and some form of transcoder would be required to interwork real time text in the data channel of WebRTC to RFC4103 real time text.

6.3. Video

Implementations MUST conform to [RFC7742] with the exception that, since backwards compatibility is desirable and older devices do not support VP8, that only H.264, as specified in [RFC7742] is Mandatory to Implement and VPB support is OPTIONAL at both the device and providers.

6.4. Audio

Implementations MUST conform to [RFC7874].

6.5. DTMF Digits

Implementations MUST support the "audio/telephone-event" [RFC4733] media type. They MUST support conveying event codes 0 through 11 (DTMF digits "0"-"9", "*", "#") defined in Table 7 of [RFC4733]. Handling of other tones is OPTIONAL.

6.6. Session Description Protocol

The SDP offers and answers MUST conform [I-D.ietf-rtcweb-jsep] except that the RUE Interface uses SIP transport for SDP.

6.7. Privacy

The RUE MUST be able to control privacy of the user by implementing a one-way mute of audio and or video, without signaling, locally, but MUST maintain any NAT bindings by periodically sending media packets on all active media sessions containing silence/comfort noise/black screen/etc. per [RFC6263].

6.8. Negative Acknowledgment, Packet Loss Indicator, and Full Intraframe Request Features

NACK SHOULD be used when negotiated and conditions warrant its use. Signaling picture losses as Packet Loss Indicator (PLI) SHOULD be preferred, as described in [RFC5104].

FIR SHOULD be used only in situations where not sending a decoder refresh point would render the video unusable for the users, as per RFC5104 subsection 4.3.1.2.

For backwards compatibility with calling devices that do not support the foregoing methods, implementations MUST implement SIP INFO messages to send and receive XML encoded Picture Fast Update messages according to [RFC5168].

7. Contacts

7.1. CardDAV Login and Synchronization

Support of CardDAV by Providers is OPTIONAL.

The RUE MUST and Providers MAY be able to synchronize the user's contact directory between the RUE endpoint and one maintained by the user's VRS provider using CardDAV ([RFC6352] and [RFC6764]).

The configuration MAY supply a username and domain identifying a CardDAV server and address book for this account.

To access the CardDAV server and address book, the RUE MUST follow Section 6 of RFC6764, using the chosen username and domain in place of an email address. If the request triggers a challenge for digest authentication credentials, the RUE MUST attempt to continue using matching "credentials" from the configuration. If no matching credentials are configured, the RUE MUST use the SIP credentials from the configuration. If the SIP credentials fail, the RUE MUST query the user.

Synchronization using CardDAV MUST be a two-way synchronization service, with proper handling of asynchronous adds, changes, and deletes at either end of the transport channel.

7.2. Contacts Import/Export Service

Implementations MUST be able to export/import the list of contacts in jCard [RFC7095] json format.

The RUE accesses this service via the "contacts" URI in the configuration. The URL MUST resolve to identify a web server resource that imports/exports contact lists for authorized users.

The RUE stores/retrieves the contact list (address book) by issuing an HTTPS POST or GET request. If the request triggers a challenge for digest authentication credentials, the RUE MUST attempt to continue using matching "credentials" from the configuration. If no credentials are configured, the RUE MUST query the user.

8. Mail Waiting Indicator (MWI)

Support of MWI by Providers is OPTIONAL

Implementations MUST support subscriptions to "message-summary" events [RFC3842] to the URI specified in the configuration.

In notification bodies, videomail messages SHOULD be reported using "message-context-class multimedia-message" defined in [RFC3458].

9. Provisioning and Provider Selection

To simplify how users interact with RUE devices, the RUE interface defines a provisioning mechanism which consist of files stored on server that are retrieved by the RUE device. Two files are supported: one provides a directory of providers so that a user interface that allows easy provider selection either for registering or for dial-around. The other file provides configuration data for the device. The RUE device would retrieve these files at boot time. No mechanism for creating the files are specified. Each of the files contains a single json object. The retrieval mechanism is HTTPS download of that object from a provisioned location.

9.1. RUE Provider Selection

To allow the user to select a relay service, the RUE MAY obtain, on startup, a list of Providers from a configured accessible URL. This file is MAY be a single file per country, containing all the providers authorized in that country, but MAY be any collection of providers.

The provider list, formatted as JSON, contains:

- * Version: Specifies the version number of the Provider list format. A new version number SHOULD only be used if the new version is not backwards-compatible with the older version. A new version number is not needed if new elements are optional and can be ignored by older implementations.
- * Providers: An array where each entry describes one Provider. Each entry consists of the following items:
 - name: This parameter contains the text label identifying the Provider and is meant to be displayed to the human VRS user.
 - domain: The domain parameter is used for configuration purposes by the RUE (as discussed in Section 9.2) and as the domain to use when targeting one-stage dial-around calls to this Provider (as discussed in Section 5.2.2).
 - operator: (OPTIONAL) The operator parameter is a SIP URL that identifies the operator "front-door" that VRS users may contact for manual (two-stage) dial-around calls.

The VRS user interacts with the RUE to select from the Provider list one or more Providers with whom the user has already established an account.

Example of a Provider list JSON object

```
{
  "version": 1,
  "providers": [
    {
      "name": "Red",
      "domain": "red.example.net",
      "operator": "sip:operator@red.example.net"
    },
    {
      "name": "Green",
      "domain": "green.example.net",
      "operator": "sip:+18885550123@green.example.net;user=phone"
    },
    {
      "name": "Blue",
      "domain": "blue.example.net"
    }
  ]
}
```

Figure 2

9.2. RUE Configuration Service

A RUE device may retrieve a configuration from a provisioned URL using HTTPS.

The data returned will include a set of key/value configuration parameters to be used by the RUE, formatted as a single JSON object and identified by the associated [RFC7159] "application/json" MIME type, to allow for other formats in the future.

The configuration data payload includes the following data items. Items not noted as (OPTIONAL) are REQUIRED. If other unexpected items are found, they MUST be ignored.

- * **version:** Identifies the version of the configuration data format. A new version number SHOULD only be used if the new version is not backwards-compatible with the older version. A new version number is not needed if new elements are optional and can be ignored by older implementations.
- * **lifetime:** Specifies how long (in seconds) the RUE MAY cache the configuration values. Values may not be valid when lifetime expires. Emergency Calls MUST continue to work.
- * **display-name:** (OPTIONAL) A user-friendly name to identify the subscriber when originating calls.

- * **phone-number:** The telephone number (in E.164 format) assigned to this subscriber. This becomes the user portion of the SIP URI identifying the subscriber.
- * **provider-domain:** The DNS domain name of the default Provider servicing this subscriber.
- * **outbound-proxies:** (OPTIONAL) A URI of a SIP proxy to be used when sending requests to the Provider.
- * **mwi:** (OPTIONAL) A URI identifying a SIP event server that generates "message-summary" events for this subscriber.
- * **videomail:** (OPTIONAL) A SIP URI that can be called to retrieve videomail messages.
- * **contacts:** An HTTPS URI that may be used to export (retrieve) the subscriber's complete contact list managed by the Provider.
- * **carddav:** (OPTIONAL) A username and domain name (separated by "@"") identifying a "CardDAV" server and user name that can be used to synchronize the RUE's contact list with the contact list managed by the Provider.
- * **sendLocationWithRegistration:** True if the RUE should send a Geolocation Header with REGISTER, false if it should not. Defaults to false if not present.
- * **ice-servers:** (OPTIONAL) An array of URLs identifying STUN and TURN servers available for use by the RUE for establishing media streams in calls via the Provider.
- * **credentials:** (OPTIONAL) TBD

Example JSON configuration payload

```
{
  "version": 1,
  "lifetime": 86400,
  "display-name" : "Bob Smith",
  "phone-number": "+18135551212",
  "provider-domain": "red.example.net",
  "outbound-proxies": [
    "sip:p1.red.example.net",
    "sip:p2.red.example.net"
  ],
  "mwi": "sip:+18135551212@red.example.net",
  "videomail": "sip:+18135551212@vm.red.example.net",
  "contacts": "https://red.example.net:443/contacts/1d5545awd",
  "carddav": "bob@red.example.com" ,
  "sendLocationWithRegistration": false,
  "ice-servers": [
    {"stun": "stun.l.google.com:19302" },
    {"turn": "turn.red.example.net:3478"}
  ],
  "credentials": [
    {
      "realm": "red.example.net",
      "username": "bob",
      "password": "reg-pw"
    },
    {
      "realm": "proxies.red.example.net",
      "username": "bob",
      "password": "proxy-pw"
    },
    {
      "realm": "cd.red.example.net",
      "username": "bob",
      "password": "cd-pw"
    },
    {
      "realm": "vm.red.example.net",
      "username": "bob",
      "password": "vm-pw"
    },
    {
      "realm": "stun-turn.red.example.net",
      "username": "bob",
      "password": "stun-turn-pw"
    }
  ]
}
```

Figure 3

The wire format of the data is in keeping with the standard JSON description in RFC7159.

The "lifetime" parameter in the configuration indicates how long the RUE MAY cache the configuration values. If the RUE caches configuration values, it MUST cryptographically protect them. The RUE SHOULD retrieve a fresh copy of the configuration before the lifetime expires or as soon as possible after it expires. The lifetime is not guaranteed: the configuration may change before the lifetime value expires. In that case, the Provider MAY indicate this by generating authorization challenges to requests and/or prematurely terminating a registration.

Note: In some cases, the RUE may successfully retrieve a fresh copy of the configuration using digest credentials cached from the prior retrieval. If this is not successful, then the RUE will need to ask the user for the username and password. Unfortunately, this authentication step might occur when the user is not present, preventing SIP registration and thus incoming calls. To avoid this situation, the RUE MAY retrieve a new copy of the configuration when it knows the user is present, even if there is time before the lifetime expires.

9.3. Schemas

The following JSON schemas are for the Provider List and the RUE Configuration.

Provider List JSON Schema

```
{
  "$schema": "http://json-schema.org/draft-07/schema",
  "type": "object",
  "title": "Providers schema",
  "description": "List of Providers",
  "required": [
    "version",
    "providers"
  ],
  "properties": {
    "version": {
      "type": "number",
      "description": "Version of this schema",
    },
    "providers": {
      "type": "array",
      "description": "provider list as an array.",
      "items": {
        "required": [
          "name",
          "domain"
        ],
        "properties": {
          "name": {
            "type": "string",
            "description": "Display Name",
          },
          "domain": {
            "type": "string",
            "description":
              "domain name used with config file",
          },
          "operator": {
            "type": "string",
            "description":
              "SIP URI for dial-around",
          }
        }
      }
    }
  }
}
```

Figure 4

RUE Configuration JSON Schema

```

{
  "$schema": "http://json-schema.org/draft-07/schema",
  "title": "The root schema",
  "description": "RUE Configuration File",
  "required": [
    "version",
    "lifetime",
    "phone-number",
    "provider-domain",
    "carddav",
  ],
  "properties": {
    "version": {
      "type": "number",
      "description": "Version of this schema",
    },
    "lifetime": {
      "type": "integer",
      "description":
        "how long (in seconds) the RUE MAY cache the configuration valu
es",
    },
    "display-name": {
      "type": "string",
      "description":
        "A user-friendly name to identify the subscriber when originati
ng calls",
    },
    "phone-number": {
      "type": "string",
      "description":
        "The telephone number (in E.164 format) assigned to this subscr
iber",
    },
    "provider-domain": {
      "type": "string",
      "description":
        "The DNS domain name of the default Provider servicing this sub
scriber",
    },
    "outbound-proxies": {
      "type": "array",
      "description":
        "List of URIs of SIP proxies to be used when sending requests t
o the Provider",
      "items": {
        "type": "string",
        "format": "uri"
      }
    },
    "mwi": {
      "type": "string",
      "format": "uri",
    }
  }
}

```

```

        "description":
            "A URI of a SIP event server that generates message-summary eve
nts",
    },
    "videomail": {
        "type": "string",
        "format": "uri",
        "description":
            "A SIP URI that can be called to retrieve videomail messages",
    },
    "contacts": {
        "$id": "#/properties/contacts",
        "type": "string",
        "format": "uri",
        "description":
            "An HTTPS URI used to manage the subscriber's contact list at t
he Provider.",
    },
    "carddav": {
        "type": "string",
        "description":
            "A username and domain name (separated by @) identifying a Card
DAV server",
    },
    "sendLocationWithRegistration": {
        "type": "boolean",
        "description":
            "True if the RUE should send a Geolocation Header with REGISTER
",
        "default": false,
    },
    "ice-servers": {
        "type": "array",
        "description":
            "An array of URLs identifying STUN and TURN servers available",
        "items": {
            "type": "object",
            "properties": {
                "servertype": {
                    "type": "string"
                },
                "url": {
                    "type": "string"
                }
            }
        }
    },
    "credentials": {
        "type": "array",
        "description": "registration credentials",
        "additionalItems": true,
        "items": {

```

```

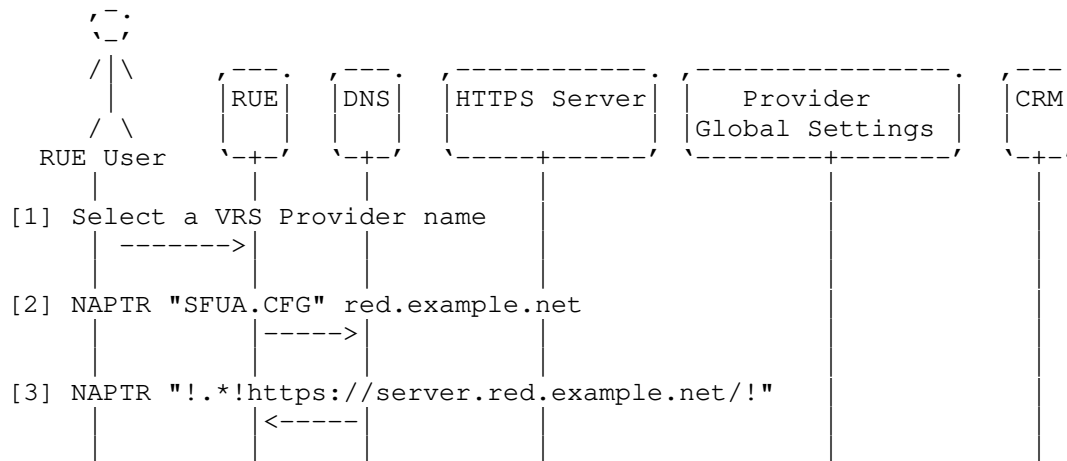
    "type": "object",
    "required": [
      "realm",
      "username",
      "password"
    ],
    "properties": {
      "realm": {
        "type": "string",
        "description":
          "domain of provider matching domain in provider list",
      },
      "username": {
        "type": "string",
        "description": "username for registration"
      },
      "password": {
        "type": "string",
        "description": "password for registration",
      }
    }
  },
}

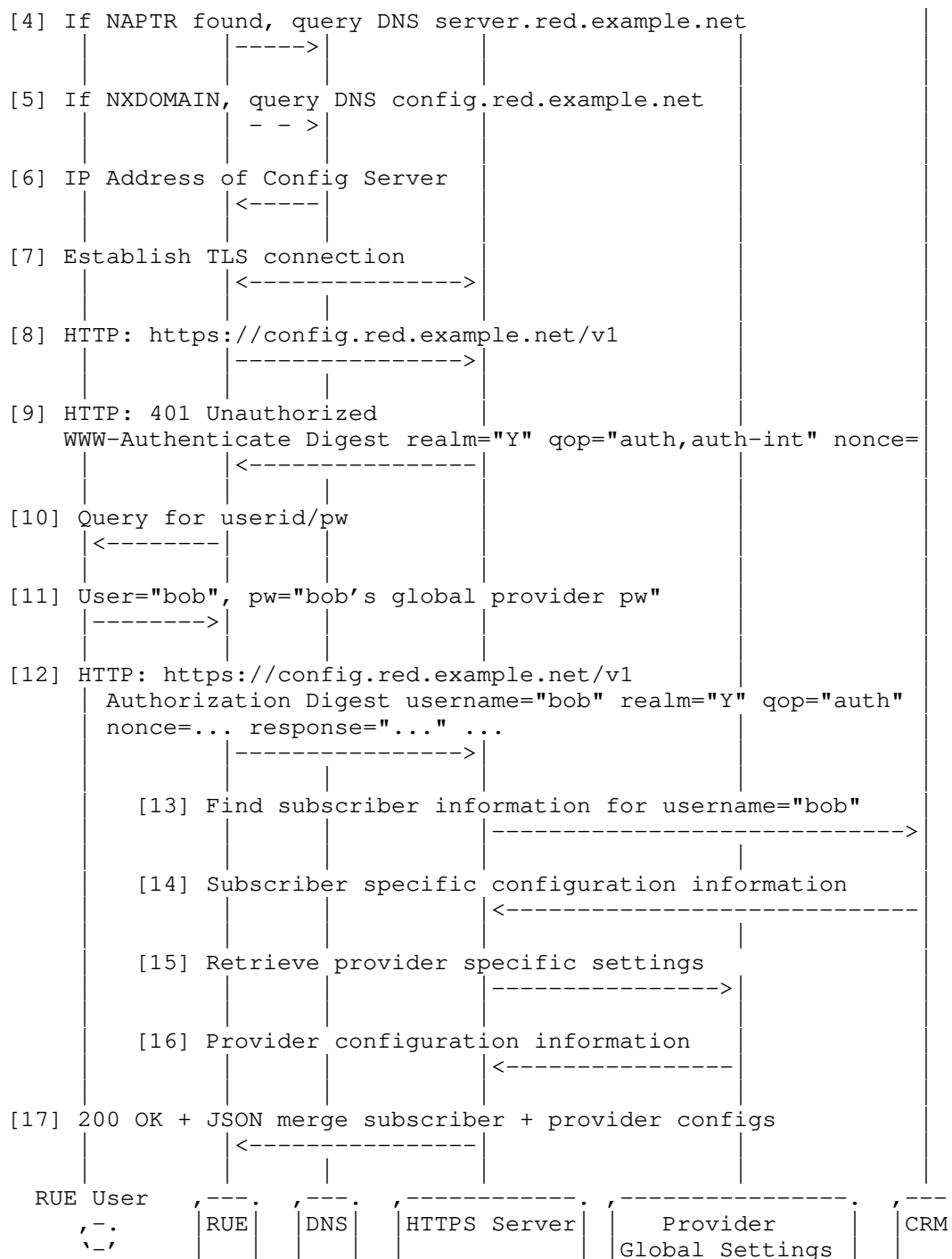
```

Figure 5

The following illustrates the message flow for retrieving a RUE automatic configuration using HTTPS Digest Authentication:

RUE Configuration Retrieval





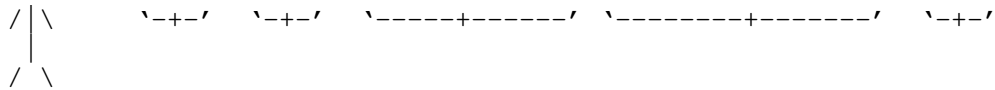


Figure 6

10. Acknowledgements

Brett Henderson and Jim Malloy provided many helpful edits to prior versions of this document.

11. IANA Considerations

This memo includes no request to IANA.

12. Security Considerations

The RUE is required to communicate with servers on public IP addresses and specific ports to perform its required functions. If it is necessary for the RUE to function on a corporate or other network that operates a default-deny firewall between the RUE and these services, the user must arrange with their network manager for passage of traffic through such a firewall in accordance with the protocols and associated SRV records as exposed by the Provider. Because VRS providers may use different ports for different services, these port numbers may differ from Provider to Provider.

13.

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rum
Internet-Draft
Intended status: Standards Track
Expires: 21 August 2022

B. Rosen
17 February 2022

Interoperability Profile for Relay User Equipment
draft-ietf-rum-rue-11

Abstract

Video Relay Service (VRS) is a term used to describe a method by which a hearing person can communicate with a deaf, hard of hearing or hearing impaired user using an interpreter ("Communications Assistant") connected via a videophone to the deaf/hard of hearing/hearing impaired user and an audio telephone call to the hearing user. The CA interprets using sign language on the videophone link and voice on the telephone link. Often the interpreters may be employed by a company or agency termed a "provider" in this document. The provider also provides a video service that allows users to connect video devices to their service, and subsequently to CAs and other deaf/hard of hearing/hearing impaired users. It is desirable that the videophones used by the deaf, hard of hearing or hearing impaired user conform to a standard so that any device may be used with any provider and that direct video calls between deaf, hard of hearing or hearing impaired users work. This document describes the interface between a videophone and a provider.

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Table of Contents

1. Introduction	3
2. Terminology	4
3. Requirements Language	6
4. General Requirements	6
5. SIP Signaling	7
5.1. Registration	8
5.2. Session Establishment	10
5.2.1. Normal Call Origination	10
5.2.2. Dial-Around Origination	10
5.2.3. RUE Contact Information	12
5.2.4. Incoming Calls	12
5.2.5. Emergency Calls	12
5.3. Mid-Call Signaling	13
5.4. URI Representation of Phone Numbers	13
5.5. Transport	14
6. Media	14
6.1. SRTP and SRTCP	14
6.2. Text-Based Communication	15
6.3. Video	15
6.4. Audio	15
6.5. DTMF Digits	15
6.6. Session Description Protocol	15
6.7. Privacy	16
6.8. Negative Acknowledgment, Packet Loss Indicator, and Full Intraframe Request Features	16
7. Contacts	16
7.1. CardDAV Login and Synchronization	16
7.2. Contacts Import/Export Service	17
8. Video Mail	17
9. Provisioning and Provider Selection	18
9.1. RUE Provider Selection	18
9.2. RUE Configuration Service	20

9.2.1.	Provider Configuration	21
9.2.2.	RUE Configuration	21
9.2.3.	Versions	23
9.2.4.	Examples	24
9.2.5.	Using the Provider Selection and RUE Configuration Services Together	25
9.3.	OpenAPI Interface Descriptions	26
9.3.1.	Provider List	26
9.3.2.	Configuration	27
10.	IANA Considerations	33
10.1.	RUE Provider List Registry	33
10.2.	Registration of rue-owner Value of the purpose Parameter	33
11.	Security Considerations	34
12.	Normative References	34
13.	Informative References	40
	Acknowledgements	41
	Author's Address	41

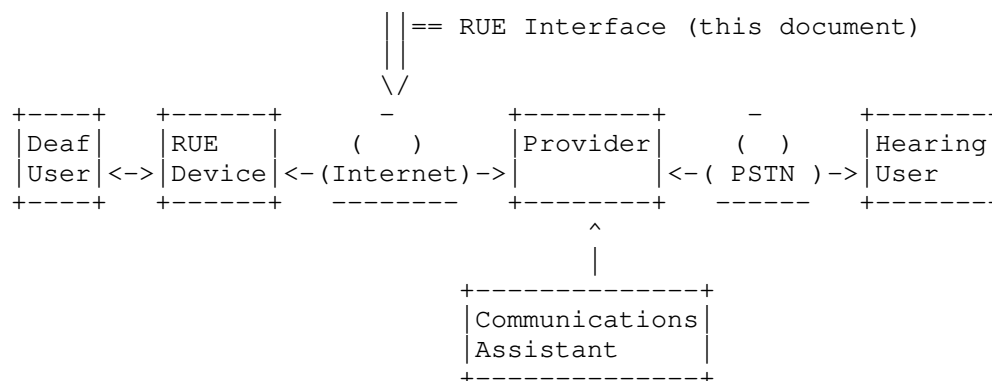
1. Introduction

Video Relay Service (VRS) is a form of Telecommunications Relay Service (TRS) that enables persons with hearing disabilities who use sign language, such as American Sign Language (ASL), to communicate with voice telephone users through video equipment. These services also enable communication between such individuals directly in suitable modalities, including any combination of sign language via video, real-time text (RTT), and speech.

This Interoperability Profile for Relay User Equipment (RUE) is a profile of the Session Initiation Protocol (SIP) and related media protocols that enables end-user equipment registration and calling for VRS calls. It specifies the minimal set of call flows, Internet Engineering Task Force (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 RUE calls.

Both deaf/HoH to provider (interpreted) and direct deaf/HoH to deaf/HoH calls are supported on this interface. While there are some accommodations in this document to maximize backwards compatibility with other devices and services that are used to provide VRS service, backwards compatibility is not a requirement, and some interwork may be required to allow direct video calls to older devices. This document only describes the interface between the device and the provider, and not any other interface the provider may have.

The following illustrates a typical relay call. The RUE device and the Communications Assistant (sign language interpreter) have videophones. The Hearing User has a telephone (mobile or fixed).



2. Terminology

Communication Assistant (CA): A sign-language interpreter working for a VRS provider, providing functionally equivalent phone service.

Communication modality (modality): A specific form of communication that may be employed by two users, e.g., English voice, Spanish voice, American Sign Language, English lip-reading, or French real-time-text. Here, one communication modality is assumed to encompass both the language and the way that language is exchanged. For example, English voice and French voice are two different communication modalities.

Default video relay service: The video relay service operated by a subscriber's default VRS provider.

Default video relay service provider (default provider): The VRS provider that registers, and assigns a telephone number to a specific subscriber, and by default provides the VRS for incoming voice calls to the user. The default provider also by default provides VRS for outgoing relay calls. The user can have more than one telephone number and each has a default provider.

Outbound Dial-around call: A relay call where the subscriber specifies the use of a VRS provider other than the default VRS provider. This can be accomplished by the user dialing a "front-door" number for a VRS provider and signing or texting a phone number to call ("two-stage"). Alternatively, this can be accomplished by the user's RUE software instructing the server of its default VRS provider to automatically route the call through the alternate

provider to the desired public switched telephone network (PSTN) directory number ("one-stage"). Dial-around is per-call -- for any call, a user can use the default VRS provider or any dial-around VRS provider.

Full Intra Request (FIR): A request to a video media sender, requiring that media 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".

Point-to-Point Call (P2P Call): A call between two RUEs, without including a CA.

Relay call: A call that allows persons with hearing or speech disabilities to use a RUE to talk to users of conventional voice services with the aid of a communication assistant (CA) to relay the communication.

Relay service (RS): A service that allow a registered subscriber to use a RUE to make and receive relay calls, point-to-point calls, and relay-to-relay calls. The functions provided by the relay service include the provision of media links supporting the communication modalities used by the caller and callee, and user registration and validation, authentication, authorization, automatic call distributor (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 (provider): An organization that operates a relay service. A subscriber selects a relay service provider to assign and register a telephone number for their use, to register with for receipt of incoming calls, and to provide the default service for outgoing calls.

Relay user: Please refer to "subscriber".

Relay user E.164 Number (user E.164): The telephone number (in ITU-T E.164 format) assigned to the user.

Relay user equipment (RUE): A SIP user agent (UA) enhanced with extra features to support a subscriber in requesting, receiving and using relay calls. A RUE may take many forms, including a stand-alone device; an application running on a general-purpose computing device such as a laptop, tablet or smartphone; or proprietary equipment connected to a server that provides the RUE interface.

RUE Interface: the interfaces described in this document between a RUE and a VRS provider who supports it

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

Subscriber: An individual who has registered with a provider and who obtains service by using relay user equipment. This is the conventional telecom term for an end-user customer, which in our case is a relay user. A user may be a subscriber to more than one VRS provider.

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 subscriber's signed conversation and relay the conversation back and forth with the other party.

3. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here. Lower- or mixed-case uses of these key words are not to be interpreted as carrying special significance.

4. General Requirements

All HTTP/HTTPS [RFC7230] and [RFC2818] connections specified throughout this document MUST use HTTPS. Both HTTPS and all SIP connections MUST use TLS conforming to at least [RFC7525] and MUST support [RFC8446].

All text data payloads not otherwise constrained by a specification in another standards document MUST be encoded as Unicode UTF-8.

Implementations MUST support IPv4 and IPv6. Dual stack support is NOT required and provider implementations MAY support separate interfaces for IPv4 and IPv6 by having more than one server in the appropriate SRV record where there is either an A or AAAA record in each server DNS record but not both. The same version of IP MUST be used for both signaling and media of a call unless ICE ([RFC8445]) is used, in which case candidates may explicitly offer IPv4, IPv6 or both for any media stream.

5. SIP Signaling

Implementations of the RUE Interface MUST conform to the following core SIP standards:

- * [RFC3261] (Base SIP)
- * [RFC3263] (Locating SIP Servers)
- * [RFC3264] (Offer/Answer)
- * [RFC3840] (User Agent Capabilities)
- * [RFC5626] (Outbound)
- * [RFC8866] (Session Description Protocol)
- * [RFC3323] (Privacy)
- * [RFC3605] (RTCP Attribute in SDP)
- * [RFC6665] (SIP Events)
- * [RFC3311] (UPDATE Method)
- * [RFC5393] (Loop-Fix)
- * [RFC5658] (Record Route fix)
- * [RFC5954] (ABNF fix)
- * [RFC3960] (Early Media)
- * [RFC6442] (Geolocation header field)

In the above documents the RUE device conforms to the requirements of a SIP user Agent, and the provider conforms to the requirements of Registrar and Proxy Server where the document specifies different behavior for different roles. The only requirement on providers for RFC6665 (Events) is support for the Message Waiting Indicator (See Section 8), which is optional and providers not supporting video mail need not support RFC6665.

In addition, implementations MUST conform to:

- * [RFC3327] (Path)
- * [RFC8445] and [RFC8839] (ICE)

- * [RFC3326] (Reason header field)
- * [RFC3515] (REFER Method)
- * [RFC3891] (Replaces Header field)
- * [RFC3892] (Referred-By)

Implementations MUST implement full ICE, although they MAY interwork with User Agents that implement ICE-lite.

Implementations MUST include a "User-Agent" header field uniquely identifying the RUE application, platform, and version in all SIP requests, and MUST include a "Server" header field with the same content in SIP responses.

Implementations intended to support mobile platforms MUST support [RFC8599] and MUST use it as at least one way to support waking up the client from background state.

The SIP signaling for registration and placing/receiving calls depends on configuration of various values into the RUE device. Section 9.2 describes the configuration mechanism which provides values that are used in the signaling. When the device starts, the configuration mechanism is run which retrieves the configuration data, and then SIP registration occurs using the values from the configuration. After registration, calls may be sent or received by the RUE device.

5.1. Registration

The RUE MUST register with a SIP registrar, following [RFC3261] and [RFC5626] at a provider it has an account with. If the configuration (see Section 9.2) contains multiple "outbound-proxies" in "RueConfigurationData", then the RUE MUST use them as specified in [RFC5626] to establish multiple flows.

The Request-URI for the REGISTER request MUST contain the "provider-domain" from the configuration. The To-URI and From-URI MUST be identical URIs, formatted as follows:

- * if "user-name" is provided: "username@provider-domain";
- * if "user-name" is not provided: as specified in Section 5.4, using "phone-number" and "provider-domain" from the configuration.

The RUE determines the URI to resolve by initially determining if one or more outbound proxies are configured. If there are, the URI will be that of one of the "outbound-proxies". If no "outbound-proxies" are configured, the URI will be the Request-URI from the REGISTER request. The RUE extracts the domain from that URI and consults the DNS record for that domain. The DNS entry MUST contain NAPTR records conforming to RFC3263. One of those NAPTR records MUST specify TLS as the preferred transport for SIP. For example, a DNS NAPTR query for "sip: pl.red.example.net" could return:

```
IN NAPTR 50 50 "s" "SIPS+D2T" "" _sips._tcp.pl.red.example.net
IN NAPTR 90 50 "s" "SIP+D2T" "" _sip._tcp.pl.red.example.net
```

If the RUE receives a 439 (First Hop Lacks Outbound Support) response to a REGISTER request, it MUST re-attempt registration without using the outbound mechanism.

The registrar MAY authenticate the RUE using SIP digest authentication. The credentials to be used MUST come from the configuration Section 9.2: "user-name" if provided or "phone-number" if user-name is not provided, and "sip-password". This "user-name"/"sip-password" combination SHOULD NOT be the same as that used for other purposes, except as expressly described below, such as retrieving the RUE configuration or logging into the Provider's customer service portal. [RFC8760] MUST be supported by all implementations and SHA-512 digest algorithms MUST be supported.

If the registration request fails with an indication that credentials from the configuration are invalid, then the RUE MUST retrieve a fresh version of the configuration. If credentials from a freshly retrieved configuration are found to be invalid, then the RUE MUST cease attempts to register and inform the RUE User of the problem.

Support for multiple simultaneous registrations with multiple providers by the RUE is OPTIONAL for the RUE (and providers do not need any support for this option).

Multiple simultaneous RUE SIP registrations from different RUE devices with the same SIP URI SHOULD be permitted by the provider. The provider MAY limit the total number of simultaneous registrations. When a new registration request is received that results in exceeding the limit on simultaneous registrations, the provider MAY then prematurely terminate another registration; however, it SHOULD NOT do this if it would disconnect an active call.

If a provider prematurely terminates a registration to reduce the total number of concurrent registrations with the same URI, it SHOULD take some action to prevent the affected RUE from automatically re-registering and re-triggering the condition.

5.2. Session Establishment

5.2.1. Normal Call Origination

After initial SIP registration, the RUE adheres to SIP [RFC3261] basic call flows, as documented in [RFC3665].

A RUE device MUST route all outbound calls through an outbound proxy if configured.

The SIP URIs in the To field and the Request-URI MUST be formatted as specified in subsection Section 5.4 using the destination phone number, or as SIP URIs, as provided in the configuration (Section 9.2). The domain field of the URIs SHOULD be the "provider-domain" from the configuration (e.g., sip:+15551234567@red.example.com;user=phone) except that an anonymous call would not use the provider domain.

Anonymous calls MUST be supported by all implementations. An anonymous call is signaled per [RFC3323].

The From-URI MUST be formatted as specified in Section 5.4, using the phone-number and "provider-domain" from the configuration. It SHOULD also contain the display-name from the configuration when present. (Please refer to Section 9.2.)

Negotiated media MUST follow the requirements specified in Section 6 of this document.

To allow time to time out an unanswered call and direct it to a videomail server, the User Agent Client MUST NOT impose a time limit less than the default SIP Invite transaction timeout of 3 minutes.

5.2.2. Dial-Around Origination

Providers and RUE devices MUST support both One-Stage and Two-Stage dial-around

Outbound dial-around calls allow a RUE user to select any provider to provide interpreting services for any call. "Two-stage" dial-around calls involve the RUE calling a telephone number that reaches the dial-around provider and using signing or DTMF to provide the called party telephone number. In two-stage dial-around, the To URI is the

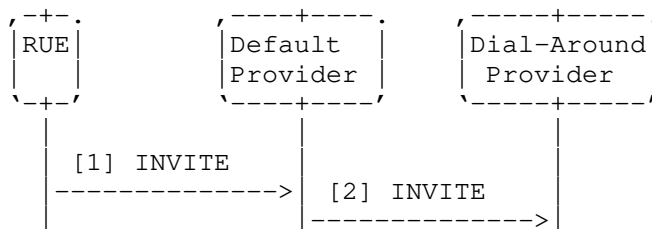
"frontDoor" URI (see Section 9.2) in "ProviderConfigurationData" of the dial-around provider. The RUE Provider Selection service (Section 9.1) can be used by the RUE to obtain a list of providers and then the Provider Configuration (Section 9.2.1) can be used to find the front door URI for each of these providers.

One-stage dial-around is a method where the called party telephone number is provided in the To URI and the Request-URI, using the domain of the dial-around provider.

For one-stage dial-around, the RUE MUST follow the procedures in Section 5.2.1 with the following exception: the domain part of the SIP URIs in the To field and the Request-URI MUST be the domain of the dial-around provider, discovered according to Section 9.1.

The following is a partial example of a one-stage dial-around call from VRS user +1-555-222-0001 hosted by red.example.com to a hearing user +1-555-123-4567 using dial-around to green.example.com for the relay service. Only important details of the messages are shown and many header fields have been omitted:

One-Stage Dial-Around



Message Details:

[1] INVITE Rue -> Default Provider

```

INVITE sip:+15551234567@green.example.net;user=phone SIP/2.0
To: <sip:+15551234567@green.example.net;user=phone>
From: "Bob Smith" <sip:+15552220001@red.example.net;user=phone>
  
```

[2] INVITE Default Provider -> Dial-Around Provider

```

INVITE sip:+15551234567@green.example.net;user=phone SIP/2.0
To: <sip:+15551234567@green.example.net;user=phone>
From: "Bob Smith" sip:+15552220001@red.example.net;user=phone
P-Asserted-Identity: sip:+15552220001@red.example.net
  
```

Figure 1

5.2.3. RUE Contact Information

To identify the owner of a RUE, the initial INVITE for a call from a RUE, or the 200 OK the RUE uses to accept a call, identifies the owner by sending a Call-Info header field with a purpose parameter of "rue-owner". The URI MAY be an HTTPS URI or Content-ID URL. The latter is defined by [RFC2392] to locate message body parts. This URI type is present in a SIP message to convey the RUE ownership information as a MIME body. The form of the RUE ownership information is a xCard [RFC6351]. Please refer to [RFC6442] for an example of using Content-Indirect URLs in SIP messages. Note that use of the Content-Indirect URL usually implies multiple message bodies ("mime/multipart"). The RUE owner is the entity that has local control over the device which is not necessarily the legal owner of the equipment. It often is the user, but that is not necessarily true. While no minimum fields in the xCard are specified, the name, address, phone number and email address of the RUE owner are expected to be supplied.

5.2.4. Incoming Calls

The RUE MUST only accept inbound calls sent to it by a proxy mentioned in the configuration.

If Multiple simultaneous RUE SIP registrations from different RUE devices with the same SIP URI exist, the provider MUST parallel fork the call to all registered RUEs so that they ring at the same time. The first RUE to reply with a 200 OK answers the call and the provider MUST CANCEL other call branches.

5.2.5. Emergency Calls

Implementations MUST conform to [RFC6881] for handling of emergency calls, except that if the device is unable to determine its own location, it MAY send the emergency call without a Geolocation header field and without a Route header field (since it would be unable to query the LoST server for a route per RFC6881). If an emergency call arrives at the provider without a Geolocation header field, the provider MUST supply location by adding the Geolocation header field, and MUST supply the route by querying the LoST server with that location.

If the emergency call is to be handled using existing country specific procedures, the provider is responsible for modifying the INVITE to conform to the country-specific requirements. In this case, location MAY be extracted from the RFC6881 conformant INVITE and used to propagate it to the appropriate country-specific entities. If the configuration specifies it, an implementation of a

RUE device MAY send a Geolocation header field containing its location in the REGISTER request. If implemented, users MUST be offered an opt-out. Country-specific procedures might require the location to be pre-loaded in some entity prior to placing an emergency call; however, the RUE may have a more accurate and timely device location than the manual, pre-loaded entry. That information MAY be used to populate the location to appropriate country-specific entities. Re-registration SHOULD be used to update the location, so long as the rate of re-registration is limited if the device is moving.

Implementations MUST implement Additional Data, [RFC7852]. RUE devices MUST implement Data Provider, Device Information and Owner/Subscriber Information blocks.

5.3. Mid-Call Signaling

Implementations MUST support re-INVITE to renegotiate media session parameters (among other uses). Per Section 6.1, implementations MUST be able to support an INFO request for full frame refresh for devices that do not support RTCP mechanisms (please refer to Section 6.8). Implementations MUST support an in-dialog REFER ([RFC3515] updated by [RFC7647] and including support for norefersub per [RFC4488]) with the Replaces header field [RFC3891] to enable call transfer.

5.4. URI Representation of Phone Numbers

SIP URIs constructed from non-URI sources (dial strings) and sent to SIP proxies by the RUE MUST be represented as follows, depending on whether they can be represented as an E.164 number. In this section "expressed as an E.164 number" includes numbers such as toll-free numbers that are not actually E.164 numbers, but have the same format.

A dial string that can be expressed as an E.164 phone number MUST be represented as a SIP URI with a URI ";user=phone" tag. The user part of the URI MUST be in conformance with 'global-number' defined in [RFC3966]. The user part MUST NOT contain any 'visual-separator' characters, as defined in [RFC3966].

Dial strings that cannot be expressed as E.164 numbers MUST be represented as dialstring URIs, as specified by [RFC4967], e.g., sip:411@red.example.net;user=dialstring.

The domain part of Relay Service URIs and User Address of Records (AoR) MUST resolve (per [RFC3263]) to globally routable IPv4 and/or IPv6 addresses.

5.5. Transport

Implementations MUST conform to [RFC8835] except for its guidance on the WebRTC data channel, which this specification does not use. See Section 6.2 for how RUE supports real-time text without the data channel.

Implementations MUST support SIP outbound [RFC5626] (please also refer to Section 5.1).

6. Media

This specification adopts the media specifications for WebRTC ([RFC8825]). Where WebRTC defines how interactive media communications may be established using a browser as a client, this specification assumes a normal SIP call. Various RTP, RTCP, SDP and specific media requirements specified for WebRTC are adopted for this document. Explicit requirements from the WebRTC suite of documents are described below .

To use WebRTC with this document, a gateway that presents a WebRTC server interface towards a browser, and a RUE client interface towards a provider is assumed. The gateway would interwork signaling, and as noted below, interwork at least any real time text media, in order to allow a standard browser based WebRTC client to be a VRS client. The combination of the browser client and the gateway would be a RUE user. This document does not specify the gateway.

The following sections specify the WebRTC documents to which conformance is required. "Mandatory to Implement" means a conforming implementation MUST implement the specified capability. It does not mean that the capability must be used in every session. For example, OPUS is a mandatory to implement audio codec, and all conforming implementations must support OPUS. However, implementation presenting a call across the RUE Interface where the call originates in the Public Switched Telephone Network, or an older, non-RUE-compatible device, which only offers G.711 audio, does not need to include the OPUS codec in the offer, since it cannot be used with that call. Conformance to this document allows end-to-end RTCP and media congestion control for audio and video.

6.1. SRTP and SRTCP

Implementations MUST support [RFC8834] except that `MediaStreamTracks` are not used. Implementations MUST conform to Section 6.4 of [RFC8827].

6.2. Text-Based Communication

Implementations MUST support real-time text ([RFC4102] and [RFC4103]) via T.140 media. One original and two redundant generations MUST be transmitted and supported, with a 300 ms transmission interval. Implementations MUST support [RFC9071] especially for emergency calls. Note that RFC4103 is not how real-time text is transmitted in WebRTC and some form of transcoder would be required to interwork real-time text in the data channel of WebRTC to RFC4103 real-time text.

Transport of T.140 real-time text in WebRTC is specified in [RFC8865], using the WebRTC data channel. RFC 8865 also has some advice on how gateways between RFC 4103 and RFC 8865 should operate. It is RECOMMENDED that RFC 8865 including multiparty support is used for communication with browser-based WebRTC implementations. Implementations MUST support [RFC9071].

6.3. Video

Implementations MUST conform to [RFC7742] with following exceptions: only H.264, as specified in [RFC7742], is Mandatory to Implement, and VP8 support is OPTIONAL at both the device and providers. This is because backwards compatibility is desirable, and older devices do not support VP8.

6.4. Audio

Implementations MUST conform to [RFC7874].

6.5. DTMF Digits

Implementations MUST support the "audio/telephone-event" [RFC4733] media type. They MUST support conveying event codes 0 through 11 (DTMF digits "0"-"9", "*", "#") defined in Table 7 of [RFC4733]. Handling of other tones is OPTIONAL.

6.6. Session Description Protocol

The SDP offers and answers MUST conform to the SDP rules in [RFC8829] except that the RUE interface uses SIP transport for SDP. The SDP for real-time text MUST specify the T.140 payload type [RFC4103].

6.7. Privacy

The RUE MUST provide for user privacy by implementing a local one-way mute, without signaling, for both audio and video. However, RUE MUST maintain any states in the network (e.g. NAT bindings) by periodically sending media packets on all active media sessions containing silence/comfort noise/black screen/etc. per [RFC6263].

6.8. Negative Acknowledgment, Packet Loss Indicator, and Full Intraframe Request Features

The NACK, FIR and PLI features as described in [RFC4585] and [RFC5104] MUST be implemented. Availability of these features MUST be announced with the "ccm" feedback value. NACK should be used when negotiated and conditions warrant its use and the other end supports it. Signaling picture losses as Packet Loss Indicator (PLI) should be preferred. FIR should be used only in situations where not sending a decoder refresh point would render the video unusable for the users, as per RFC5104 subsection 4.3.1.2.

For backwards compatibility with calling devices that do not support the foregoing methods, implementations MUST implement SIP INFO messages to send and receive XML encoded Picture Fast Update messages according to [RFC5168].

7. Contacts

7.1. CardDAV Login and Synchronization

Support of CardDAV by providers is OPTIONAL.

The RUE MUST and providers MAY be able to synchronize the user's contact directory between the RUE endpoint and one maintained by the user's VRS provider using CardDAV ([RFC6352] and [RFC6764]).

The configuration (see Section 9.2) RueConfigurationData MAY supply a "carddav-username" and "carddav-domain" identifying a CardDAV server and address book for this account, plus an optional "carddav-password".

To access the CardDAV server and address book, the RUE MUST follow Section 6 of RFC6764, using the configured carddav-username and carddav-domain in place of an email address. If the request triggers a challenge for digest authentication credentials, the RUE MUST continue using matching carddav-username and carddav-password from the configuration. If no carddav-username and carddav-password are configured, the RUE MUST use the SIP user-name and sip-password from the configuration. If the SIP credentials fail, the RUE MUST query the user.

Synchronization using CardDAV MUST be a two-way synchronization service, with proper handling of asynchronous adds, changes, and deletes at either end of the transport channel.

The RUE MAY support other CardDAV services.

7.2. Contacts Import/Export Service

Implementations MUST be able to export/import the list of contacts in xCard [RFC6351] XML format.

The RUE accesses this service via the "contacts-uri" in the configuration. The URL MUST resolve to identify a web server resource that imports/exports contact lists for authorized users.

The RUE stores/retrieves the contact list (address book) by issuing an HTTPS POST or GET request. If the request triggers a challenge for digest authentication credentials, the RUE MUST attempt to continue using the "contacts-username" and "contacts-password" from the configuration. If no contacts-username is configured, the sip user-name from the configuration is used, and if the sip user-name is not configured, the phone-number is used. If user-name or phone-number is used, the sip-password is used to authenticate to the contact list server.

8. Video Mail

Support for video mail includes a retrieval mechanism and a Message Waiting Indicator (MWI). Message storage is not specified by this document. RUE devices MUST support message retrieval using a SIP call to a specified SIP URI using DTMF to manage the mailbox, as well as a browser based interface reached at a specified HTTPS URI. If a provider supports video mail at least one of these mechanisms MUST be supported. RUE devices MUST support both. See Section 9.2 for how the URI to reach the retrieval interface is obtained.

Implementations MUST support subscriptions to "message-summary" events [RFC3842] to the URI specified in the configuration. Providers MUST support MWI if they support video mail. RUE devices MUST support MWI.

The "videomail" and "mwi" properties in the configuration (see RueConfigurationData in Section 9.2.2) gives the URIs for message retrieval and "message-summary" subscription.

In notification bodies, if detailed message summaries are available, messages with video MUST be reported using "message-context-class multimedia-message" defined in [RFC3458] .

9. Provisioning and Provider Selection

To simplify how users interact with RUE devices, the RUE interface separates provisioning into two parts. One provides a directory of providers so that a user interface can allow easy provider selection either for registering or for dial-around. The other provides configuration data for the device for each provider.

9.1. RUE Provider Selection

To allow the user to select a relay service, the RUE MAY at any time obtain (typically on startup) a list of Providers that provide service in a country. IANA has established a registry that contains a two-letter country code and an entry point string (See Section 10.1). The entry point, when used with the following OpenAPI interface, returns a list of provider names for a country code suitable for display, with a corresponding entry point to obtain information about that provider. No mechanism to determine the country the RUE is located is specified in this document. Typically the country is the home country of the user, but may be a local country while traveling. Some countries allow support from their home country when traveling abroad. Regardless, the RUE device will need to allow the user to choose the country.

Each country that supports video relay service using this specification MAY support the provider list. This document does not specify who maintains the list. Some possibilities are a regulator or entity designated by a regulator, an agreement among providers to provide the list, or a user group.

The interface to obtain the list of providers is described by an OpenApi [OpenApi] interface description. In that interface description, the "servers" component includes an occurrence of "localhost". The value from the registry of the "list entry point" string for the desired country is substituted for "localhost" in the

"servers" component to obtain the server URI prefix of the interface to be used to obtain the list of providers for that country. The "Providers" path then specifies the rest of the URI used to obtain the list. For example, if the list entryPoint is "example.com/api", the provider list would be obtained from `https://example.com/api/rum/v1/Providers`.

The V1.0 "ProviderList" is a JSON object consisting of an array where each entry describes one provider. Each entry consists of the following items:

- * name: This parameter contains the text label identifying the provider and is meant to be displayed to the human VRS user.
- * providerEntryPoint: A string used for configuration purposes by the RUE (as discussed in Section 9.2). The string **MUST** start with a domain, but **MAY** include other URI path elements after the domain.

The VRS user interacts with the RUE to select from the provider list one or more providers with whom the user has already established an account, wishes to establish an account, or wishes to use the provider for a one-stage dial around.

```
{
  "providers": [
    {
      "name": "Red",
      "entryPoint": "red.example.net"
    },
    {
      "name": "Green",
      "entryPoint": "green.example.net"
    },
    {
      "name": "Blue",
      "entryPoint": "blue.example.net"
    }
  ]
}
```

Figure 2: Example of a ProviderList JSON object

9.2. RUE Configuration Service

A RUE device may retrieve a provider configuration using a simple HTTPs web service. There are two entry points. One is used without user credentials, and the response includes configuration data for new user sign up and dial around. The other uses locally stored username and password that results from a new user sign up to authenticate to the interface and returns configuration data for the RUE.

The interface to obtain configuration data is described by an OpenApi [OpenApi] interface description. In that interface description, the "servers" component string includes an occurrence of "localhost". The entry point string obtained from the provider list (Section 9.1) is substituted for "localhost" to obtain the server prefix of the interface. The path then specifies the rest of the URI used to obtain the list. For example, if the entryPoint from the provider list is "red.example.net", the provider configuration would be obtained from `https://red.example.net/rum/V1/ProviderConfig` and the RUE configuration would be obtained from `https://red.example.net/rum/V1/RueConfig`.

In both the queries, an optional parameter may be provided to the interface which is an API Key (apiKey). The implementation MAY have an apiKey obtained from the provider and specific to the implementation. The method used to obtain the apiKey is not specified in this document. The provider MAY refuse to provide service to an implementation presenting an apiKey it does not recognize.

Also in both queries, the RUE device provides a client provided, required parameter, which contains an instance identifier (instanceId). This parameter MUST be the same value each time this instance (same implementation on same device) queries the interface. This MAY be used by the provider, for example, to associate a location with the instance for emergency calls. This should be globally unique. A UUID is suggested.

For example, a query for the RUE configuration could be `https://red.example.net/rum/V1/RueConfig?apiKey="t65667Ajjss90uuuDisKt8999"&instanceId="5595b5a3-0687-4b8e-9913-a7f2a04fb7bd"`

The data returned is a JSON object consisting of key/value configuration parameters to be used by the RUE.

The configuration data payload includes the following data items. Items not noted as (OPTIONAL) are REQUIRED. If other unexpected items are found, they MUST be ignored.

9.2.1. Provider Configuration

- * **signup:** (OPTIONAL) an array of JSON objects consisting of:
 - **language:** entry from the IANA language subtag registry (<https://www.iana.org/assignments/language-subtag-registry/language-subtag-registry>). Normally, this would be a written language tag.
 - **uri:** a URI to the website for creating a new account in the supported language. The new user signup URI may only initiate creation of a new account. Various vetting, approval and other processes may be needed, which could take time, before the account is established. The result of creating a new account would be account credentials (e.g. username and password), which would be manually entered into the RUE device which form the authentication parameters for the RUE configuration service described below in Section 9.2.2.
- * **dialAround:** an array of JSON objects consisting of:
 - **language:** entry from the IANA language subtag registry. Normally, this would be a sign language tag.
 - **frontDoor:** a URI to a queue of interpreters supporting the specified language for a two-stage dial-around
 - **oneStage:** a URI that can be used with a one-stage dial-around Section 5.2.2 using an interpreter supporting the specified language
- * **helpDesk:** (OPTIONAL) an array of JSON objects consisting of:
 - **language:** entry from the IANA language subtag registry. Normally this would be a sign language tag, although it could be a written language tag if the help desk only supports a chat interface
 - **uri:** URI that reaches a help desk for callers supporting the specified language. The URI MAY be a SIP URI for help provided with a SIP call, or MAY be an HTTPS URI for help provided with a browser interface.

A list is specified so that the provider can offer multiple choices to users for language and interface styles.

9.2.2. RUE Configuration

- * **lifetime:** (optional) Specifies how long (in seconds) the RUE MAY cache the configuration values. Values may not be valid when lifetime expires. If the RUE caches configuration values, it MUST cryptographically protect them against unauthorized disclosure (e.g. by other applications on the platform the RUE is built on). The RUE SHOULD retrieve a fresh copy of the configuration before the lifetime expires or as soon as possible after it expires. The lifetime is not guaranteed: the configuration may change before the lifetime value expires. In that case, the Provider MAY indicate this by generating authorization challenges to requests and/or prematurely terminating a registration. Emergency Calls MUST continue to work. If not specified, the RUE MUST fetch new configuration data every time it starts.
- * **sip-password:** (optional) a password used for SIP, STUN and TURN authentication. The RUE device retains this data, which it MUST cryptographically protect against unauthorized disclosure (e.g. by other applications on the platform the RUE is built on). If it is not supplied, but was supplied on a prior invocation of this interface, the most recently supplied password MUST be used. If it was never supplied, the password used to authenticate to the configuration service is used for SIP, as well as STUN and TURN servers mentioned in this configuration.
- * **phone-number:** The telephone number (in E.164 format) assigned to this subscriber. This becomes the user portion of the SIP URI identifying the subscriber.
- * **user-name:** (optional) a username used for authenticating to the provider. If not provided, the phone-number is used.
- * **display-name:** (optional) a human readable display name for the subscriber
- * **provider-domain:** the domain for the provider. This becomes the server portion of the SIP URI identifying the subscriber.
- * **outbound-proxies:** (optional) An array of URIs of SIP proxies to be used when sending requests to the provider.
- * **mwi:** (optional) A URI identifying a SIP event server that generates "message-summary" events for this subscriber.
- * **videomail:** (optional) An SIP or HTTPS URI that can be used to retrieve video mail messages.

- * `contacts`: (optional) An HTTPS URI ("`contacts-uri`"), (optional) "`contacts-username`" and "`contacts-password`" that may be used to export (retrieve) the subscriber's complete contact list managed by the provider. At least the URI MUST be provided if the subscriber has contacts. If `contact-username` and `contacts-password` are not supplied, the sip credentials are used. If the `contacts-username` is provided, `contacts-password` MUST be provided. If `contacts-password` is provided, `contacts-username` MUST be provided.
- * `carddav`: (optional) An address ("`carddav-domain`"), (optional) "`carddav-username`" and "`carddav-password`" identifying a "CardDAV" server and account that can be used to synchronize the RUE's contact list with the contact list managed by the provider. If `carddav-username` and `carddav-password` are not supplied, the sip credentials are used. If the `carddav-username` is provided, `carddav-password` MUST be provided. If `carddav-password` is provided, `carddav-username` MUST be provided.
- * `sendLocationWithRegistration`: (optional) True if the RUE should send a Geolocation header field with REGISTER, false if it should not. Defaults to false if not present.
- * `ice-servers`: (optional) An array of server types and URLs identifying STUN and TURN servers available for use by the RUE for establishing media streams in calls via the provider. If the same URL provides both STUN and TURN services, it MUST be listed twice, each with different server types.

9.2.3. Versions

Both web services also have a simple version mechanism that returns a list of versions of the web service it supports. This document describes version 1.0. Versions are described as a major version, the period "." and a minor version, where major and minor versions are integers. A backwards compatible change within a major version MAY increment only the minor version number. A non-backwards compatible change MUST increment the major version number. Backwards compatibility applies to both the server and the client. Either may have any higher or lower minor revision and interoperate with its counterpart with the same major version. To achieve backwards compatibility, implementations MUST ignore any object members they do not implement. Minor version definitions SHALL only add objects, non-required members of existing objects, and non-mandatory-to use functions and SHALL NOT delete or change any objects, members of objects or functions. This means an implementation of a specific major version and minor version is backwards compatible with all minor versions of the major version. The version mechanism returns

an array of supported versions, one for each major version supported, with the minor version listed being the highest supported minor version.

Unless the per-country provider list service is operated by a provider at the same base URI as that provider's configuration service, the version of the configuration service MAY be different from the version of the provider list service.

```
{
  "versions": [
    {
      "major": 1,
      "minor": 6
    },
    {
      "major": 2,
      "minor": 13
    },
    {
      "major": 3,
      "minor": 2
    }
  ]
}
```

Figure 3: Example of a Version JSON object

9.2.4. Examples

Example JSON provider configuration payload

```
{
  "signUp": [
    { "language" : "en", "uri" : "https:hello-en.example.net"},
    { "language" : "es", "uri" : "https:hello-es.example.net"} ] ,
  "dialAround": [
    { "language" : "en", "frontDoor" : "sip:fd-en.example.net",
      "oneStage" : "sip:1stg-eng.example.com" } ,
    { "language" : "es", "frontDoor" : "sip:fd-es.example.net",
      "oneStage" : "sip:1stg-spn.example.com" } ] ,
  "helpDesk": [
    { "language" : "en", "uri" : "sip:help-en.example.net"} ,
    { "language" : "es", "uri" : "sip:help-es.example.net"} ]
}
```

Figure 4

Example JSON RUE configuration payload

```

{
  "lifetime": 86400,
  "display-name" : "Bob Smith",
  "phone-number": "+15551234567",
  "provider-domain": "red.example.net",
  "outbound-proxies": [
    "sip:p1.red.example.net",
    "sip:p2.red.example.net" ],
  "mwi": "sip:+15551234567@red.example.net;user=phone",
  "videomail": "sip:+15551234567@vm.red.example.net;user=phone",
  "contacts": {
    "contacts-uri":
      "https://red.example.net:443/c/3617b719-2c3a-46f4-9c13",
    "contacts-username": "bob",
    "contacts-password": "XhOT4ch@ZEi&3u2xEYQNMO^5UGb"
  },
  "carddav": {
    "carddav-domain": "carddav.example.com",
    "carddav-username": "bob",
    "carddav-password": "sj887%dd*jJty%87hyys5hHT"
  },
  "sendLocationWithRegistration": false,
  "ice-servers": [
    {"stun": "stun.red.example.com:19302" },
    {"turn": "turn.red.example.com:3478"}
  ]
}

```

Figure 5

9.2.5. Using the Provider Selection and RUE Configuration Services Together

One way to use these two services is:

- * At startup, the RUE retrieves the provider list for the country it is located in.
- * For each provider in the list:
 - If the RUE does not have credentials for that provider, if requested by the user, use the ProviderConfig path without credentials to obtain signup, dial around and helpdesk information.
 - If the RUE has credentials for that provider, use the RueConfig path with the locally stored credentials to configure the RUE for that provider.

9.3. OpenAPI Interface Descriptions

The interfaces in Section 9.1 and Section 9.2 are formally specified with OpenAPI 3.0 ([OpenApi]) descriptions in YAML form.

The OpenAPI description below is normative. If there is any conflict between the text or examples and this section, the OpenAPI description takes precedence.

9.3.1. Provider List

```
openapi: 3.0.1
info:
  title: RUM Provider List API
  version: "1.0"
servers:
  - url: https://localhost/rum/v1
paths:
  /Providers:
    get:
      summary: Get a list of providers and domains to get
        config data from
      operationId: GetProviderList
      responses:
        '200':
          description: List of providers for a country
          content:
            application/json:
              schema:
                $ref: '#/components/schemas/ProviderList'
  /Versions:
    servers:
      - url: https://localhost/rum
        description: Override base path for Versions query
    get:
      summary: Retrieves all supported versions
      operationId: RetrieveVersions
      responses:
        '200':
          description: Versions supported
          content:
            application/json:
              schema:
                $ref: '#/components/schemas/VersionsArray'
components:
  schemas:
    ProviderList:
      type: object
```

```
required:
  - providers
properties:
  providers:
    type: array
    items:
      type: object
      required:
        - name
        - providerEntryPoint
      properties:
        name:
          type: string
          description: Human readable provider name
        providerEntryPoint:
          type: string
          description: provider entry point for interface
VersionsArray:
  type: object
  required:
    - versions
  properties:
    versions:
      type: array
      items:
        type: object
        required:
          - major
          - minor
        properties:
          major:
            type: integer
            format: int32
            description: Version major number
          minor:
            type: integer
            format: int32
            description: Version minor number
```

Figure 6: Provider List OpenAPI description (RueProviderList.yaml)

9.3.2. Configuration

```
openapi: 3.0.1
info:
  title: RUM Configuration API
  version: "1.0"
servers:
  - url: https://localhost/rum/v1
paths:
  /ProviderConfig:
    get:
      summary: Configuration data for one provider
      operationId: GetProviderConfiguration
      parameters:
        - in: query
          name: apiKey
          schema:
            type: string
            description: API Key assigned to this implementation
        - in: query
          name: instanceId
          schema:
            type: string
            required: true
            description: Unique string for this implementation
              on this device
      responses:
        '200':
          description: configuration object
          content:
            application/json:
              schema:
                $ref:
                  '#/components/schemas/ProviderConfigurationData'
  /RueConfig:
    get:
      summary: Configuration data for one RUE
      operationId: GetRueConfiguration
      parameters:
        - in: query
          name: apiKey
          schema:
            type: string
            description: API Key assigned to this implementation
        - in: query
          name: instanceId
          schema:
            type: string
            required: true
            description: Unique string for this implementation
```

```

        on this device
    responses:
      '200':
        description: configuration object
        content:
          application/json:
            schema:
              $ref: '#/components/schemas/RueConfigurationData'
  /Versions:
    servers:
      - url: https://localhost/rum
        description: Override base path for Versions query
    get:
      summary: Retrieves all supported versions
      operationId: RetrieveVersions
      responses:
        '200':
          description: Versions supported
          content:
            application/json:
              schema:
                $ref: '#/components/schemas/VersionsArray'
  components:
    schemas:
      ProviderConfigurationData:
        type: object
        required:
          - dialAround
        properties:
          signup:
            type: array
            items:
              type: object
              required:
                - language
                - uri
              properties:
                language:
                  type: string
                  description: entry from IANA language-subtag-registry
                uri:
                  type: string
                  format: uri
                  description: URI to signup website supporting
                    this language
          dialAround:
            type: array
            items:
```

```
    type: object
    required:
      - language
      - frontDoor
      - oneStage
    properties:
      language:
        type: string
        description: entry from IANA language-subtag-registry
      frontDoor:
        type: string
        format: uri
        description: SIP URI for two-stage dial around
      oneStage:
        type: string
        format: uri
        description: SIP URI for one-stage dial around
  helpDesk:
    type: array
    items:
      type: object
      required:
        - language
        - uri
      properties:
        language:
          type: string
          description: entry from IANA language-subtag-registry
        uri:
          type: string
          format: uri
          description: SIP URI of helpdesk supporting language
  RueConfigurationData:
    type: object
    required:
      - phone-number
      - provider-domain
    properties:
      lifetime:
        type: integer
        description: how long (in seconds) the RUE MAY cache the
          configuration values
      sip-password:
        type: string
      phone-number:
        type: string
        description: telephone number assigned this subscriber in
          E.164 format
```

```
user-name:
  type: string
  description: a username assigned this subscriber.
display-name:
  type: string
  description: display name for the subscriber
provider-domain:
  type: string
  description: domain of the provider for this subscriber
outbound-proxies:
  type: array
  items:
    type: string
    format: uri
    description: SIP URI of a proxy to be used when sending
                  requests to the provider
mwi:
  type: string
  format: uri
  description: A URI identifying a SIP event server that
               generates "message-summary" events for this subscriber.
videomail:
  type: string
  format: uri
  description: An HTTPS or SIP URI that can be used to
               retrieve video mail messages.
contacts:
  type: object
  description: server and credentials for contact
               import/export
  required:
    - contacts-uri
  properties:
    contacts-uri:
      type: string
      format: uri
      description: An HTTPS URI that may be used to export
                   (retrieve) the subscriber's complete contact list
                   managed by the provider.
    contacts-username:
      type: string
      description: username for authentication with CardDAV
                   server. Use sip user-name if not provided
    contacts-password:
      type: string
      description: password for authentication. Use provider
                   sip-password if not provided
carddav:
```

```
type: object
description: CardDAV server and user information that can
             be used to synchronize the RUE's contact list with
             the contact list managed by the provider.
required:
  - carddav-domain
properties:
  carddav-domain:
    type: string
    description: CardDAV server address
  carddav-username:
    type: string
    description: username for authentication with CardDAV
                 server. Use sip user-name if not provided
  carddav-password:
    type: string
    description: password for authentication to the CardDAV
                 server. Use provider sip-password if not provided
  sendLocationWithRegistration:
    type: boolean
    description: True if the RUE should send a Geolocation
                 header field with REGISTER, false if it should not.
                 Defaults to false if not present.
  ice-servers:
    type: array
    items:
      type: object
      required:
        - server-type
        - uri
      properties:
        server-type:
          type: string
          description: server type ("stun" or "turn")
        uri:
          type: string
          format: uri
          description: URIs identifying STUN and TURN servers
                       available for use by the RUE for establishing
                       media streams in calls via the provider.
VersionsArray:
  type: object
  required:
    - versions
  properties:
    versions:
      type: array
      items:
```

```
type: object
required:
  - major
  - minor
properties:
  major:
    type: integer
    format: int32
    description: Version major number
  minor:
    type: integer
    format: int32
    description: Version minor number
```

Figure 7: Configuration OpenAPI description (RueConfiguration.yaml)

10. IANA Considerations

10.1. RUE Provider List Registry

IANA has created the "RUE Provider List" registry. The management policy for this registry is "Expert Review" [RFC8126]. The expert should prefer a regulator operated or designated list interface operator. Otherwise, evidence that the proposed list interface operator will provide a complete list of providers is required to add the entry to the registry. Updates to the registry are permitted if the expert judges the new proposed URI to provide a more accurate list than the existing entry. Each entry has two fields, values for both of which MUST be provided when registering or updating an entry:

- * country code: a two-letter ISO93166 country code
- * list entry point: a string is used to compose the URI to the provider list interface for that country

10.2. Registration of rue-owner Value of the purpose Parameter

This document defines the new predefined value "rue-owner" for the "purpose" header field parameter of the Call-Info header field. The use for rue-owner is defined in Section 5.2.3. This modifies the "Header Field Parameters and Parameter Values" subregistry of the "Session Initiation Protocol (SIP) Parameters" registry by adding this RFC as a reference to the line for the header field "Call-Info" and parameter name "purpose"

- * Header Field: Call-Info
- * Parameter Name: purpose

* Predefined Values: Yes

11. Security Considerations

The RUE is required to communicate with servers on public IP addresses and specific ports to perform its required functions. If it is necessary for the RUE to function on a corporate or other network that operates a default-deny firewall between the RUE and these services, the user must arrange with their network manager for passage of traffic through such a firewall in accordance with the protocols and associated SRV records as exposed by the provider. Because VRS providers may use different ports for different services, these port numbers may differ from provider to provider.

This document requires implementation and use of a number of other specifications in order to fulfill the RUE profile; the security considerations described in those documents apply accordingly to the RUE interactions.

When a CA participates in a conversation they have access to the content of the conversation even though it is nominally a conversation between the two endpoints. There is an expectation that the CA will keep the communication contents in confidence. This is usually defined by contractual or legal requirements.

Since different providers (within a given country) may have different policies, RUE implementations MUST include a user interaction step to select from available providers before proceeding to actually register with any given provider.

12.

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Acknowledgements

Brett Henderson and Jim Malloy provided many helpful edits to prior versions of this document. Paul Kyzivat provided extensive reviews and comments.

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