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Interoperability Profile for Relay User Equipment
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Abstract

This document identifies a minimum set of standards and requirements that must be supported by a Video Relay Service (VRS) Video Access Technology Reference Platform (VATRP)-compliant client and United States Telecommunications Relay Service providers required to be VATRP compliant. This Relay User Equipment specification only specifies a minimum set of requirements. It does not prohibit VRS providers or endpoint developers from developing or deploying additional capabilities, provided that doing so will not prevent compliance with the requirements specified here.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on February 8, 2020.

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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.

This RUE profile supports the requirements of relay services in the United States, as described in 47 CFR 64.601 et seq., but may be applicable to similar uses elsewhere.

2. Scope

This RUE Specification documents the standards and controls associated with the Video Access Technology Reference Platform (VATRP). This RUE specification identifies the minimum set of standards for the interface between the VATRP and Providers' networks to which the VATRP adheres. This RUE Specification does not prohibit the implementation of additional features or functionality by any Provider. It also contains some Provider-optional features. If a Provider offers the feature described by the optional specification on at least one endpoint, the Provider MUST supply the standardized interface described in this document for that feature. This edition of the RUE specification does not address Provider-to-Provider communication (covered in the US VRS Provider Interface Profile) or the user interface to the RUE.

3. 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".

NANP: North America Numbering Plan (please refer to: http://nationalnanpa.org).

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 number database (RND): The iTRS Relay Number Database (RND) functions as a 10-digit NANP phone number lookup for SIP and H.323 URLs for TRS subscribers.

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.

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.

4. 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]

5. General Requirements

All HTTP/HTTPS connections specified throughout this document MUST use HTTPS. Both HTTPS and all SIP connections MUST use TLA conforming to [RFC7525]

During the establishment of secure connections with a provider, the RUE MAY be asked by the server for a client certificate. In that case it SHOULD provide the provisioned client certificate (See Section 10.2. Providers MAY reject requests that fail to provide a recognized certificate.

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

6. SIP Signaling

The RUE and Providers 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 addition, the RUE MUST, and Providers MAY, conform to [RFC3327] (Path), [RFC5245] (ICE), [RFC3326] (Reason header), [RFC3515] (REFER Method), [RFC3891] (Replaces Header), [RFC3892] (Referred-By).

RUEs 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.

6.1. Registration

The RUE MUST register with a SIP registrar, following [RFC3261] and [RFC5626]. 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: p1.red.example.netv" could return:

```
IN NAPTR 50  50 "s" "SIPS+D2T" "" _sips._tcp.p1.red.example.net
IN NAPTR 90  50 "s" "SIP+D2T"  "" _sip._tcp.p1.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 both the RUE and Providers 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 by Providers is OPTIONAL, as described in Section 2.

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.

6.2. Session Establishment

6.2.1. Normal Call Origination

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

The RUE MUST route all calls through the outbound proxy of the default Provider.

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 both the RUE and Providers. An anonymous call is signaled per [RFC3323].

The From-URI MUST be formatted as specified in Section 6.4, 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 10.2.)

Negotiated media MUST follow the guidelines specified in Section 7 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.

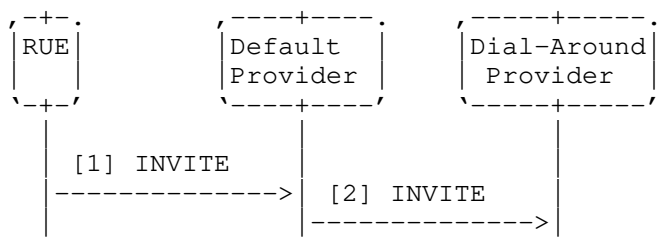
6.2.2. One-Stage Dial-Around Origination

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 6.2.1 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 10.1.

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:



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
```

One Stage Dial-Around

6.2.3. 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 an 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").

6.2.4. 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.

6.2.5. Emergency Calls

The RUE MUST comply with [RFC6881] for handling of emergency calls.

Providers MAY comply with RFC6881 for handling of emergency calls. In addition, they MUST:

- Accept RUE emergency calls complying with the specifications in this document;
- Recognize such calls as emergency calls and properly handle them as such;
- Address other behavior not specified by RFC6881 as specified in Section 6.2.

Specifically, if the emergency call is to be handled using E9-1-1 (VPC) procedures, the Provider is responsible for modifying the INVITE to conform to the VPC requirements. In this case, location MAY be extracted from the RFC6881 conformant INVITE and used to propagate it to the VPC where possible with the emergency call. Because the RUE may have a more accurate and timely location of the device than the typical manual entry location for nomadic RUE devices, the RUE MUST send a Geolocation header containing its location in the REGISTER request if the configuration specifies it. The Provider MAY use that information to populate the location of the device in the VPC before any emergency call.

6.3. Mid Call Signaling

The RUE and Providers MUST support re-INVITE to renegotiate media session parameters (among other uses). Per Section 7.1, the RUE MUST, and providers SHOULD, be able to support an INFO request for full frame refresh for devices in a call with the RUE that do not support RTCP mechanisms (please refer to Section 7.8). The RUE 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.

6.4. 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.

6.5. Transport

The RUE and providers MUST conform to [I-D.ietf-rtcweb-transports] with the understanding that this specification does not use the WebRTC data channel.

The RUE and providers MUST support SIP outbound [RFC5626] (please also refer to Section 6.1).

7. 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 following sections specify the WebRTC documents to which conformance is required.

7.1. SRTP and SRTCP

The RUE and Providers MUST support [I-D.ietf-rtcweb-rtp-usage] with the understanding that RUE does not specify an API and therefore MediaStreamTracks are not used. Implementations MUST conform to Section 6.4 of [I-D.ietf-rtcweb-security-arch].

7.2. Text-Based Communication

The RUE MUST and Providers 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.

7.3. Video

The RUE and Providers MUST conform to [RFC7742].

7.4. Audio

The RUE and Providers MUST conform to [RFC7874].

7.5. DTMF Digits

The RUE and Providers 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.

7.6. Session Description Protocol

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

7.7. 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].

7.8. 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, the RUE MUST implement and use SIP INFO messages to send and receive XML encoded Picture Fast Update messages according to [RFC5168].

8. Contacts

8.1. CardDAV Login and Synchronization

Support of CardDAV by Providers is OPTIONAL, as described in Section 2.

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.

8.2. Contacts Import/Export Service

Each Provider MUST supply a standard xCard import/export interface and the RUE 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 matching "credentials" from the configuration. If no credentials are configured, the RUE MUST query the user.

9. Mail Waiting Indicator (MWI)

Support of MWI by Providers is OPTIONAL, as described in Section 2

The RUE MUST and Providers SHOULD support subscriptions to "message-summary" events [RFC3842] to the URI specified in the configuration if the Provider supports message waiting indicator on any endpoint.

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

10. Provisioning and Provider Selection

10.1. 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.

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 10.2) and as the domain to use when targeting one-stage dial-around calls to this Provider (as discussed in Section 6.2.2).

  * 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.

```
{
  "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"
    }
  ]
}
```

Example of a Provider list JSON object

10.2. RUE Configuration Service

The RUE is provisioned with one or more URIs that may be queried for configuration with HTTPS.

The data returned will include a set of key/value configuration parameters to be used by the RUE, formatted as a 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.

- turn-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

```
{
  "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/1dess45awd"
  "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"
  }
]
}
```

Example JSON configuration payload

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.

10.3. Schemas

The following JSON schemas are for the Provider List and the RUE Configuration. These are represented using the JSON Content Rules [JCR] schema notation.

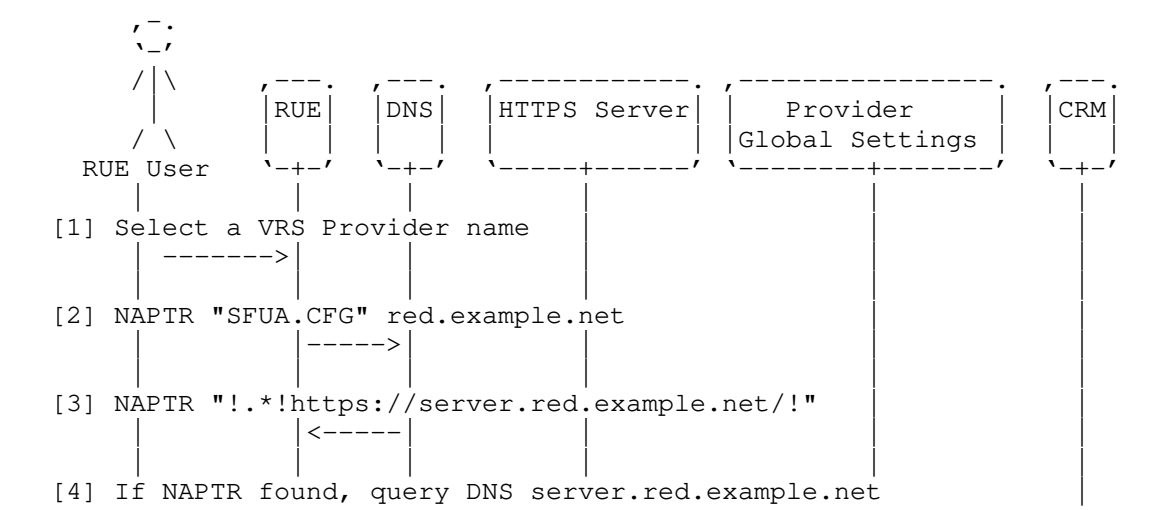
```
{
  "version": 1,
  "providers": [
    1*
      {
       "name": string,
       "domain": fqdn,
       ?"operator":           ; "front-door" access to provider
           uri,              ; (sip uri)
           * /^.*$/ : any        ; (allow future extensions)
      }
    ] ,
    * /^.*$/ : any            ; (allow future extensions)
}
```

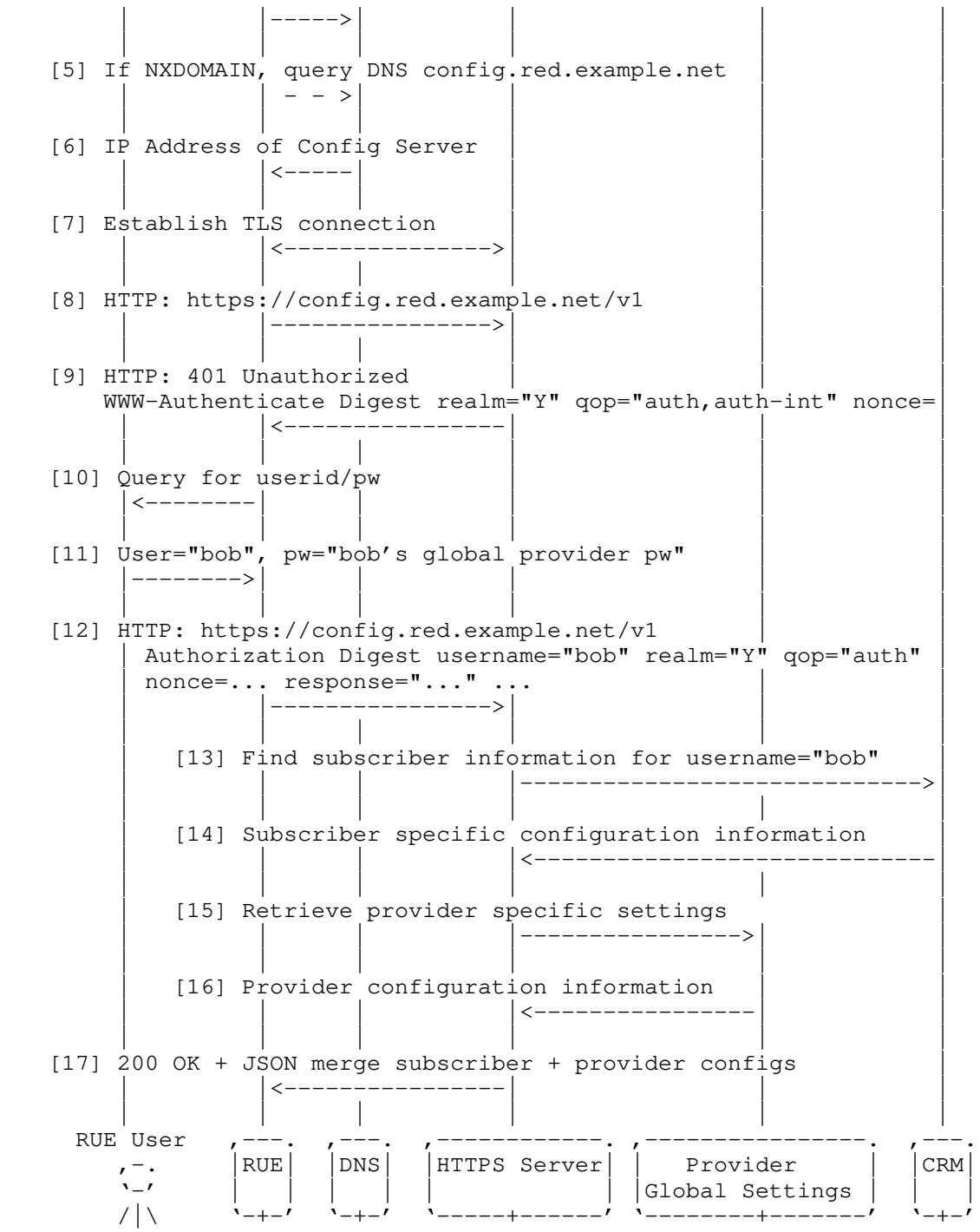
Provider List JSON Schema

```
{
  "version": 1,            ; Interface version
  "lifetime": integer,     ; Deadline (in seconds) for
                           ; refreshing this config without
                           ; user input.
  "phone-number": /^\+[0-9]+$/ , ; E.164 phone number
                          ; for this user
  ?"display-name" : string,; display name for From: header
  "provider-domain": fqdn, ; SHOULD match that in Provider List
  ?"outbound-proxies": [ 1* : uri ], ; sip URIs
  ?"mwi": uri ,            ; sip URI for MWI subscriptions
  ?"videomail": uri ,      ; sip URI for videomail retrieval
  "contacts": uri ,        ; https URI for contact list retrieval
  ?"carddav": /^[^@]+@[^@]+$/ , ; for contact list synch
  ?"sendLocationWithRegistration": boolean , ; send location y/n
  ?"ice-servers":          ; (Required for ICE use)
     [ 1* : uri ],         ; (stun[s] & turn[s] URIs
  ?"credentials":          ; for digest authentication
    [ 1* {
      "realm": string,
      "username": string,
      "password": string
      } ],
  * /^.*$/ : any           ; (allow future extensions)
}
```

RUE Configuration JSON Schema

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




[5] If NXDOMAIN, query DNS config.red.example.net
[6] IP Address of Config Server
[7] Establish TLS connection
[8] HTTP: https://config.red.example.net/v1
[9] HTTP: 401 Unauthorized
WWW-Authenticate Digest realm="Y" qop="auth,auth-int" nonce=
[10] Query for userid/pw
[11] User="bob", pw="bob's global provider pw"
[12] HTTP: https://config.red.example.net/v1
Authorization Digest username="bob" realm="Y" qop="auth"
nonce=... response="..." ...
[13] Find subscriber information for username="bob"
[14] Subscriber specific configuration information
[15] Retrieve provider specific settings
[16] Provider configuration information
[17] 200 OK + JSON merge subscriber + provider configs
RUE User
RUE
DNS
HTTPS Server
Provider
Global Settings
CRM

RUE Configuration Retrieval

11. Acknowledgements

12. IANA Considerations

This memo includes no request to IANA.

13. 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.

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