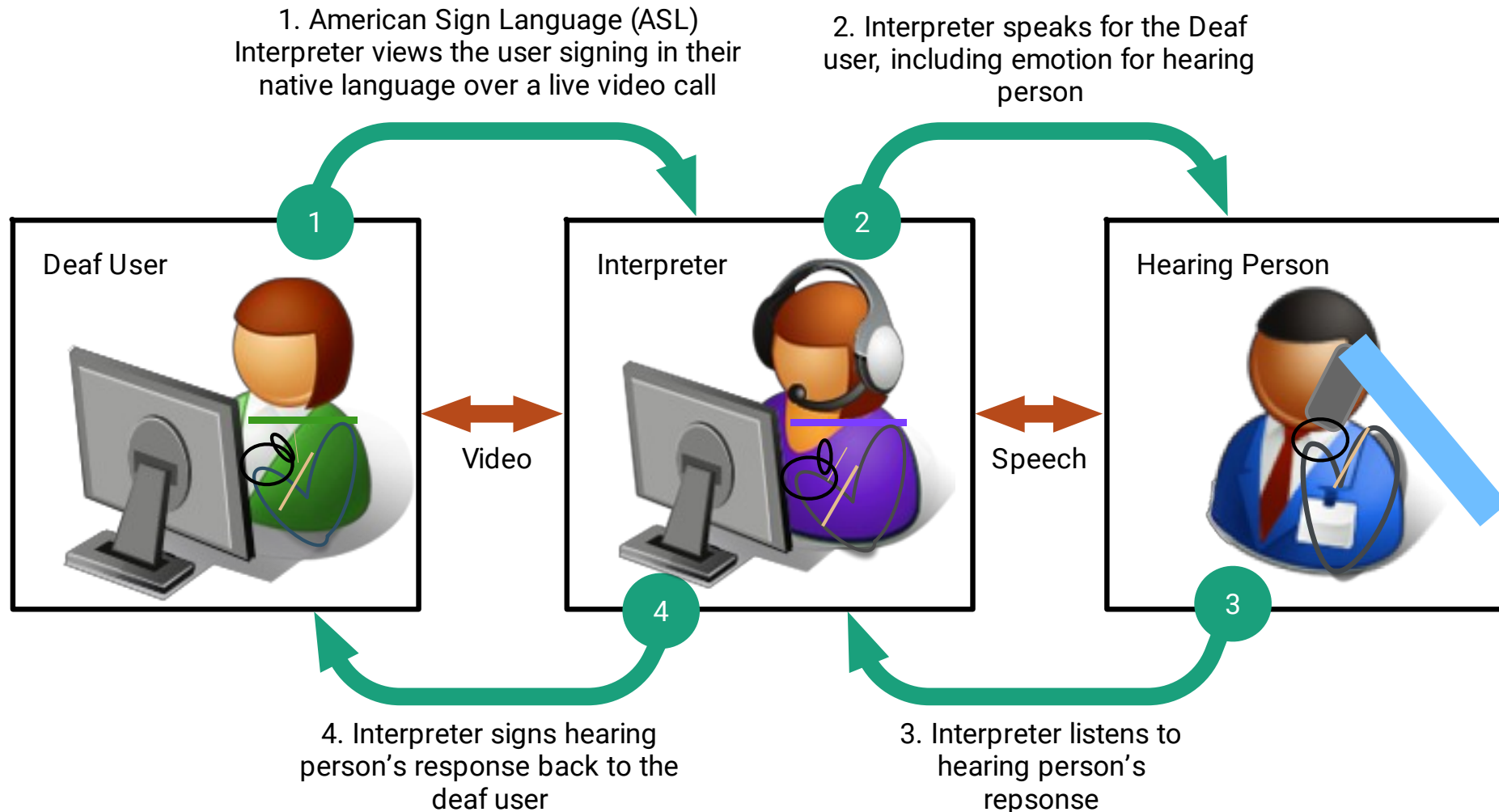


RUM – a bit of history and background

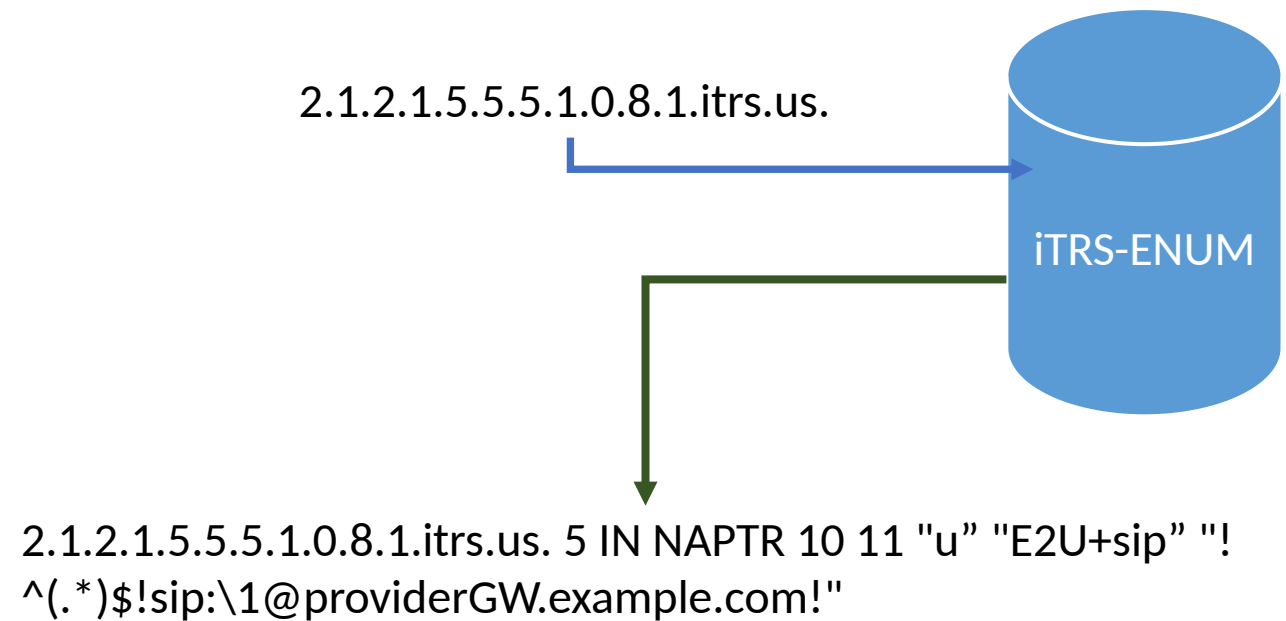
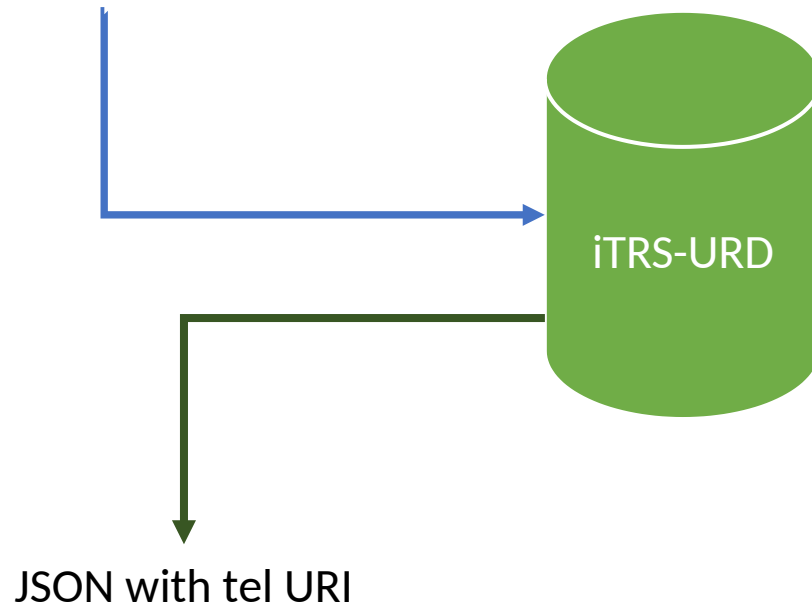
Henning Schulzrinne
Columbia University

What is a Video Relay Service Call?



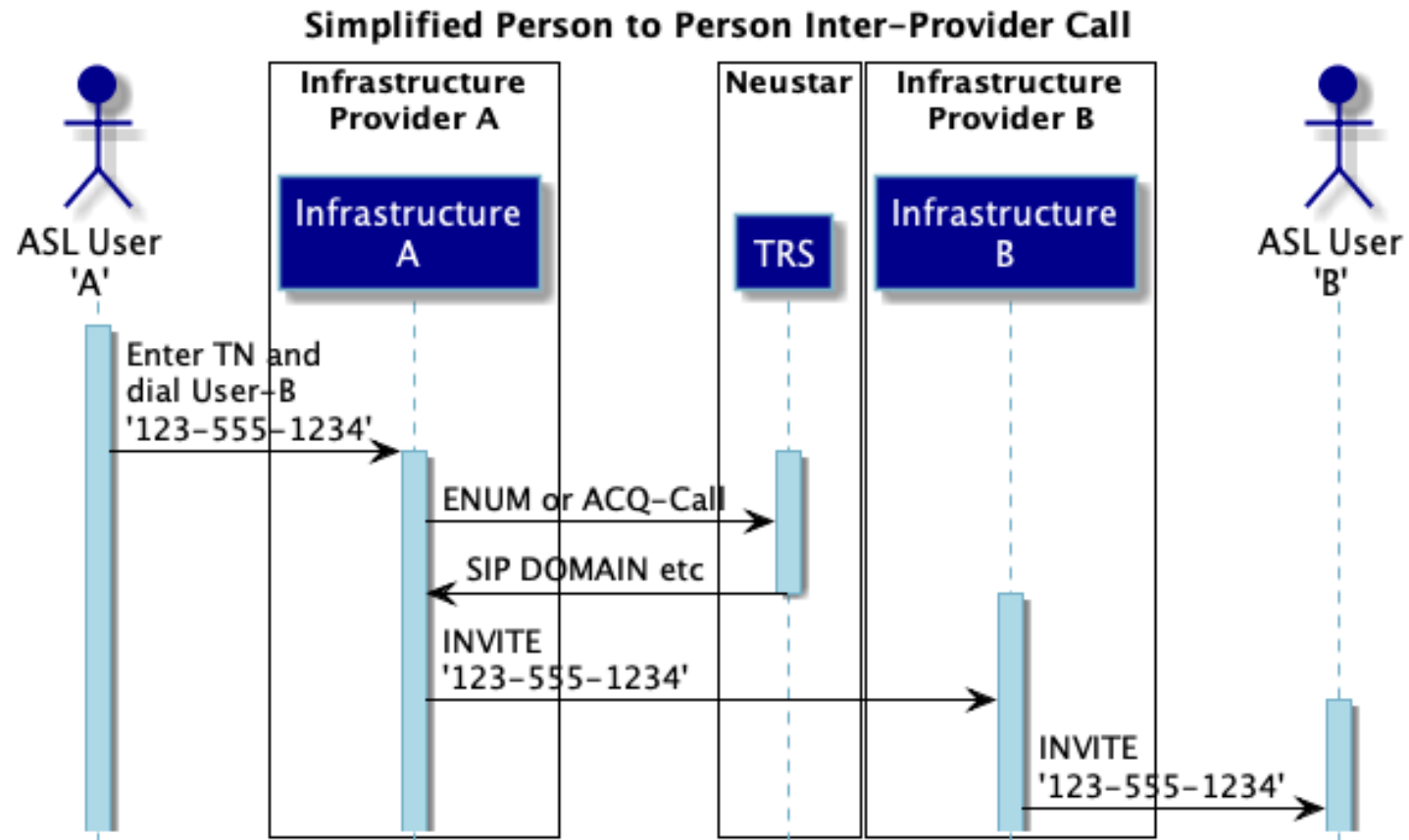
Databases

https://itrs-location.neustar.biz/itrs-urd/v1/urd/call-query/queried_tn/5714341010/other_tn/9284587966/service/IP_RELAY/direction/INBOUND



All relay services are just media combinations



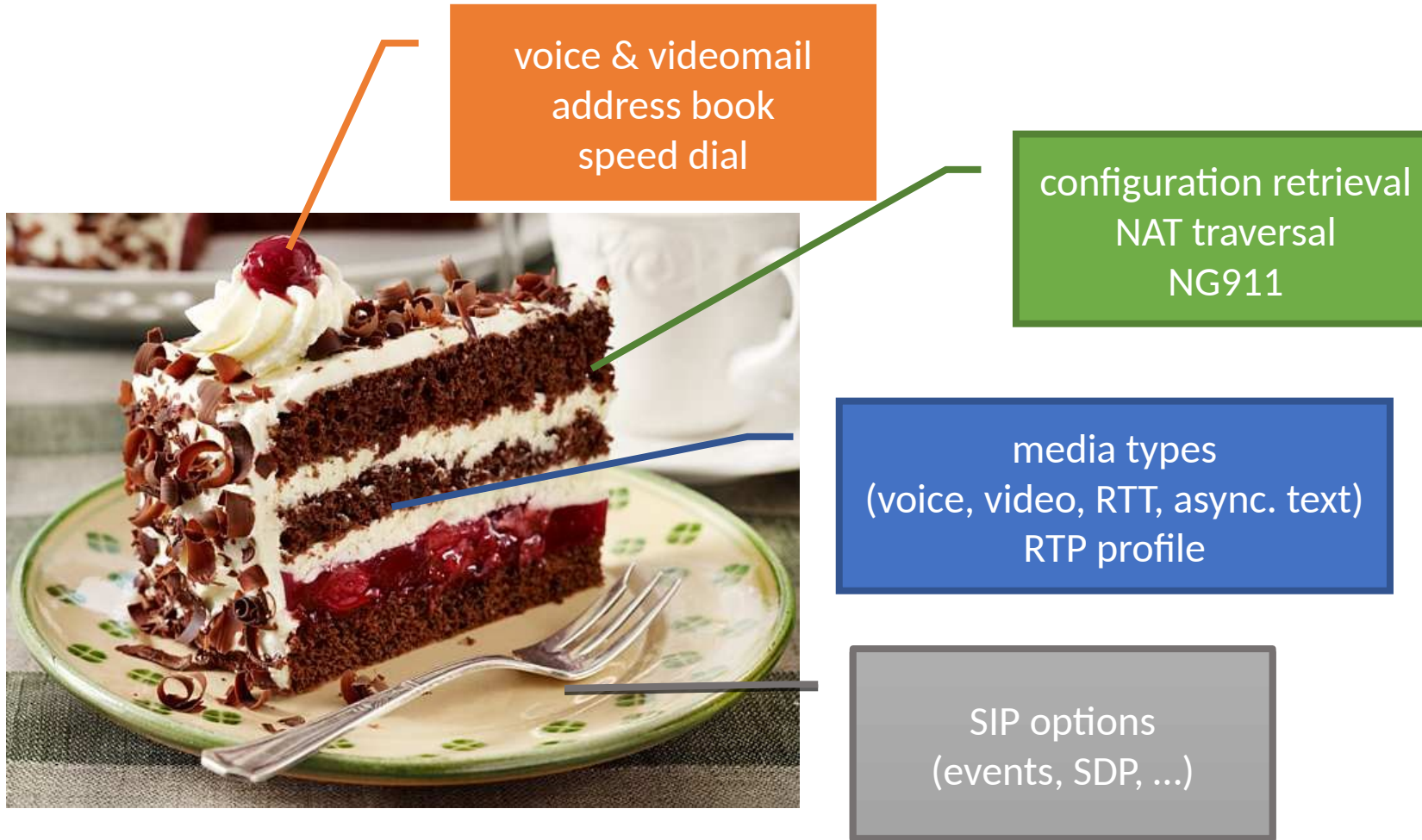


Note: Sequence diagram is indicative only. It does not reflect the full call flow.

What are the recurring problems?

- UA interoperability vs. carrier/provider interoperability
 - “device porting”
- Currently, software tied to VRS provider
 - provided for “free” (i.e., by US phone bill contributions) to VRS users
- Innovation in end user functionality
 - similar to portability of SIMs among Android and IOS devices
 - specialty functionality
- Additional platforms
- Ability of users to switch between providers on a call-by-call basis
 - for outbound calls
 - or multiple numbers for incoming calls with one device

The full interoperability cake



Incomplete timeline

- Sept. 2012: SIP Forum Video Relay Service Task Group Charter
- June 2013: FCC VRS Reform Order
- January 2015: MITRE starts work on VRS for FCC
- May 2015: VTCSecure contracted to develop VATRP
- Sept. 2015: VRS US Providers Profile TWG-1.0
- 2016: transition of VATRP and call routing effort to MITRE
- July 2016: **draft-vrs-rue-dispatch-00**
- Oct. 2018: VRS providers request halting effort
- Jan. 2019: Initial MITRE release of VATRP
- Fall 2018 - Spring 2019: NANC IVC (interoperable video calling) group
 - report pending
 - IMS model or gateway model (proprietary user agents)

From: [vrs-bounces at sipforum.org](mailto:vrs-bounces@sipforum.org) [vrs-bounces at sipforum.org] on behalf of Richard Shockey [richard at shockey.us]
Sent: Friday, September 28, 2012 6:14 PM
To: vrs at sipforum.org
Subject: [vrs] Some ideas for a charter...

SIP Forum Video Relay Service Task Group Charter

Task Group Chairs: TBD

Document Editors: TBD

Overview:

Video Relay Service [VRS] is a form of Telecommunications Relay Service that enables hearing impaired persons who use American Sign Language to communicate with hearing voice telephone users. A video communications session connects the VRS user with a VRS communications assistant [CA] , enabling the VRS user and the CA to converse with each other in signed conversation. The CA acts as an interpreter, relaying the VRS users signed conversation as a voice conversation with a voice telephony user. The video communications between the VRS user and CA may be supplemented by the user's voice (e.g., for VRS users who can speak but cannot hear) and by real-time text (e.g., to facilitate communicating otherwise cumbersome detail). The legal basis for the VRS lies in various national laws.[1] In the United States the VRS is authorized and governed by several laws including the 1996 Communications Act, as amended, The Americans for Disabilities Act [ADA] and the Twenty-First Century Communications and Video Accessibility Act [CVAA].

VRS users and telephony users to reach each other through the Video Relay Service using the standard fully expressed E.164 phone number. The system also enables deaf or hard-of-hearing users to reach each other directly at their phone numbers in order to have a signed conversation, referred to here as point-to-point calling. Number translation to a VRS CUA is handled through queries to a centralized database, in some cases based on RFC 6116 [ENUM].

Various sources have indicated that there are substantial interoperability problems with the existing VRS service[2] [3]and that that a comprehensive interoperability profile for the VRS based on SIP is now required.

Expected Outputs:

The task group will produce one or more SIP Forum recommendations that define a common set of implementation rules for VRS SIP client user agents and SIP proxy's deployed by VRS providers. These recommendations will specify which standards must be supported, provide price guidance in the areas where the standards leave multiple options and supplement functional gaps in existing protocols.

The task group will not attempt to modify or define SIP technical standards as it might relate to the VRS but will document possible requirements that may require further standardization. Should modifications or clarifications to the existing SIP standards be warranted the task group will fully document these new requirements and forward them to the IETF DISPATCH Working Group for further action.

Specific objectives of the task group are:

- First develop a comprehensive requirements document that sets forth the common network elements for the VRS service.

- Specify the basic protocols and protocols extensions that must be supported by each element in the VRS system. Specify the exact RFC or other existing standards to be used.

- Mandate specific video and audio codec's that all VRS client user agents and proxies must support for real-time communications

- Mandate specific video and audio codec's that all VRS client user agents and proxies must support for Video/Audio mail applications and potential user interfaces.

- Recommend minimum broadband connectivity requirements.

- Develop credentials for both Client User Agents and Proxy in the VRS system.

SIPForum Video Relay Service (VRS)

US VRS Provider Interoperability Profile

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

5 **1 Abstract**

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

2 Table of Contents

1 Abstract.....	1
2 Table of Contents.....	1
3 Introduction.....	3
4 Scope.....	3
5 Conventions and Terminology.....	3
5.1 Terminology from Requirements.....	3
5.2 Normative Language.....	6
6 Reference Architecture.....	6
7 Key Assumptions and Limitations of Scope.....	7
8 Use Cases.....	7
8.1 PSTN to RUE: two stage manual dial around.....	8
8.1.1 End to End Overview.....	8
8.1.2 Detail on the RS1 – RS2 leg.....	9
8.2 RUE – PSTN: two stage manual dial around.....	10
8.2.1 End to End Overview.....	10
8.2.2 Detail on the RS1 – RS2 leg.....	11
8.3 RUE to RUE Point-to-Point call between users of different providers.....	11
8.4 Video Mail.....	11
8.5 Muting (Privacy).....	12
9 Relay Service.....	13

VRS interoperability events



April 17, 2019

Ex Parte

Ms. Marlene H. Dortch
Secretary
Federal Communications Commission
445 12th Street SW
Washington, DC 20554

Re: Tenth Industry-wide VRS Interoperability Event (CG Docket Nos. 10-51 & 03-123)

Dear Ms. Dortch,

Each year, U.S. Video Relay Service (VRS) providers meet in the spring and fall to complete in-depth testing of endpoints and backend systems. The tenth industry-wide VRS interoperability event was held in Salt Lake City, Utah from April 8 through 11. Five VRS providers participated in the event: Convo, ASL Services Holdings, LLC (Global VRS), Purple Communications, Sorenson VRS, and ZVRS. In addition, nWise participated as a technology provider for Global VRS. MITRE, a Federal Communications Commission contractor, also attended the conference.

On the first day of the event, the group held a round table discussion on a new revision to the U.S. VRS Provider Interoperability Profile, or SIP Profile, planning the documentation for and implementation of enhancements including encryption, 911 geo-location, and other service improvements. The providers also discussed support for STIR/SHAKEN to address caller ID spoofing by robocallers.

Over the four-day event, the VRS providers and MITRE completed over 1,800 interoperability tests conducted between all participating providers using SIP. These tests included point-to-point and VRS dial-around calls between 33 different VRS endpoints, with over 50 test cases per endpoint.

Other related efforts

- Related = profiles targeting SIP system-level interoperability
 - but different use cases (carriers, mostly)
- SIPConnect 2.0 (SIP features, voice codecs)
- ATIS IP-NNI (voice only)
- DOCSIS VoIP spec (?)
- WebRTC (codecs, mostly)
- RCS (IR.65, IR.92, IR.94, ...)

2 IMS Feature Set

- 2.1 General
- 2.2 Support of Generic IMS Functions
 - 2.2.1 SIP Registration Procedures
 - 2.2.1a Addressing
 - 2.2.2 Call Establishment and Termination
 - 2.2.3 Early Media
 - 2.2.4 SIP OPTIONS
- 2.3 Supplementary Services
 - 2.3.1 General
 - 2.3.2 Communication Hold
 - 2.3.3 Ad-Hoc Multi Party Conference
- 2.4 Call Set-up Considerations for Calls with Video Media
 - 2.4.1 Integration of Resource Management and SIP
 - 2.4.2 Video Media Considerations
 - 2.4.3 SIP Precondition Considerations

3 IMS Media

- 3.1 General
- 3.2 Voice Media
- 3.3 Video Media
 - 3.3.1 Video Codec
 - 3.3.2 RTP Profile and Data Transport
 - 3.3.3 RTCP Usage
 - 3.3.4 RTP Payload Format Considerations for Video