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Video Access Technology Reference Platform (VATRP) Release Documentation

Version 1.0

January 2, 2019

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Record of Changes

Version	Date	Author / Owner	Description of Change
1.0	January 2, 2019	CAMH	Additional information corresponding to v1.0 release, including an updated test plan. Version 1.0 for publication by Sponsor.

Executive Summary

The Federal Communications Commission (FCC) Telecommunications Relay Service (TRS) Center of Expertise (COE) Project promotes the Commission’s goal to foster innovations that advance functionally equivalent telecommunications. Toward that end, the project ensures that the TRS employs improved technology for persons who are d/Deaf, hard of hearing, deaf-blind, and/or have speech disabilities. In this document, “d/Deaf” describes individuals who are deaf in the audiological sense as well as those who identify as culturally Deaf.

The FCC has embraced a research-based approach to achieve this goal by engaging the Centers for Medicare & Medicaid Services (CMS) Alliance to Modernize Healthcare (CAMH) Federally Funded Research and Development Center (FFRDC), operated by The MITRE Corporation (MITRE), to conduct independent engineering assessments that promote and demonstrate TRS’s functional equivalence. CAMH, sponsored by CMS and all divisions of the Department of Health and Human Services (HHS), is the first FFRDC dedicated to strengthening the nation’s healthcare system. MITRE, an objective not-for-profit organization, operates CAMH in partnership with CMS and all HHS agencies to implement innovative ideas to solve our nation’s toughest health problems.

CAMH is independently assessing voice telephone services, video access services, and Internet Protocol (IP)-based captioning technology; improvements to TRS efficiency; solutions for direct communication between people with communication disabilities and other telephone users; and the effectiveness, efficiency, and consumer response to current and future approaches for delivering TRS.

At the FCC’s request, CAMH developed a Video Access Technology Reference Platform (VATRP) in support of the FCC’s Accessible Communications for Everyone (ACE) program. This platform was developed in accordance with the Relay User Equipment (RUE) Specification to serve as a standards-based test platform for interoperability. Table ES-1 presents the new VATRP features for this release.

Table ES-1. New VATRP Features

Version	Release Date	New Feature or Capability
0.1	December 3, 2018	<ul style="list-style-type: none"> • One stage dial-around • SIP over TLS encryption • Changed application logo from “ACE” to “VATRP” • Registration patches for Proxy server and configurable port
1.0	January 2, 2019	<ul style="list-style-type: none"> • Adaptive Rate Control • Patch for DTLS support and for calls placed using an unsupported encryption • Additional test cases in the test plan

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1. Introduction

The Federal Communications Commission (FCC) Telecommunications Relay Service (TRS) Center of Expertise (COE) Project promotes the Commission's goal to foster innovations that advance functionally equivalent telecommunications. Toward that end, the project ensures that the TRS project employs improved technology for persons who are d/Deaf, hard of hearing, deaf-blind, and/or have speech disabilities. In this document, "d/Deaf" describes individuals who are deaf in the audiological sense as well as those who identify as culturally Deaf.

The FCC has embraced a research-based approach to achieve this goal by engaging the CMS Alliance to Modernize Healthcare (CAMH) Federally Funded Research and Development Center (FFRDC), sponsored by the Centers for Medicare & Medicaid Services (CMS) and all divisions of the Department of Health and Human Services (HHS). CAMH is the first FFRDC dedicated to strengthening the nation's healthcare system. The MITRE Corporation (MITRE), an objective not-for-profit organization, operates CAMH in partnership with CMS and all HHS agencies to implement innovative ideas to solve our nation's toughest health problems.

1.1 Background

As part of the Accessible Communications for Everyone (ACE) program, CAMH independently assesses voice telephone services, video access services, and Internet Protocol (IP)-based captioning technology; improvements to TRS efficiency; solutions for direct communication between people with communication disabilities and other telephone users; and the effectiveness, efficiency, and consumer response to current and future approaches for delivering TRS. At the FCC's request, CAMH developed a Video Access Technology Reference Platform (VATRP) in support of the ACE program. This platform was developed in accordance with the Relay User Equipment (RUE) Specification to serve as a standards-based test platform for interoperability.

1.2 Purpose and Scope

This document presents an overview of the VATRP's release history, features, installation and user guides, and test cases.

2. Release Notes

2.1 Release History

Table 1 describes the version history of the VATRP. The v0.0.75 Preview release coincided with finalizing the RUE Specification. The v0.0.76 Patch was released in preparation for the Video Relay Services (VRS) Session Initiation Protocol (SIP) Interoperability Virtual Conference, which was held November 12–16, 2018. The v0.1 Release Candidate precedes the January v1.0 release.

Table 1. VATRP Version History

Version	Release Date	Enhancements / Features Introduced
0.0.75 Preview	October 11, 2018	<ul style="list-style-type: none"> • VATRP Preview released to VRS Providers and the FCC • Used for testing of registration • Enabled use of anonymous calls • Enabled Content Data Network (CDN) Endpoint configuration through User Interface (UI) • Enabled xCard Endpoint configuration through UI
0.0.76 Patch	November 8, 2018	<ul style="list-style-type: none"> • VATRP Preview patch for registration issues • Includes support for using a configuration file for registration • Additional features for testing include calling, mute, privacy, and Real-Time Text (RTT) • User ID added as registration input parameter • Enabled Call-Info and Geolocation endpoint configuration through UI
0.1.0 Release Candidate	December 5, 2018	<ul style="list-style-type: none"> • One-stage dial-around • SIP over Transport Layer Security (TLS) encryption • Changed application logo from “ACE” to “VATRP” • Registration patches for Proxy server and configurable port
1.0 Release	January 2, 2019	<ul style="list-style-type: none"> • Adaptive Rate Control • Patch for DTLS support and for calls placed using an unsupported encryption • Additional test cases in the test plan

2.2 Known Issues

Table 2 describes the known issues that are open as of this release.

Table 2. VATRP Known Issues

No.	Description	Status
1	Audio and video codec settings do not persist between sessions.	Open
2	Selected media encryption not displayed in settings.	Open
3	The Contact header is incorrect in the INVITE.	Open
4	VATRP is unable to connect to some Providers' videomail servers.	Open
5	User needs to add leading 1 to 10-d number for logging in/dialing.	Open
6	Call history is not saved when logging into a different device.	Open
7	The timestamp for call duration in call history is not correct.	Open
8	The "Provider" field box is blank when adding a new contact.	Open
9	There are several feedback forms that will freeze the application when the user clicks on "Send."	Open
10	Some of the text to be displayed on the settings summary tab is cut off when clicking "View TSS."	Open
11	The Resources view under the More option will present "None Found."	Open
12	VATRP does not properly handle registration to an outbound proxy.	Open
13	Auth-ID and username are not displayed correctly if loaded in from a config.	Open
14	SIP Encryption checkbox in General settings is disabled.	Open
15	Media encryption checkbox is not configurable from login window and does not persist between sessions.	Open
16	Some call metrics in the Info icon view are not accurately reported.	Open
17	vCard is presented as an option for the local file import/export for xCard.	Open
18	xCard server import requires user to manually enter credentials, is not optional, and does not try to retrieve credentials from the configuration.	Open
19	VATRP freezes after xCard server import.	Open
20	Dialpad disappears after holding down a key for DTMF transmit for more than 2 seconds; for an 8-second transmit nothing is sent.	Open
21	RTT uses rtpmap for h264 instead of t140.	Open
22	TCP and TLS port are not configurable.	Open

2.3 Resolved Issues

Table 3 describes the issues that have been resolved since the last release.

Table 3. VATRP Resolved Issues

No.	Description
1	STUN server settings entry cannot be null.
2	FQDN check for server does not allow ".us" extension.
3	VATRP login is missing fields for Proxy server and account.
4	VATRP does not allow configuration file upload for registration.

No.	Description
5	Legacy non-RTT messaging box needs to be removed.
6	Call Info header is missing in the INVITE.
7	Wrong h264 library causing application crash.
8	Registration with proxy server in configuration file has incorrect URI.
9	Crash caused when encrypted call is placed to an endpoint that does not support that encryption.
10	SRTP media encryption uses SDES instead of DTLS.
11	If the Message Waiting Indicator (MWI) endpoint is set from the settings menu located on the login page, the application will crash.
12	Missing DLLs when installing for the first time.

3. Installation Guide

This section provides a quick installation guide with an installer file and instructions for building and installing from the VATRP source code. In both installations, the VATRP must be installed on a Windows 10 platform.

The minimum requirements for VATRP installation consist of:

- Windows 10.X Operating System (OS)
- Webcam
- Microphone
- Microsoft Visual C++ 2013 Redistributable
- Microsoft Visual Studio Community 2015 Update 3 (for building from source code)

3.1 Quick Installation

Download the VATRP.msi file on a Windows 10 machine. Open the file, using Windows Installer as the default application. The VATRP Setup Wizard will appear as shown in Figure 1.

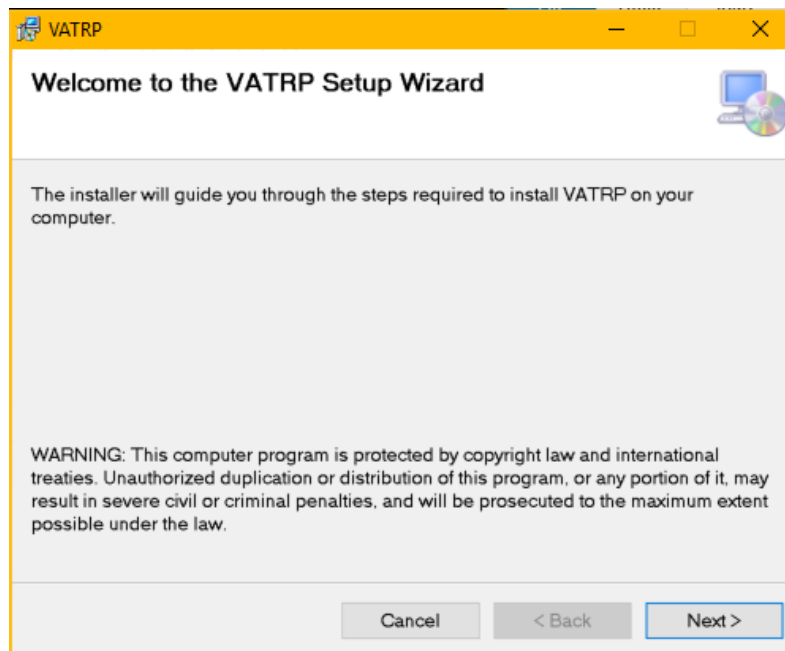


Figure 1. Screenshot of VATRP Setup Wizard

Click “Next” and then click through the prompts to install the VATRP. The VATRP Setup Wizard can be closed after successful installation. The VATRP can be opened via the shortcut in the start menu or by running ACE.exe from the directory chosen during the installation.

3.2 Installation from Source Code

Building from the source code requires Windows 10, Microsoft Visual Studio 2015 Update 3, and Git Bash or a Git client for Windows.

3.2.1 Installing Microsoft Visual Studio

When installing the Microsoft Visual Studio IDE, navigate to the “Older Versions” sections of the download page and then continue with Visual Studio Community 2015 Update 3.

Downloading the IDE may require creation of a Microsoft 360 account. During the installation process, select a custom install.

On the following page, check the box that selects all the available installation options and then de-select the C++ package. The installation may take several hours to complete.

The VATRP has been provisioned with an installer project that simplifies the process of installing the endpoint client device. Execute the following steps to install Microsoft Visual Studio:

1. Run Visual Studio as Administrator and select the “Extensions and Updates” option from the Tools dropdown.
2. Search 'installer' while in the online section.
3. Download and install Microsoft Visual Studio 2015 Installer Projects.
4. Close the running instance of Visual Studio and run the VSI_bundle.exe that was just downloaded.

3.2.2 Building the VATRP

Once Visual Studio has been installed and this repository has been cloned, take the following steps to build the VATRP:

1. Open the VATRP.sln file within the IDE.
2. Open the solution folder and right-click the VATRP solution.
3. Clean and rebuild the application before running it for the first time.

By default, the VATRP App project should already be selected as the default project to run. The navigation bar at the top of Visual Studio contains a green “Play” arrow icon that, when clicked, will build and run the application in debug mode. To run the application outside this mode, press Ctrl F5 as a short cut.

3.2.2.1 Reference Errors

If there are any issues regarding an assembly reference to Windows Devices or the IAsync type when building the project solution for the first time, then complete the following:

4. Navigate to the “Uninstall a Program” window from the Start menu.
5. Select “Microsoft Visual Studio Community 2015” and click “Change.”
6. Install the Windows 10 SDK.

7. Open the VATRP.App project in Visual Studio.
8. Find the References item under the VATRP.App Project.
9. Right-click and select “Add reference”.
10. Select the option to Browse.
11. In the Browse popup, select the following file: C:\Program Files (x86)\Windows Kits\10\UnionMetadata\Windows.md or Windows.winmd.
12. Make sure the Windows.md or Windows.winmd reference file is checked off, then click “OK”.
13. Clean and rebuild the application. Any reference errors should be resolved.

4. User Guide

The following subsections describe how to register the VATRP to a SIP server and interact with its features.

4.1 Registration

When launching the VATRP, a login screen appears as shown in Figure 2. The user can choose to provide the relevant information in the UI text fields or upload a JSON configuration file that populates the fields found in the JSON file.

If the Provider is not listed under the dropdown, select “Custom” and enter the fully qualified domain name (FQDN) in the Server field. The username and the password should be entered as well.

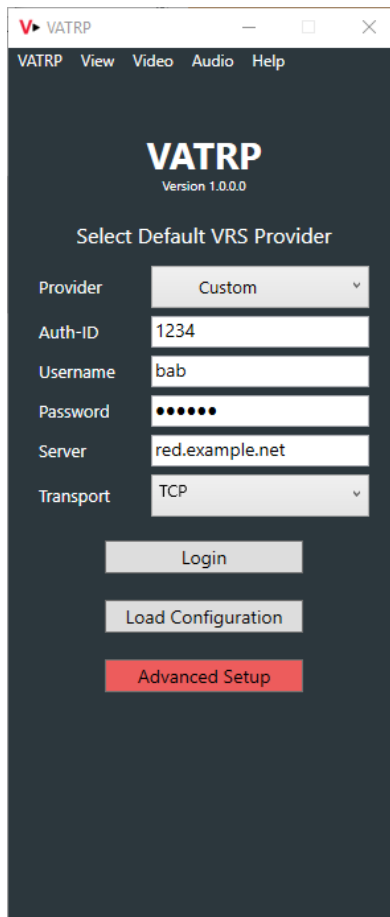
The screenshot shows the VATRP login interface within a web browser window. The window title is 'VATRP'. The interface has a dark blue background with white text. At the top, there is a navigation bar with 'VATRP', 'View', 'Video', 'Audio', and 'Help'. Below this, the 'VATRP' logo and 'Version 1.0.0.0' are displayed. The main section is titled 'Select Default VRS Provider'. It contains several input fields: 'Provider' (a dropdown menu currently showing 'Custom'), 'Auth-ID' (text field with '1234'), 'Username' (text field with 'bab'), 'Password' (password field with six dots), 'Server' (text field with 'red.example.net'), and 'Transport' (a dropdown menu currently showing 'TCP'). Below these fields are three buttons: 'Login' (grey), 'Load Configuration' (grey), and 'Advanced Setup' (red).

Figure 2. Screenshot of VATRP Login Screen

Selecting the “Advanced Setup” button can open the Settings window. In the Settings window, navigate to the Account tab. Several fields can be modified in this window, including the CDN Uniform Resource Identifier (URI), which can import the list of Providers.

To populate the “Provider” dropdown menu, go to the CDN URI field and place a URI that references a JSON document containing a list of Providers. If a configuration file was loaded, the Settings window should reflect the loaded content. Many of the fields in the Settings menu will remain unalterable and only activate once the login process has been completed.

After populating all the desired fields in the login screen, click “Login” to proceed to the VATRP main window depicted in Figure 3.

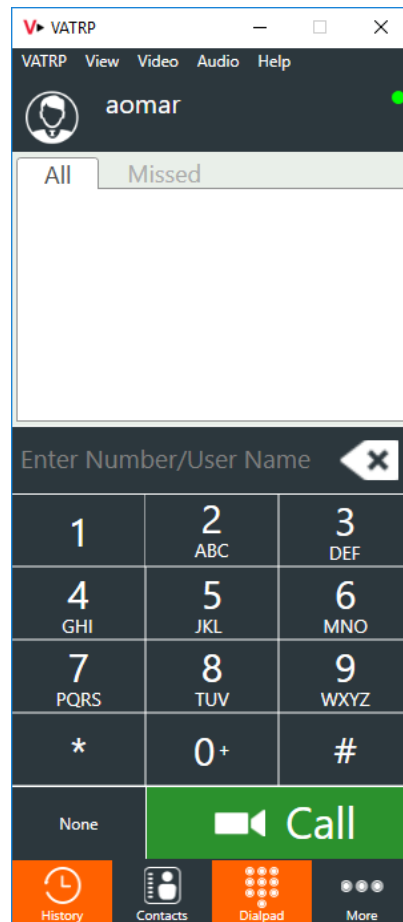


Figure 3. Screenshot of VATRP Main Window

The state of the registration attempt can be determined by the circular status indicator at the top right of the application. Table 4 lists the possible states. Note that even in a failed registration, although the main window will be reached, the user will not be able to make or receive calls.

Table 4. Registration Statuses

Color	Status
Green	Successful
Red	Failed
Gray	In Progress

4.2 Placing a Call

To place a call, select the dialpad tool from the navigation menu at the bottom of the application's display, enter a contact's phone number, and click "Call." If the callee answers, an on-screen menu will appear allowing for use of the VATRP's call features. Figure 4 presents a view of the VATRP when in a call. The following features are displayed in the grayed out overlay buttons from the viewer's left to right:

- Video Privacy
- Microphone Mute
- Audio Mute
- Dual-Tone Multi-Frequency (DTMF) Dialpad
- Real-Time Text
- Pause Call

An "End Call" button appears at the bottom of the video window (and below the overlay buttons).

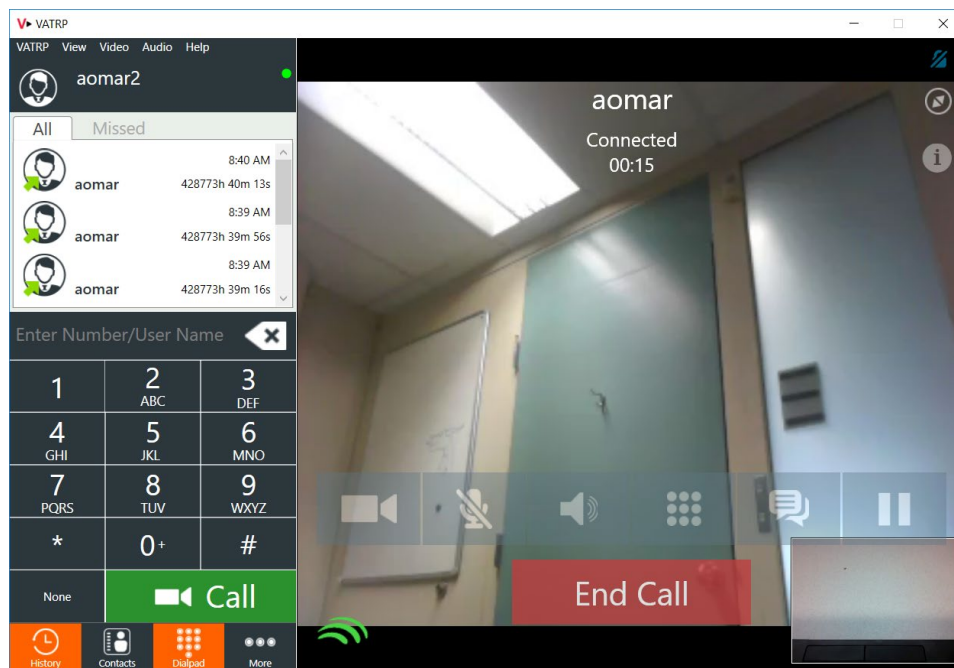


Figure 4. Screenshot of Call View

4.3 Features

4.3.1 Real-Time Text

When a user has selected the RTT feature from the on-screen menu bar, a new dialog box attaches to the right side of the video stream view as shown in Figure 5. As a message is typed, a

character-by-character stream is sent to the other user. When the ENTER key or SEND button has been clicked, the message remains in the dialog view.

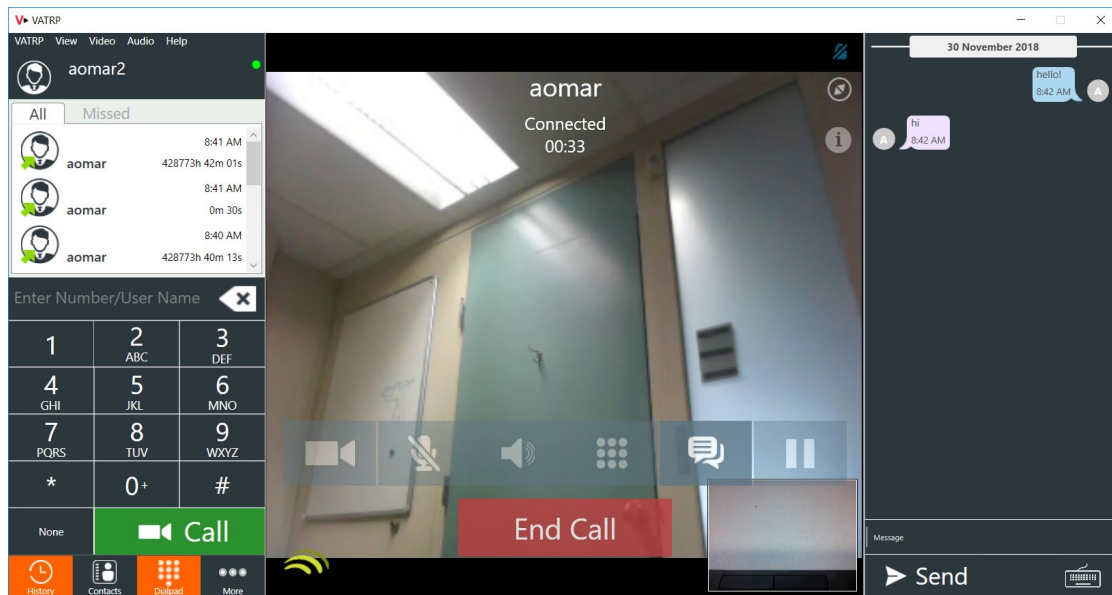


Figure 5. Screenshot of Call View with Chat

To mute RTT, click the keyboard icon on the bottom right side of the RTT dialog. This hides the message bar and prevents the user from sending additional messages or text characters until RTT is unmuted.

4.3.2 Contacts

To view the current list of saved contacts that can be used to place calls, select the “Contacts” option in the bottom pane of the main window. To manually add a new contact, select the “+” icon near the top right of the contacts window. This will open a new window where the new contact’s information can be inputted and saved. For a contact with a 10-digit number, the phone number must be entered without spaces, dashes, or any leading characters such as “+1”.

Contacts can be imported by selecting the down-arrow icon near the top right of the contacts window. To import contacts, one of two options must be selected:

14. **Import from a remote URI to the local machine through a HTTPS web request.** This URI can be set through the Settings window under the Account tab.
15. **Import from the local file system.** Either a vCard file or xCard (XML) formatted list of contacts can be selected.

If there are contacts present in the VATRTP account, they can be exported like the contact import options described. This feature can be used by selecting the up-arrow icon near the top right of the contacts window.

4.3.3 Videomail

The VATRP has a control flow in place for handling the selection of videomail from the user's settings menu tab. When this selection is made, the VATRP attempts to start a SIP session with the videomail server specified in the application's account settings tab. If the user's videomail box contains media that has not been viewed, a status indicator appears in the settings menu tab, alerting the user to the total number of their outstanding messages. Viewing the unviewed messages removes the MWI mailbox notification. This feature is in development.

5. VATRP Test Plan

This section contains test cases to measure interoperability with the VATRP and assess RUE Specification compliance. While all tests have user-based verification, some also involve analyzing packet captures or REST calls. All GUI-related actions and results are based on the VATRP as User 1. For any testing that involves a non-VATRP endpoint as User 2, the tester should find the appropriate place on the endpoint's GUI to complete the test case.

For each Provider, test calls will be placed between the VATRP registered to that Provider's server and the other VRS endpoints. The results will be documented in two matrices—a Requirements Traceability Matrix for tests involving multiple calls and a Requirements Traceability Matrix for tests conducted once per Provider. Figure 6 and Figure 7 show examples of these matrices.

There will be a unique matrix for Figure 6 corresponding to each of the VRS Providers, namely, Convo, Global, Purple, Sorenson, and ZVRS. The matrix in Figure 7 will be used for all Providers. Each set of requirements in Figure 6 will be tested against all Provider endpoints over each of the Provider server environments, unless designated as "Not Tested." Each set of requirements in Figure 7 will be tested once per Provider on the VATRP registered to the designated Provider.

VATRP TRACEABILITY MATRIX: Tests Involving Multiple Calls		<div><div><div><div><div><div></div><div>FEDERAL COMMUNICATIONS COMMISSION</div><div>USA</div></div></div><div><div></div><div>FC</div><div></div></div></div></div></div>																																Convo					Global				Purple							Sorenson						ZVRS									
		Provider Endpoint																																																															
		Convo Softphone (iOS)	Convo Softphone (Android)	Convo Softphone (Mac/OSX)	Convo Softphone (PC/ Windows 10)	Convo VATRP Softphone (PC/Windows 10)	Global Softphone (PC/Windows 10)	Global Softphone (iOS)	Global Softphone (Android)	Global VATRP Softphone (PC/Windows 10)	Purple Smart VP (Desktop)	Purple P70 (Desktop)	Purple P3 Softphone (iOS)	Purple P3 Softphone (Android)	Purple P3 Softphone (Mac/OSX)	Purple P3 Softphone (PC/Windows 10)	Purple VATRP Softphone (PC/Windows 10)	Sorenson nTouch VP (Desktop)	Sorenson nTouch VP2 (Desktop)	Sorenson nTouch Softphone (iOS)	Sorenson nTouch Softphone (Android)	Sorenson nTouch Softphone (Mac/OSX)	Sorenson nTouch Softphone (PC/Windows 10)	Sorenson VATRP Softphone (PC/Windows 10)	ZVRS Z20 - Cisco E20 (Desktop)	ZVRS Z5 Softphone (iOS)	ZVRS Z5 Softphone (Android)	ZVRS i3	ZVRS Z5 Softphone (Mac/OSX)	ZVRS Z5 Softphone (PC/Windows 10)	ZVRS Z70 - Cisco (Desktop)	ZVRS VATRP Softphone (PC/Windows 10)																																	
VATRP Test Case for Provider: _____		1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31																																	
5.3.1 - Call Quality (Video - Effective ASL)	1																																																																
5.3.1 - Call Quality (Audio - Comprehension)	2																																																																
5.3.2 - Media Encryption	3																																																																
5.3.3 - Video Privacy	4																																																																
5.3.3 - Audio Mute	5																																																																
5.3.4 - DTMF Transfer*	6																																																																
5.3.5 - Real-Time Text (RTT)*	7																																																																
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5.7.1 - One-stage Dial-Around	11																																																																

Test Case Legend:

*: Required if the Provider supports the functionality on at least one endpoint
All other test cases are required.

Results Legend:


Passed RUE Test Requirement

Passed with caveats - Video or Audio Quality Issues (or other specified issue)

Failed RUE Test Requirement

Not Tested

Figure 6. VATRP Traceability Matrix for Tests Involving Multiple Calls

VATRP TRACEABILITY MATRIX: Tests Conducted Once Per Provider						
		Provider				
VATRP Test Case		Convo	Global	Purple	Sorenson	ZVRS
5.1.1 - Registration	1					
5.1.2 - SIP over TLS Encryption	2					
5.2.1 - Geolocation Transfer	3					
5.2.2 - Contact Info Transfer	4					
5.5.2 - Message Waiting Indicator (MWI)*	5					
5.6.1 - Import/Export Contact List via xCard file format	6					
5.6.2 - Import/Export Contact List via CardDAV Server*	7					
5.8 - Emergency Call Handling	8					

Test Case Legend:

*: Required if the Provider supports the functionality on at least one endpoint
 All other test cases are required.

Results Legend:





 Passed RUE Test Requirement
 Passed with caveats - Video or Audio Quality Issues (or other specified issue)
 Failed RUE Test Requirement
 Not Tested

Figure 7. VATRP Traceability Matrix for Tests Conducted Once Per Provider

Many of the tests require the user to be in a call scenario; therefore, Table 5 presents the typical call procedure.

Table 5. Point-to-Point Call Procedure

Step	Action	Expected Result
1	User 1 navigates to call screen.	User 1 should see input box for phone number or SIP URI.
2	User 1 inputs phone number of User 2 in the input box and clicks a button to place a call.	User 1 should see or hear ring-back and visual feedback describing connection status information.
3	User 2 sees visual and/or audio notification of an incoming call and clicks button to receive the call.	User 1 and User 2 should see connection status information associated with the call connection.
4	Both users interact for 2 minutes.	Video and audio quality should be acceptable for duration of call.
5	User 1 hangs up.	Verify that the call terminates correctly from both ends.

The following subsections describe test cases for each of the outlined VATRP Test Areas.

5.1 Registration

This subsection contains test cases for registering and using TLS encryption.

5.1.1 Registration Test

The Provider has a choice of registering with the fields in the login screen or by uploading a configuration file. Table 6 presents a procedure for testing registration by either method. The registration test must be performed once for each Provider on the VATRP only.

Table 6. Registration Test

Step	Action	Expected Result
1	User 1 opens the VATRP application.	The VATRP login screen appears.
2	(For configuration file. If using UI only, skip to Step 4.) User 1 clicks "Load Configuration."	A window pops up that allows the user to select a file.
3	User 1 selects a properly formatted JSON configuration file and clicks "Open."	The file selection window closes. For fields that are present in the configuration file and the UI, their values are populated in the UI.

Step	Action	Expected Result
4	User 1 enters information in any necessary login fields that were not populated by the configuration file. Then the user clicks "Login."	The VATRP main window appears, and the login status circle changes from gray to green.
5	(Ongoing) Periodically check the status circle during other tests.	The login status circle should remain green to indicate that User 1 is still logged in.

5.1.2 SIP over TLS Encryption

Table 7 presents the test case for SIP over TLS encryption. This test must be performed once per Provider on the VATRP only.

Table 7. SIP over TLS Encryption Test

Step	Action	Expected Result
1	User 1 starts a Wireshark capture.	Wireshark captures the packet flow for the VATRP endpoint registering to the Provider server.
2	User 1 completes registration as described in the registration test case, choosing TLS for Transport.	Registration takes User 1 to the VATRP main window and is successful.
3	User 1 waits until the registration status indicator turns green and then ends the Wireshark capture.	In the Wireshark capture, the last (most recent) "TLS SERVER HELLO" message before the "REGISTER" message should show version 1.2 or higher, with Cipher suite TLS_RSA_WITH_AES_256_CBC_SHA256 or TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384.

5.2 Geolocation and Contact Information

The geolocation and contact information test is conducted once per Provider on the VATRP only. Table 8 describes the test case.

Table 8. Geolocation and Contact Information Test

Step	Action	Expected Result
1	User 1 opens the VATRP and navigates to "Advanced Setup", then to "Account", to enter a Geolocation URI. User 1 then clicks "Close".	The Geolocation URI is saved in the Advanced Setup menu and User 1 returns to the login screen.
2	User 1 starts a Wireshark capture.	Wireshark captures the packet flow for the VATRP endpoint registering to the Provider server.
3	User 1 completes registration as described in the registration test case.	Registration takes User 1 to the VATRP main window and is successful.

Step	Action	Expected Result
4	User 1 waits until the registration status indicator turns green, and then ends the Wireshark capture.	In the Wireshark capture, the last (most recent) the "REGISTER" message should contain a Geolocation header with the Geolocation URI provided in the Advanced Setup menu. In addition, the message should contain a Contact header with the correct contact information for User 1.

5.3 Current Call Features

The current call feature tests include tests for call quality, media encryption, audio mute, video privacy, DTMF, RTT, and call on hold. The following subsections provide a description of these test procedures.

5.3.1 Call Quality

The call quality tests include subjective measures of video and audio quality as outlined in Table 9 and Table 10. The tests must be conducted on all endpoints.

Table 9. Subjective Video Quality Test Criteria

Criterion	Video Quality Criteria
1	Visible: The tester reports whether the other endpoint sees the incoming video (e.g., not a black or green screen).
2	Conversational: The tester reports whether video quality is acceptable for an ASL conversation.
3	Blurry / Pixelated: The tester reports any instances where the video is blurry, blocky, or pixelated at any point during the call.
4	Stutter / Stalls: The tester reports any video stutter or stalls, and the maximum delay effect observed.
5	Frame Rate: The tester reports the frame rate.

Table 10. Subjective Audio Quality Test Criteria

Criterion	Audio Quality Criteria
1	Audible: The tester reports whether the other endpoint hears the incoming audio.
2	Clear: The tester reports any noticeable distortions to the audio or echo.
3	Timing: The tester reports any significant latency to the audio stream and whether the audio appears in sync with the video.
4	Dropouts: The tester reports any dropouts in the audio.
5	Background noise: The tester reports whether background noise (loud intermittent, or white noise) negatively affects the ability to hear the other person clearly.

5.3.2 Media Encryption

The test case for SRTP media encryption is in development.

5.3.3 Audio Mute and Video Privacy

Table 11 describes the test cases for audio mute and video privacy. Mute and privacy tests shall be conducted on all endpoints.

Table 11. Audio Mute and Video Privacy Test

Step	Action	Expected Result
1	User 1 initiates a call using the Point-to-Point Call Procedure.	Call connects. Both User 1 and User 2 can communicate visually and by voice.
2	User 1 hits the "Privacy" button.	User 2's device indicates that User 1 is in privacy mode. User 2 cannot see User 1's video stream but can still hear User 1's audio stream.
3	User 1 turns off the Privacy option.	Call resumes normally with two-way video.
4	User 1 hits the "Mute" button.	User 2 can no longer hear audio from User 1's device but can still see User 1's video stream. The User 2 device may or may not indicate that the remote audio is muted.
5	User 1 turns off the Mute option.	Both parties are again able to communicate verbally.
6	Switch sides and try again. (User 2 initiates privacy and mute options.)	Results should be the same as those in Steps 2 through 5, with the user switched.

5.3.4 Dual-Tone Multi-Frequency

Table 12 presents the test case for DTMF. The DTMF test results for VATRP compliance are only measured in VATRP to VATRP tests. Additional DTMF tests with other VRS endpoints are conducted as part of a general interoperability survey but will not impact compliance.

Table 12. DTMF Test

Step	Action	Expected Result
1	User 1 starts a packet capture in Wireshark and initiates a call using the Point-to-Point Call Procedure.	Call connects. Both User 1 and User 2 can communicate visually and by voice.
2	User 1 clicks the DTMF dialpad button.	A dialpad appears on the screen.
3	User 1 presses 0-9, *, and # on the dialpad for short transmissions. Then User 1 presses 0-9, *, and # on the dialpad for long transmissions.	Audio may be transmitted from User 1 to User 2 but is not required for the DTMF telephony event. There may be no noticeable change for the user.

Step	Action	Expected Result
4	Switch sides and try again. (User 2 sends DTMF.)	Results should be the same as those in Steps 2 and 3, with the user switched.
5	User 1 ends the Wireshark capture and navigates to Statistics, and then chooses Flow Graph.	The DTMF tones appear in the flow graph as “RTP (telephony event) DTMF” followed by the number that was pressed.

5.3.5 Real-Time Text

Table 13 contains the test cases for RTT and RTT mute. The RTT tests are conducted on all endpoints that support RTT.

Table 13. RTT Test

Step	Action	Expected Result
1	User 1 initiates a call using the Point-to-Point Call Procedure.	Call connects. Both User 1 and User 2 can communicate visually and by voice.
2	User 1 clicks the chat icon.	A RTT chat tab appears next to the video.
3	User 1 types a message but does not click “Send.”	User 2 can see the message coming in, character by character, with less than a 1-second delay.
4	User 1 and User 2 type messages at the same time. (Note: at the end of this step, User 2 may stop typing.)	Each user sees the message from the other user coming in, character by character, with less than a 1-second delay.
5	User 1 hits “Enter” and starts to type a new message.	User 2 sees the first message fixed on the screen with no more changes, and a second message window with the new message coming in character by character, with less than a 1-second delay.
6	User 1 clicks the keyboard icon to initiate RTT mute.	User 1 cannot type anything new. User 2 does not see any incoming RTT from User 1.
7	User 1 clicks the keyboard icon to unmute RTT, and then begins typing a message.	User 2 sees the incoming message from User 1, character by character, with less than a 1-second delay.
8	Switch sides and try again. (User 2 repeats RTT test.)	Results should be the same as those in Steps 2 through 7, with the user switched.

5.3.6 Pause Call

Table 14 presents the pause call test. This test is conducted on all endpoints.

Table 14. Pause Call Test

Step	Action	Expected Result
1	User 1 initiates a call using the Point-to-Point Call Procedure.	Call connects. Both User 1 and User 2 can communicate visually and by voice.
2	Both users interact for 1 minute.	Video and audio quality should be acceptable for duration of call.
3	User 1 clicks the pause button to pause the call for 30 seconds and then resumes the call.	While paused, User 1 sees the pause sign in the video window and hears music. User 2 sees a frozen video of the last frame and receives no audio. When the call resumes, video and audio resume as in a typical call scenario.
4	User 2 pauses the call for 30 seconds and then resumes the call.	Video and audio should cease while paused, then continue when the call resumes.

5.4 Additional Call Features

5.4.1 Multiple Registered RUEs

The test case for multiple registered RUEs involves two VATRP endpoints and one additional endpoint registered to the same account that receives a call from another VATRP endpoint. Table 15 describes this test case. This test is conducted on all endpoints that support multiple registrations.

Table 15. Multiple Registered RUEs Test

Step	Action	Expected Result
1	User 1 registers on the VATRP to a unique account. Users 2 and 3, on the VATRP, and User 4, on another endpoint, all register to the same account.	Users 2, 3, and 4 can successfully register and remain registered.
2	User 1, on the VATRP, places a call to the phone number shared by Users 2, 3, and 4.	Users 2, 3, and 4 all receive audio and/or visual notification of an incoming call.
3	User 2 answers the call.	The call is established correctly between User 1 and User 2. User 3 and User 4's endpoints stop ringing.
4	Repeat steps 2 and 3 twice, with the remaining Users (3 and 4) taking turns answering the call.	The same expected behavior should occur but with a different endpoint answering.
5	Users 2 through 4 initiate a call to User 1.	The call is established correctly between Users 2 through 4 and User 1.
6	User 4 de-registers. Repeat steps 2 through 5, with only Users 2 and 3 registered.	The expected behavior of steps 2 through 5 should occur for Users 2 and 3.

Step	Action	Expected Result
7	User 3 de-registers. User 1 calls User 2.	The call is established correctly between User 1 and User 2.
8	User 2 calls User 1.	The call is established correctly between User 1 and User 2.

5.4.2 Anonymous Calls

Table 16 describes the test case for anonymous calls. The anonymous call tests are conducted with the VATRP placing outbound anonymous calls to all endpoints. The non-VATRP endpoints will not place outbound anonymous calls for this test case.

Table 16. Anonymous Call Test

Step	Action	Expected Result
1	User 1 enables anonymity. (In the VATRP, navigate to “Settings,” then “General,” and then check the “Privacy” box.)	User 1 should see that Privacy is enabled in the Settings menu.
2	User 1 starts a Wireshark capture of the call. User 1 inputs phone number of User 2 in the input box and clicks a button to place a call.	User 1 should see or hear ring-back or visual feedback describing connection status information.
3	User 2 receives the call.	User 2 sees visual and/or audio notification of an incoming call, with “Anonymous” displayed in place of the caller ID or phone number.
4	Switch sides and try again. (User 2 repeats the anonymous call test.)	Results should be the same as those in Steps 1 through 3, with the user switched.
5	User 1 ends the Wireshark capture and finds the SIP INVITE message.	In the header of the SIP INVITE, the “From” header field should have ‘ “Anonymous” sip:anonymous@anonymous.invalid ’ before the “;tag=...”. User 1’s ID and telephone number should not appear in any header field in the SIP INVITE.

5.5 Message Waiting Indicator

Videomail can be initiated through three different methods on the callee side: timeout, call rejection, and user offline. Table 17 presents the test case for leaving a videomail through these methods and then tracking the MWI throughout viewing the messages. MWI tests are conducted on the VATRP only for Providers where at least one client supports a videomail notification.

Table 17. Message Waiting Indicator Test

Step	Action	Expected Result
1	User 1 navigates to call screen.	User 1 sees input box for phone number or URL.
2	User 1 inputs the phone number of User 2 and clicks a button to place a call.	User 1 sees or hears ring-back or visual feedback describing connection status information.
3	(Timeout) User 2 sees visual and/or audio notification of an incoming call and does not answer. User 2 lets the call ring until it times out.	User 1 leaves a videomail message by following the videomail Interactive Voice & Video Response (IVVR) presented to them. User 2 sees an MWI count of one.
4	(Reject) User 2 sees visual and/or audio notification of an incoming call and rejects the call.	User 1 leaves a videomail message by following the videomail IVVR presented.
5	(Offline) Step 2 is conducted but with User 2 offline.	User 1 leaves a videomail message by following the videomail IVVR presented. User 2 sees an MWI count of two.
6	User 2 sees the message waiting indicator on the screen of the endpoint for videomail.	User 2 receives the indication for videomail. User 2 sees an MWI count of three.
7	User 2 views the first videomail.	User 2 records observations regarding status connecting to mail server for retrieving videomail, including delay, date/time, connection status, video/audio quality, protocol information (as needed), and time-stamps for specific information to estimate delay.
6	User 2 returns to the call view.	The MWI is still present for the remaining two messages in the inbox.
7	User 2 views the second videomail.	The MWI is still present for the remaining one message in the inbox.
8	User 2 views the third videomail.	The MWI is no longer present.
9	Repeat Steps 5 and 6 one more time. Between two and four hours later, User 2 accesses the client to check for a MWI.	The MWI is still present with one message in the inbox.

5.6 Contact List Management

5.6.1 xCard

Table 18 presents the test procedure for xCard. This test is conducted once per Provider on the VATRP only. The User 2 mentioned in this test case must be another VATRP endpoint registered to the same Provider.

Table 18. xCard Test

Step	Action	Expected Result
1	User 1 navigates to Settings and then to Account and provides a Contacts URI that will upload an XML file containing User 2's contact information.	The contacts URI is successfully loaded in (the user can click "Close").
2	User 1 navigates to the Contacts tab and clicks the down-arrow icon for import.	User 1 sees a pop-up window with import options.
3	User 1 clicks "Yes" to import from the server.	User 1 sees the imported contacts in the contacts list.
4	User 1 places a call to one of the imported contacts, which is User 2.	User 1's call to User 2 is successful.
5	User 1 navigates to the contact view and clicks the up-arrow icon for export.	User 1 sees a pop-up window with export options.
6	User 1 clicks "Yes" to export to the server.	The contacts are successfully exported, which can be verified by checking the XML file. All contacts in the imported file are correctly represented in the exported file.
7	User 1 opens Postman and creates a GET request for the exported contacts.	User 1 can use the response to verify that the contact export was successful.

5.6.2 CardDAV

The test case for CardDAV is in development.

5.7 One-Stage Dial-Around

The one-stage dial-around test case is in development.

5.8 Emergency Calls

The emergency calls test case is in development.

Acronyms

ACE	Accessible Communications for Everyone
ASL	American Sign Language
CA	Communication Assistant
CAMH	CMS Alliance to Modernize Healthcare
CDN	Content Data Network
CMS	Centers for Medicare & Medicaid Services
DLL	Dynamic-Link Library
FCC	Federal Communications Commission
FFRDC	Federally Funded Research and Development Center
FQDN	Fully Qualified Domain Name
HHS	Department of Health and Human Services
HTTPS	HyperText Transport Protocol Secure
IDE	Integrated Development Environment
IP	Internet Protocol
IVVR	Interactive Video & Voice Response
JSON	JavaScript Object Notation
MWI	Message Waiting Indicator
NAT	Network Address Translation
OS	Operating System
REST	Representational State Transfer
RFC	Request for Comment
RUE	Relay User Equipment
RTT	Real-Time Text
SDK	Software Development Kit
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRTP	Secure Real-Time Transport Protocol
STUN	Session Traversal Utilities for NAT
TCP	Transmission Control Protocol

UI	User Interface
URI	Uniform Resource Identifier
URL	Universal Resource Locator
VATRP	Video Access Technology Reference Platform
VRS	Video Relay Service
XML	Extensible Markup Language

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