

My Military Health

Patient Troubleshooting Guide – Virtual Visits, Scheduled and On-Demand

14 NOVEMBER 2024

Version 1.0

Table of Contents

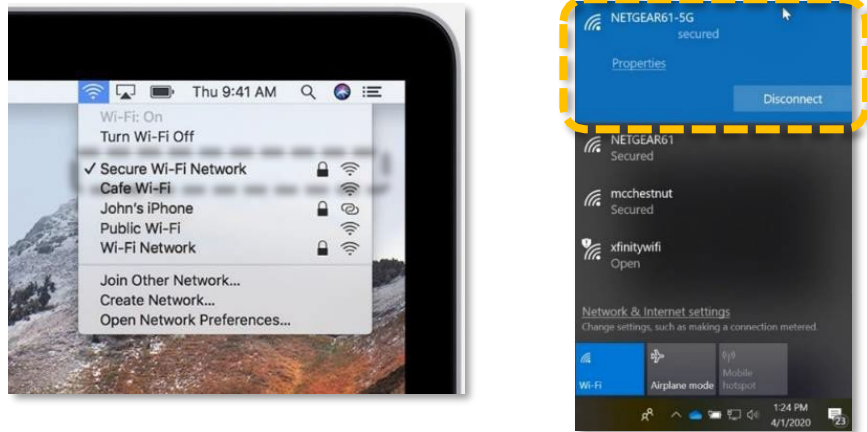
Purpose	2
Connection Issues	2
Device Issues	3
Updating your Internet Browser & Clearing your Cache and Browser History	4
Camera and Audio Issues	5
Browser Permissions	5
Wrong Camera, Microphone, or Speaker Device Pulled into Visit.....	5
In Video Technical Experience	5
Black Screen	5
Overloaded Connection	6
High Latency	6
High Packet Loss.....	6
High Jitter.....	7
In Video Behavioral Experience	7
Location	7
Devices	7
Network Indicator	8
Other Methods of Communication	8
Reporting Issues.....	9
In-Visit	9
Post-Visit.....	9
Virtual Visits Audio and Video Troubleshooting Tips	10
Glossary of Common Video Terms	11

Purpose

The My Military Health Patient Troubleshooting Guide (hereafter referred to as the User Guide) is a reference document to provide support to patients on the access, use, and troubleshooting of Virtual Visits. This document will help users resolve common issues while navigating a Virtual Visit.

Connection Issues

- If you encounter an issue launching a visit, first confirm that you are connected to the internet. If using a Mac, click the internet icon in the toolbar located on the top right of your screen. If using a PC, click the internet icon located on the toolbar in the bottom right of your screen. Click on the icon to confirm you are connected to your Wi-Fi.



- Test your internet connection speed. Use <https://www.speedtest.net> or something equivalent, to see what your download, upload, and ping values are.
 - The minimum bandwidth required for video conferencing that's effective is 8 Mbps for downloading and 1.5 Mbps for uploading. For higher quality video you will want higher bandwidth for both.
 - Ping Speed (or latency) is the time it takes for a small data set to be transmitted from your device to a server on the internet and back to your device again. You want your latency to be below 150 ms.
 - If your internet speed doesn't meet the requirements above or comes close to the minimum requirements, try moving closer to your router and make sure to limit any extraneous internet use that can consume bandwidth. Make sure nothing is streaming or being downloaded/uploaded to your Wi-Fi during your call.

- Make sure you are also using the preferred browsers - Chrome, Safari, Edge, Mobile Chrome (<https://www.twilio.com/docs/voice/client/javascript#supported-browsers>)
- If the steps above do not resolve your connection issues, try restarting your device.
- Twilio provides a website that tests low level connectivity. Run the tests at <https://networktest.twilio.com> and share any failures with your Internet Service Provider (ISP).

Device Issues

- Check browser permissions. You may need to give permission to your browser to enable the camera/mic/speaker. This can be done through settings for that browser or each time you open the video application.
- If you have multiple input devices (camera/mic/speaker), make sure that you have selected the one you want as your default prior to joining. Some video applications will allow you to change devices once in the room.
 - If you are having issues, try changing devices to see if that helps
 - Still having issues, try refreshing the page
- Make sure volume is turned up, microphone is unmuted, and camera is enabled.
- Video WebRTC uses significant system memory - if your hardware has less than 4 GB of memory, close any other apps or browser tabs to free up as much memory as possible. Twilio recommends at least 2 GB be available for the Video app.
- Some examples of mobile devices with sufficient memory (not an exhaustive list):
 - iPhone 11+
 - iPad/iPad Air (2019+)
 - iPad Pro (2017+)
 - Google Pixel 1+
 - Samsung S8+

Updating your Internet Browser & Clearing your Cache and Browser History

Confirm you are using the latest version of your internet browser. It is also a best practice to clear your cache and browsing history to help resolve technical issues. Please follow the steps below.



Updating your Browser

Chrome:

1. On your computer, open Chrome.
2. At the top right, click More [...] (3 vertical dots).
3. Click Update Google Chrome.
4. If you do not see this button, you are already on the latest version.
5. Click Relaunch.

Safari:

1. On your computer, go to Apple Menu.
2. Select Software Update.
3. If there are updates available, click Update Now.

Microsoft Edge:

1. On your computer, open Microsoft Edge.
2. At the top right, click More [...] (3 vertical dots).
3. Select Settings.
4. Click About Microsoft Edge
5. If the page does not show an update option, you are already on the latest version. If an update is available, select download and install and follow the prompts.



Clearing Cache & Browser History

Chrome:

1. On your computer, open Chrome.
2. At the top right, click More (3 vertical dots).
3. Click More tools and select Clear Browsing Data.
4. At the top, choose the time range.
5. To delete everything, select All Time.
6. Check the boxes next to Cookies and Other Site Data and Cached Images and Files.
7. Click Clear Data.

Safari:

1. On your computer, go to Settings.
2. Select Safari.
3. Click Clear History and Website Data.

Microsoft Edge:

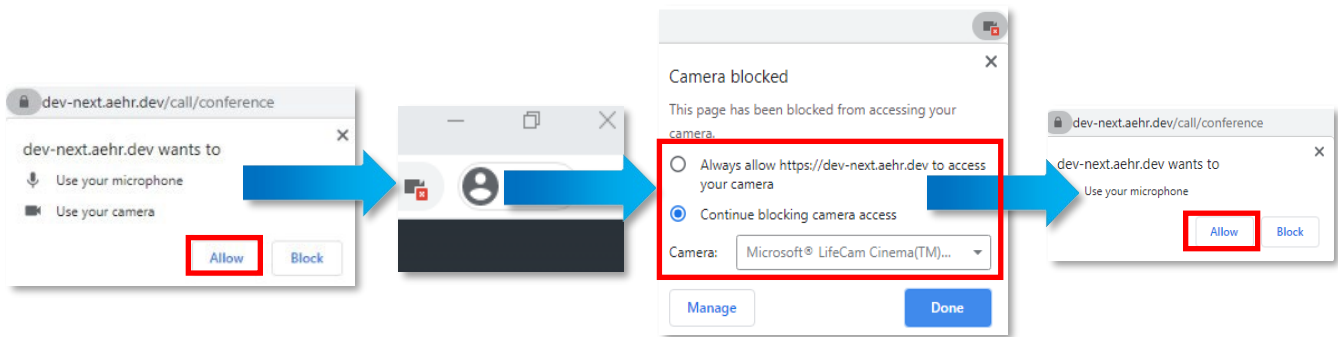
1. On your computer, open Microsoft Edge.
2. At the top right, click More [...] (3 vertical dots).
3. Select Settings.
4. Click Privacy, search, and services.
5. Below Clear Browsing Data, click Choose what to Clear
6. Click Clear now.

Camera and Audio Issues

Browser Permissions

As the visit opens click “Allow” to grant your browser permission to access your camera and microphone. If you do not grant your browser permission, you will not be able to see or hear your provider and will receive the Browser Permission error.

To resolve Browser Permission error, click the camera icon in the top right of your toolbar and change the setting to “Always Allow”. Reload the page. Click “Allow” to grant access to your microphone.



Wrong Camera, Microphone, or Speaker Device Pulled into Visit

To change devices, follow the steps below.

1. While in your visit, select “More”
2. Click “Device Settings”
3. Select the appropriate “camera”, “microphone”, and “speaker” for your visit.

In Video Technical Experience

Black Screen

If you are in a video conference and someone's screen goes black, they may have minimized or backgrounded the video application. This can happen if they are on a mobile device, and they received a phone call/notification/opened another application.

Overloaded Connection

If you notice the following:

- Permanent video freezing
- Choppy audio
- Dropped connections

If the connection overloaded, check your bandwidth or turn off your camera, to reduce bandwidth use.

High Latency

If you notice the following:

- Lag in audio/video playback
- Occasional video freezing
- While still hearing smooth audio

This is likely due to high latency (ping speed is high). Disconnect from a VPN if you are using one. Try using a wired connection instead of a wireless connection. Close background programs and applications. If you see this consistently over many days, you may want to upgrade your router.

High Packet Loss

If you notice the following:

- Frequent video freezing
- Choppy video
- Choppy audio

You may be experiencing high packet loss. This could be due to congestion on the network or faulty hardware. If you change your device's video to a lower resolution via system settings, that may help. Close any applications and browser tabs/windows. You can try resetting your router or restarting your computer. If this continues to persist, you may consider replacing your router or ethernet cable (if you are not on Wi-Fi). You can also talk with your internet provider for additional support.

High Jitter

If you notice the following:

- Bursts of video freezing
- Bursts of choppy audio

Your video application may be experiencing jitter. Close out applications and browser tabs/windows that you don't need. Upgrade your ethernet cable if you are not on Wi-Fi. If you are using a mobile device, make sure you are on Wi-Fi and not cellular data.

In Video Behavioral Experience

Location

- Find a location that's isolated from background noise and is well lit.
 - Background noise can interfere with audio quality
 - Being in a room that's well-lit will enable a better video experience
- It's best if you stay in one spot instead of moving from one location to another.
 - If on mobile, the connection can switch from Wi-Fi to a cell tower to another cell tower and can reduce the bandwidth available.

Devices

- Using a headset will improve audio quality
 - This can be helpful if you are hearing an echo
 - This can also reduce the background noise that the other person hears
- Mobile devices have different microphones in different locations on the phone, selecting the right one can improve both your ability to hear the other person as well as improve your voice quality.
 - Devices can be changed during the tech check, but also in visit. Try modifying the microphone device if the other person has trouble hearing you.
- If you are having difficulty communicating, there is a chat functionality that you can use to communicate your issues.

Network Indicator

- During the virtual visit, if you or the other person experiences a poor network condition, an indicator will show up next to your name in red. Since network quality can vary, this may fix itself. Otherwise, follow the suggestions in the “Preparing for your Visit” section of the Patient Experience Guide.

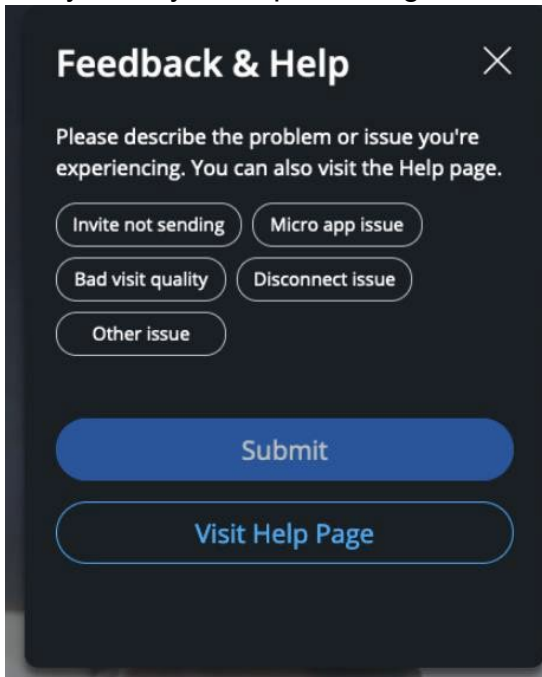
Other Methods of Communication

- If you are having difficulty with video/audio, there is a chat functionality that you can use to communicate those issues.
- If the video quality is poor, another alternative is to have the provider dial you into the visit via your phone.

Reporting Issues

In-Visit

- Click “More” on the bottom of the screen, which will open a Feedback section to report issues you may be experiencing.

A screenshot of a dark-themed mobile application dialog box titled "Feedback & Help" with a close button (X) in the top right corner. The dialog contains the text: "Please describe the problem or issue you're experiencing. You can also visit the Help page." Below this text are five buttons: "Invite not sending", "Micro app issue", "Bad visit quality", "Disconnect issue", and "Other issue". At the bottom of the dialog are two larger buttons: "Submit" and "Visit Help Page".

Post-Visit

- At the end of the visit, you will be given the option to rate the technical performance of the application: Thumbs Up or Thumbs Down. If Thumbs Down is chosen, you will be able to select from a list of options or “other” to create your own or provide additional issue details.

Virtual Visits Audio and Video Troubleshooting Tips

In Scheduled Virtual Visits, working audio and video capability are key to conducting successful video appointments. Reference this table if you are encountering technical issues during an appointment.

Microphone Issues

Cause	Resolution
Microphone not allowed on browser (user)	Look in browser upper left for permissions window. Select "Allow"
Microphone not allowed (policy block)	Submit a ticket to local IMD/IT/systems
No microphone connected	Check plug connection
Microphone mute selected (headset or standalone microphone)	Unmute microphone or headset
No feed/sound from mic	Select "Microphone", click down arrow and choose desired microphone

Camera Issues

Cause	Resolution
Camera not allowed on browser (user)	Look in browser upper left for permissions window. Select "Allow"
No camera connected	Check USB connection if USB Locally request camera
Self-image appears black	Selected camera of closed laptop Click the "Select Camera" down arrow and choose "Other Camera". If camera is closed, open it

Technical Support for MHS GENESIS and My Military Health Issues

GSC Help Desk: For MHS GENESIS-related issues (e.g., Issues with scheduling My Military Health virtual visits)	My Military Health Help Desk: For access related issues (e.g., password reset, access,)
Hours: 24/7 • Phone: 1-800-600-9332 • Website: https://gsc.health.mil	Hours: M-F:8AM – 10PM Sat & Sun: 8AM – 6PM ET Phone: (844-342-5664) Email: DHAsupport.mymilitary@health.mil

Glossary of Common Video Terms

Bandwidth	<p>This is the rate at which data can be transferred between endpoints. Usually expressed as “bits per second” (bps) or “bytes per second” (Bps), often with a “k” or “m” to show kilo or mega, respectively. 40 mbps, for example.</p> <p>Bandwidth is constrained by the most constrained network leg between the endpoints. So, while one end user may have a very high bandwidth connection, overloaded sections between users, poor Wi-Fi connections, or slow VPN systems between users can reduce usable bandwidth.</p> <p>Larger values are better.</p>
Codec	<p>Refers to software that Encodes and Decodes audio and video from their raw data streams. Codecs provide encryption and compression of the data streams to reduce bandwidth usage.</p>
H.264	<p>A video codec that is available on nearly every platform. H.264 is not as “efficient” (results in larger data streams) as VP8.</p>
Jitter	<p>Jitter is a measure of the consistency of timing within a network stream. In other words, how much does latency deviate over time. Twilio Video apps manage jitter using a jitter buffer. The jitter buffer collects incoming packets, so the next stage of the processing pipeline can have a consistent, properly ordered flow of packets. Larger buffers can smooth out larger jitter values, but larger buffers also introduce latency!</p> <p>Smaller values are better.</p>
Latency	<p>Latency is the measured time a packet of data takes to travel from one network interface endpoint to another. Small latency values (less than 100ms) may not be noticeable in a video call. Higher latencies will cause issues with audio and video synchronization - causing the speaker’s words and mouth movements to “not line up.” Smaller values are better.</p>

Media Server or Media Plane	The Media Server deals with the media information itself. Media packets typically transport encoded and encrypted audio and video bits.
NAT	Network Address Translation (NAT) is the modification of in-transit network packets to map one IP address space to another. It is most used in IP Masquerading, where a large private IP network shares a relatively small number of publicly facing IP addresses behind a router or gateway.
Opus	An audio codec that results in smaller data streams than PCMU but is not as universally available.
Packet Loss	Network data traffic is sent in “packets” -- small collections of bytes. Packet loss refers to the number of packets that do not reach the other endpoint. Audio and video media streams can tolerate a small percentage of lost packets. Up to 2% should not impact audio/video quality. High latency can exacerbate packet loss problems however, so packet loss and latency issues should be reviewed together. Smaller values are better.
PCMU	An audio codec that is available on nearly every platform. PCMU is not as “efficient” (results in larger data streams) as Opus.
Signaling Server or Signaling Plane	The Signaling Server deals with the control information. The communicating entities typically exchange signaling messages for agreeing on what’s to be communicated (e.g., audio, video, etc.) and how’s to be communicated (e.g., codecs, formats, etc.). Signaling Server tests are pass/fail.
STUN	STUN, and TURN, are IETF standard protocols for negotiating traversing NATs when establishing peer-to-peer communication sessions. A host uses Session Traversal Utilities for NAT (STUN) to discover its public IP address when it is located behind a NAT/Firewall.

TURN	<p>STUN, and TURN, are IETF standard protocols for negotiating traversing NATs when establishing peer-to-peer communication sessions. Traversal Using Relay around NAT (TURN) allows two endpoints behind firewalls to connect. It may be used with the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) and can be secured/encrypted using TLS.</p> <p>TURN connection tests are pass/fail - each variant serves a purpose; more information is below. If a TURN test fails, a user's IT Helpdesk would need to verify and open firewall ports. Detailed information here (What are the network connectivity and bandwidth requirements for the Client JS and Mobile SDKs?) and here (Programmable Video IP Addresses).</p>
------	----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

TURN UDP	<p>Connection to a TURN server, using UDP (User Datagram Protocol). UDP is faster than TCP but does not allow for the retransmission of lost packets. Twilio Programmable Video uses UDP to transfer media streams to the Media Server. A pixel lost here or there due to packet loss should not degrade video quality, compared to the speed benefits of UDP.</p>
TURN TCP	<p>Connection to a TURN server, using TCP (Transmission Control Protocol). TCP is slower than UDP but can retransmit lost packets. Twilio Programmable Video uses TCP to transfer signaling data to the Signaling Server. TCP being relatively slower than UDP shouldn't impact data transfers on the Signaling Plane, with TCP's retransmit functionality being critical.</p>
TURN TLS	<p>TURN server using TLS. TLS, or Transport Layer Security, is a widely used cryptographic protocol that ensures data security during communication over a network. The TLS protocol, like its predecessor SSL (Secure Sockets Layer), is primarily designed to enable reliable, authenticated, and secure communication between two or more computer applications. In modern browsers, connections secured with TLS are usually indicated by a lock icon next to the URL. Twilio's Client SDKs do not support TLS currently, so this test is only important to direct uses of the Twilio API.</p>
VP8	<p>A video codec that results in smaller data streams than H.264 but is not as universally available. VP8 Simulcast is a related codec option that sends multiple video streams simultaneously, at different resolutions.</p>

WebRTC	Web Real-Time Communication (WebRTC) is a collection of communications protocols and APIs originally developed by Google that enable real-time voice and video communication over peer-to-peer connections.
--------	-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------