SIP Soft Phone - Common problems

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Article Type Configuration Guide

Title SIP Soft Phone - Common problems

Summary This article contains an overview of some of the common issues experienced when deploying and utilizing the SIP Soft Phone.

Question/ Problem This article contains an overview of some of the common issues experienced when deploying and utilizing the SIP Soft Phone.

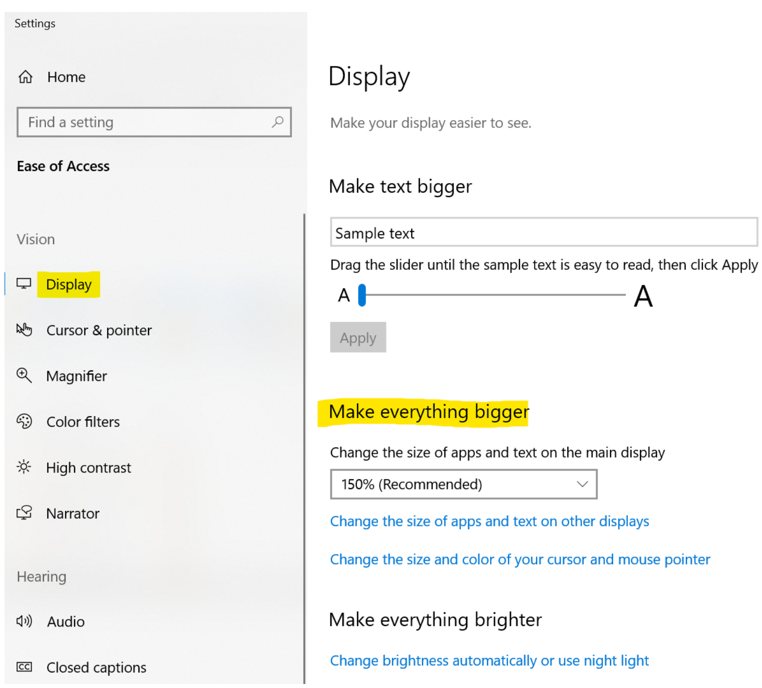
Answer/ Details

1) Unable to provision the SIP Softphone

- Verify DHCP Option 160 configuration on the DHCP server contains the record for the IC server.
- Verify through DNS that the hostname configured in DHCP resolves to the IP address of the correct CIC server.
- Ensure that the PC has access to the corporate DNS servers and is able to resolve for the IC server the short name, FQDN, and reverse IP to the FQDN
    - For more information follow this article: https://genesys.my.salesforce.com/articles/TECHNICAL_ARTICLE/Requirements-for-SIP-Soft-Phone-to-Provision/
- Verify that the Windows User account that is running the SIP Soft Phone has full access to the following registry keys:
    - HKLMSoftwarePoliciesMicrosoftSystemCertificates
    - HKLMSoftwarePoliciesMicrosoftCryptography
- Verify that the Windows User account that is running the SIP Soft Phone has full access to the following file paths:
    - %ALLUSERSPROFILE%\Application Data\Microsoft\Crypto\Keys
    - C:\ProgramData\Microsoft\Crypto\RSA
- If there is a firewall between the client workstation running the SIP Soft Phone and the CIC servers, verify that the firewall rules allow communication to the CIC servers on ports 8088 (http) and 8089 (https).

2) SIP Soft Phone crashes on startup / when acquiring DHCP Options

- The SIP Softphone crashes if the user's scaling in their display settings is over 100% on some versions.
- This is addressed in IC-152010 and fixed in 2018R1 P37, 2018R2 P35, 2018R3 P25, 2018R4 P18, 2018R5 P12, 2019R1 P6, and 2019R2+. Requires installing an updated IC User Apps version on the end-user machine.
- The workaround is to have the user turn their scaling down to 100, log out and log back in (may require a restart of the PC), and then the application should now be able to provision without crashing.

- 

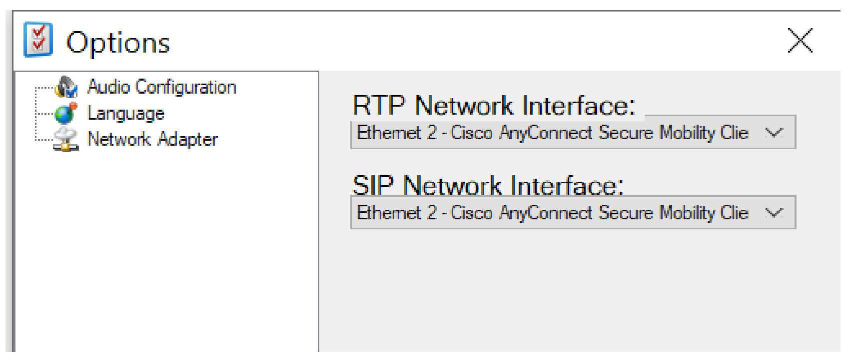
3) No audio when making an internal Station to Station call

If your users are able to call each other while in the office but when remote users call each other they do not have audio, this is normally the result of the stations Audio Path being set to Dynamic. When a Dynamic Autio Path is used, RTP will be sent directly between stations. Given that this works in the office, there is most likely a networking issue occurring with the VPN (or other remote-specific infrastructure) and the routes that need to be in place to allow these devices to communicate with each other. In the office, there are typically routes in place to allow this traffic to occur but some additional network work may be needed to enable the same routes when using a VPN to allow endpoints to transmit RTP between them.

Another option is to change the line that IC is using from Dynamic to Always-In:

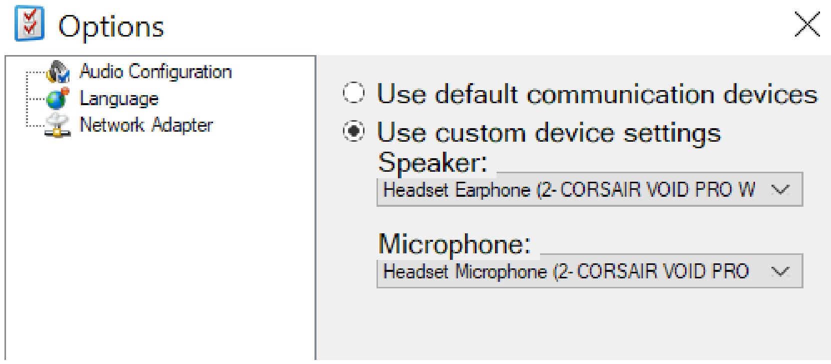
- This forces the RTP traffic through a Media Server for **all** calls using the line
- This allows you to have a defined network path for RTP traffic
    - Allow all end devices to communicate **to** and **from** the Media Server IPs
    - Forces **all** RTP through to the central office or wherever the Media servers are
    - Uses more Media Server resources
- Since the lines were dynamic, the Media servers were not servicing all of the internal calls and would only have been used to perform other functions like service inbound and outbound calls, play prompts, perform conferencing, and record calls
- For more information about the differences between Always-In and Dynamic, review this article: https://genesys.my.salesforce.com/articles/Tech_Tutorial/Differences-between-Always-In-and-Dynamic-audio

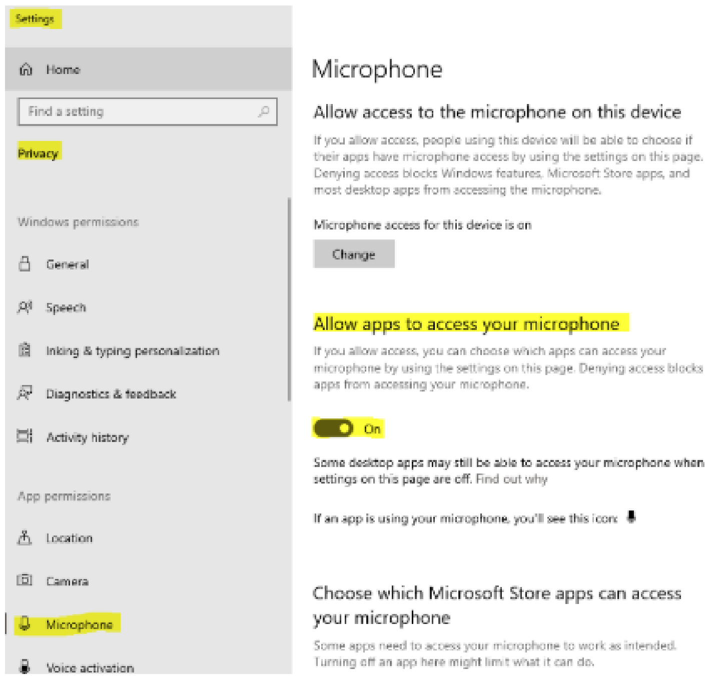
4) User clicks pickup but call doesn't connect / unable to place an outbound call

- No IP Response / There Was No IP Response From Your Station / Your Station Is Unreachable
    - This message occurs when the station was previously Registered and IC sends a SIP INVITE to the IP address that was used for registration but does not receive a response
    - After exhausting all retry attempts, the call is disconnected with this error code and is placed back in queue
- There Is No Contact Address For Your Station
    - This occurs if the station's Connection Address is configured as Dynamic and the station has not sent a REGISTER or INVITE yet
    - Check the status of the station in Interaction Administrator.
        - If it says "Not Registered" then it has provisioned but has not registered to IC
            - Provisioning occurs via HTTP/S (ports 8088 and 8089) where Registration occurs via SIP (ports 8060 and 8061)
            - Ensure that the Network Adapters are set appropriately for RTP and SIP in Options -> Network Adapter
            - 

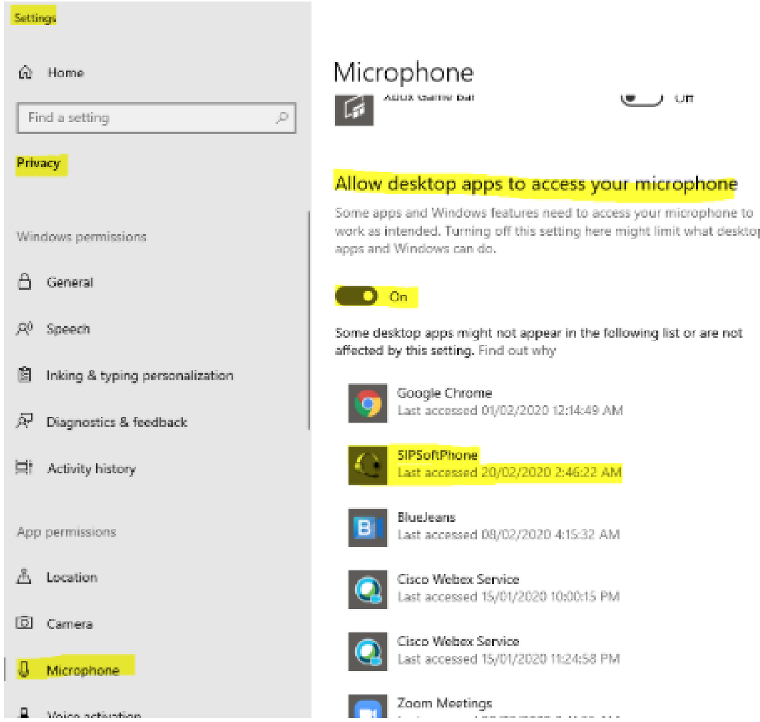
5) There Are No Lines Available To Reach Your Station / 480 SIPSoftphone audio device failure / Device call failed. There was an error dialing the call

- This error commonly occurs when the SIP Softphone does not have access to the speaker or microphone plugged into the PC
- This can occur if Windows is not detecting a speaker or microphone
- This can occur if the appropriate audio device is not set as the default audio device
    - You can got to Options -> Audio Configuration -> Use custom device settings to manually set a Speaker and a Microphone

- 

- This can also occur if SIP Soft Phone application is not granted access to the microphone or speaker.
  - To check for this go to Settings->Privacy->Microphone->Allow apps to access your microphone
    - Make sure that this is set to "ON"
    - 

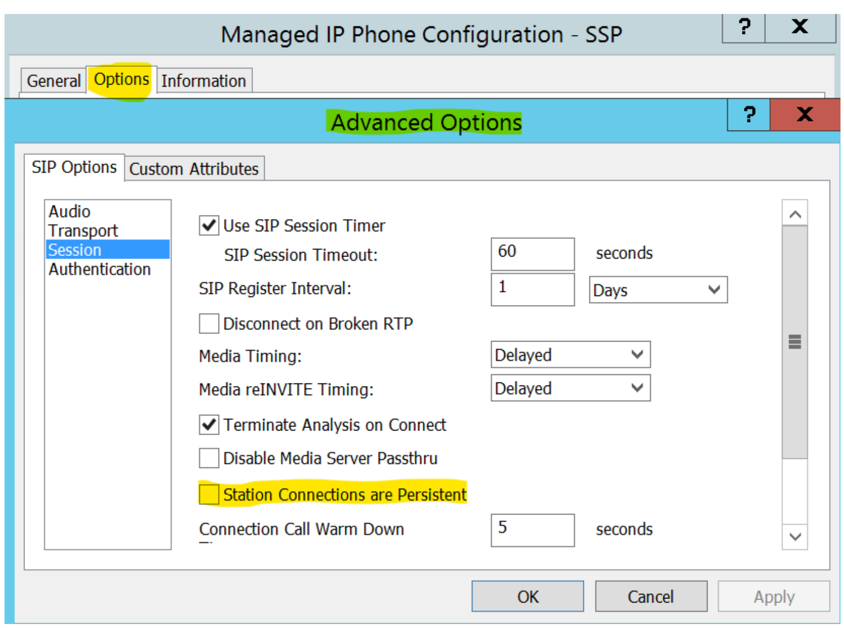
  - Scroll down on the lower part of the same microphone settings->Allow desktop apps to access your microphone
    - Make sure that this is also turned "ON". Also, make sure that the SIPSoftPhone application is listed here as well.



- This can also occur if there is something else blocking microphone or speaker access, such as an antivirus application

6) The SIP Softphone is able to take calls after provisioning, but after some time can no longer receive calls until it is re-provisioned

- A symptom of this is receiving the message "Your Station Is Unreachable" in the client
- The issue is typically with a Firewall/Layer3 device which is closing the ports when SIP Soft Phone is idle. We have seen this with many customer's using VPN.
  - Provisioning and Registration opens a connection from the SIP Softphone to the IC Server.
  - Bi-directional traffic can be limited by the TTL settings within the VPN network or the Firewall rule
  - When the TTL timer expires, traffic is not allowed from the outside (IC) to the phone (client that established the initial connection)
  - Re-provisioning the SIP Softphone "re-opens" this connection and allows bi-directional traffic until the connection timeout is reached or refreshed by further client-initiated traffic
- If this occurs, involve your Network admin to investigate the firewall configuration and if there are any Idle timeouts or any SIP Inspection enabled, as this likely needs to be disabled.
- Until the issue is resolved by your Network admin, you can enable persistent station connection call in Interaction Administrator, which will keep the station call connected all the time as a workaround. You can enable persistent connection in Interaction Administrator (`Advanced Options -> Session -> Station Connections are Persistent`)



Supplemental Information https://help.inin.com/cic/mergedprojects/wh_tr/desktop/pdfs/managed_ip_phones_ag.pdf

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