Possible problems and how to solve them

Some possible problems you may experience during the WCS server operation and recommendations to solve the issues are below. Please ask all technical questions about Web Call Server configuration and functions on the technical support forum https://forum.flashphoner.com: Technical support

1. WCS server does not start

Problem

Seems like WCS server does not start properly

Solution

1. Make sure the server process is created.
2. Check log files `/usr/local/FlashphonerWebCallServer/logs/startup.log` and `/usr/local/FlashphonerWebCallServer/logs/server_logs/flashphoner.log` for errors.
3. Make sure the name of your server host (Linux command `hostname`) correctly resolves to the IP address in `/etc/hosts` file. The `ping yourhostname` command must work, where `yourhostname` is the name of your host displayed by the `hostname` Linux command.
4. Check free disk space

2. WCS server does not accepts web client connections

Problem

Seems like WCS server is running, but does not accepts web client connections

Solution

1. Make sure you have activated the license key.
2. Check log file `/usr/local/FlashphonerWebCallServer/logs/server_logs/flashphoner.log` for errors.
3. Check the Websocket port of the server (by default `8080` for WS and `8443` for WSS) using the `telnet` command. Make sure the server uses this specific port for Websocket. To do this, you can use the 'netstat' Linux command and check the `ws_port` and `wss_port` server settings.
4. Make a traffic dump and check if the Websocket traffic is running through.

3. Registration on the SIP server does not work

Problem

Seems like WCS server is running, but registration on the SIP server does not work when trying to make a call

Solution

1. Check log file `/usr/local/FlashphonerWebCallServer/logs/server_logs/flashphoner.log` for errors.
2. Make sure the SIP port range on the WCS server (by default 30000-31000) is not blocked by Firewall, and if the WCS server is behind NAT additionally check that UDP packets sent to the external IP address reach the corresponding ports of the WCS server. Check `port_from` and `port_to` settings.
3. Make a traffic dump and check if SIP traffic is running through.

4. One-way audio during a WebRTC-SIP call or completely no audio

Problem

One-way audio during a WebRTC-SIP call or completely no audio

Solution

Configure extended logging with `client_dump_level=2` and check the logs and traffic dumps created after a test call followed by disconnect of the user. Make sure SIP, RTP and WebRTC traffic flows normally and there are no serious errors in the web browser console.

5. Lack of audio or video stream when working with streaming video and WebRTC

Problem

Lack of audio or video stream when streaming via WebRTC

Solution

1. Check the stream metrics for `VIDEO_LOST`, `AUDIO_LOST`, `NACK_COUNT` values. If they grow, consider to use lower publishing resolution/bitrate or use TCP transport
2. Collect traffic dumps. Make sure WebRTC traffic flows normally and there are no serious errors in the web browser console.

6. The server stopped responding to requests

Problem

The server stopped responding to requests. Seems like the server was shut down.

Solution

1. Check if the server process is alive.
2. If there is no server process, check crash dumps in the `/usr/local/FlashphonerWebCallServer/logs` directory of the server. A crash dump files may look like `error3677.log`, where `3677` is the PID of the server process that was shut down. Send these dumps to the Flashphoner technical support.
3. If the server process is alive while the server looks not responding and does not process connections, make a dump of system streams of the server using the jstack command. Send this dump to the Flashphoner technical support along with the server logs.