Ports used by the system

Sip signaling is running through ports 5060, 5070 5080 (TCP\UDP), 5061 and 5071 - TLS
Voice traffic is running through ports 16384-32768/udp

WebRTC, WebSocket and API connection is running through the standard protocol HTTP (80/tcp) and HTTPS (443/tcp)

There should be network connection from the telephony server through protocols HTTP (80/tcp) and HTTPS (443/tcp) for getting updates and Text-To-Speech services