# How to Forward WhatsApp User Initiated Calls to a WhatsApp Business Number to an IP-PBX

WhatsApp can now forward WhatsApp user calls made to WhatsApp business phone numbers to a SIP server. You need to have a WhatsApp Business Number with voice calling enabled via the Meta Cloud API. The SIP server must support the SIP-TLS transport protocol, SDES media encryption, and the Opus codec.

TekSIP provides the necessary protocol compatibility to forward calls from WhatsApp users to your existing SIP-based systems. Using the SIP-TLS transport protocol, incoming calls are converted to UDP and TCP transport protocols, while SDES-encrypted media streams are converted to unencrypted RTP streams. The Opus codec can also be converted to G.711 A-Law or G.711 U-Law.

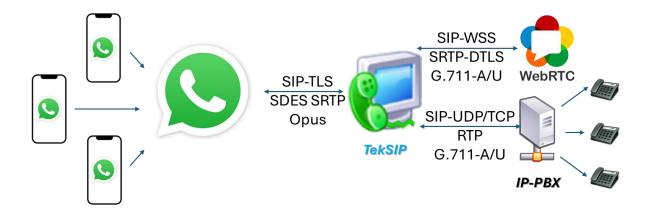


Figure - 1. Sample Topology

## **TekSIP Configuration**

You can use TekSIP in three different scenario:

#### **Simple Proxy Configuration**

TekSIP provides only SIP protocol proxy between WhatsApp and the IP-PBX. Transport interworking (*TLS to UDP or TCP*) will be automatically provided if the IP-PBX does not support TLS transport. Your IP PBX must handle NAT traversal for media streams in this type of configuration. You need to have a destination entry for the IP-PBX and a route entry for the incoming calls to the IP-PBX.

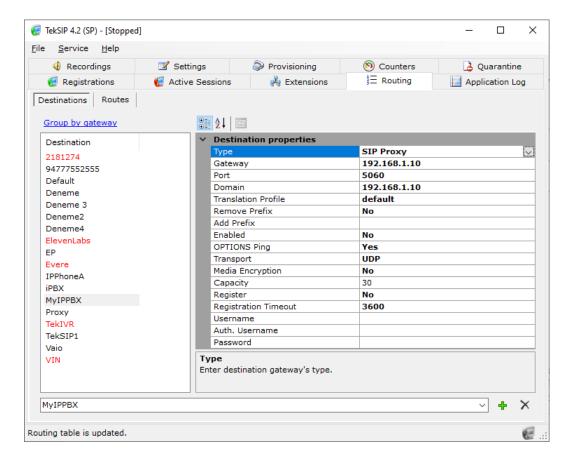


Figure - 2. Destination entry for the IP-PBX

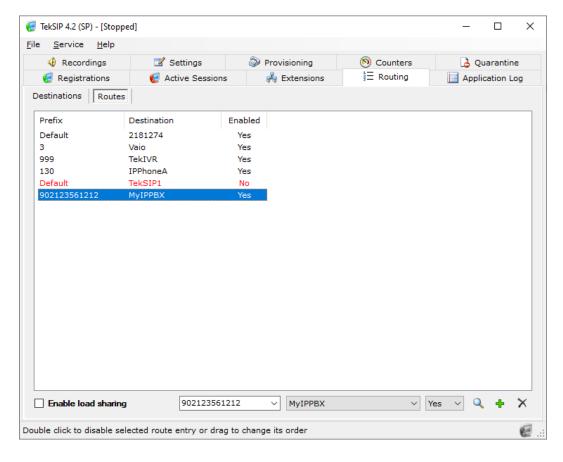


Figure - 3. Route entry for the WhatsApp Business number

You also need Internet access with a static IP address, corresponding DNS A record for this static IP address and a valid server certificate signed for the hostname specified in the A record for the TLS transport.

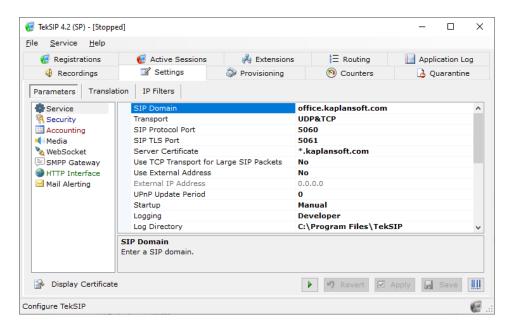


Figure - 4. TekSIP Settings

Please see instructions at <a href="https://developers.facebook.com/docs/whatsapp/cloud-api/calling/sip/">https://developers.facebook.com/docs/whatsapp/cloud-api/calling/sip/</a> how to redirect incoming calls to you WhatsApp business numbers to your TekSIP installation. You must set <a href="mailto:srtp\_key\_exchange\_protocol">srtp\_key\_exchange\_protocol</a> = SDES for this setup.

#### **SIP Proxy with Media Proxy**

You may enable media proxy if you need to handle NAT traversal in TekSIP. TekSIP media proxy will also handle SRTP-RTP interworking if your IP-PBX does not support SDES media encryption.

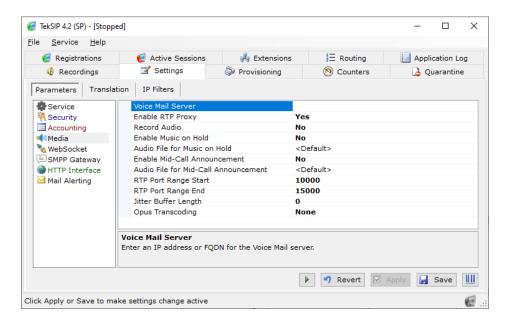


Figure - 4. TekSIP Media Settings

### SIP Proxy with Media Proxy including Opus Transcoder

You can also enable media proxy for transcoding Opus media. TekSIP can transcode incoming Opus media packets to G.711 U or A-Law packets vice versa. Please note that sampling will be reduced to 8000 Hz, 16 bits.

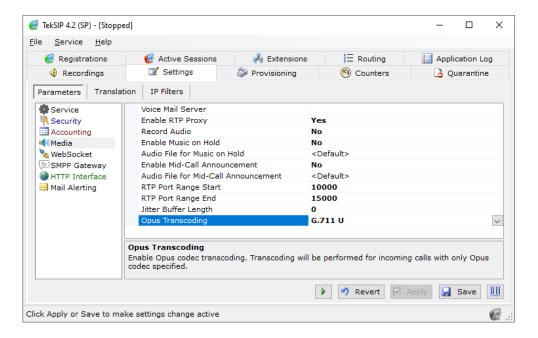


Figure - 5. TekSIP Media Settings - Opus Transcoding

You must enable STUN if your TekSIP installation is behind a NAT gateway.

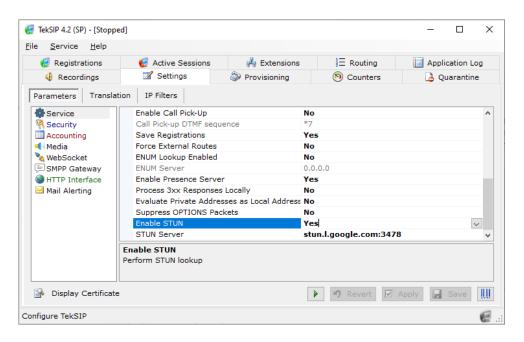


Figure - 6. TekSIP Service Settings - STUN