SIP SERVER SDK v8.0

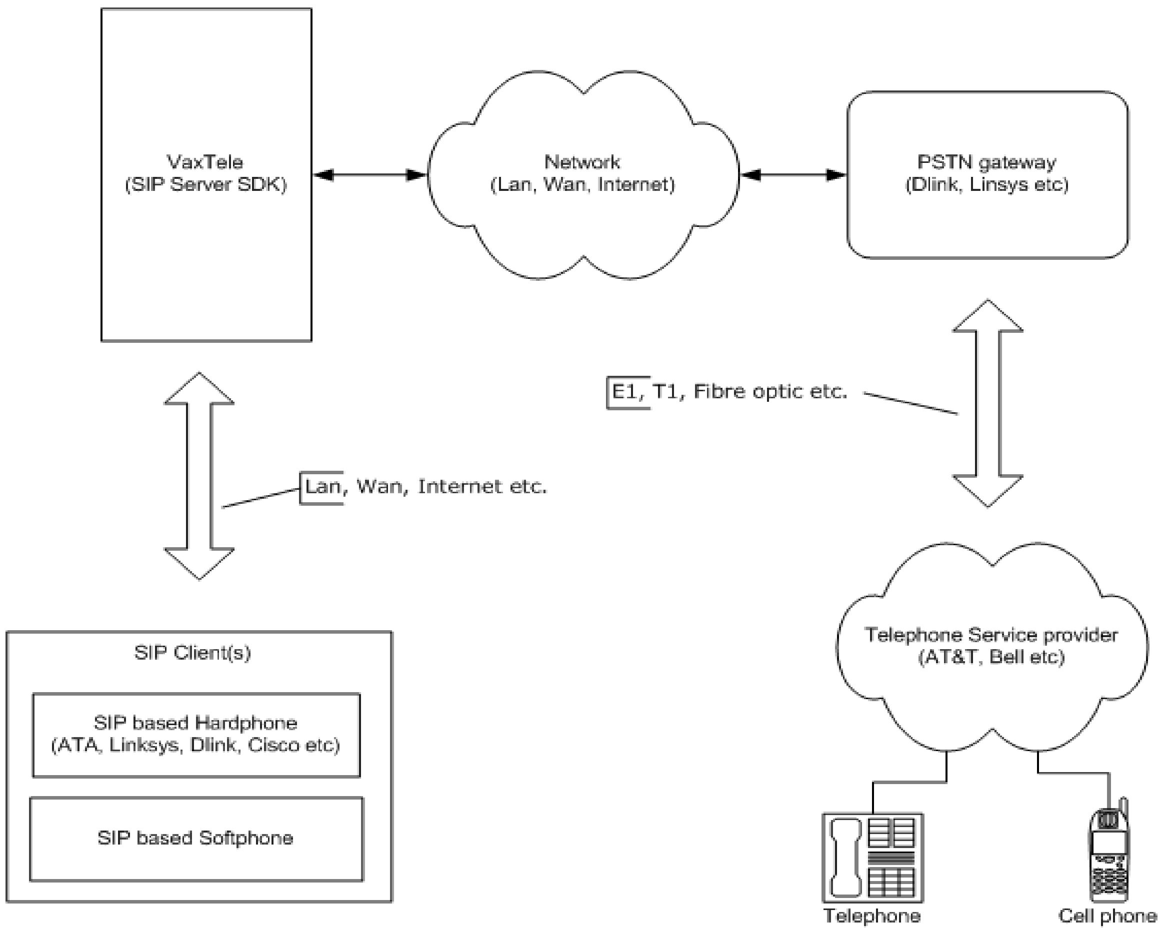
CONNECT TO PSTN/GSM NETWORK

VERSION 8.0.2.4

CONNECT TO PSTN/GSM NETWORK

Numerous SIP-based PSTN/GSM based IP-Telephony gateway devices are available in the market; a search for “SIP-based gateways” on the internet will reveal various options.

These gateways enable the connection of the VaxVoIP SIP server to the PSTN/GSM network, including E1, T1, GSM and other PSTN interfaces. The PSTN/GSM network allows for dialing and receiving phone/mobile calls to and from other telephone and mobile numbers, facilitating integration with traditional telephony systems.



A telephony gateway device can be configured in the following ways:

1. **Gateway as SIP Client:** The device is set up to function as a SIP client, connecting to a SIP server to handle calls over the IP network.

2. **Gateway for Direct IP-to-IP Communication:** The device is configured for direct IP-to-IP SIP communication.

TELEPHONY GATEWAY AS SIP CLIENT

The telephony gateway is configured to act as a SIP client, registering with the VaxVoIP SIP server. Once registered, it sends and receives call requests, facilitating communication between the telephony network and the VaxVoIP SIP Server.

DIAL CALL TO TELEPHONY GATEWAY

Method: AddUser() User Name for SIP client **Method:** AddUser() User Name for gateway
Gateway and SIP client registers to VaxVoIP successfully.
SIP client (softphone/hardphone/ATA etc) dials phone number and sends call request to VaxVoIP.
Event: OnCallSessionCreated()
Event: OnIncomingCall() FromPeerType = USER FromPeerName = Client User-Name
Method: AcceptCallSession() ToPeerName = Gateway User-Name.
VaxVoIP sends call request to gateway.
Event: OnCallSessionConnecting() ChannelId = Channel-ONE (Gateway call)
Mobile/PSTN phone starts ringing.
Mobile/PSTN phone user accepts the incoming call.
Event: OnCallSessionConnected()
Audio streaming starts between SIP client and Telephony gateway.
SIP client or Mobile/PSTN phone user disconnects the Call.

Event: OnCallSessionHangUp() ChannelId = Channel-ZERO (if SIP client hangs up) ChannelId = Channel-ONE (if gateway hangs up)
Event: OnCallSessionClosed()

RECEIVE CALL FROM TELEPHONY GATEWAY

Method: AddUser() User-Name for SIP client **Method:** AddUser() User-Name for gateway
Telephony gateway and SIP client registers to VaxVoIP successfully.
Gateway receives Mobile/PSTN phone call. Gateway sends SIP call request to VaxVoIP.
Event: OnCallSessionCreated()
Event: OnIncomingCall() FromPeerType = USER, FromPeerName = Gateway User-Name
Method: AcceptCallSession() ToPeerName = Client User-Name.
VaxVoIP sends call request to SIP client (softphone/hardphone/ATA etc).
Event: OnCallSessionConnecting() ChannelId = Channel-ONE (SIP Client)
SIP client starts ringing.
SIP client accepts the incoming call.
Event: OnCallSessionConnected()
Audio streaming starts between SIP client and Telephony gateway.
SIP client or Mobile/PSTN phone user disconnects the Call.
Event: OnCallSessionHangUp() ChannelId = Channel-ZERO (if gateway hangs up) ChannelId = Channel-ONE (if SIP client hangs up)
Event: OnCallSessionClosed()

TELEPHONY GATEWAY AS DIRECT IP TO IP COMMUNICATION

The telephony gateway is configured to directly receive call requests on its listening IP and port, and to send call requests directly to the VaxVoIP SIP Server's listening IP and port.

DIAL CALL TO TELEPHONY GATEWAY

Method: AddUser() User-Name for SIP client **Method:** AddLine("DirectLINE", VAX_LINE_TYPE_UDP, "", "", "", "", "", "192.168.0.24", 5060, "") Add DirectLINE to send call requests directly to the gateway's IP and port.
SIP client registers successfully.
SIP client (softphone/hardphone/ATA etc) dials a phone number.
Event: OnCallSessionCreated()
Event: OnIncomingCall() FromPeerType = USER FromPeerName = Client User-Name
Method: AcceptCallSession() ToPeerName = DirectLINE
VaxVoIP sends call request to gateway's IP and port.
Event: OnCallSessionConnecting() ChannelId = Channel-ONE (Gateway call)
Mobile/PSTN phone starts ringing.
Mobile/PSTN phone user accepts the incoming call.
Event: OnCallSessionConnected()
Audio streaming starts between SIP client and Telephony gateway.
SIP client or Mobile/PSTN phone user disconnects the Call.
Event: OnCallSessionHangUp() ChannelId = Channel-ZERO (if SIP client hangs up) ChannelId = Channel-ONE (if gateway hangs up)
Event: OnCallSessionClosed()

RECEIVE CALL FROM TELEPHONY GATEWAY

Method: AddUser() User-Name for SIP client **Method:** AddLine("DirectLINE", VAX_LINE_TYPE_UDP, "", "", "", "", "", "192.168.0.24", 5060, "") Add DirectLINE to receive call requests directly from gateway.
SIP client registers successfully.
Mobile/PSTN phone user dials phone number.
Gateway receives Mobile/PSTN phone call. Gateway sends SIP call request directly on IP and Port of VaxVoIP.
Event: OnCallSessionCreated()
Event: OnIncomingCall() FromPeerType = LINE FromPeerName = DirectLINE
Method: AcceptCallSession() ToPeerName = Client User-Name
VaxVoIP sends call request to SIP client (softphone/hardphone/ATA etc).
Event: OnCallSessionConnecting() ChannelId = Channel-ONE (SIP client)
SIP client starts ringing.
SIP client accepts the incoming call.
Event: OnCallSessionConnected()
Audio streaming starts between SIP client and Telephony gateway.
SIP client or Mobile/PSTN phone user disconnects the Call.
Event: OnCallSessionHangUp() ChannelId = Channel-ZERO (if gateway hangs up) ChannelId = Channel-ONE (if SIP client hangs up)
Event: OnCallSessionClosed()