



## **SIP SERVER SDK v4.0**

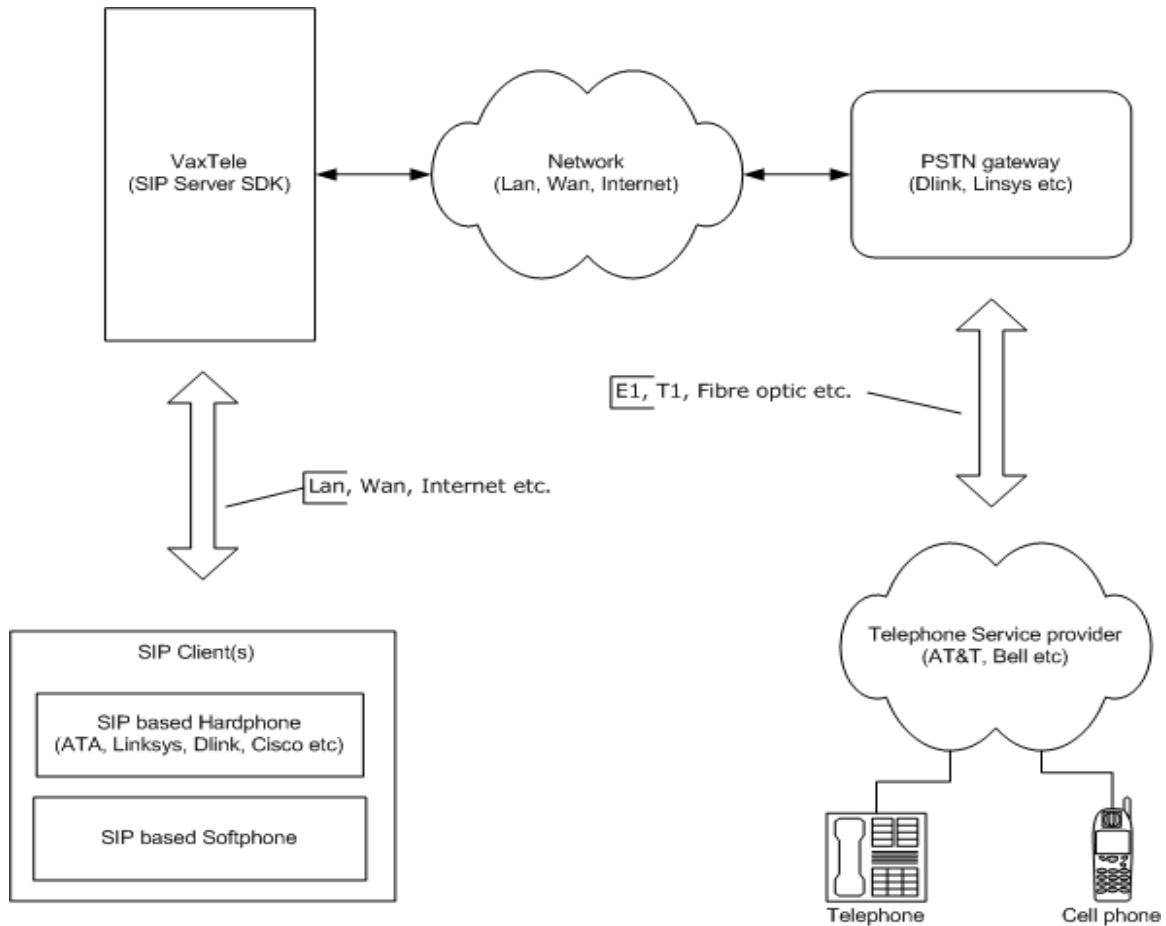
**CONNECT TO PSTN/GSM NETWORK**

**VERSION 2.6**

## **CONNECT TO PSTN/GSM NETWORK**

There are many SIP based PSTN gateways available in the market, you may search on Internet with "SIP based PSTN gateways". Those gateways can be used to connect VaxTele SIP server to the PSTN (E1, T1, etc) Network.

PSTN network is used to dial/receive phone calls to other telephone and mobile numbers.



PSTN gateway can be configured;

1. PSTN GATEWAY AS SIP CLIENT
2. PSTN GATEWAY AS DIRECT IP TO IP COMMUNICATION

## **PSTN GATEWAY AS SIP CLIENT**

PSTN gateway is configure in a way that it acts as SIP client, registers to the VaxTele SIP server and then send and receive call requests.

### **DIAL CALL TO PSTN GATEWAY**

<b>Method:</b> AddUser()	User Name for SIP client
<b>Method:</b> AddUser()	User Name for PSTN gateway
PSTN gateway and SIP client registers successfully.	
SIP client (softphone/hardphone/ATA etc) dials phone number.	
<b>Event:</b> OnCallSessionCreated()	
<b>Event:</b> OnIncomingCall()	FromPeerType = USER FromPeerName = Client User-Name
<b>Method:</b> AcceptCallSession()	ToPeerType = USER, ToPeerName = Gateway User-Name.
VaxTele sends dial call request to PSTN gateway.	
<b>Event:</b> OnCallSessionConnecting()	
ChannelId = Channel-ONE (Gateway user)	
<b>Method:</b> CallSessionSendStatusResponse()	
Mobile/PSTN phone starts ringing.	
Mobile/PSTN phone user accepts the incoming call.	
<b>Event:</b> OnCallSessionConnected()	
Voice streaming starts between SIP client and Mobile/PSTN phone user.	
SIP client or Mobile/PSTN phone user disconnects the Call.	
<b>Event:</b> OnCallSessionHangUp()	
ChannelId = Channel-ZERO (if SIP client hangup phone) ChannelId = Channel-ONE (if PSTN user hangup phone)	
<b>Event:</b> OnCallSessionClosed()	

**RECEIVE CALL FROM PSTN GATEWAY**

<b>Method:</b> AddUser()	User Name for SIP client
<b>Method:</b> AddUser()	User Name for PSTN gateway
PSTN gateway and SIP client registers successfully.	
Mobile/PSTN phone user dials phone number.	
<b>Event:</b> OnCallSessionCreated()	
<b>Event:</b> OnIncomingCall()	FromPeerType = USER, FromPeerName = Gateway User-Name
<b>Method:</b> AcceptCallSession()	ToPeerType = USER, ToPeerName = Client User-Name.
VaxTele sends dial call request to SIP client (softphone/hardphone/ATA etc).	
<b>Event:</b> OnCallSessionConnecting()	
ChannelId = Channel-ONE (SIP Client)	
<b>Method:</b> CallSessionSendStatusResponse()	
SIP client starts ringing.	
SIP client accepts the incoming call.	
<b>Event:</b> OnCallSessionConnected()	
Voice streaming starts between SIP client and Mobile/PSTN phone user.	
SIP client or Mobile/PSTN phone user disconnects the Call.	
<b>Event:</b> OnCallSessionHangUp()	
ChannelId = Channel-ZERO (if SIP client hangup phone) ChannelId = Channel-ONE (if PSTN user hangup phone)	
<b>Event:</b> OnCallSessionClosed()	

## **PSTN GATEWAY AS DIRECT IP TO IP COMMUNICATION**

PSTN gateway is configured in a way that it directly receives call requests on listen IP and sends call requests directly to the VaxTele SIP Server IP.

### **DIAL CALL TO PSTN GATEWAY**

<b>Method:</b> AddUser() User Name for SIP client	
<b>Method:</b> AddDirectProxySIP() Direct Proxy Name to add IP address and other info for PSTN gateway	
SIP client registers successfully.	
SIP client (softphone/hardphone/ATA etc) dials a phone number.	
<b>Event:</b> OnCallSessionCreated()	
<b>Event:</b> OnIncomingCall()	FromPeerType = USER FromPeerName = Client User-Name
<b>Method:</b> AcceptCallSession()	ToPeerType = DIRECT-PROXY ToPeerName = Direct-Proxy-Name
VaxTele sends dial call request to PSTN gateway.	
<b>Event:</b> OnCallSessionConnecting()	ChannelId = Channel-ONE (PSTN Gateway)
<b>Method:</b> CallSessionSendStatusResponse()	
Mobile/PSTN phone starts ringing.	
Mobile/PSTN phone user accepts the incoming call.	
<b>Event:</b> OnCallSessionConnected()	
Voice streaming starts between SIP client and Mobile/PSTN phone user.	
SIP client or Mobile/PSTN phone user disconnects the Call.	
<b>Event:</b> OnCallSessionHangUp()  ChannelId = Channel-ZERO (if SIP client hangup phone) ChannelId = Channel-ONE (if PSTN user hangup phone)	
<b>Event:</b> OnCallSessionClosed()	

**RECEIVE CALL FROM PSTN GATEWAY**

<b>Method:</b> AddUser() User Name for SIP client	
<b>Method:</b> AddDirectProxySIP() Direct Proxy Name to add IP address and other info for PSTN gateway	
SIP client registers successfully.	
Mobile/PSTN phone user dials phone number.	
<b>Event:</b> OnCallSessionCreated()	
<b>Event:</b> OnIncomingCall()	FromPeerType = DIRECT-PROXY, FromPeerName = Direct-Proxy-Name
<b>Method:</b> AcceptCallSession()	ToPeerType = USER ToPeerName = Client User-Name
VaxTele sends dial call request to SIP client (softphone/hardphone/ATA etc).	
<b>Event:</b> OnCallSessionConnecting()	ChannelId = Channel-ONE (SIP client)
<b>Method:</b> CallSessionSendStatusResponse()	
SIP client starts ringing.	
SIP client accepts the incoming call.	
<b>Event:</b> OnCallSessionConnected()	
Voice streaming starts between SIP client and Mobile/PSTN phone user.	
SIP client or Mobile/PSTN phone user disconnects the Call.	
<b>Event:</b> OnCallSessionHangUp()  ChannelId = Channel-ZERO (if SIP client hangup phone) ChannelId = Channel-ONE (if PSTN user hangup phone)	
<b>Event:</b> OnCallSessionClosed()	