SIP Customer Configuration Manual

Circuit Name:

XXXX-XXXX SIPXXXX

SIP Configuration Manual

how to start your SIP trunk

dear valuable customer
thank you for choosing SIP trunk services, in order to start your SIP phones, you need to implement your IP-PBX and interconnect it with our SIP trunk. And that is happening by connecting the IP-PBX to stc SIP-Server through the provided access to you.

connectivity:

be sure that you configured your parameters as mentioned below.

CUSTOMER IP:	XXX.XX.XXX.XXX	Subnet Mask	255.255.255.252
stc / GW IP :	XXX.XX.XXX.XXX		
SIP Server IP#:	**RIYADH 10.154.15.49**		
	DAMMAM 10.154.15.1		
	JEDDAH 10.154.15.25		

connectivity trouble shooting:

- make it sure that your link is up and your IP “CUSTOMER IP ADDRESS” is defined at your end
- make it sure you can reach stc IP which is your gateway. “stc IP address”
- define a static route for Sip Server “10.154.15.0” Subnet Mask “255.255.255.0” and next hope “stc IP address”

registration method:

Host domain/IP:	**10.154.15.X (SIP SERVER)**
Transport:	**UDP**
Domain:	**fmc.stc.com.sa**
SIP Port:	**5060**

we are sending option packet from stc side to customer's gateway as customer must reply with 200 ok message accordingly in order to get the trunk registered.

also, customer should stop re Invite and customer should reply on the option which he gets from stc side, do not send any option from his PBX.

find below example as reference,

option message sent from stc to customer

SIP Message Trace

```
OPTIONS sip:10.170.38.162:5060 SIP/2.0
Via: SIP/2.0/UDP 10.154.15.1:5060;branch=z9hG4bK5uyg5qhb9c7gygjydjb7gshci;Role=3;Hpt=8e42_
Call-ID: w7sidjiwsiddbhjqiw59ff9b7fybshjb@10.154.15.1
From: <sip:10.154.15.1>;tag=jhcchgdw
To: <sip:10.170.38.162>
CSeq: 1 OPTIONS
Contact: <sip:10.154.15.1;Hpt=13c4_16>;expires=65535
Accept: application/sdp
Max-Forwards: 70
Content-Length: 0
```

reply from customer end

SIP Message Trace

```
SIP/2.0 200 OK
Allow: INVITE, ACK, CANCEL, BYE, PRACK, NOTIFY, SUBSCRIBE, OPTIONS, UPDATE, INFO
Contact: sip:10.170.38.162:5060;isfocus
Supported: timer,path,100rel
User-Agent: OmniPCX Enterprise R100.1 n2.514.30.a
P-Alcatel-CSBU: fb=notransfer;categparty=external
To: <sip:10.170.38.162>;tag=2d560c4f85a023e41eed956579387aa1
From: <sip:10.154.15.1>;tag=jhcchgdw
Call-ID: w7sidjiwsiddbhjqiw59ff9b7fybshjb@10.154.15.1
CSeq: 1 OPTIONS
Via: SIP/2.0/UDP 10.154.15.1:5060;branch=z9hG4bK5uyg5qhb9c7gygjydjb7gshci;Role=3;Hpt=8e42_
Content-Length: 0
```

Once the trunk gets registered then Customer should send global format +966(Area code) (7 digits of range) for all incoming and outgoing calls

parameters:

You have to be sure that the following parameters are defined:

Protocol= SIP

SIP Port = 5060

Enable **User** = **Phone** for IN and OUT calls

Voice Codec = G711 A-Law 1st Priority with Ptime:20

G711 U-Law 2nd priority

DTMF = IN-Band DTMF with RFC3261

A = fmtp:97 0-15

Incoming & Outgoing Parameters:

- for incoming and outgoing format Ex: +966112108670
- IPPBX send invite message to stc, From header should be sip format like sip:+96611443XXXX@fmc.stc.com.sa;user=phone
- IPPBX send invite message to stc, to/Req URI should be sip format and user=phone appended
- not required transport and port in URI
- not required to send subscriber message to stc
- not required PAI nor PPI headers
- IPPPX side should support URIs in Sip and Tel format in all messages/headers
- IPPBX side should support sip like sip:+9661144XXXXX@fmc.stc.com.sa;user=phone with all domain included
- IPPBX should support handling ReqURI in priority over to header and PAI over from respectively

fax configuration:

- should be G711A pass through
- EC cancellation setting (default settings are OK)
- type of fax used (should enable high & low speed rate)

encryption configuration:

- the encryption box should be compatible with fax machine
- in NETWORK prospective normal fax and encrypted fax are not different

converters configuration:

- there might be many types of converters serving to the customers, most commonly used are TDMOIP (TDM over IP) used to convert Traditional signaling (R2, Qsig,) to IP
- a type of converter is used to provide RJ11 lines from IP are ATA device (Most popular devices are Cisco and Linksys) customer should be aware of ATA setting used
- we strongly advice to let the vendor support in installing and interconnecting the converters

other issues:

please capture END TO END trace and share with 909 team in order to save the time. Contact us on our 24/7 line services 909 directly