YAYFON ASTERISK INTEGRATION

YayFon is a cloud-based technologically advanced communication platform. Encrypted!

Interconnection instructions:

Our system support SIP UDP, TCP as well TSL, however we recommend using TLS. Also RTP/AVP (aka RTP), RTP/SAVP (aka SRTP) and RTP/SAVPF (aka RTP with DTLS) supported. If your system is behind a NAT you must give full internet access (outgoing from server) to your server and map one port (that you will place in sip.conf externtlsport=XXXX) to your internal servers ip and port. Don't forget to allow UDP 10000-40000 for RTP.
IPv6 supported.

sip.conf

[general]
udpenable=yes
tcpenable=yes

;we are highly recommend to use tls for signaling.
tlsenable=yes
tlsbindaddr=0.0.0.0:8443 ;using not standard port for tls is best practice for security.
tlscertfile=/etc/asterisk/keys/asterisk.pem

localnet=172.31.0.0/16 ; Your private IP range. Multiple line accepted
externaddr=XX.XX.XX.XX ;Your public IP
externtlsport=8443 ; Any port different 5060 and 5061. If your server is behind a NAT you must forward this port to internal TLS port.

[yayfon-trunk](!)
transport=tls
type=peer
context=from-yayfon ;Which dialplan to use for incoming calls. set to your own. Note that we are sending DID as e164 (without + prefix)
dtmfmode=auto ; we are supporting rfc2833, rfc4733 and SIP INFO
canreinivite=no
insecure=port,invite
directmedia=no
directrtpsetup=no
disallow=all
allow=g722
allow=alaw

allow=ulaw
encryption=yes ; SRTP recommended
qualify=no ; it is no need to send OPTIONS to our servers, we have auto scale and there are no chance to go to down :)

[yayfon](yayfon-trunk)
host=callme.yayfon.com

This is necessary to increase rtp port range. Edit rtp.conf with following params, also dont forget to allow udp from 10000 to 40000 on your firewall from whole world.

rtpstart=10000
rtpend=40000