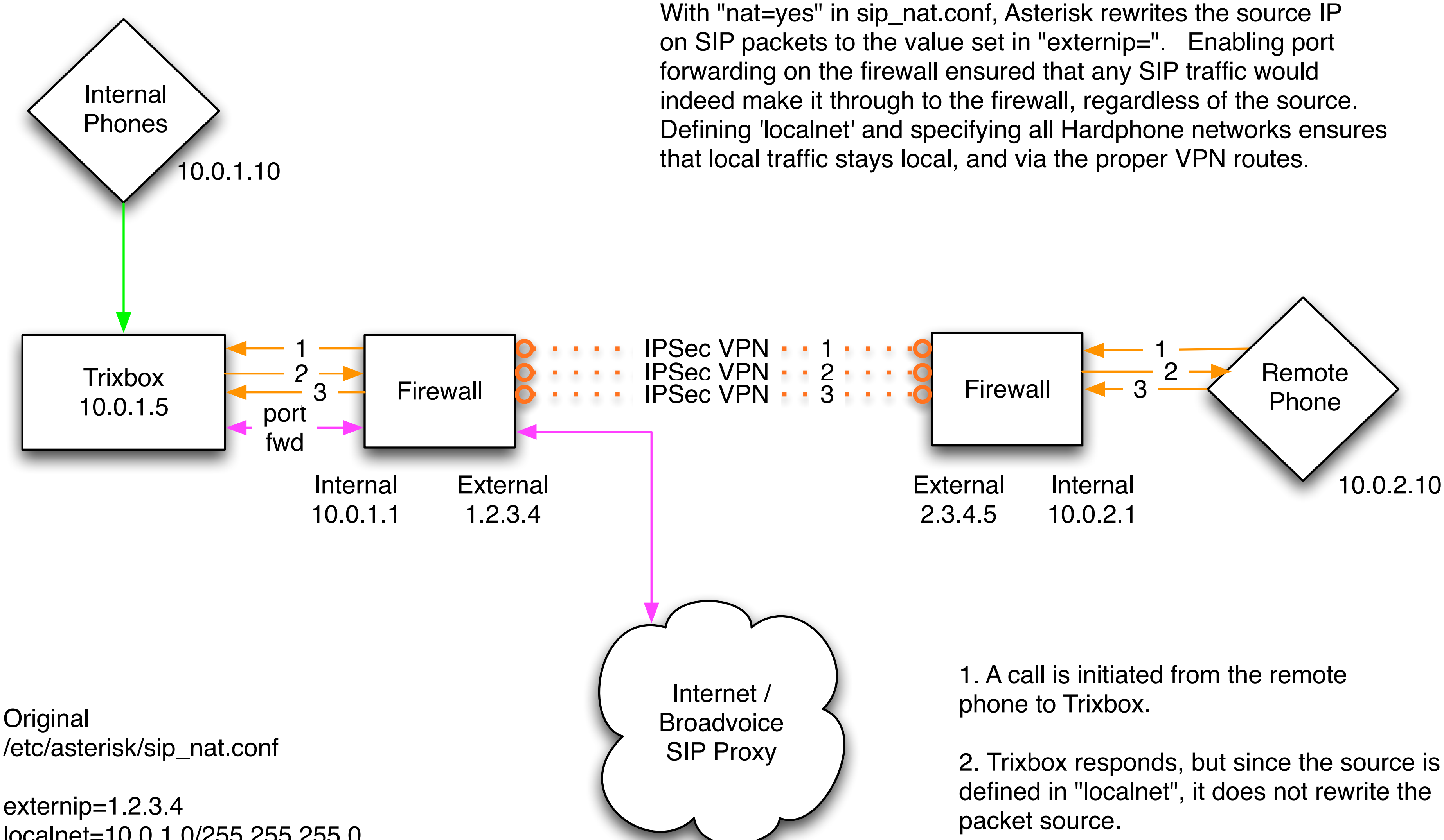
"The Right Way"

With "nat=yes" in sip_nat.conf, Asterisk rewrites the source IP on SIP packets to the value set in "externip=". Enabling port forwarding on the firewall ensured that any SIP traffic would indeed make it through to the firewall, regardless of the source. Defining 'localnet' and specifying all Hardphone networks ensures that local traffic stays local, and via the proper VPN routes.



Original
/etc/asterisk/sip_nat.conf

externip=1.2.3.4
localnet=10.0.1.0/255.255.255.0
localnet=10.0.2.0/255.255.255.0
nat=yes

1. A call is initiated from the remote phone to Trixbox.

2. Trixbox responds, but since the source is defined in "localnet", it does not rewrite the packet source.

3. The remote phone is now able to respond to stateful requests from Trixbox.