

"The Wrong Way"

With "nat=yes" in sip_nat.conf, Asterisk rewrites the source IP on SIP packets to the value set in "externip=". Being that we did not have portforwarding turned on, all packets sent to the remote phones had a source of "1.2.3.4", which thus bypasses VPN. Since Asterisk didn't receive a reply from the remote phone when it was asked, it then terminates the call in 20 seconds assuming the remote peer is dead.

