SBC Requirements

Who we are:

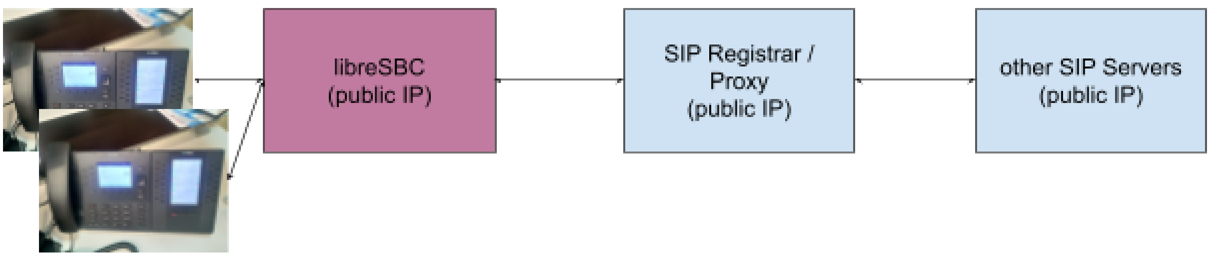
ImproWare AG is a Voice Service Provider providing feature rich SIP telephony to residential customers (single number sip subscriber UA) and businesses (PBX with sip-trunking and DDI ranges)
We are looking for a solution to replace our commercial SBC product, preferably open source.

Actual situation:

We have replaced our core routing, service, announcement, CDR gathering (billing) and registrar infrastructure with Kamailio and Asterisk.
A commercial SBC solution is still in place between the customer devices and our core infrastructure and between our core infrastructure and interconnections to upstream carriers.

What we need:

Our commercial SBC in use is EOL and soon out of support by the vendor. We prefer an open source solution as we have been disappointed by the lack of flexibility and support by commercial vendors.
We need an SBC in between our customer devices and our core infrastructure to perform topology hiding (**Main issue is to reduce the size of SIP messages, especially the number of route and via header as we have a lot of UAs which can only handle 4 route header**) and optionally handle RTP and NAT.



Our **priority** is to replace the SBC between our customers and the SIP Registrar/Proxy servers as shown above.
In a second stage, we will look into replacing the SBC between our Kamailio routing core and interconnections to upstream voice carriers.

Requirements:

Ideally the SBC would work transparently and route/track all SIP messages from UAs to the registrar and vice versa.

General

- Support for IPv4/IPv6 [required]
- Support for TLS/RTP encryption [required]
- Topology hiding of internal network from the UA [mandatory]
- RTP handling (e.g. RTP Engine) [optional]

REGISTER

- Route incoming REGISTER requests from SIP UAs to next hop (internal SIP Registrar/Proxy) [mandatory]
- Route replies from the registrar to SIP UAs [mandatory]
- NAT detection / handling [mandatory]
- NAT Ping / Options (keep connection open, active from SBC) [required]
- Register Multiple Contacts per AOR [mandatory]

INVITE, PRACK, UPDATE, ACK

- Route requests and replies between SIP UAs and SIP Registrar/Proxy [mandatory]
- NAT detection / handling [mandatory]
- Parallel Forking (call to multiple registered contacts or multiple AOR) [mandatory]
- Serial Forking (call forwarding on busy etc) [mandatory]
- Looping Calls (eg CFW from registrar via routing core, routed back to other customer on same registrar) [mandatory]
- Session timer [required]

Other Methods

- OPTIONS (for keep alive) [required]
- NOTIFY (Voicemail MWI) [mandatory]

Failed attempts to fulfill requirements with existing products

Kamailio with Topology Hiding Modules

- Fails to handle looping calls situations, example: call to registered UA runs through topo hiding module, times out, CFW engages and routes call back to core, which routes to another customer registered on same registrar and is handled again through topo hiding.

OpenSIPs with B2BUA UA

- Forking legs are aggregated and thus indistinguishable (to-tag not retained)
    - Functions depending on a specific forked leg fail
        - e.g., parallel or serial forking: RSeq of replies from different destinations are not tracked and independently increased causing other devices to drop them as duplicates.