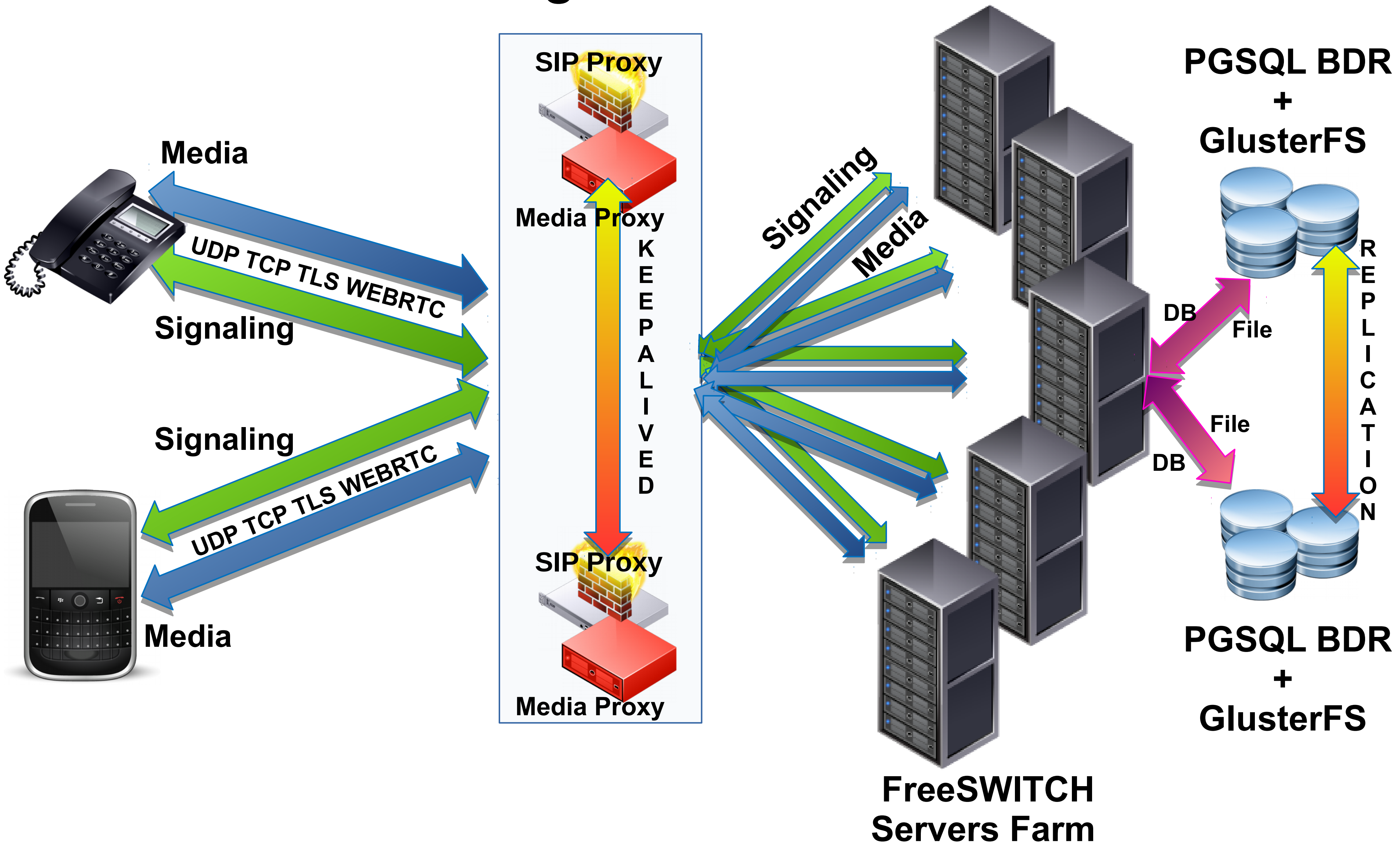
SCALING

FreeSWITCH(es)

Beyond the single machine: special cases and differences between single domain and multi tenant

Giovanni Maruzzelli
gmaruzz@OpenTelecom.IT

SIP
Scaling FreeSWITCHes

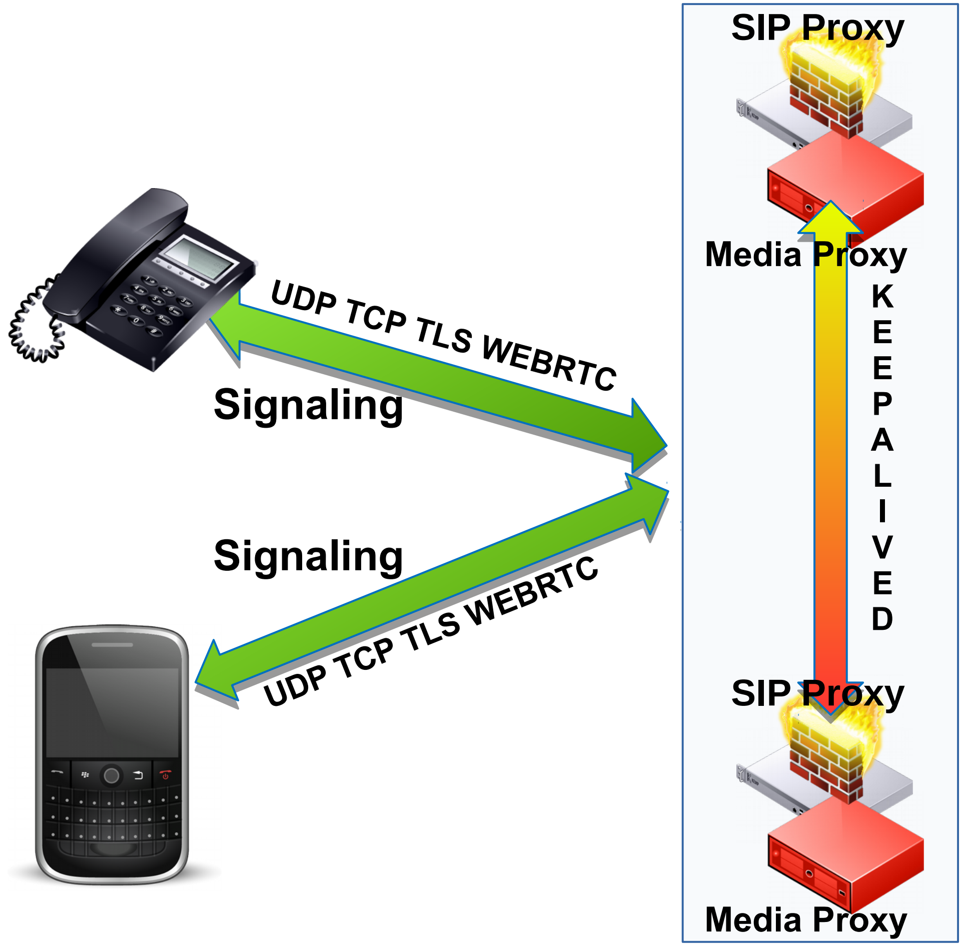


SIP and NAT

- Client is behind NAT, not directly reachable by server
- Client sends from its own IP:port a REGISTER request to Location Server IP:port, and in doing so it **opens a pinhole** in the NAT, waiting for server's answer
- NAT pinhole is only able to receive packets from **same IP:port** couple (Client/Server) it was open by, **and for a limited period** of time (30 seconds?)
- Location Server **sends periodically from same IP:port an OPTIONS** message to Client IP:port, Client answers, and in doing so it maintains the **pinhole open**
- When there is an incoming call for Client, Server sends the **INVITE** from **same IP:port** to Client IP:port

SIP

Load Balancing and Proxies



Where to put the **SIP** Registrar

- **ON LB (SIP Proxy) MACHINE**, directly interacting with Clients
  - REGISTER and NAT Keepalive (OPTIONS) are high volume, low load transactions
  - One robust box (in active-passive HA) will be able to serve tens of thousands clients
  - This is the most straightforward topology

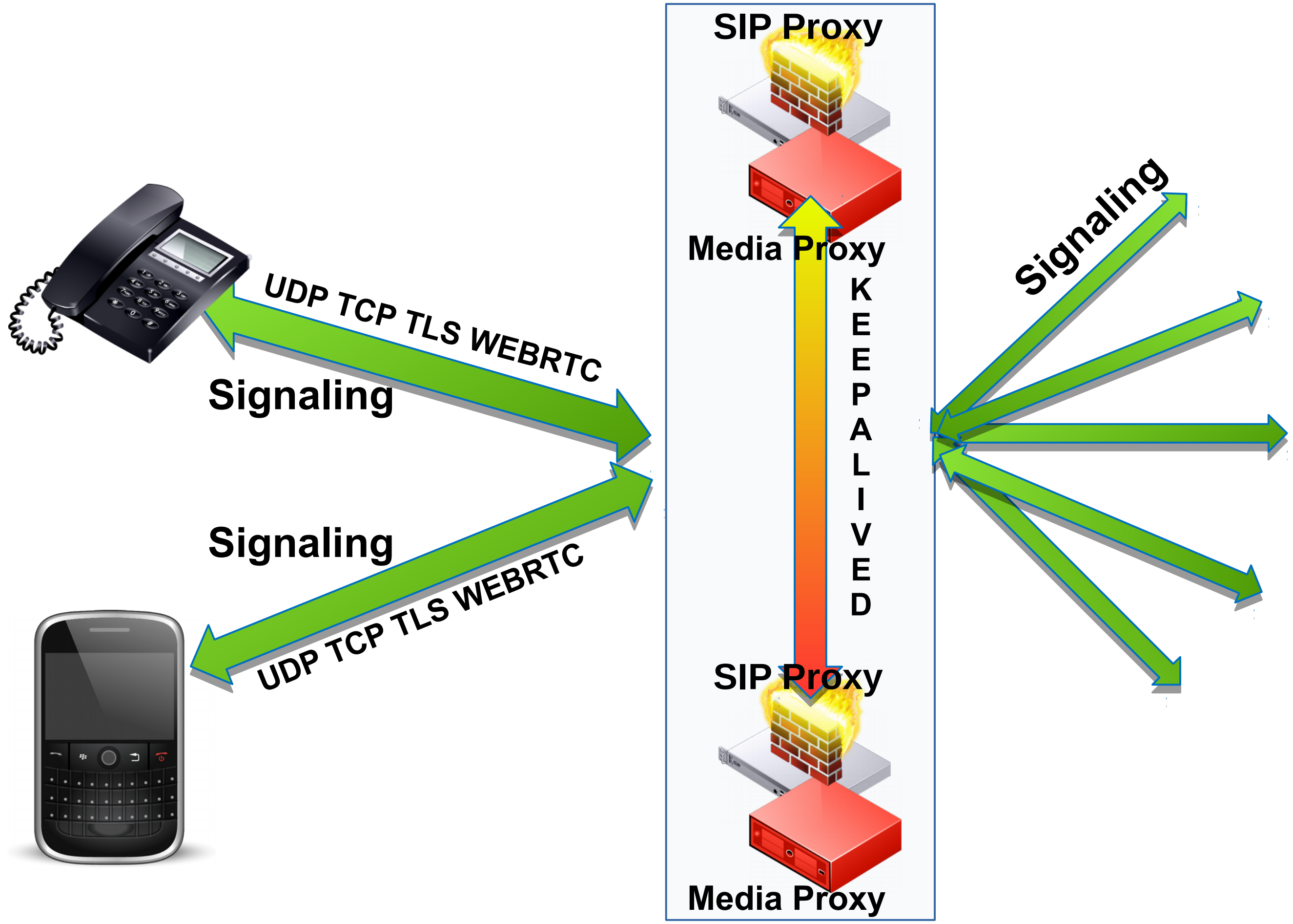
- **REGISTRATION is then Forwarded to FreeSWITCH MACHINES**, load balanced by LB
  - FreeSWITCHes are made aware of registration (eg, where the phone is) created and deleted
  - No registration traffic, no NAT keepalive traffic

SIP Call Distribution: DISPATCHER & LOAD BALANCER

- SIP Proxy can be used for relaying requests to multiple boxes using “static” algorithms (eg: round robin or weighted) or “dynamic” algorithms (that take care of actual number of active calls on each machine)
- All proxy's algorithms are able to “ping” destinations, retry on failed destination, disable the failed box from list, and re-enable it when is back in order

SIP

Load Balancing and Proxies

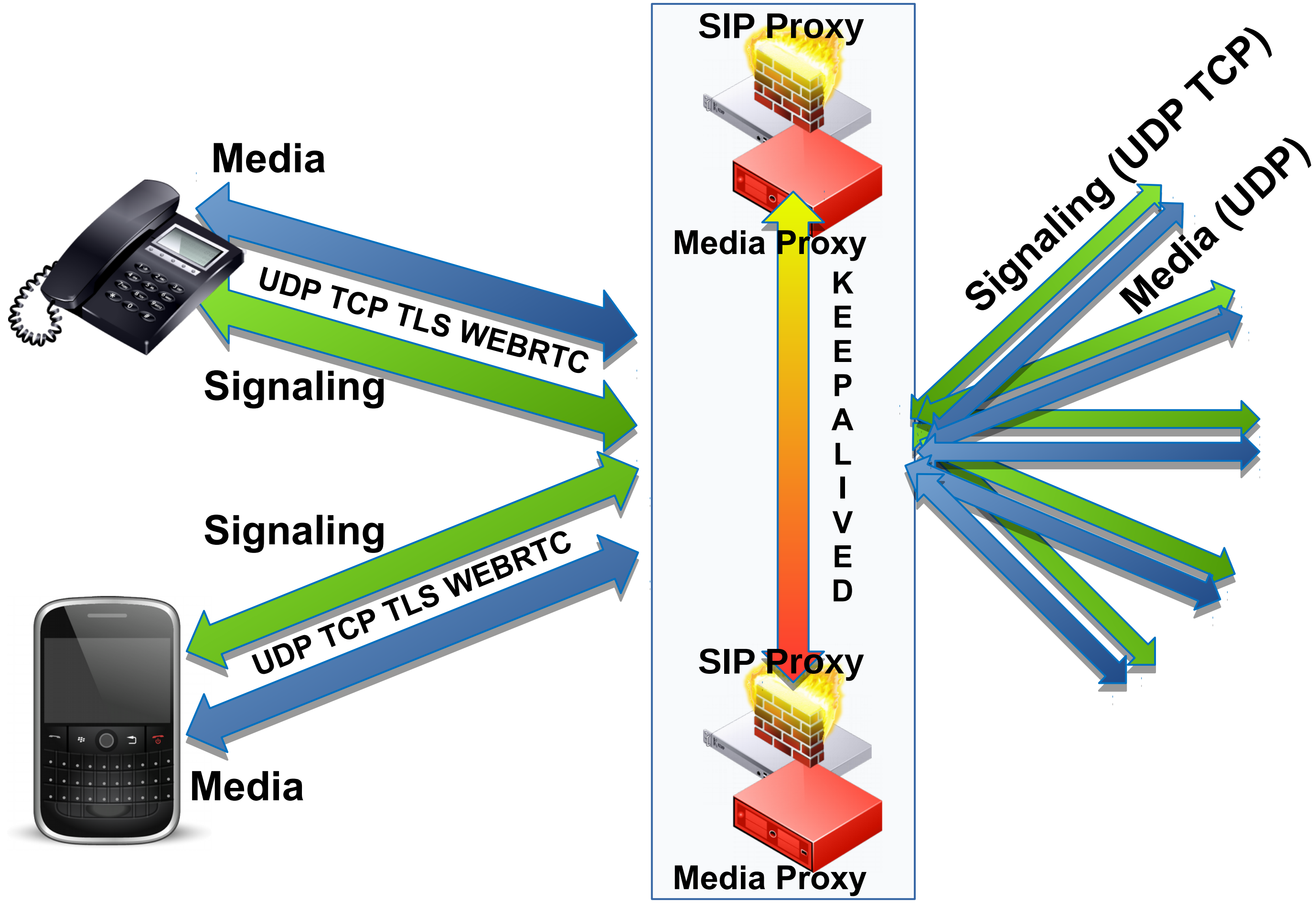


SIP Media Relaying

- SIP (signaling) proxy has nothing to do with media flow, it does not touch RTP
- It can modify SIP headers, and SDP bodies, so clients behind restrictive NATs will use a third party as a relay, and it can pass commands to that relay (eg: so the relay knows which client must be relayed to which)
- Original relay software is “Rtpproxy”
- More recent and advanced (eg: kernel space, etc):
  - MediaProxy
  - RtpEngine
- All of them can scale indefinitely

SIP

Load Balancing and Proxies



Adapted from: https://txlab.wordpress.com/tag/xlab1/ pass-through proxy by Stanislav Sinyagin

```
route {
        force_rport();
        if (!ds_is_in_list("$si", "$sp"))
        {
            # SIP request packet client->backend
            if( !loose_route() )
            {
                if ( !ds_select_dst("1","0") )
                {
                    send_reply("500","No Destination available");
                    exit;
                }
            }
            if (nat_uac_test("19")) {
                if (method=="REGISTER") {
                    fix_nated_register();
                } else {
                    fix_nated_contact();
                }
            }
            add_path_received();
        }
        else
        {
            # SIP request packet backend->client
            loose_route();
        }
        if (method=="INVITE") {
              rtpproxy_engage("cw");
        }
        record_route();
        t_relay();
}
onreply_route {
        if (!ds_is_in_list("$si", "$sp"))
        {
            # SIP reply packet client->backend
            fix_nated_contact();
        }
    return(1);
}
                                                   163,1          Bot
```

Standard Calls (one to one, straightforward)

- Registered Phone to Registered Phone
  (eg “Internal Calls”)
- Registered Phone to ITSP gw
  (eg “Outbound Calls”)
- ITSP’s to Registered Phone
  (eg “Inbound DID Calls”)

Special Cases (must be managed)

- Load Balancing is predicated on a server farm of equivalent and equipollent (eg: interchangeable) servers

- There are cases for which this is not true:
  - Conferences
  - Call Queues
  - Call Park – Unpark
  - Call/Group Pickup (Intercept)
  - And so on, and so on (quot. Zizek)

Conferences, Call Queues, Call Parks (must be local to one FS machine)

- Conferences are multiple calls' media streams switched or mixed together (think multitrack video/audio editing software), result stream is then broadcasted to all participants
- Call Queues are stacks of incoming live calls, all of them listening to Music on Hold, waiting to be dispatched to answering agents. It is possible to inject streams to single callers (eg "You are 3rd in line, your average waiting time is 9 minutes")
- Call Parks are named stalls where you put a call, and after a while you or someone else pick it up

Special Cases
(hash on destination)

opensips.cfg (~) - VIM

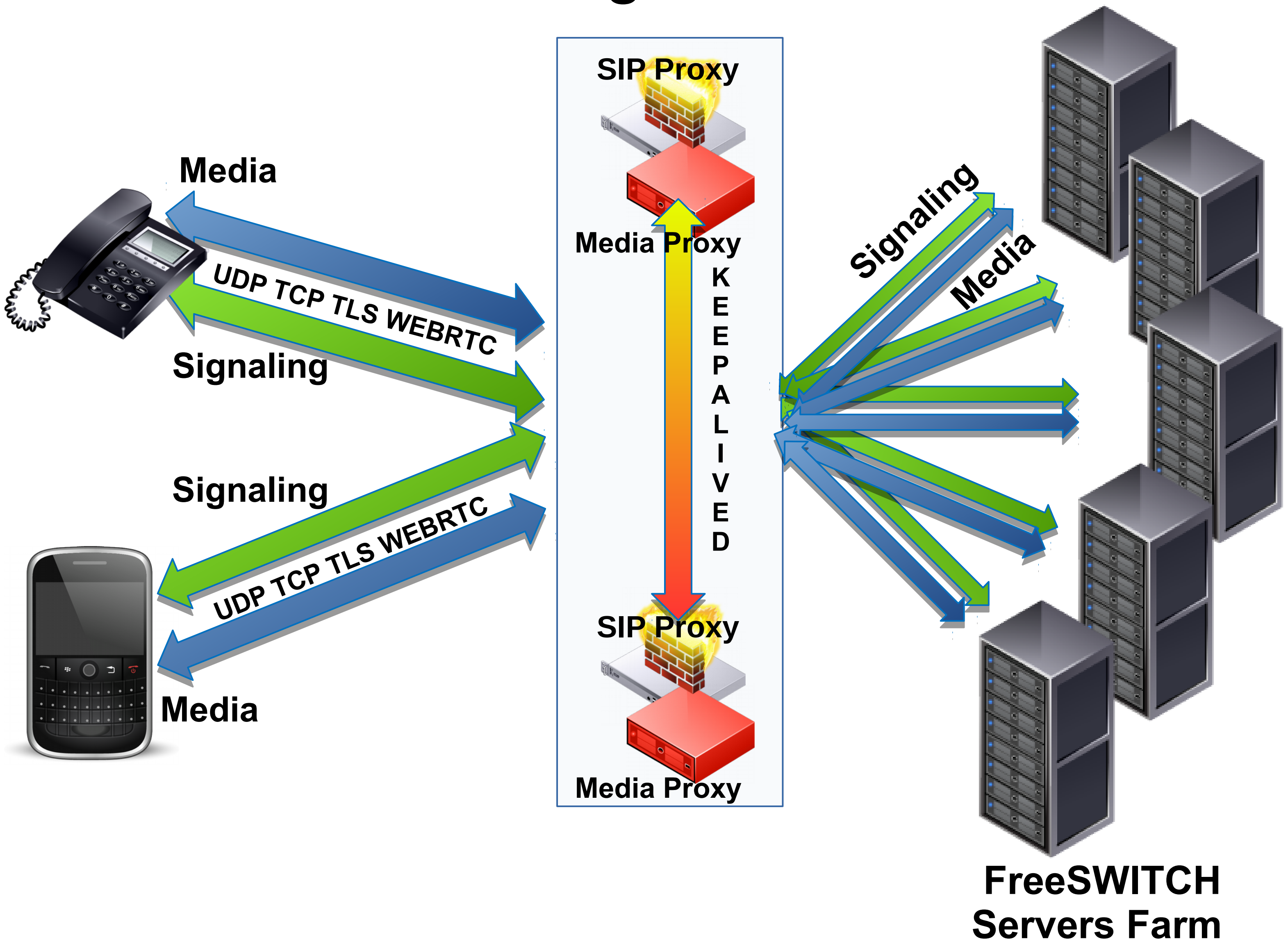
```
        $var(destination) = "" + $rU;
        $var(port) = "5060";

        $var(destinationmd5) = $(var(destination){s.md5}{s.substr,0,1}) ;
        $var(destinationmd5hex)="0x" + $var(destinationmd5);
        $var(destinationmd5int) = (int)$var(destinationmd5hex);
        $var(destinationmd5intmodulo) = $var(destinationmd5int) mod 3;

        switch ($var(destinationmd5intmodulo)) {
                case 0:
                        $du = "sip:192.168.1.117:" + $var(port) ;
                        break;
                case 1:
                        $du = "sip:192.168.1.116:" + $var(port) ;
                        break;
                case 2:
                        $du = "sip:192.168.1.113:" + $var(port) ;
                        break;
                default:
                        $du = "N/A";
        }
                                                              3,0-1          Top
```

SIP

Scaling FreeSWITCHes



Call/Group Pickups

- Someone answered a call, you are in his same group, he stares at you, you press *8 and pickup the call
- A call is ringing on a phone a desk away on your same group, you press *4EXT and answer the call

- Those two cases are to be managed at FreeSWITCH dialplan level, inserting into a DB table info on call groups’ belonging

Special Cases
(Multi Tenancy)

- Multi Tenant = Multiple SIP/WebRTC domains, managed independently
- Farm is partitioned on Domains by the Proxy, each domain goes to a particular machine
- This solves the conferencing-queues-transfer-pickup issues (eg locality of calls/users)

- High Availability by one or more SPARE machines, ready to take the role of the failed machine

Special Cases (hash on domain)

opensips.cfg + (~) - VIM

```
        $var(domain) = "" + $td;
        $var(port) = "5060";

        $var(domainmd5) = $(var(domain){s.md5}{s.substr,0,1}) ;
        $var(domainmd5hex)="0x" + $var(domainmd5);
        $var(domainmd5int) = (int)$var(domainmd5hex);
        $var(domainmd5intmodulo) = $var(domainmd5int) mod 3;

        switch ($var(domainmd5intmodulo)) {
                case 0:
                        $du = "sip:192.168.1.117:" + $var(port) ;
                        break;
                case 1:
                        $du = "sip:192.168.1.116:" + $var(port) ;
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                case 2:
                        $du = "sip:192.168.1.113:" + $var(port) ;
                        break;
                default:
                        $du = "N/A";
        }
                                                        1,1           Top
```
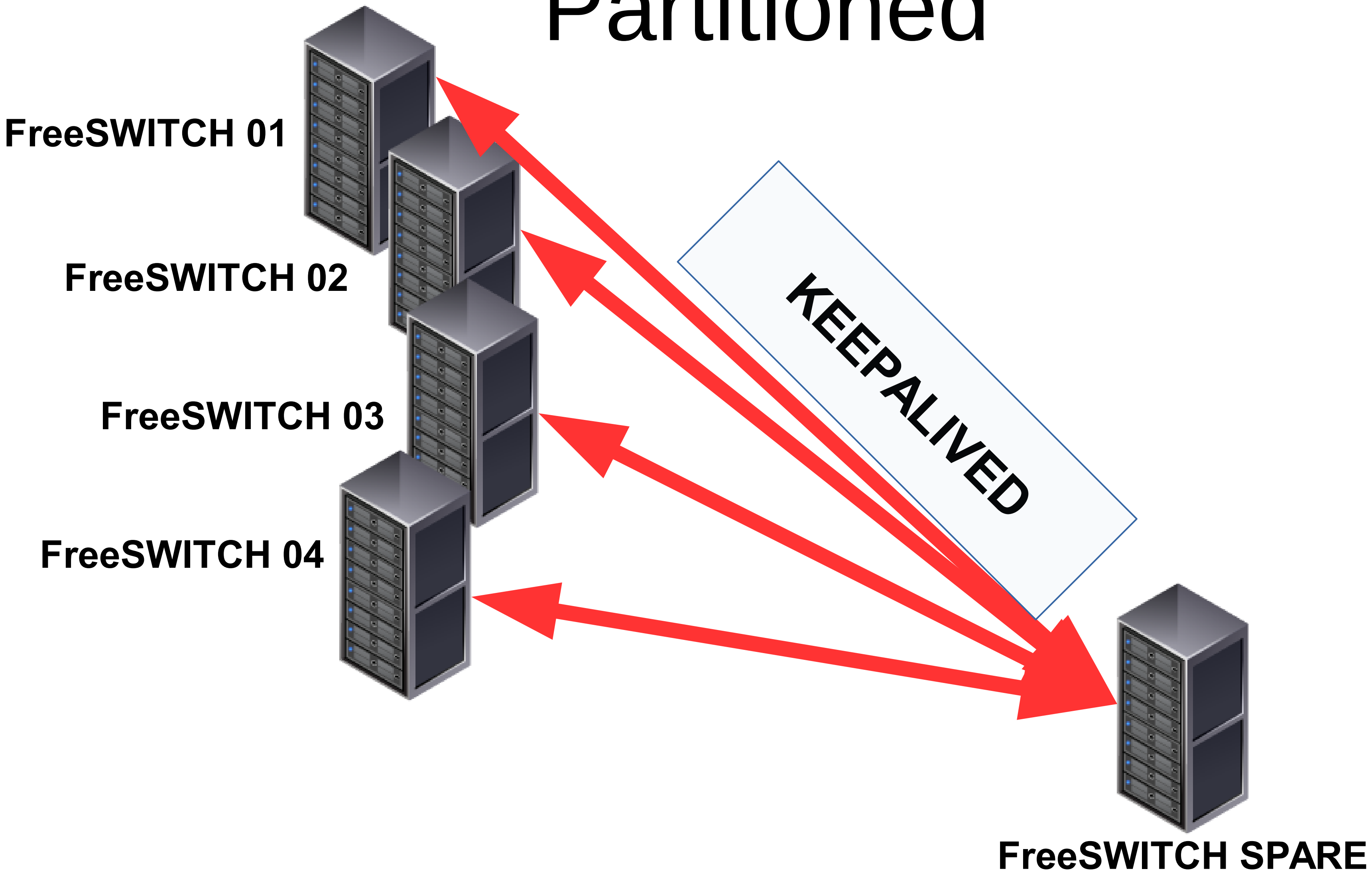

FreeSWITCHes' Farm Partitioned



Signaling, Again
(Presence, BLF, Messaging)

ALL PURE SIGNALING ARE BELONG TO SIP PROXY

- Presence
  - **SUBSCRIBE PUBLISH NOTIFY**
    - Event: State (Available, Busy, Do Not Disturb, Away)

- Blinking Field Lamp (BLF)
  - **SUBSCRIBE PUBLISH NOTIFY**
    - Event: Dialog (Idle, Ringing, Calling, in a call)

- Messaging
  - **MESSAGE (SIMPLE)**

OpenSIPS
Enhancements

- Clustering Proxies
- Mid-Registar
- Balancing on FreeSWITCH Load

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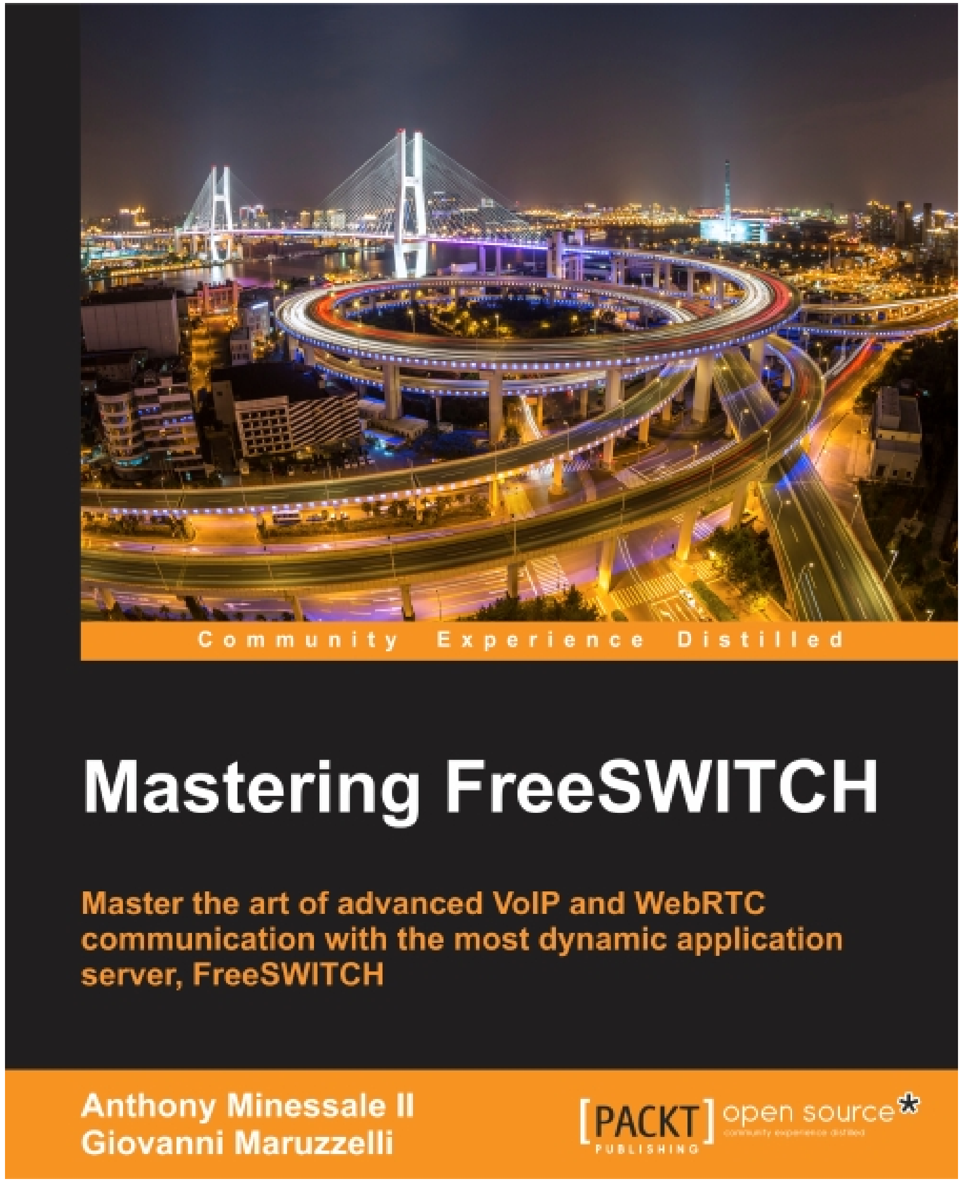

Thank You

QUESTIONS ?

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