

Kamailio for Building an IMS Core for VoLTE



Today's schedule

- Last year with Kamailio & IMS in review
- Basic IMS Infrastructure overview
- Installation of the network components
 - Proxy-CSCF (with SEMS for AMR)
 - Interrogating-CSCF
 - Serving-CSCF
 - Fraunhofer's OpenHSS (FhoSS)
- Walkthrough of HSS-Webinterface

Why VoLTE?

- Spectrum is limited
- Traditional 2G / 3G Networks use appr. 40% of the available spectrum for Voice
- With Voice-over-LTE it's down to 10%
- Cost of “pure” LTE networks are appr. 80% compared to 3G (5% compared to 2G)

40% OFF
SPECIAL SALE

Kamailio & IMS: Last year in review

- added support for RAVEL
- added (proper) support for 3GPP 23.228 annex U
- stability, stability, stability
- performance, performance, performance

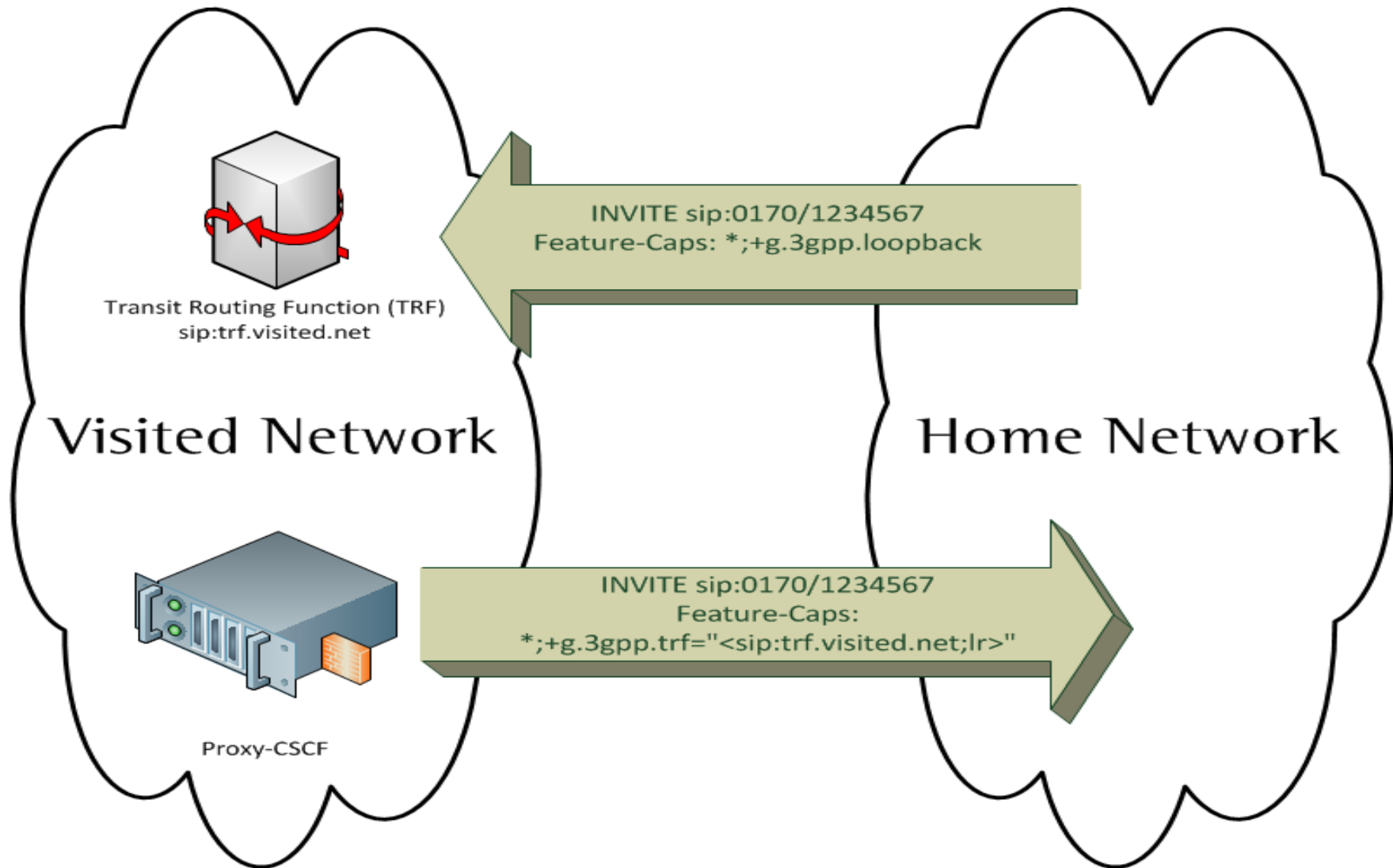
We've been busy!

RAV-What????

- RAVEL describes a mechanism for using a local breakout in Roaming scenarios

RAVEL = Roaming Architecture for Voice over IMS with Local break-out

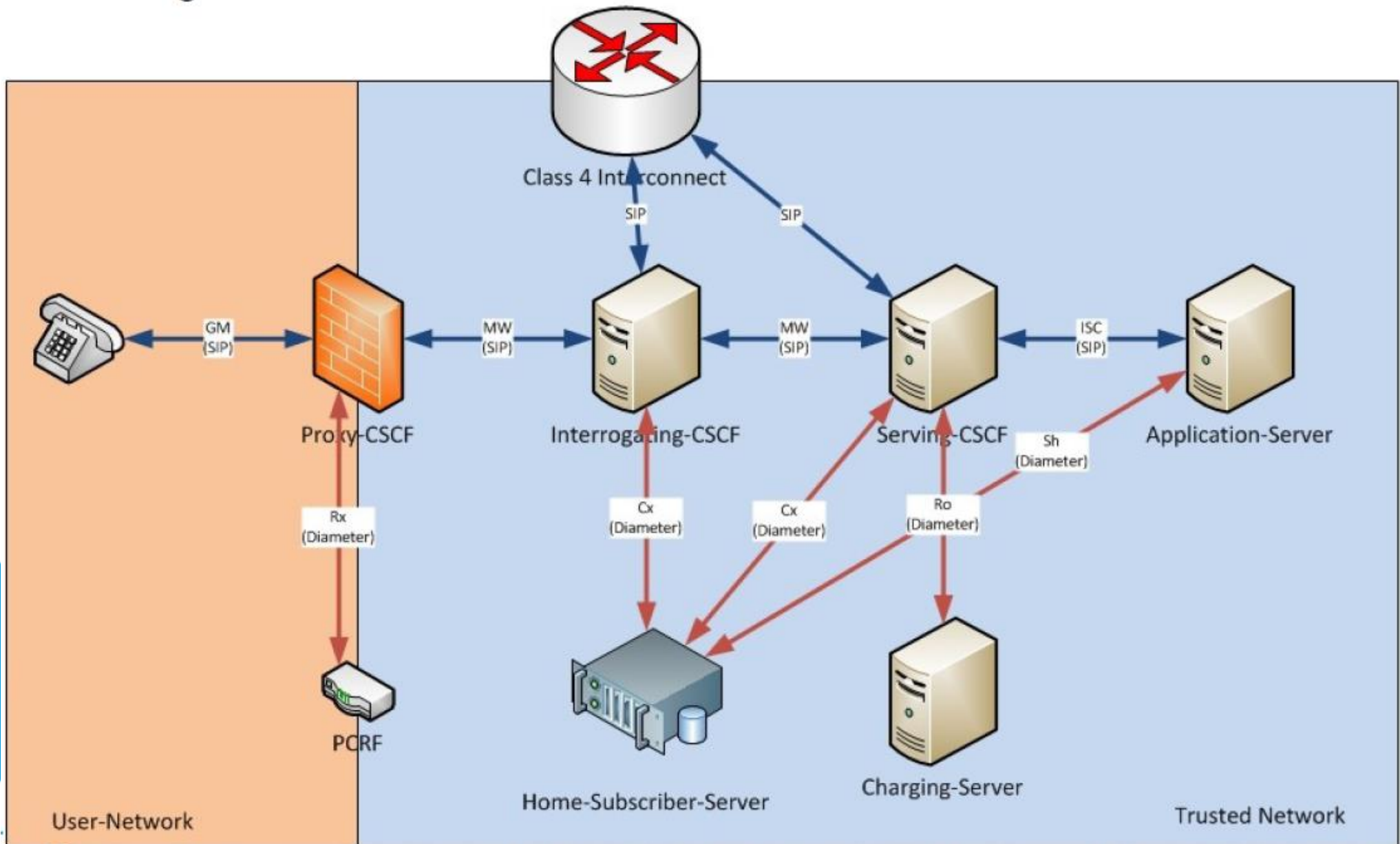
RAVEL



3GPP 23.228 annex U????

- Quite simple: It's WebRTC for IMS!!!
(nothing more, nothing less)
- With IMS, Kamailio & VoLTE supports:
 - Voice over LTE (VoLTE)
 - Voice over Wifi (VoWiFi)
 - OTT Apps (both LTE and Wifi)
 - Fixed-Lines devices
 - WebRTC-Endpoints

Basic IMS Infrastructure overview



Preparations: DNS / Bind

```
$ORIGIN mnc001.mcc001.3gppnetwork.org.
$TTL 1H
@           5M IN SOA      localhost. root.localhost. (
                                4           ; serial
                                5M         ; refresh
                                15M        ; retry
                                1W         ; expiry
                                5M )       ; minimum

                                6H IN NS    ns1.ng-voice.com.
                                6H IN NS    ns2.ng-voice.com.
ns1         6H IN A        109.239.50.66
ns2         6H IN A        109.239.50.67

;kamailio-ims.org. 5M IN NAPTR 10 10 "s" "SIPS+D2T"      ""      _sips._tcp.pcscf
kamailio-ims.org. 5M IN NAPTR 10 30 "s" "SIP+D2U"      ""
        _sip._udp.pcscf
kamailio-ims.org. 5M IN NAPTR 10 20 "s" "SIP+D2T"      ""      _sip._tcp.pcscf
```

Preparations: DNS / Bind (2)

```
pcscf 5M IN A 46.101.144.112
pcscf 5M IN NAPTR 10 10 "s" "SIP+D2T" "" _sip._tcp.pcscf
pcscf 5M IN NAPTR 10 20 "s" "SIP+D2U" "" _sip._udp.pcscf
_sip._tcp.pcscf 5M SRV 10 1 4060 pcscf
_sip._udp.pcscf 5M SRV 10 1 4060 pcscf

icscf 5M IN A 46.101.144.112
icscf 5M IN NAPTR 10 50 "s" "SIP+D2U" "" _sip._udp.icscf
_sip._udp.icscf 5M SRV 20 0 5060 icscf

scscf 5M IN A 46.101.144.112
scscf 5M IN NAPTR 10 50 "s" "SIP+D2U" "" _sip._udp.scscf
_sip._udp.scscf 5M SRV 10 0 6060 scscf

hss 5M IN A 46.101.144.112
```

Basic Installation

Kamailio:

- Hint: Take the trunk version!
- HSS: OpenHSS from Fraunhofer is a good start, but it can be replaced with any other HSS (Kamailio is tested with Nokia-Siemens Networks (NSN), Ericsson, ZTE, Huawei, ...)
- Installation of the trunk version is described in the Kamailio Wiki
- SEMS with AMR-Codec is available here:<https://github.com/ngvoice/sems-amr>
 - Note: Usage of the AMR Codec requires patent licensing from Nokia, Ericsson and others

Configuring the Proxy-CSCF (1)

SIP Express Media Server (SEMS) – for AMR-NB

- Apply provided configurations
(in the examples folder of Kamailio)
- Edit /etc/default/sems:
 - `RUN_SEMS="yes"`

Configuring the Proxy-CSCF (2)

Configure SIPWise' RTPEngine

- We need two instances of RTPEngine
 - for Originating traffic (MO)
 - for Terminating traffic (MT)
- Configs can be found in `/etc/default/rtpengine`

Configuring the Proxy-CSCF (3)

Configure Kamailio for use as a Proxy-CSCF:

- Add the SEMS-SBC to the dispatcher.list file
- Modify pcscf.cfg to fit to your needs (IP-Adresses, Hostnames, ...)
- Create the database for the Proxy-CSCF

Configuring the Interrogating-CSCF

- Modify `icscf.cfg` (Kamailio-Settings)
- Modify `icscf.xml` (Diameter-Connection)
- Create the database for the Interrogating-CSCF

Configuring the Serving-CSCF

- Modify scscf.cfg (Kamailio-Settings)
- Modify scscf.xml (Diameter-Connection)
- Create the database for the Serving-CSCF

Adding PSTN-Interconnects

- Inbound calls need to point to the I-CSCF
- Outbound gateways are defined in Dispatcher List on the Serving-CSCF
- ENUM is required for number to user mapping

Adding Applications

- The difficult/complex part is to add the proper rules
- Any SIP-Endpoint can be an application

Download: Configurations

All configurations, Zone-Files, etc.:

<https://github.com/kamailio/kamailio>

Check the examples folder, it's just been updated!