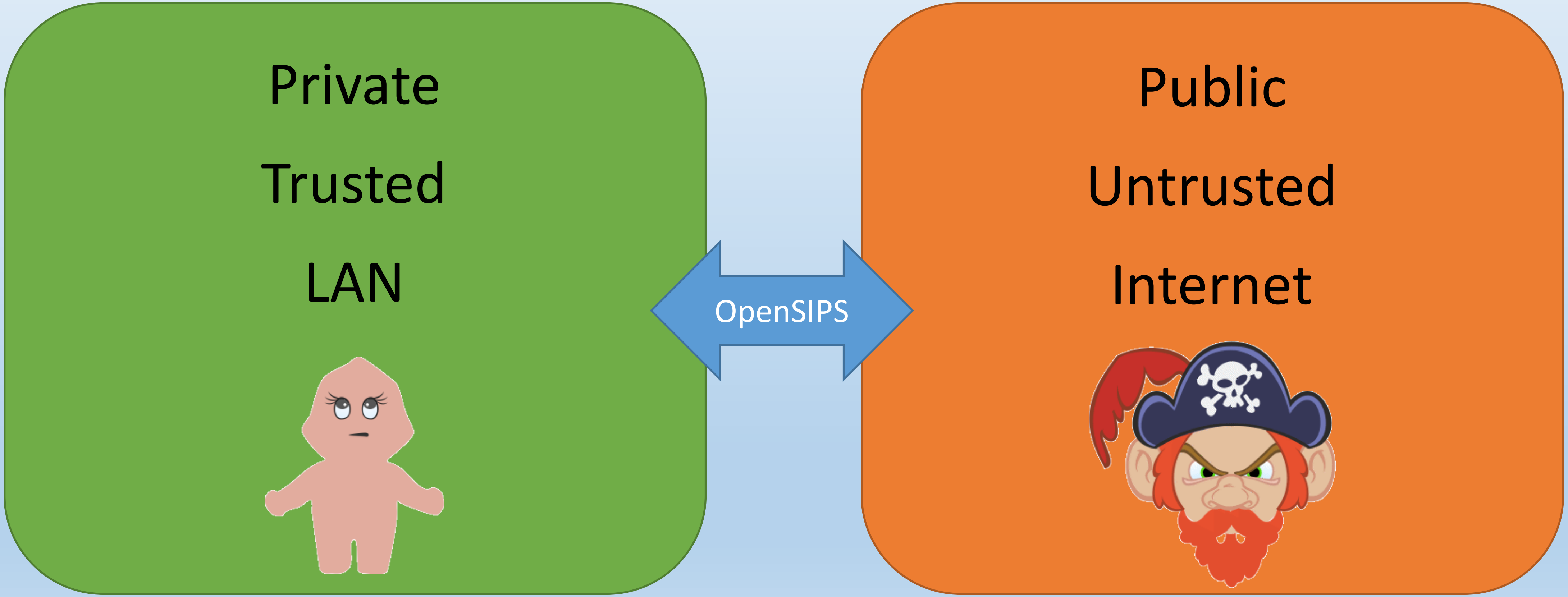
Securing the network edge with OpenSIPS

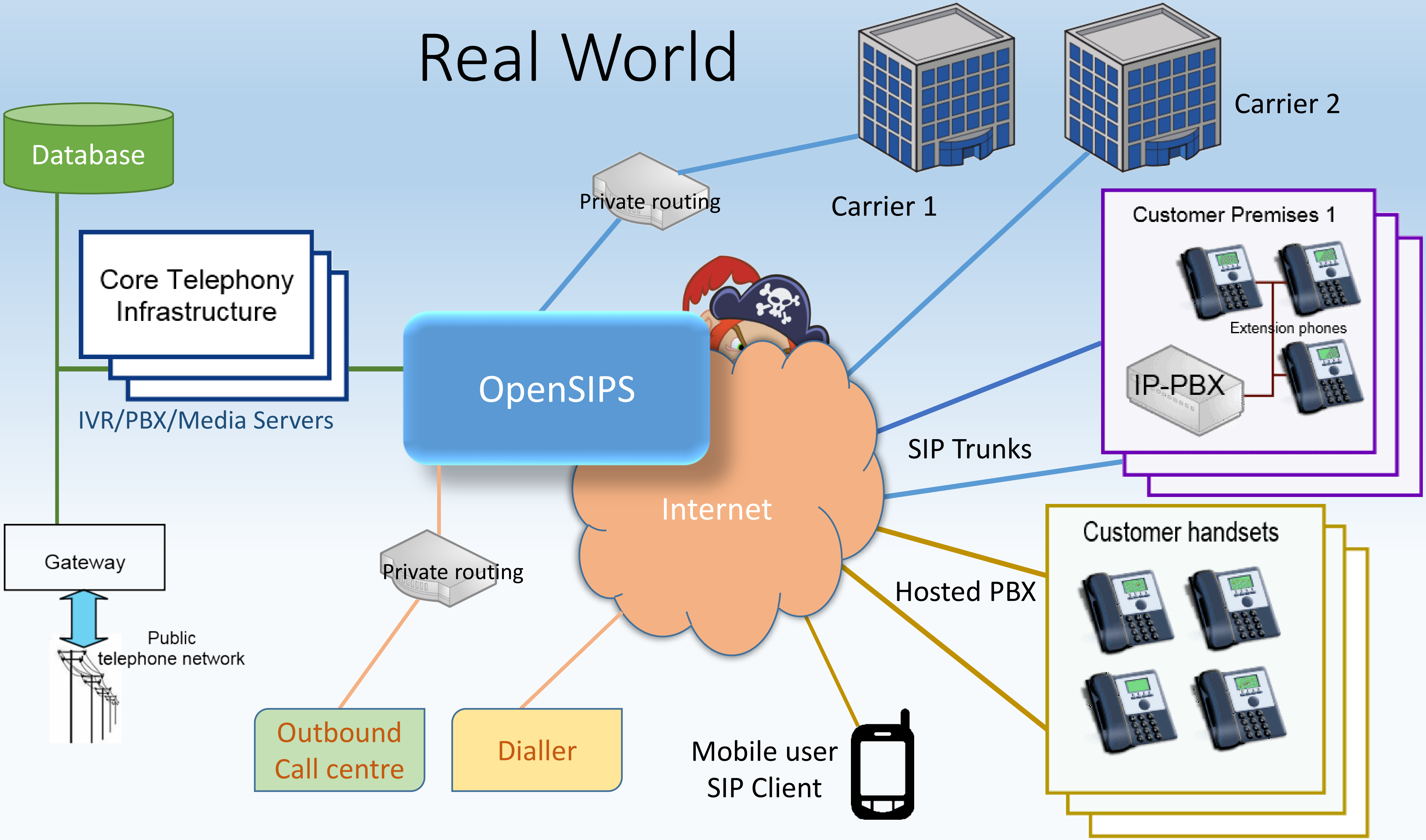
John Quick

Smartvox Limited

Network Edge – a simplistic view

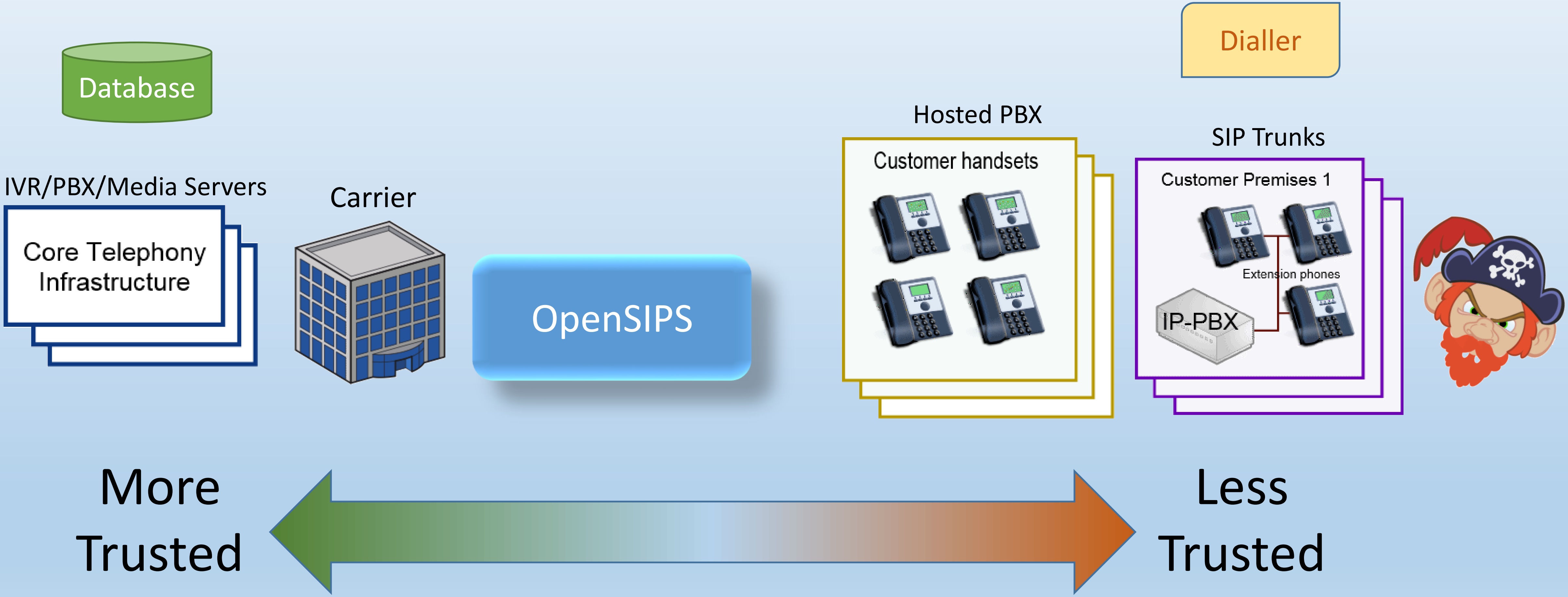


Real World

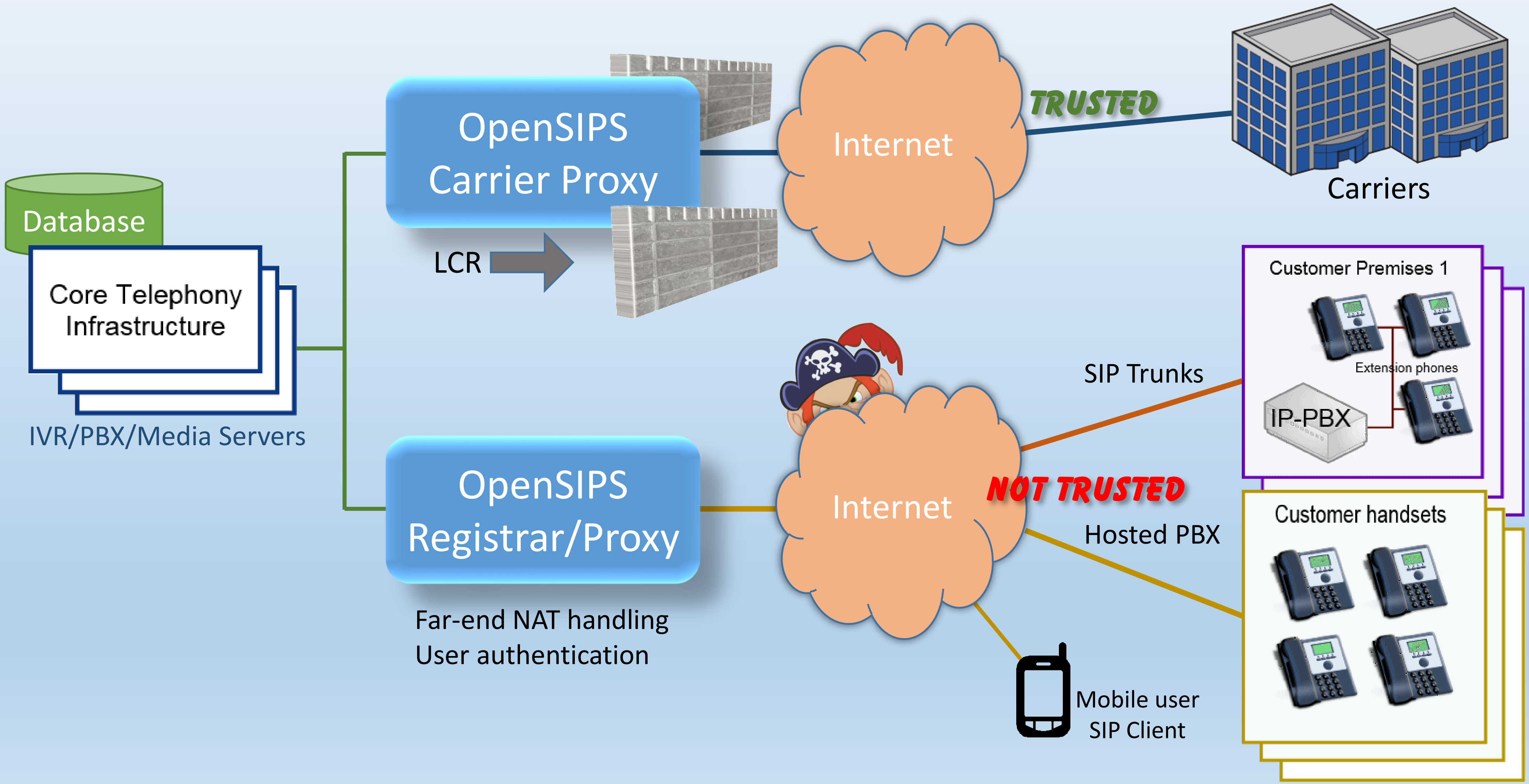


Database
Core Telephony Infrastructure
IVR/PBX/Media Servers
OpenSIPS
Private routing
Carrier 1
Carrier 2
Customer Premises 1
Extension phones
IP-PBX
SIP Trunks
Internet
Customer handsets
Hosted PBX
Gateway
Public telephone network
Private routing
Outbound Call centre
Dialler
Mobile user SIP Client

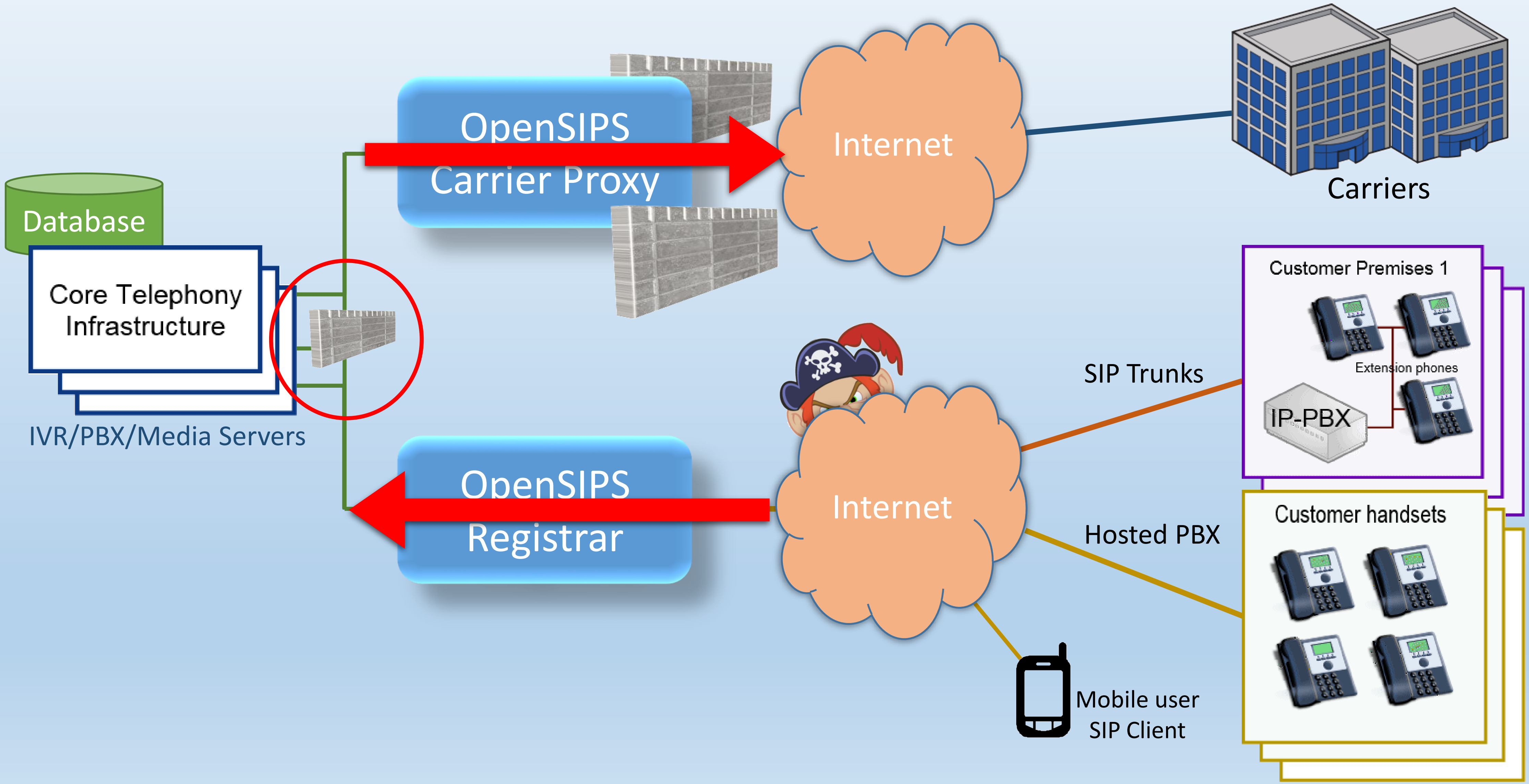
Who can be trusted?



OpenSIPS: Role separation



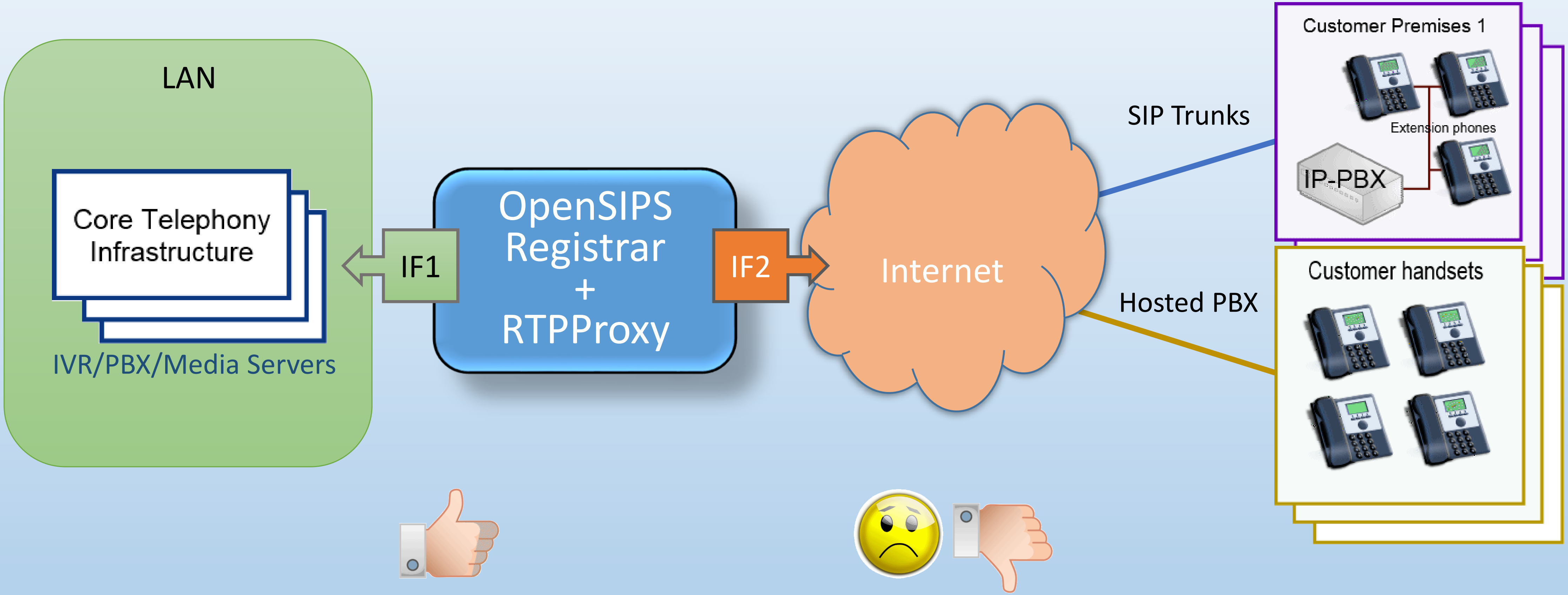
Call Routing: Clear identification of risk



Possible Network Configuration Scenarios

- Dual network interfaces on the OpenSIPS server
- All VoIP servers in a DMZ
- OpenSIPS in the DMZ and Media Servers on the LAN
- OpenSIPS in the DMZ and Media Servers behind 1-to-1 NAT
- OpenSIPS in the Cloud

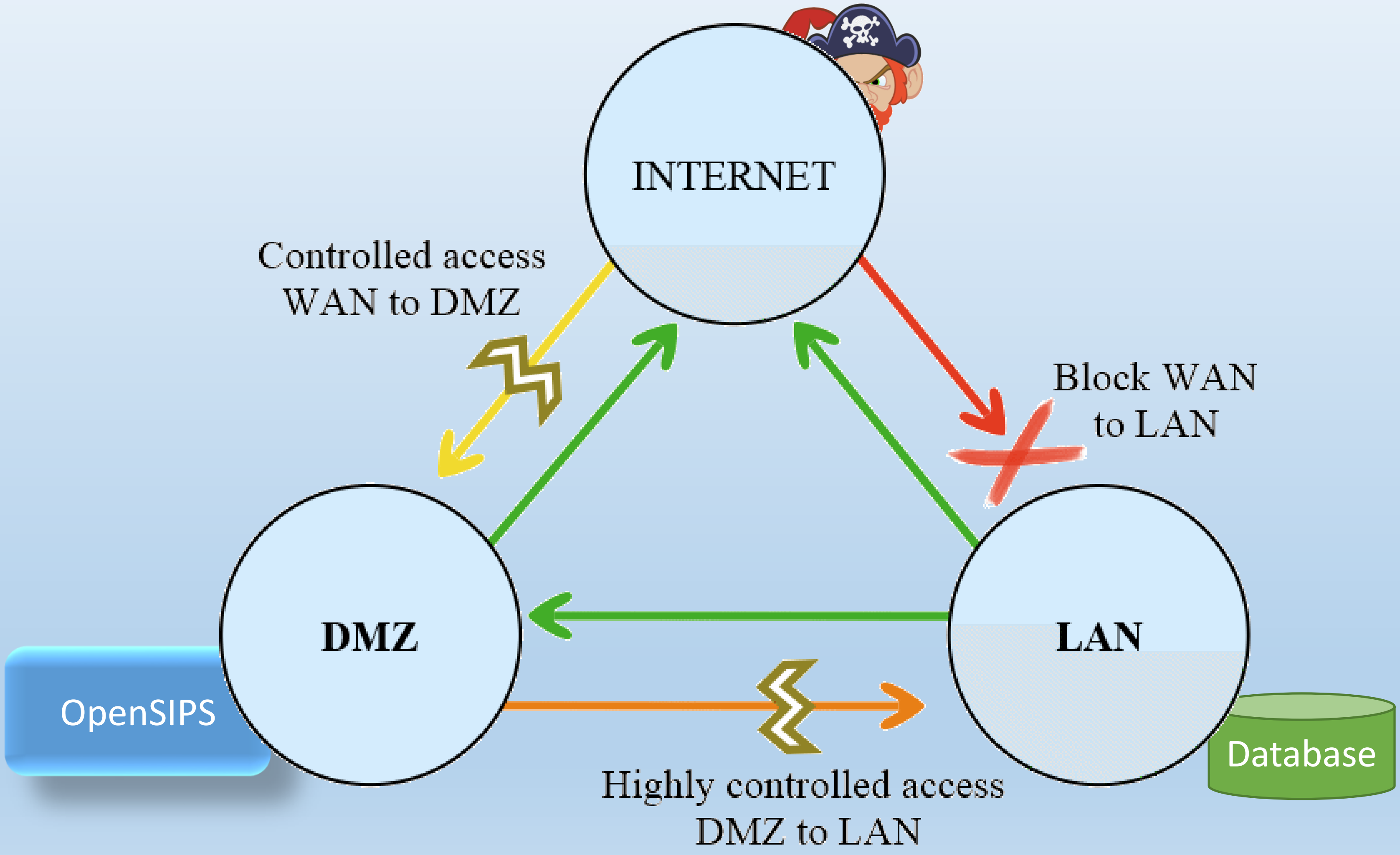
Network Zoning: Dual interface solution



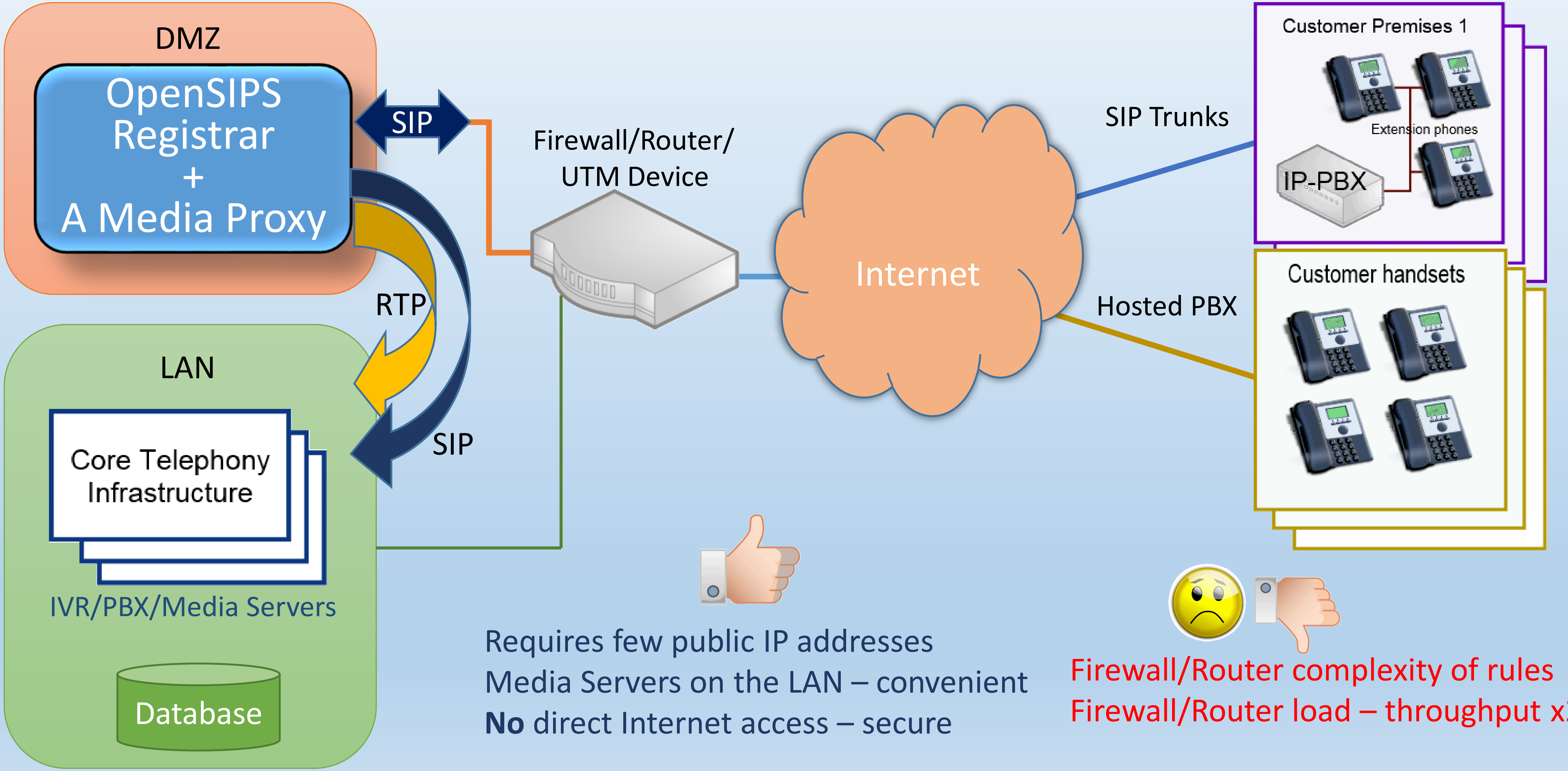
Requires few public IP addresses
No complex firewall configuration

Requires RTPProxy in bridging mode
RR headers need fixing in both directions
Security vulnerability if server is hacked

Network Edge – traditional zoning model



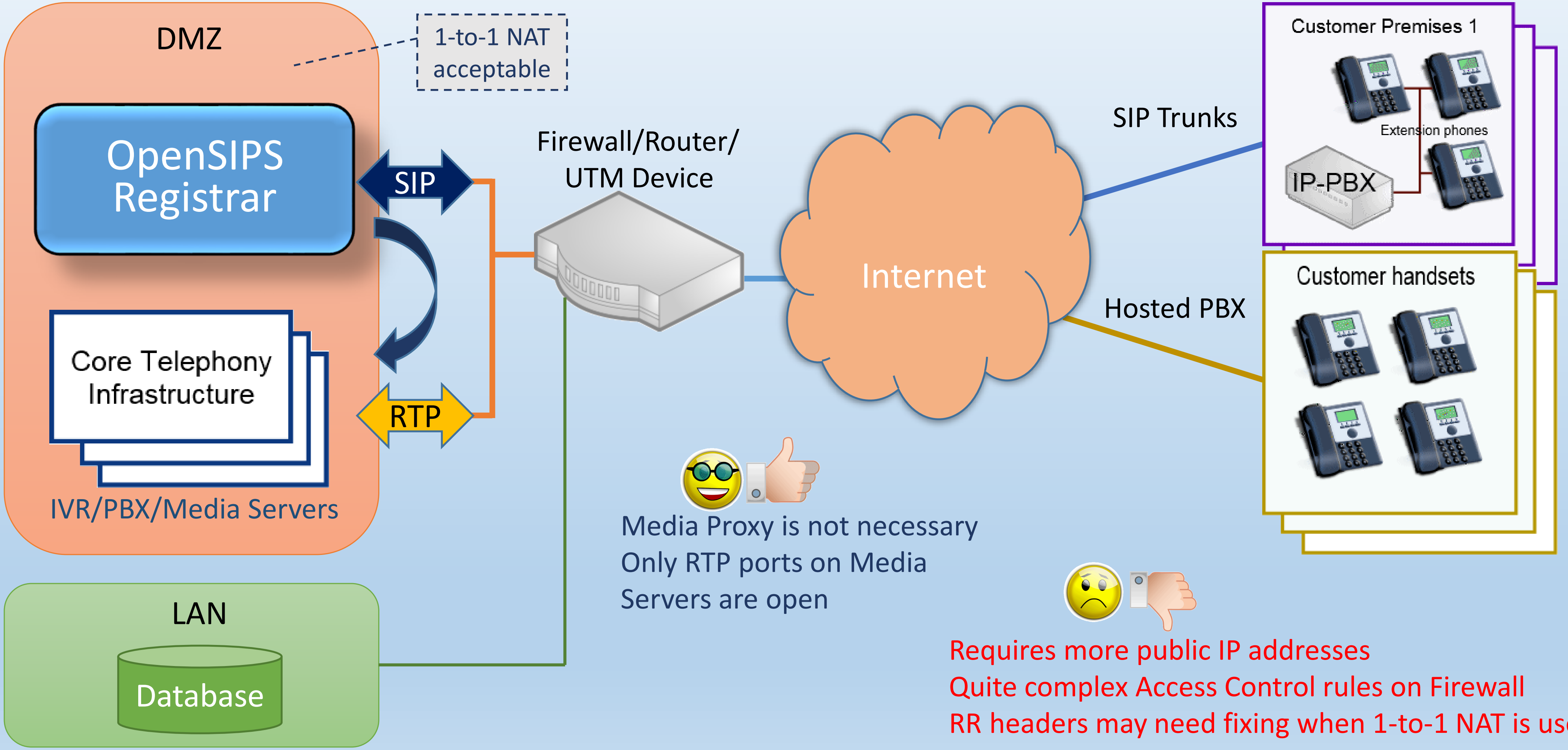
Network Zoning: OpenSIPS in the DMZ



Requires few public IP addresses
Media Servers on the LAN – convenient
No direct Internet access – secure

Firewall/Router complexity of rules
Firewall/Router load – throughput x2

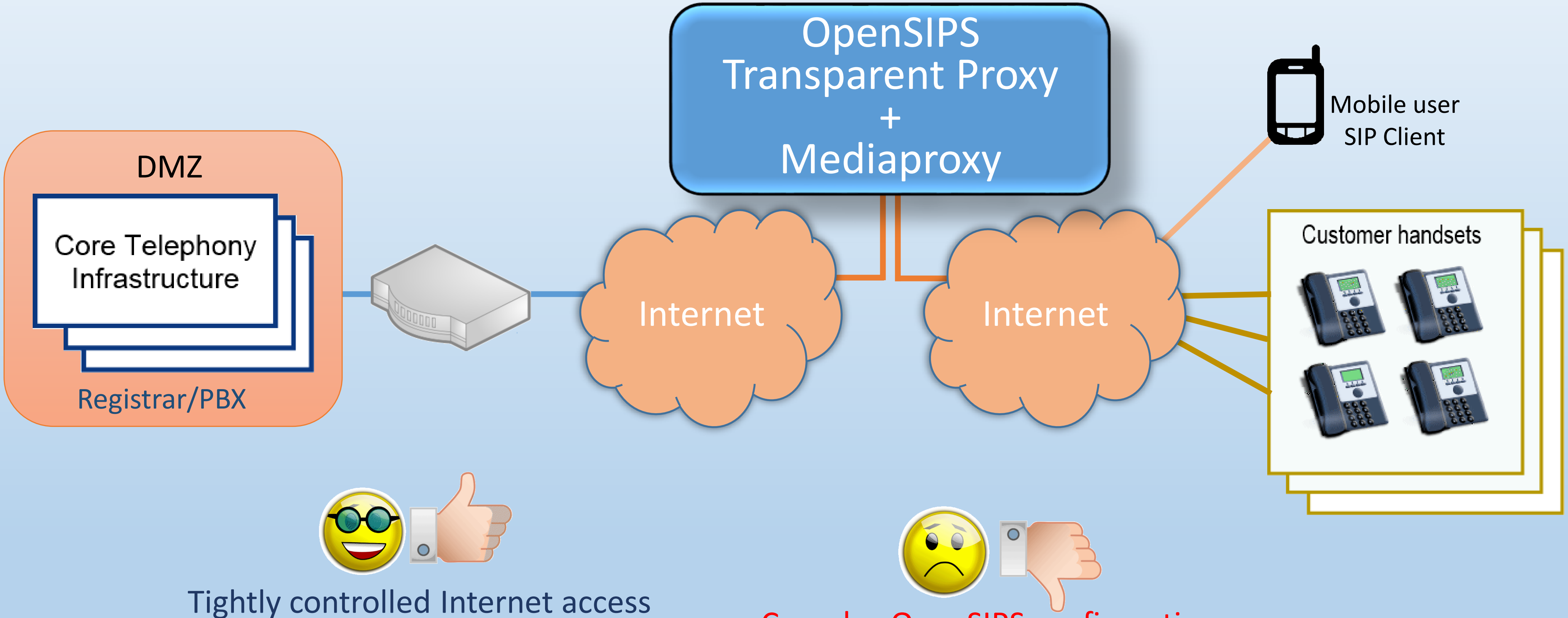
Network Zoning: VoIP Servers in DMZ



Media Proxy is not necessary
Only RTP ports on Media Servers are open

Requires more public IP addresses
Quite complex Access Control rules on Firewall
RR headers may need fixing when 1-to-1 NAT is used

Network Zoning: OpenSIPS in the Cloud

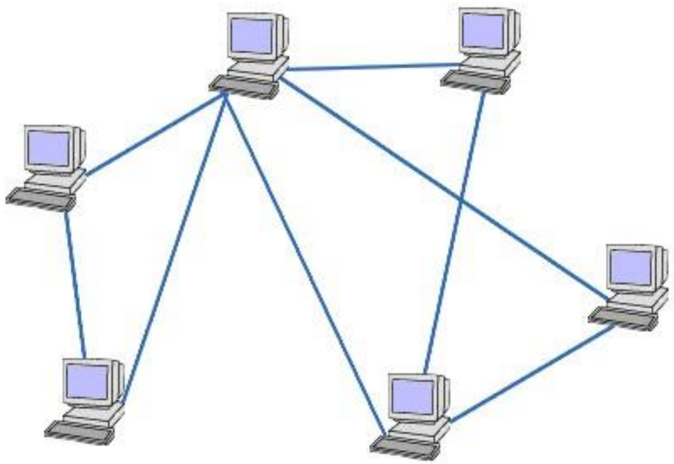


Tightly controlled Internet access
Multi-tenant possibilities
Potential as Value Added solution

Complex OpenSIPS configuration

A presentation of two halves

- Network Topology



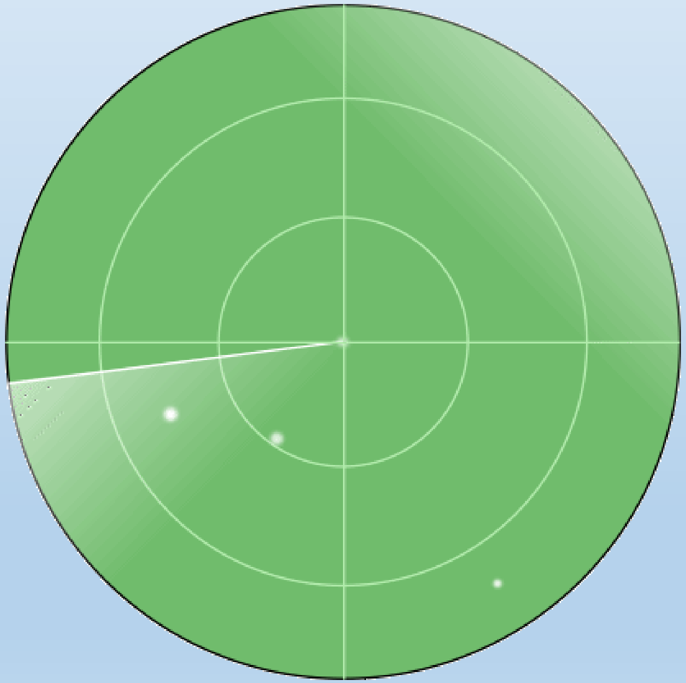
- Application level Security

OPTIONS SIP Request

OPTIONS sip:100@123.45.251.24 SIP/2.0.
Via: SIP/2.0/UDP 127.0.0.1:6241;branch=z9hG4bK-2252719110;rport.
Content-Length: 0.
From: "sipvicious"<sip:100@1.1.1.1>;tag=303539631363413939373134.
Accept: application/sdp.
User-Agent: friendly-scanner.
To: "sipvicious"<sip:100@1.1.1.1>.
Contact: sip:100@127.0.0.1:6241.
CSeq: 1 OPTIONS.
Call-ID: 10971752559881716145531.
Max-Forwards: 70.

Phase 1: Server Discovery

OPTIONS Ping like radar

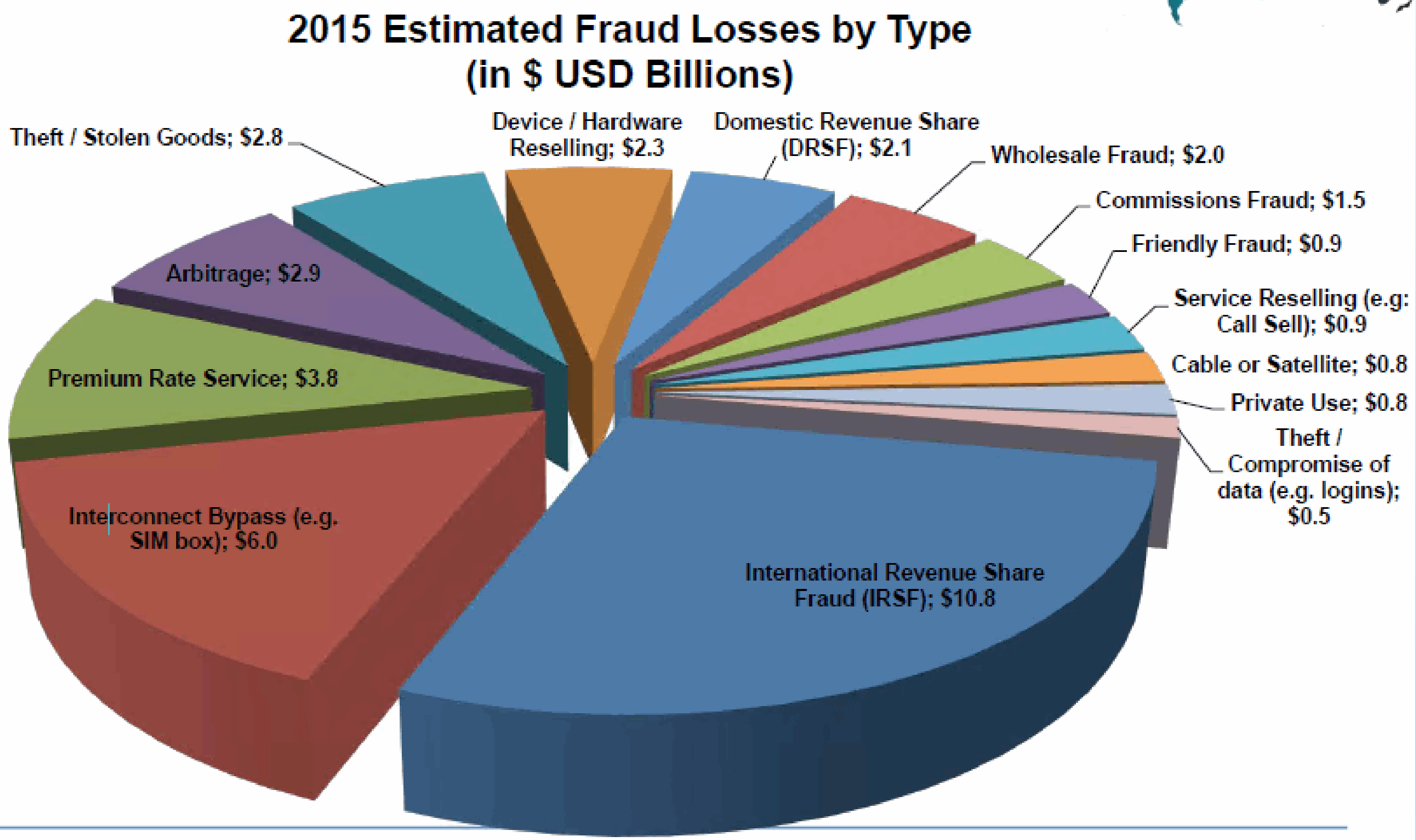


Response may indicate:

A SIP server is at this address

The type of product and version of software installed

2015 Survey


2015 Estimated Fraud Losses by Type
(in $ USD Billions)
Theft / Stolen Goods; $2.8
Device / Hardware Reselling; $2.3
Domestic Revenue Share (DRSF); $2.1
Wholesale Fraud; $2.0
Commissions Fraud; $1.5
Friendly Fraud; $0.9
Service Reselling (e.g: Call Sell); $0.9
Cable or Satellite; $0.8
Private Use; $0.8
Theft / Compromise of data (e.g. logins); $0.5
Arbitrage; $2.9
Premium Rate Service; $3.8
Interconnect Bypass (e.g. SIM box); $6.0
International Revenue Share Fraud (IRSF); $10.8

$10,800,000,000

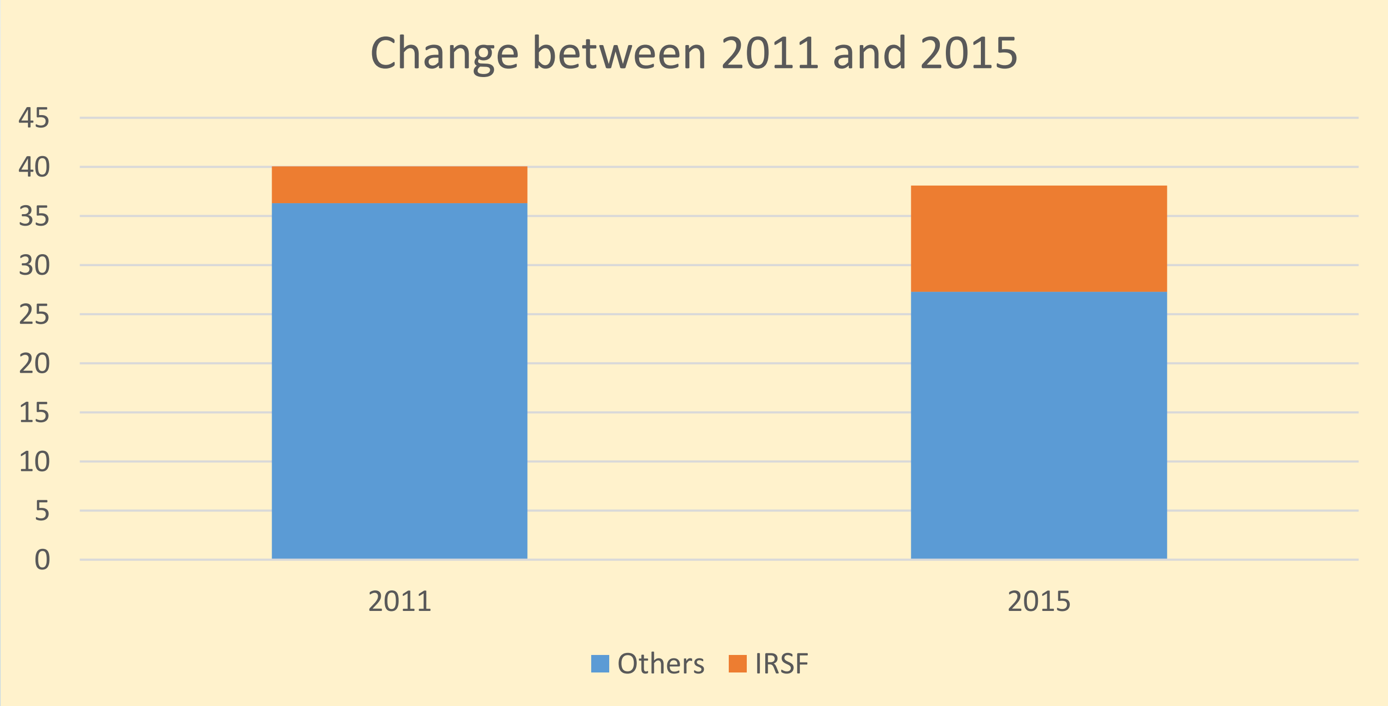
International Revenue Share Fraud

$9,890,000,000

Annual Government Expenditure Budget for Estonia for 2016*

* Source: Wikipedia

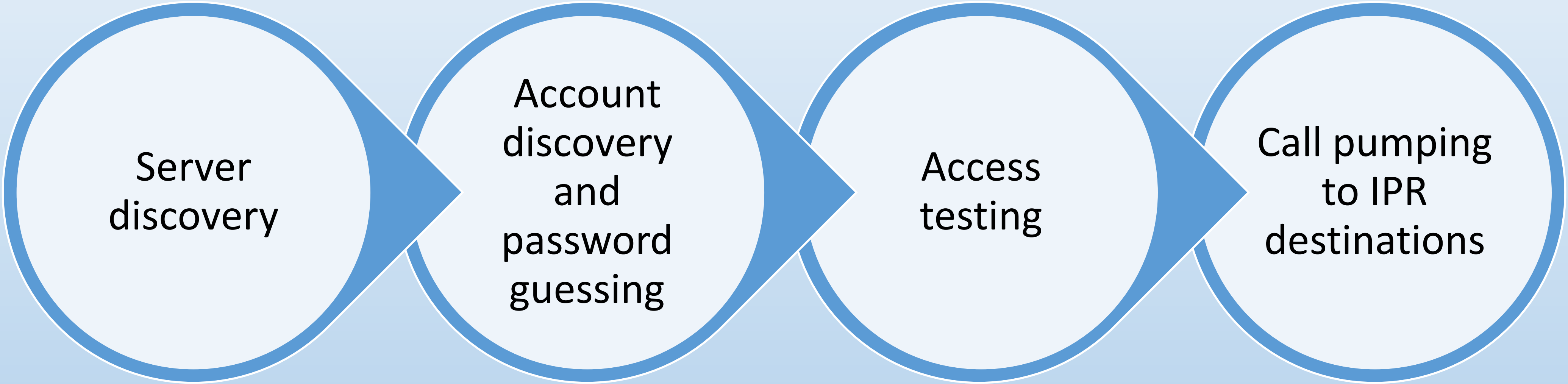
IRSF: Fastest growing telecoms fraud



	2011	2015
Estimated Total Global fraud loss	$ **40.1** Billion	$ **38.1** Billion
International Revenue Share fraud	$ **3.8** Billion	$ **10.8** Billion

Source: Communications Fraud Control Association

The phases of a typical attack



- OPTIONS / INVITE
- Within 1 hour
- 2 requests / hour
- Multiple sources

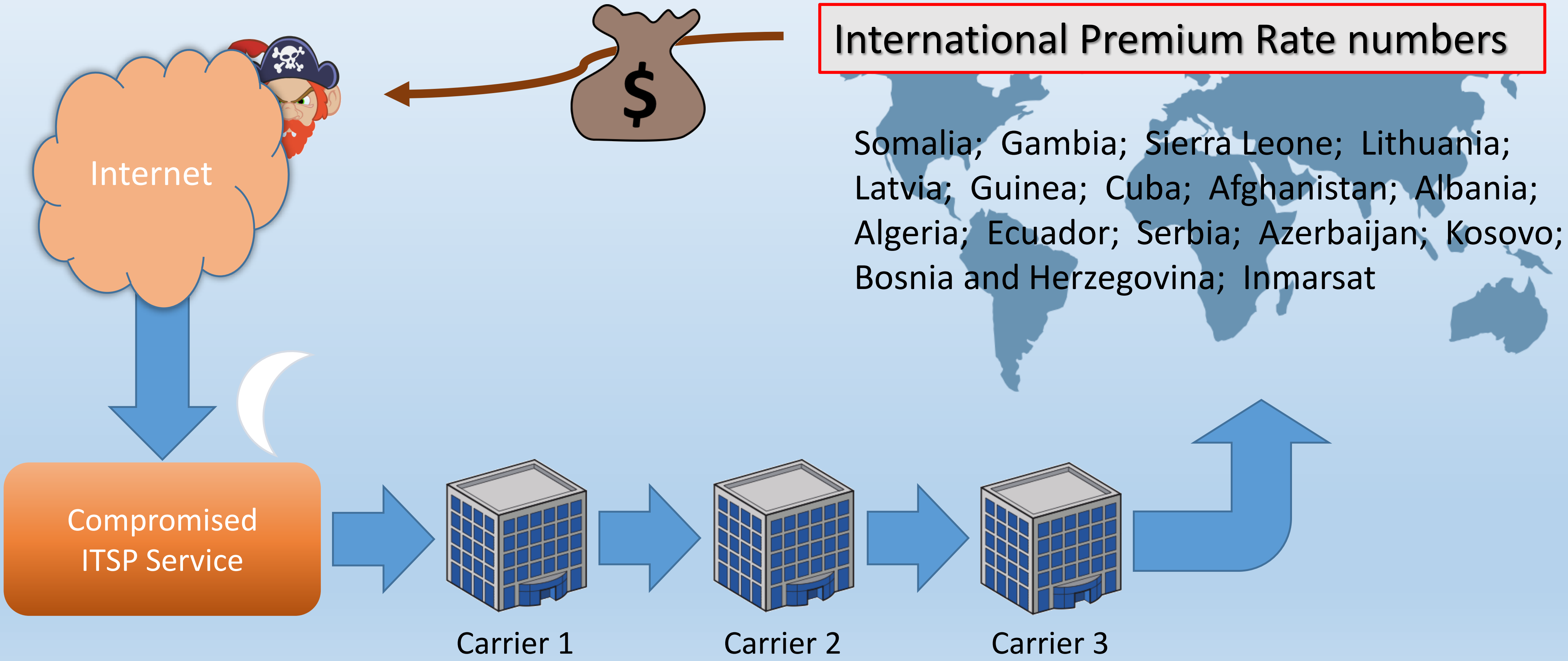
- REGISTER / INVITE
- After approx. 6 hours*
- 10,800 requests / hour*

- INVITE
- Depends on Phase 2
- 10's requests / day

- INVITE
- Could wait days or even weeks for the best opportunity
- As fast as your system will allow

* Source: Measured rate on an Asterisk server used for honeypot testing

Mechanics of the fraud (phase 4)



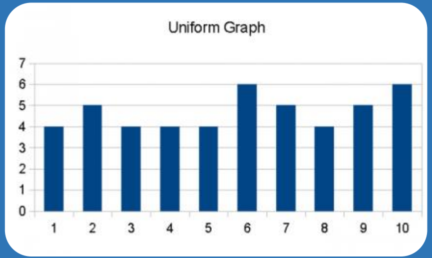
Some precautions possible with OpenSIPS

Detection of malicious SIP requests

- Identifying characteristics

Logging and automated inspection of logs

- xlog
- fail2ban

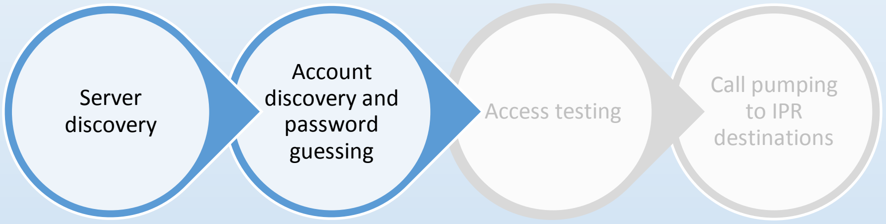


Call traffic analysis

- Automated analysis
- Fraud Detection Module; White and Black lists

Detect and Reject Malicious SIP Requests

Useful for: Phase 1 and Phase 2



Principles used: User-Agent header contains characteristic string

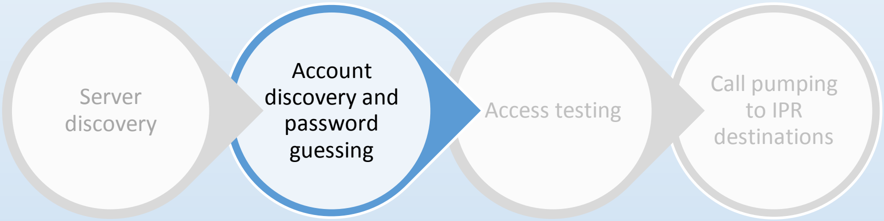
Coding example:

```
if ($ua =~ "friendly|sipcli|VaxSIPUser|VoIP.*v1")
        exit;

if ($fd == "1.1.1.1" || $fU == "nm")
        exit;
```

Logging Suspicious SIP Requests

Useful for: Phase 2



Principles used: Write to different log streams depending on category

L_INFO, L_NOTICE, L_WARN, L_ERROR, L_ALERT

Coding example:

```
xlog("L_INFO", "--> $rm  R-URI=$ru  LR Request (Rcvd $si:$sp) Call-ID=$ci\n");

xlog("L_WARN", "!--> !!WARNING: $rm request to $ru; Sender is NOT Registered.
                             (From $fu ($si:$sp) UA=$ua)  Call-ID=$ci\n");
```

Check return code from authorize functions

```
if (!www_authorize("","subscriber")) {
      switch ($retcode) {
        case -1:
          xlog("L_ALERT", "$si:$sp  Register_Unknown_User  From=$fu\n");
          break;
        case -2:
          xlog("L_ALERT", "$si:$sp  Register_Bad_Password  From=$fu\n");
      }
      www_challenge("","0");
      exit;
};
```

Mapping log_level parameter to log files

/etc/rsyslog.conf

```
# Send opensips messages to their own log files
$template MyTimeStamp,"%TIMESTAMP::date-pgsql% %msg%\n"
local0.=warn,local0.=notice                   -/var/log/opensips.log;MyTimeStamp
local0.=alert                                 -/var/log/osips_alert.log;MyTimeStamp
local0.err,local0.=info                       -/var/log/osips_error.log;MyTimeStamp
```

Log output designed for fail2ban

```
2016-04-20 19:04:13  195.154.182.164:5074  Invite_from_unknown_source  From=sip:115@72.15.16.29  RURI=sip:0041445209698@72.15.16.29  UA=sipcli/v1.8
2016-04-20 20:09:02  195.154.182.164:5074  Invite_from_unknown_source  From=sip:115@72.15.16.29  RURI=sip:00041445209698@72.15.16.29  UA=sipcli/v1.8
2016-04-20 21:17:00  195.154.182.164:5087  Invite_from_unknown_source  From=sip:115@72.15.16.29  RURI=sip:90041445209698@72.15.16.29  UA=sipcli/v1.8
2016-04-20 22:22:52  195.154.182.164:5070  Invite_from_unknown_source  From=sip:115@72.15.16.29  RURI=sip:090041445209698@72.15.16.29  UA=sipcli/v1.8
2016-04-20 23:29:12  195.154.182.164:5070  Invite_from_unknown_source  From=sip:115@72.15.16.29  RURI=sip:41445209698@72.15.16.29  UA=sipcli/v1.8
2016-04-21 07:39:26  212.129.2.189:9916  Register_Wrong_Domain  Contact=sip:7865432789@212.129.2.189:9916  UA=VoIP v11.2.4
2016-04-21 07:39:26  212.129.2.189:9916  Register_Wrong_Domain  Contact=sip:7865432789@212.129.2.189:9916  UA=VoIP v11.2.4
2016-04-21 13:43:00  163.172.194.75:6746  Register_Wrong_Domain  Contact=sip:202@163.172.194.75  UA=eyeBeam release 3006o stamp 17551
2016-04-21 13:43:20  163.172.194.75:6746  Register_Wrong_Domain  Contact=sip:202@163.172.194.75  UA=eyeBeam release 3006o stamp 17551
2016-04-21 13:44:00  163.172.194.75:6746  Register_Wrong_Domain  Contact=sip:202@163.172.194.75  UA=eyeBeam release 3006o stamp 17551
2016-04-21 14:53:22  46.166.165.79:5392  Register_Wrong_Domain  Contact=  From=sip:100@1.1.1.1  To=sip:100@1.1.1.1  UA=friendly-scanner
2016-04-21 20:16:19  80.241.209.130:5010  Register_Wrong_Domain  Contact=sip:100@80.241.209.130:5010  UA=VoIP SIP v12.1.0
2016-04-21 20:16:19  80.241.209.130:5010  Register_Wrong_Domain  Contact=sip:100@80.241.209.130:5010  UA=VoIP SIP v12.1.0
2016-04-21 20:16:20  80.241.209.130:5010  Register_Wrong_Domain  Contact=sip:100@80.241.209.130:5010  UA=VoIP SIP v12.1.0
2016-04-22 04:47:52  113.240.250.156:5495  Request_to_unknown_destination  RURI=sip:nm  Method=OPTIONS  UA=<null>
```

Fail2ban configuration

/etc/fail2ban/filter.d/osips-alert.conf

```
# Regexp to block repeat attempts sending invalid requests to OpenSIPS
[Definition]
failregex = <HOST>.*Invite_from_unknown.*$
            <HOST>.*Request_to_unknown_dest.*$
            <HOST>.*Register_Wrong_Domain.*$
            <HOST>.*Register_Bad_Credential.*
ignoreregex =
```

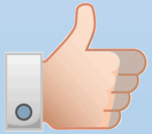
/etc/fail2ban/jail.d/osips-alert.conf

```
[osips-alert]
enabled  = true
filter   = osips-alert
action   = iptables[name=osips-alert, port="5060", protocol=udp]
logpath  = /var/log/osips_alert.log
maxretry = 3
findtime = 7200
bantime  = 10800
```

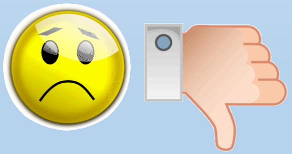
Avoiding the weaknesses of fail2ban

```
avp_db_query("select username from location where received like 'sip:$si%'","$avp(usrname)");
if (is_avp_set("$avp(usrname)"))
    # Address is being used by a registered device so should not be blocked

avp_db_query("select username from location where contact like 'sip:%$si%'", "$avp(usrname)");
if (is_avp_set("$avp(usrname)"))
    # Address found in the contact field of the location table – don’t block
```



Avoids blocking a legitimate customer



Might give hackers a way around your security

Why do we need Call Traffic Analysis?

Because Customers are a weak link

- Their equipment may get hacked
- User Account Credentials may get stolen

Because we make silly mistakes

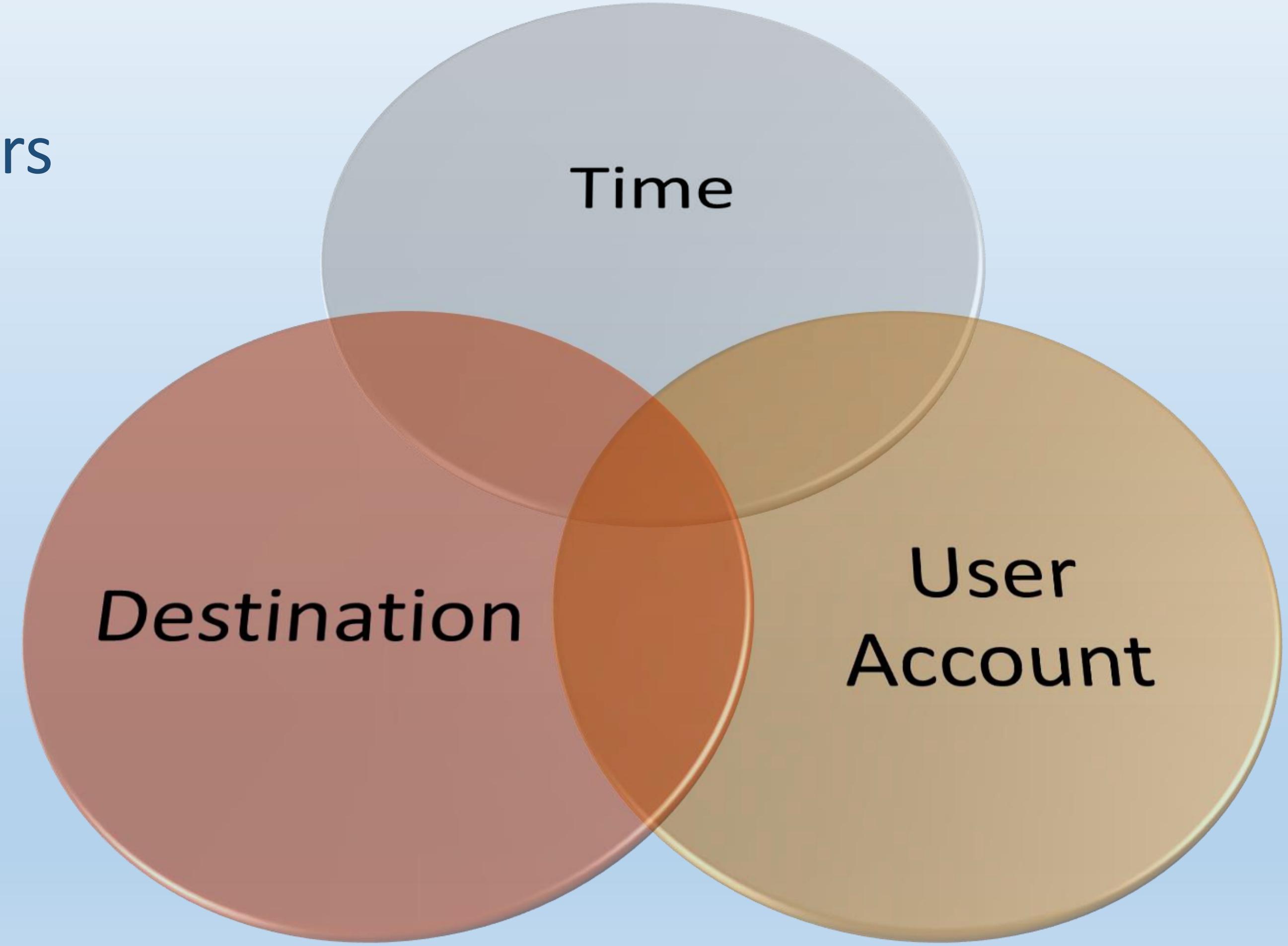
- Maintenance of firewall rules
- Vulnerability when infrastructure is changed

Because the hackers are clever and inventive

- May hack your Provisioning Server
- May set up a customer account with no intention of paying

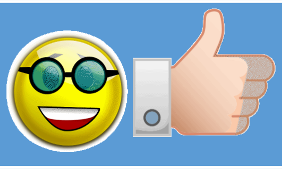
Fraud Detection through Traffic Analysis

3 qualifying parameters



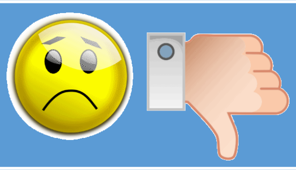
OpenSIPS Fraud Detection Module

Good



- Includes Time (hour and day), Destination and User Account
- Measures Total & Concurrent Calls; also Calls/Minute
- Data driven

Not so good!



- Only available in v2
- Users or Destinations: Cannot be assigned to a group (categorized)
- Destinations: Prefix only – not a regex
- Metric for Total does not persist through a restart

Concurrent Call Counting using Dialog Module

```
loadmodule "dialog.so"

modparam("dialog", "log_profile_hash_size", 8)          # allows up to 256
modparam("dialog", "profiles_with_value", "concurrent")

route[]
  # Remember the User a/c (using $au) and set a flag for onreply
  store_dlg_value("UserAccount", "$au");
  set_dlg_flag("12");
  # Get the number of concurrent calls already active for this user account
  get_profile_size("concurrent", "$au", "$var(numcalls)");
  # Test if number of concurrent calls is within permitted limit..

onreply_route[]
  if ($rm=="INVITE" && status=="200" && is_dlg_flag_set("12")) {
      fetch_dlg_value("UserAccount", "$var(profvalue)");
      set_dlg_profile("concurrent", "$var(profvalue)");
      reset_dlg_flag("12");
  }
```

Bespoke solution: Cached Categorisation List

Startup_route

- Load data into local cache – e.g. from a DB table
- Each record is a value/category pair

Main route
INVITE handler

- Set 'User Account' – e.g. using $au (Auth User) or $si (Source IP)
- Use cached data to set a category – e.g. using $rU (dialed number)
- Create a compound "key" to use in set_dlg_profile()
- Use 'Concurrent Call Counting' to track number of calls in selected category from that User

Bespoke solution: Coding ideas/suggestions

```
loadmodule “cached_local.so"

modparam("cachedb_local", "cache_table_size", 10)   # Allows over 1000 entries

startup_route {
    # Create a cached list of Destination prefixes – e.g. Sierra Leone & Somalia
    cache_store("local", "+616", "HIGHRISK");
    cache_store("local", "+686", "HIGHRISK");
}

route[]
     # Check destination number against cached list of prefixes – for example:
     $var(testval) = $(rU{s.substr,0,4});
     cache_fetch("local","$var(testval)",$avp(retval))
...
     $var(compoundkey) = $au + $avp(retval)
     store_dlg_value("UserAccount", "$var(compoundkey)");
     set_dlg_flag("12");
     get_profile_size("concurrent", "$var(compoundkey)", "$var(numcalls)");
```

Third Party Solutions

Is fraud protection provided by your Carrier?

How does it work – technically and commercially?

Do they actively research/improve fraud prevention solutions?

e.g. Simwood

Bolt-on Solutions

e.g. FRS Labs PRISM database

Please don't get caught by hackers

The End

Thank you