OpenSIPS For Asterisk Users

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Who we are

- 3 Companies sitting on top of VoIP Network
  - Localphone
    - Retail ITSP offering (VoIP accounts, apps, DIDs in UK, US, Europe, Worldwide)
    - Over 1,000,000 users
  - Voxbeam
    - Wholesale, A-Z Termination, VoIP reseller
    - US CLEC
    - Terminate ~20,000,000 mins/week internationally

- We use
  - OpenSIPS
  - Asterisk
  - FreeSWITCH
  - RabbitMQ
  - Redis
  - Hadoop
  - Homer (awesome!)
  - Sangoma (Transcoding)

I have Asterisk…

… so why do I need OpenSIPS?

OpenSIPS Summit / Astricon 2013, Atlanta

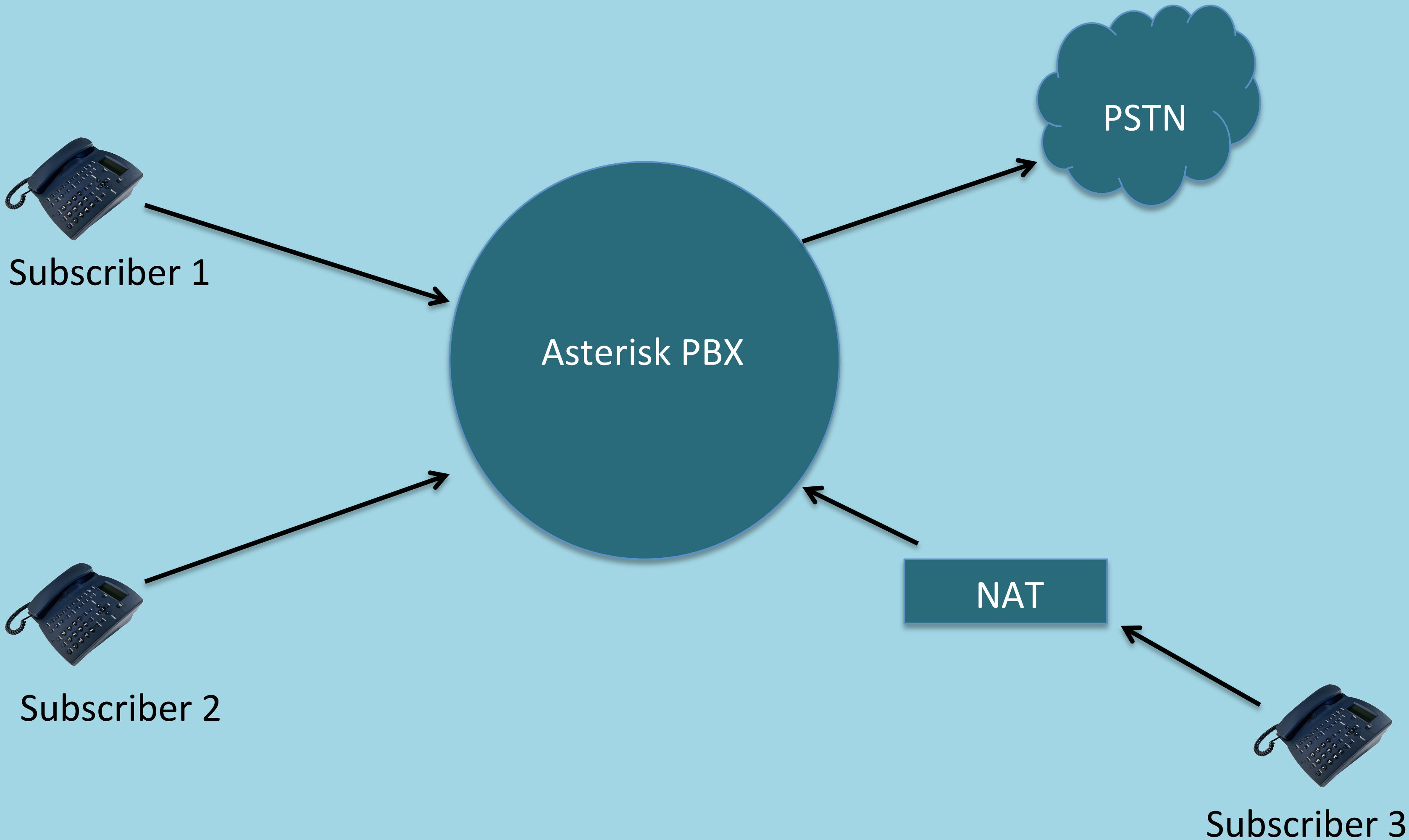
Asterisk is a:

- UAC/UAS
- B2BUA
- PBX
- Media Endpoint
- SIP Registrar

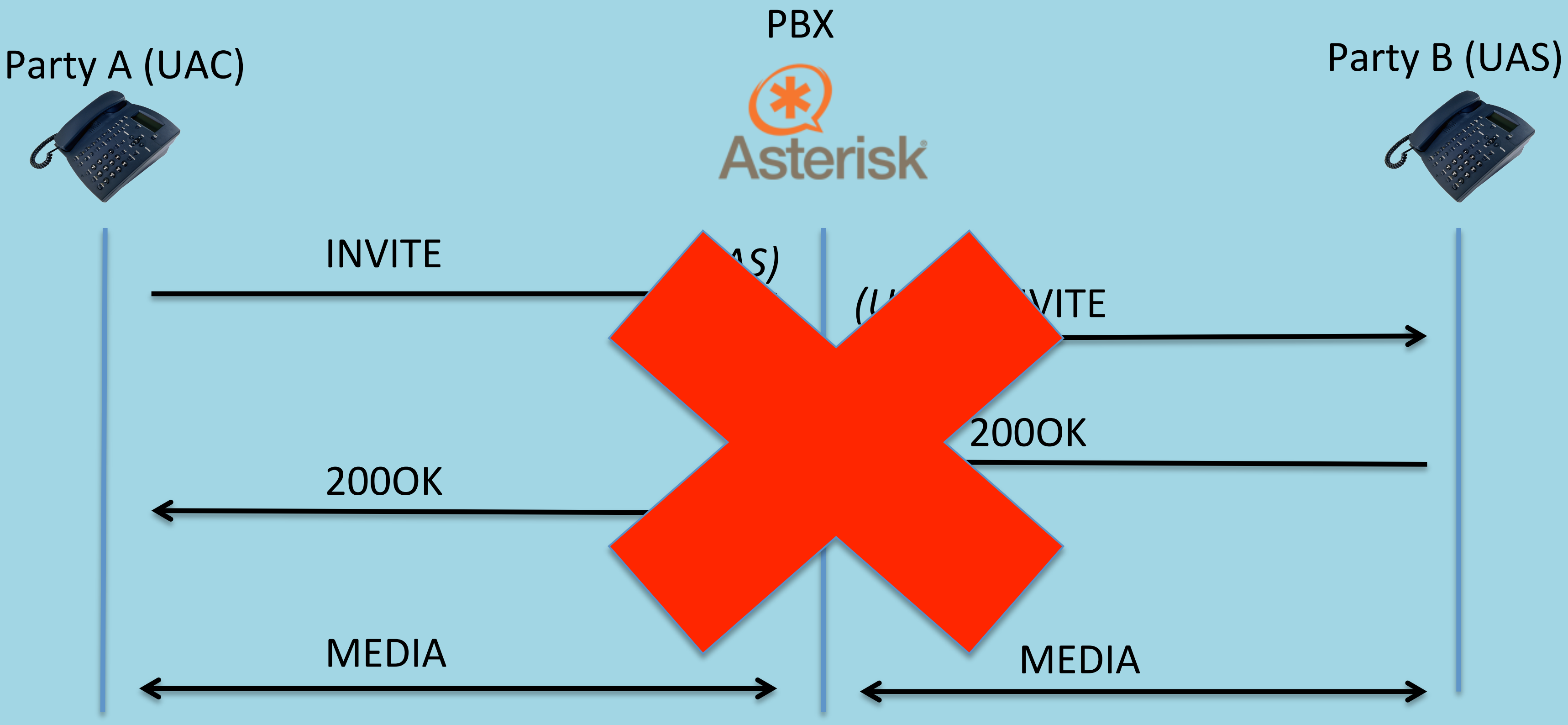
OpenSIPS is a:

- SIP Proxy
- SIP Registrar

“Basic” Asterisk setup

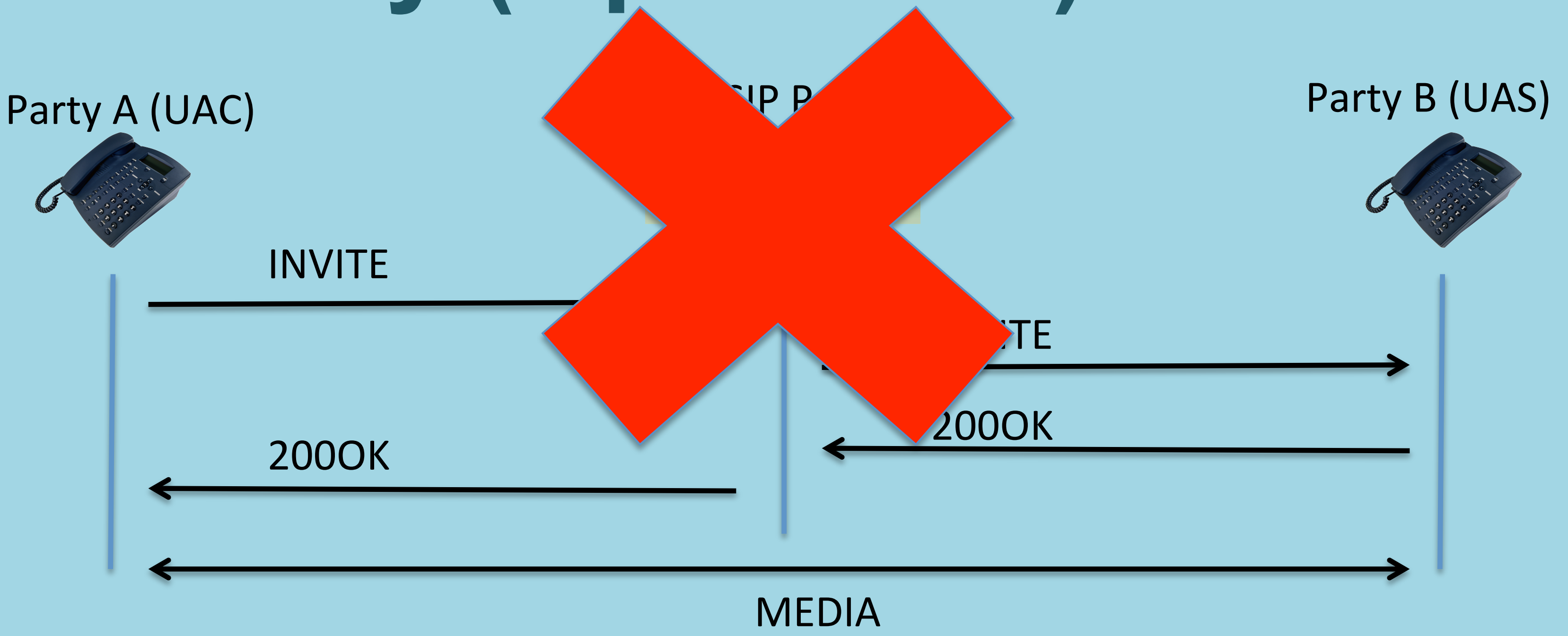


PBX (Asterisk)



- **2 Calls**
  - **Call 1: Party A -> Asterisk**
  - **Call 2: Asterisk -> Party B**

SIP Proxy (OpenSIPS)



- Only 1 call (Party A > Party B)
- Media keeps flowing
- Proxy is Stateless
  - Once the call is established, it stays up.
- When OpenSIPs comes back up, the BYE will still proxy!

OpenSIPS Flexibility (some features)

Load Balancing

B2BUA

SIP Manipulation

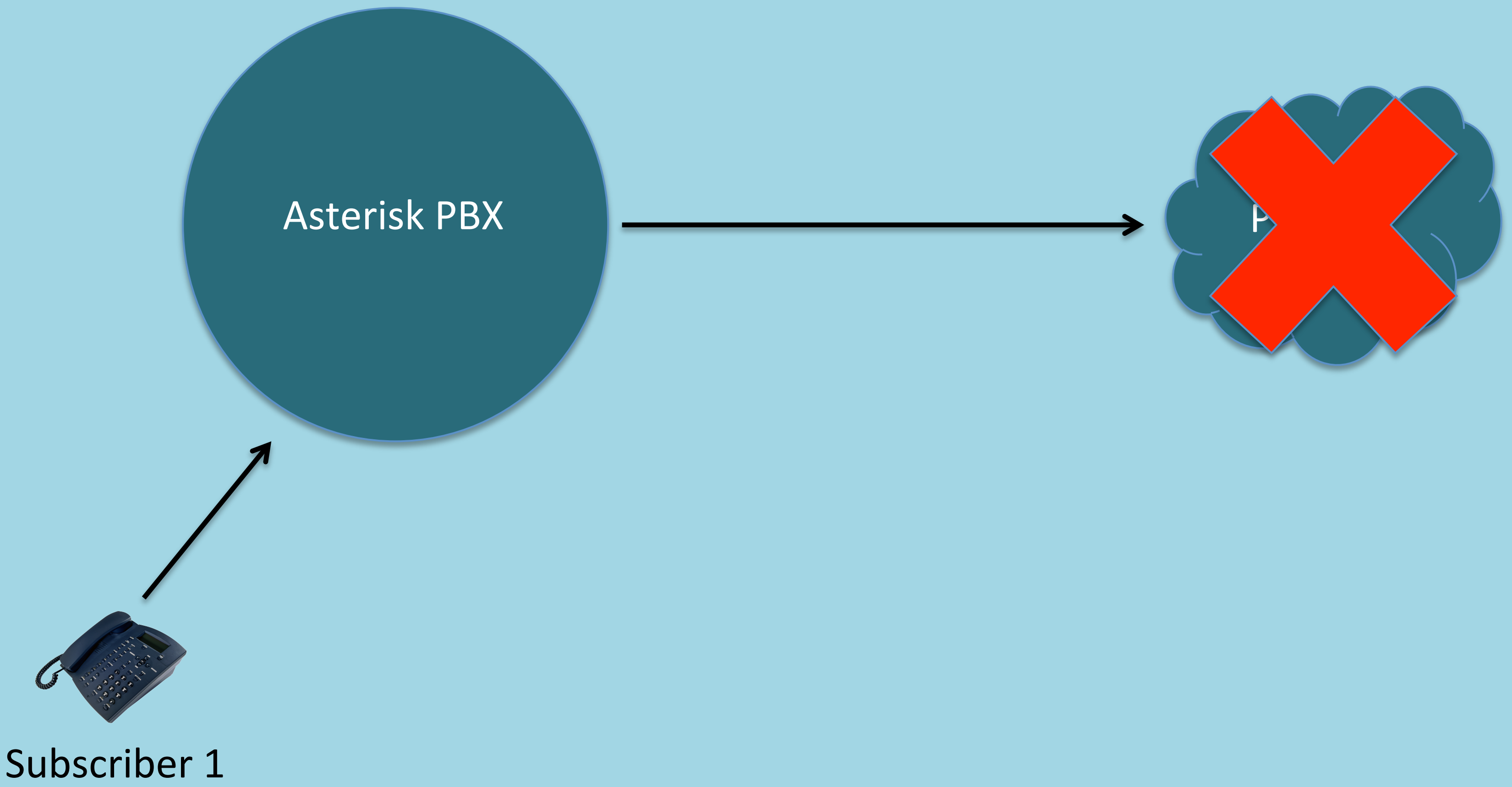
Topology Hiding

Dialplans

REGISTRAR Support

Dynamic Routing (LCR)

Dynamic Routing



Dynamic Routing

Gateways

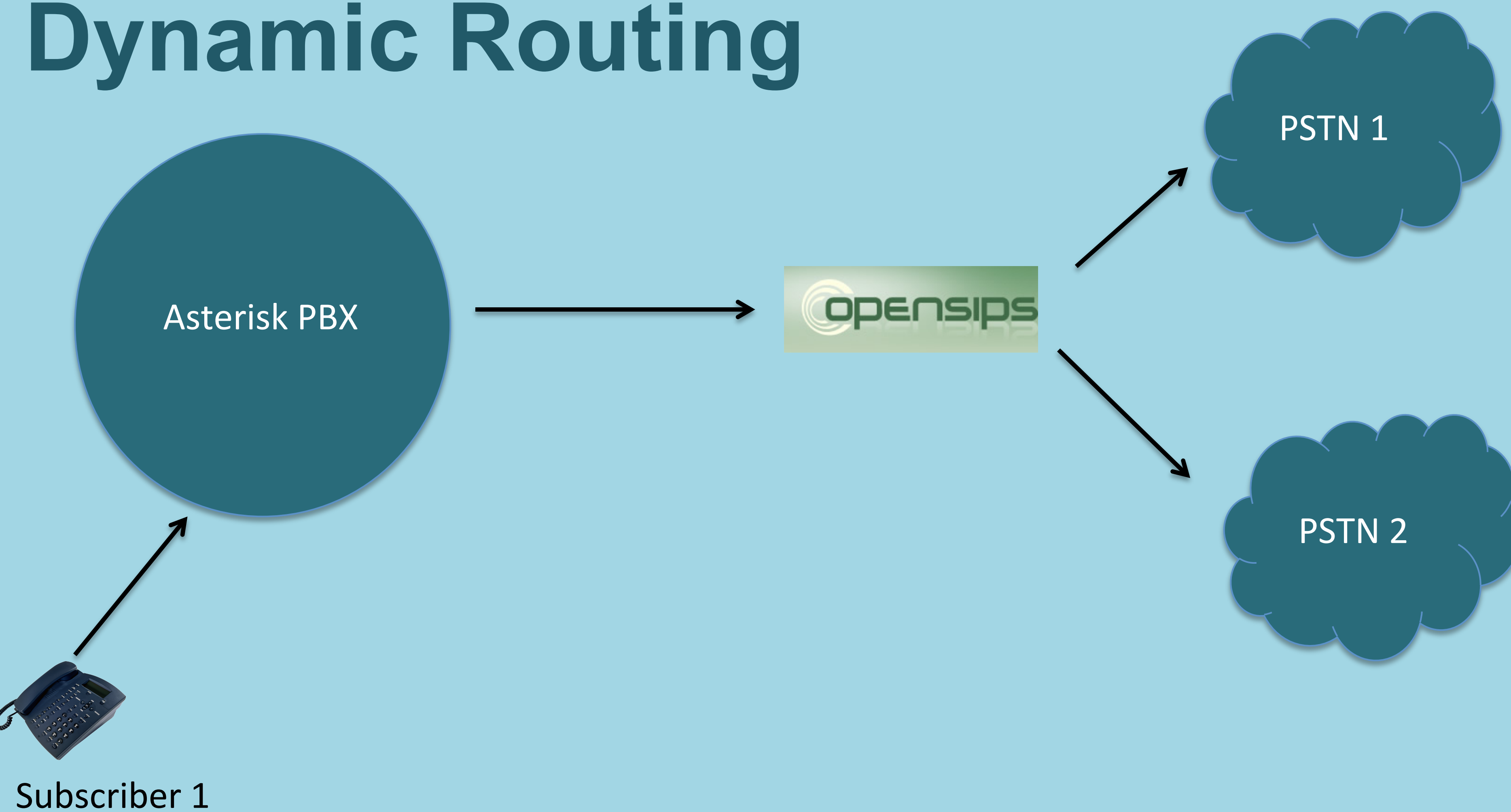
Gateway ID	Gateway Name
1	PSTN 1
2	PSTN 2

Prefixes

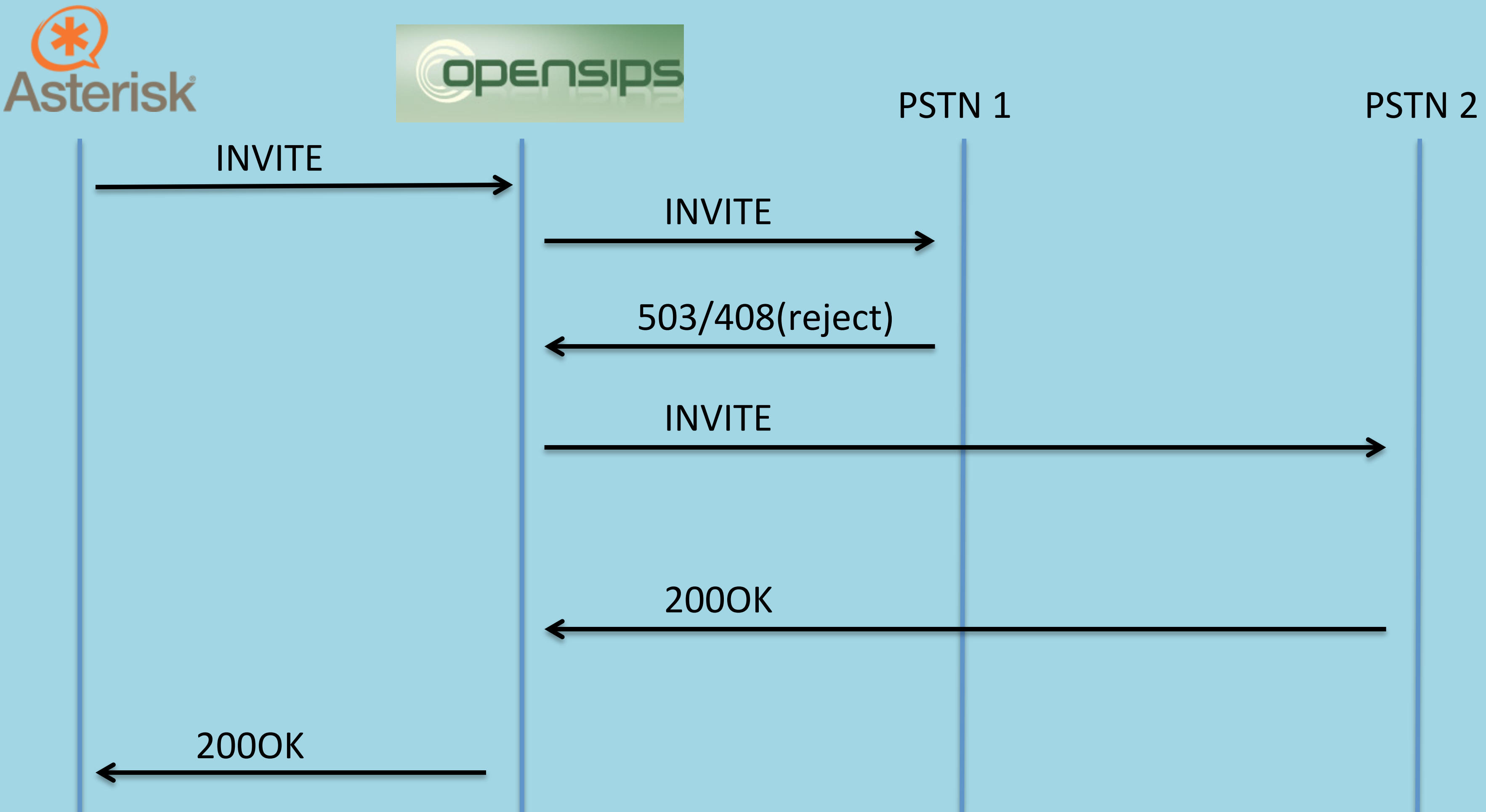
Prefix	Gateway List
1407	1,2
44	2
33	1,2
91	2,1

- Call Orlando(1407) – both carriers can be used. Try 1 then 2
- Call the UK, only carrier 2 can be used
- Call France or India – both carriers can be used. Try 2 then 1

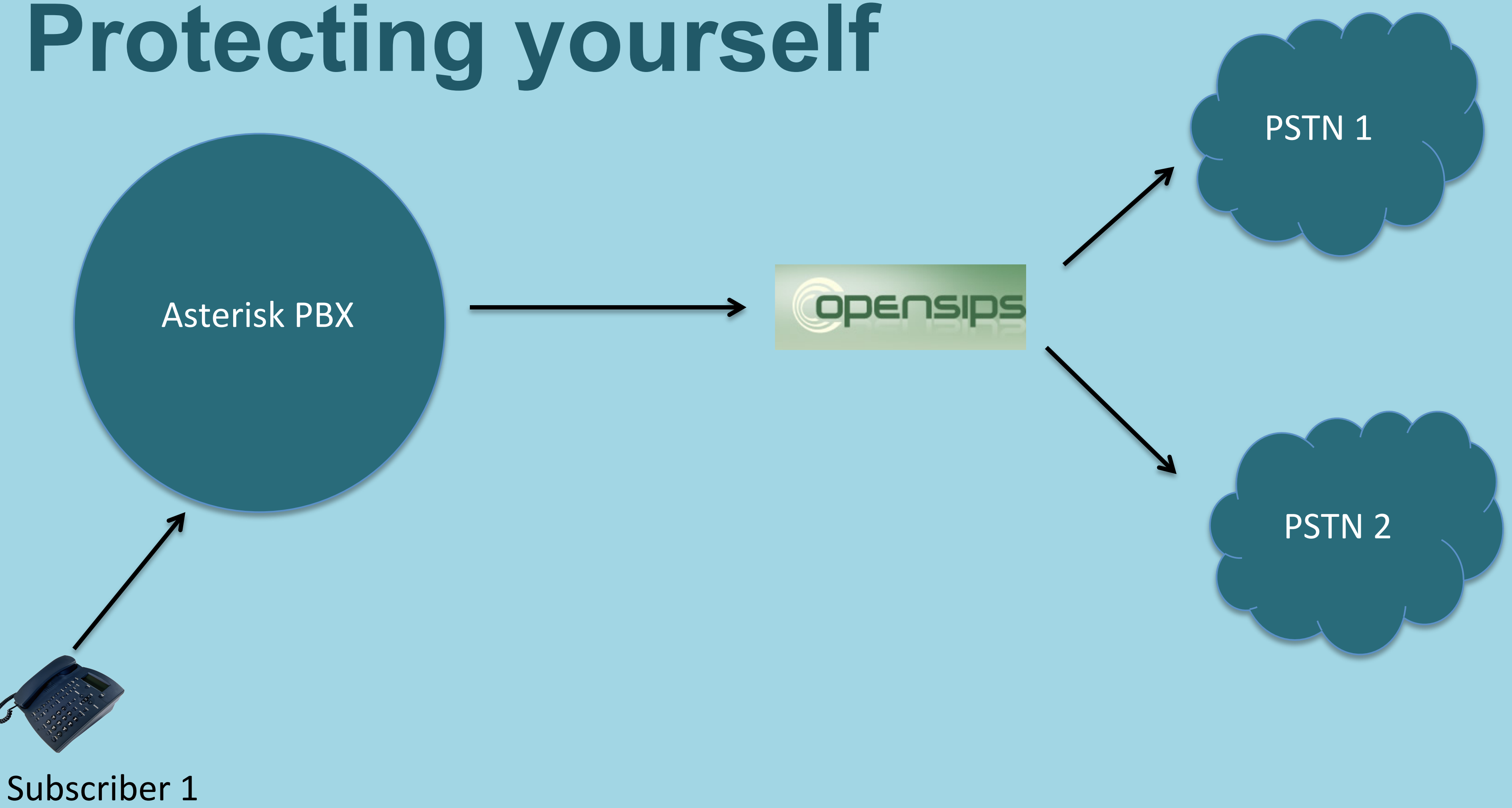
Dynamic Routing



Dynamic Routing



Protecting yourself



Protecting yourself

```
INVITE sip:22796650943@pstn1.voxbeam.com:5060 SIP/2.0
Record-Route: <sip:192.168.1.135;lr;ftag=as77e746e2;did=14b.c50b5e54>
Via: SIP/2.0/UDP 192.168.1.135;branch=z9hG4bK11ae.d61e2086.0
Via: SIP/2.0/UDP 192.168.1.133;branch=z9hG4bK11ae.7e078732.0
Max-Forwards: 67
From: "unknown" <sip:2348129246861@154.50.206.66>;tag=as77e746e2
To: <sip:0011101227966509430192.168.1.133>
Contact: <sip:caller@192.168.1.133>
Call-ID: 6a76c149049997c47da398ad2393691c@154.50.206.66:5060
CSeq: 102 INVITE
User-Agent: Asterisk
Date: Thu, 03 Oct 2013 19:52:18 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY,
INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 321
```

Protecting yourself

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INVITE sip:22796650943@pstn1.voxbeam.com:5060 SIP/2.0
Max-Forwards: 67
From: "unknown" <sip:2348129246861@154.50.206.66>;tag=as77e746e2
To: <sip:001110122796650943@192.168.1.133>
Contact: <sip:caller@192.168.1.135;did=14b.a2521543>
Call-ID: 6a76c149049997c47da398ad2393691c@154.50.206.66:5060
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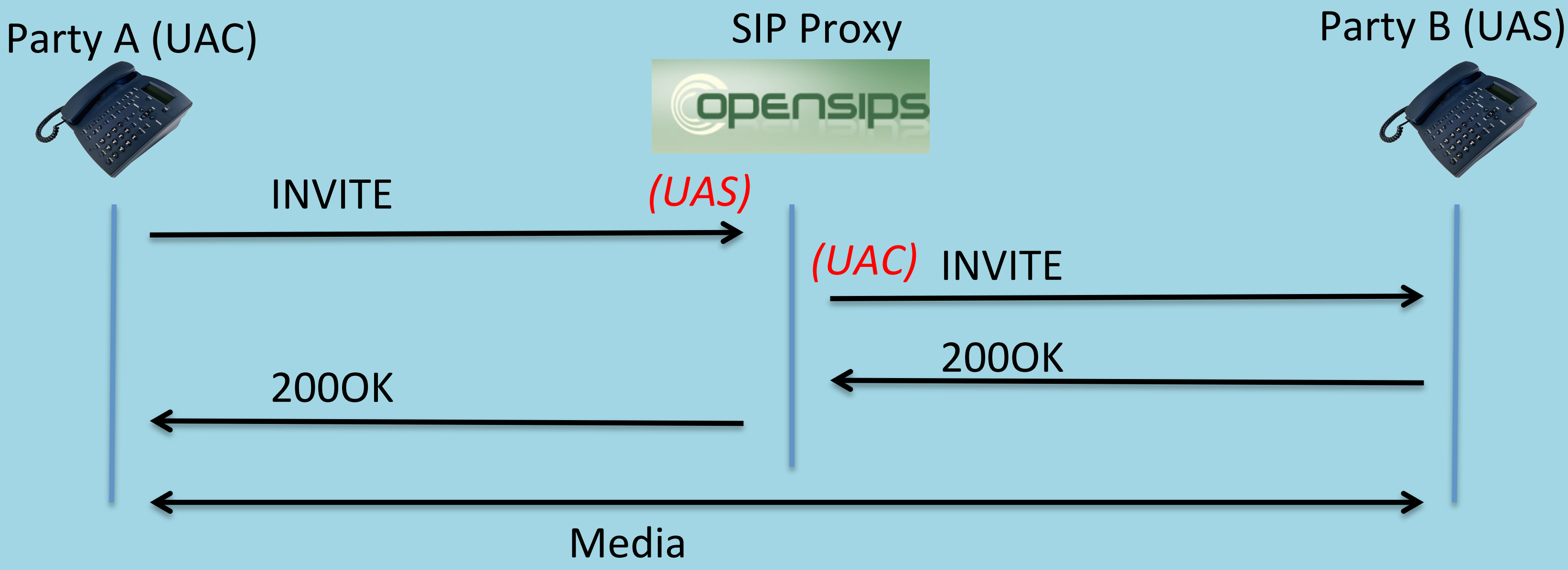
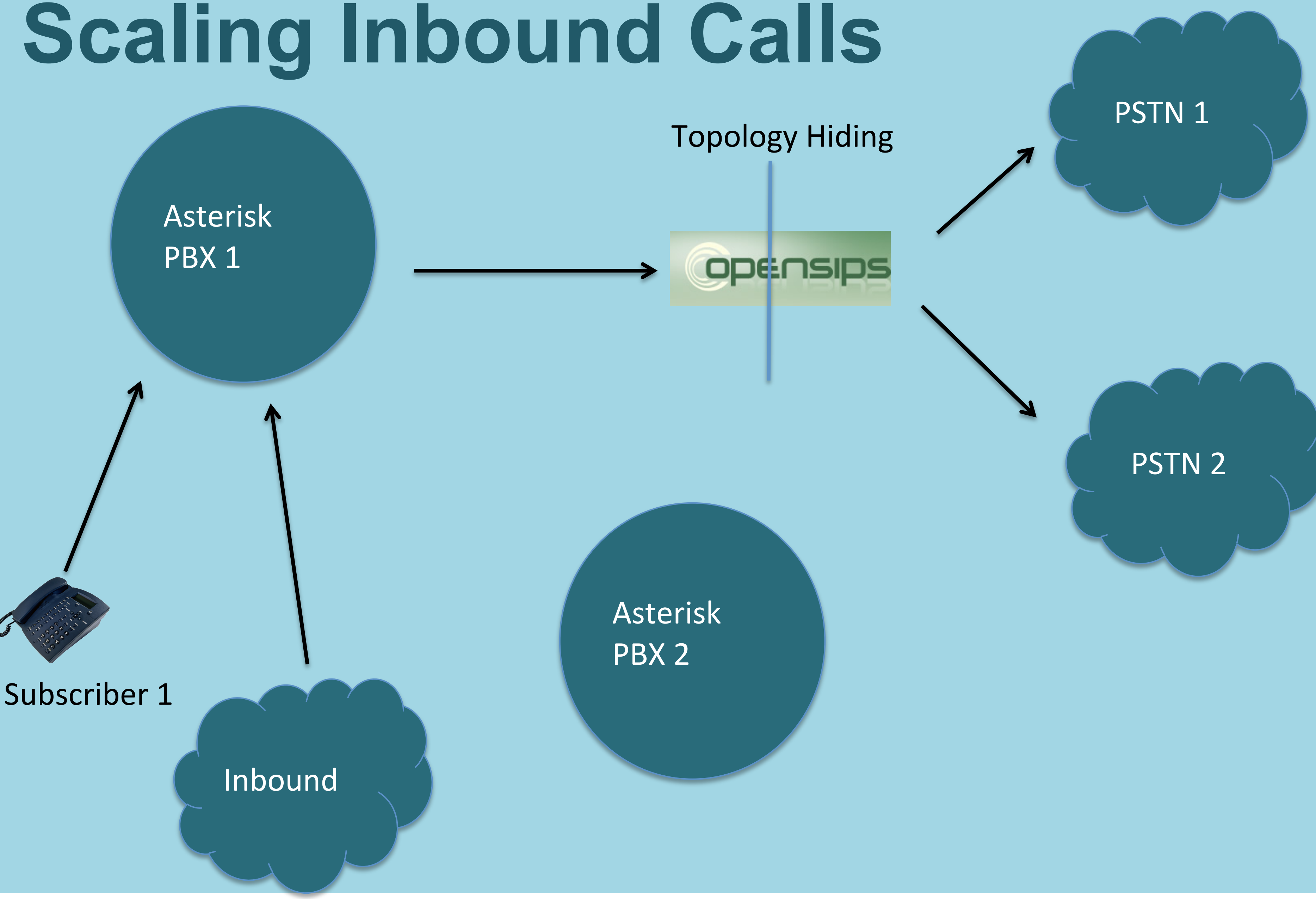
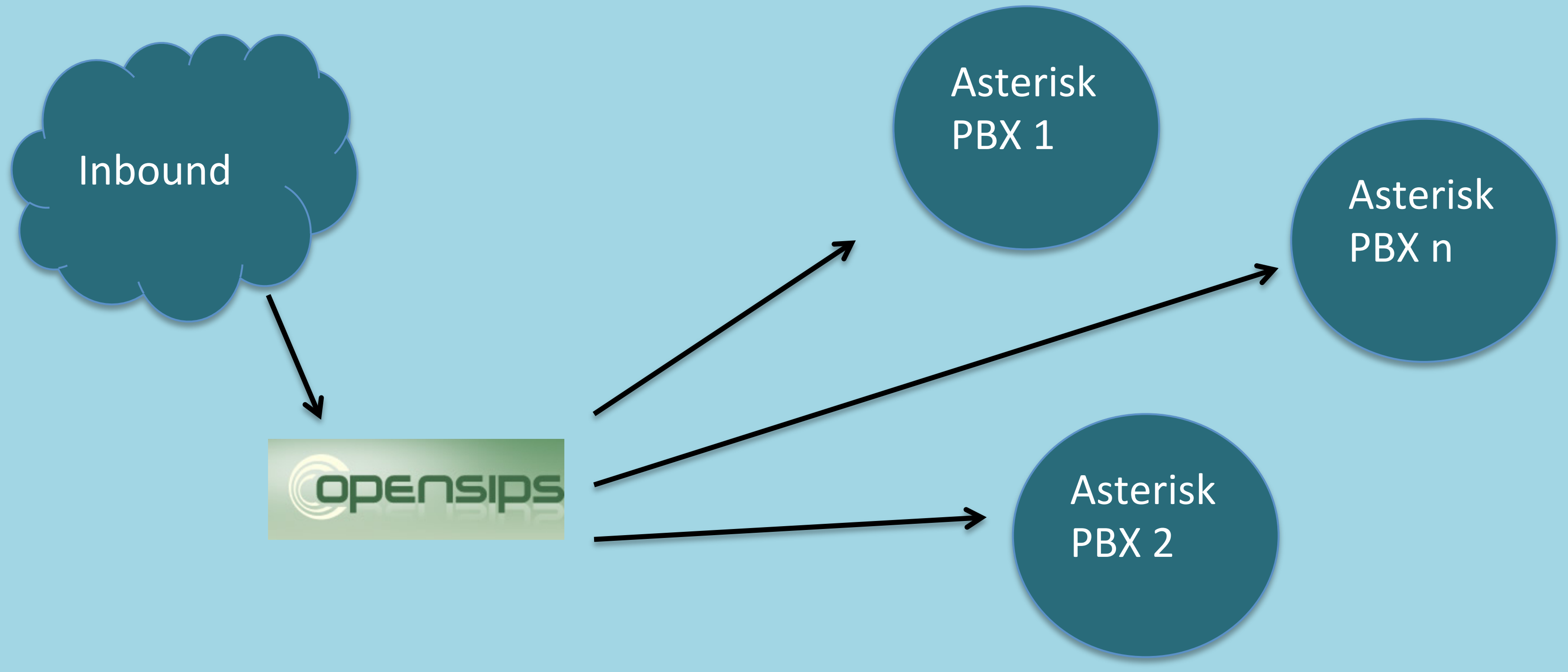
Protecting yourself



Scaling Inbound Calls

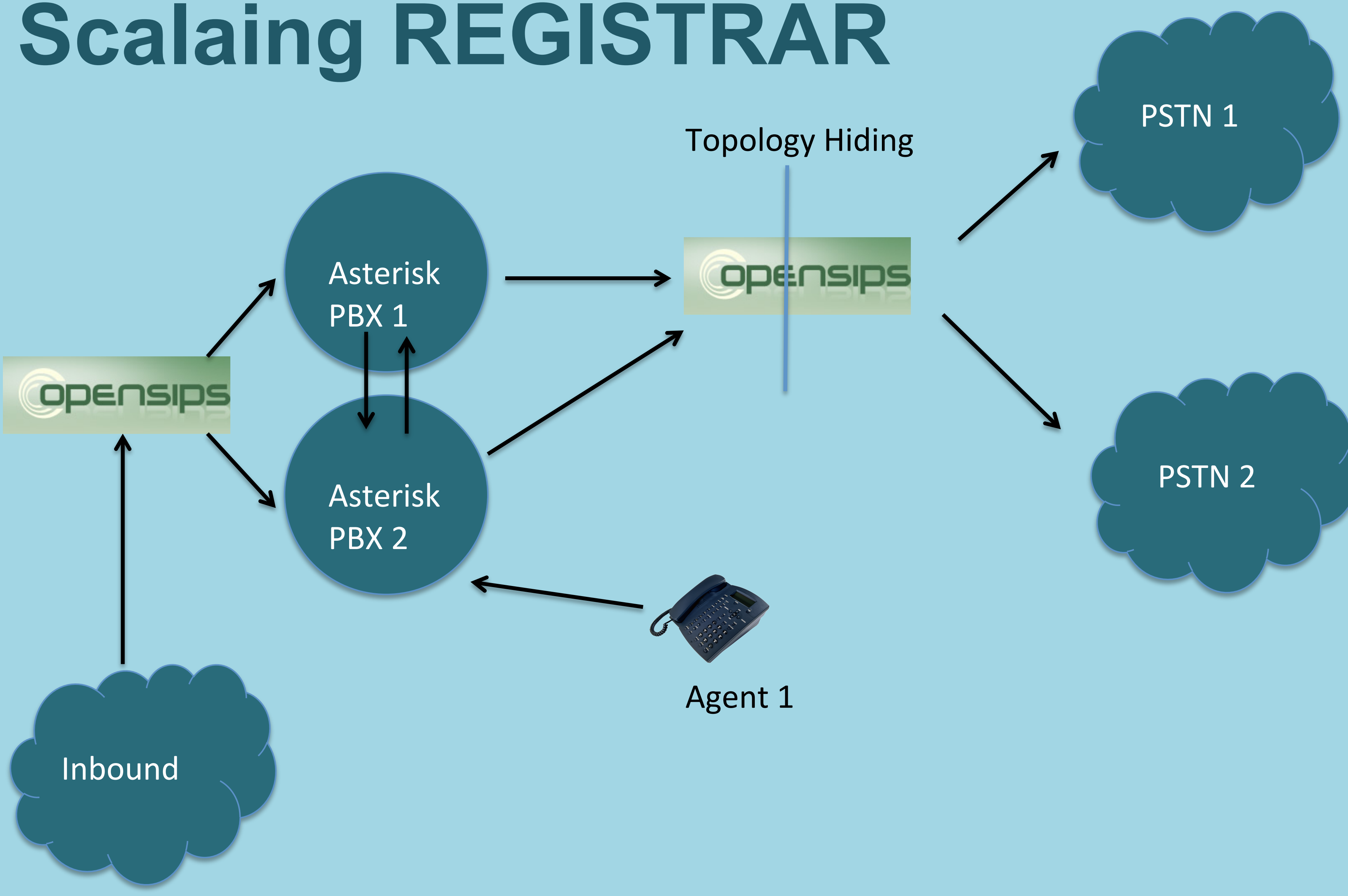


Scaling Inbound Calls

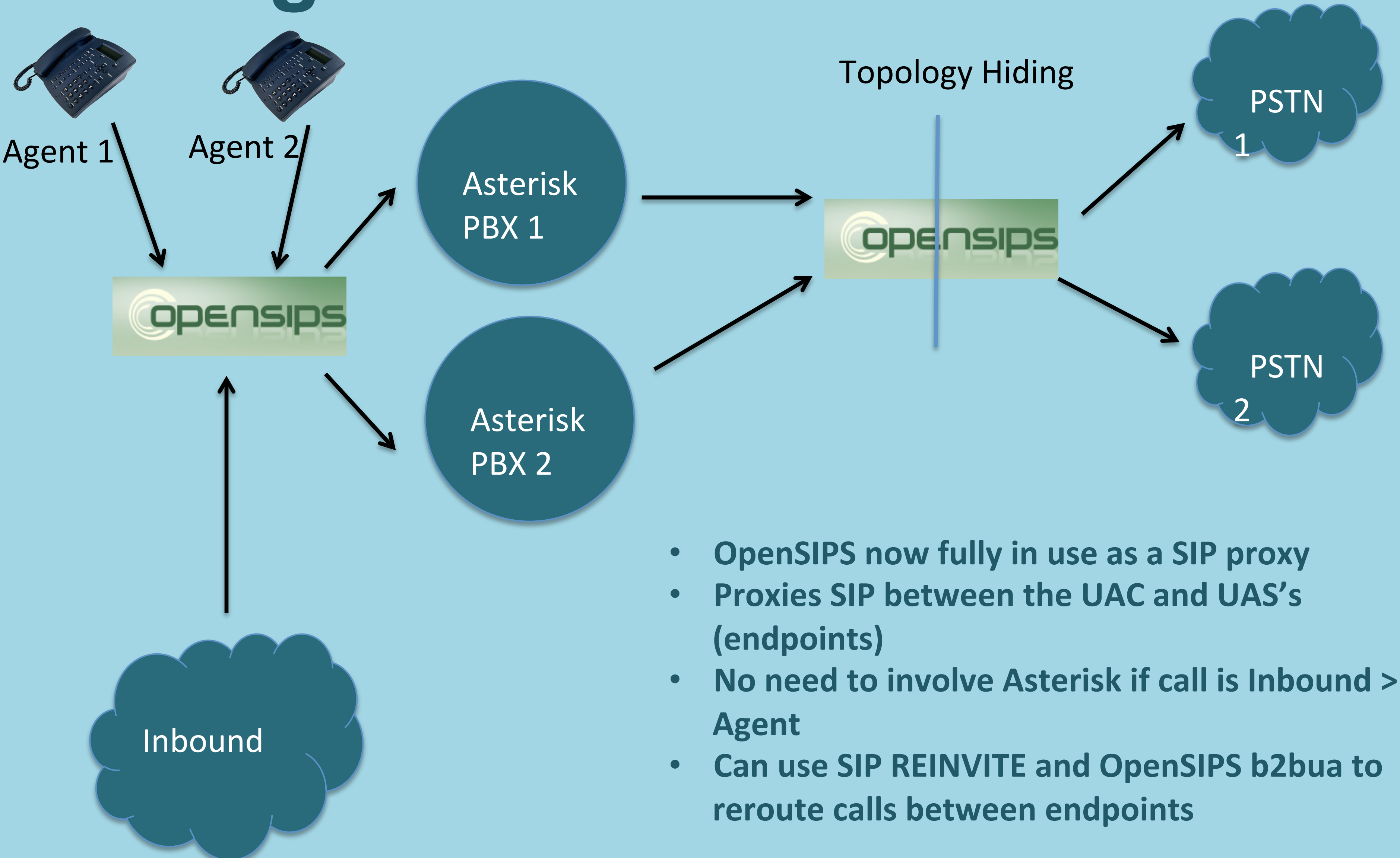


- Use Dispatcher module
  - Round Robin
  - Other algorithms available
- Check Asterisk is alive with OPTIONS ping

Scalaing REGISTRAR



Scaling REGISTRAR



- OpenSIPS now fully in use as a SIP proxy
- Proxies SIP between the UAC and UAS's (endpoints)
- No need to involve Asterisk if call is Inbound > Agent
- Can use SIP REINVITE and OpenSIPS b2bua to reroute calls between endpoints

Other Cool Stuff…High Availability

- OpenSIPS is Stateless
  - This means OpenSIPS server can be restarted and calls still flow!
- In case of hardware failure… lots of options
  - Pair your OpenSIPS box with another
  - Share REGISTRAR/call information using DB/Redis
  - Share an IP between the two boxes
    - Use open source VIP tools do to this
- In case of Data Centre failure…!
  - Use redis/cassandra support to share REGISTER between DC

Other Cool Stuff… SIP Manipulatio

- Add headers
- Remove headers
- Change request URI's
- Modify From headers
- Modify To headers
- Extremely Flexible scripting language
    - Bogdan to show some examples later

The Best Part… OpenSIPS is *FAST

- OpenSIPS is really really fast
  - Regularly over 8000 CPS
- OpenSIPs is really really reliable,
  - Typically months between restarts (usually to add feature)
- Tens of thousands of call setups/teardowns per second
- All config loaded into memory
- Optimised for reduced IO

- See OpenSIPS website for speed and CPS lab tests

Thank You…

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