Scaling Asterisk with OpenSIPS

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Asterisk

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Asterisk Features


Hell yeah!
All I need is !

Asterisk Platform

Asterisk Platform

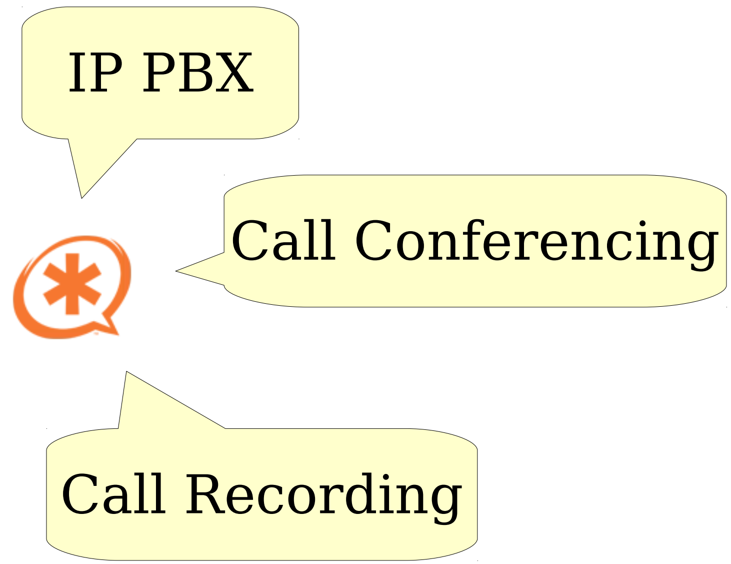
IP PBX

Asterisk Platform

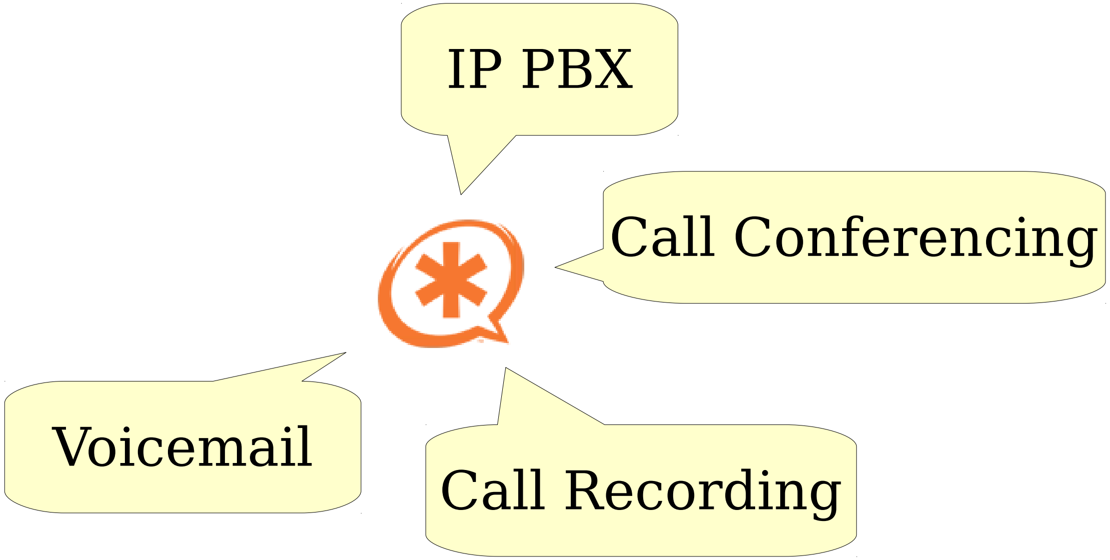
IP PBX

Call Conferencing

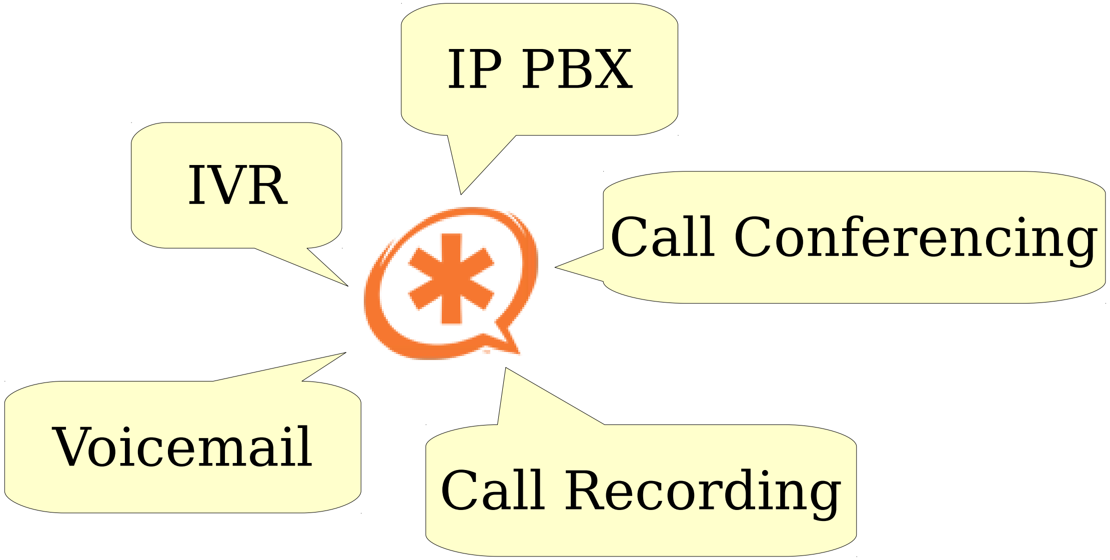
Asterisk Platform



Asterisk Platform

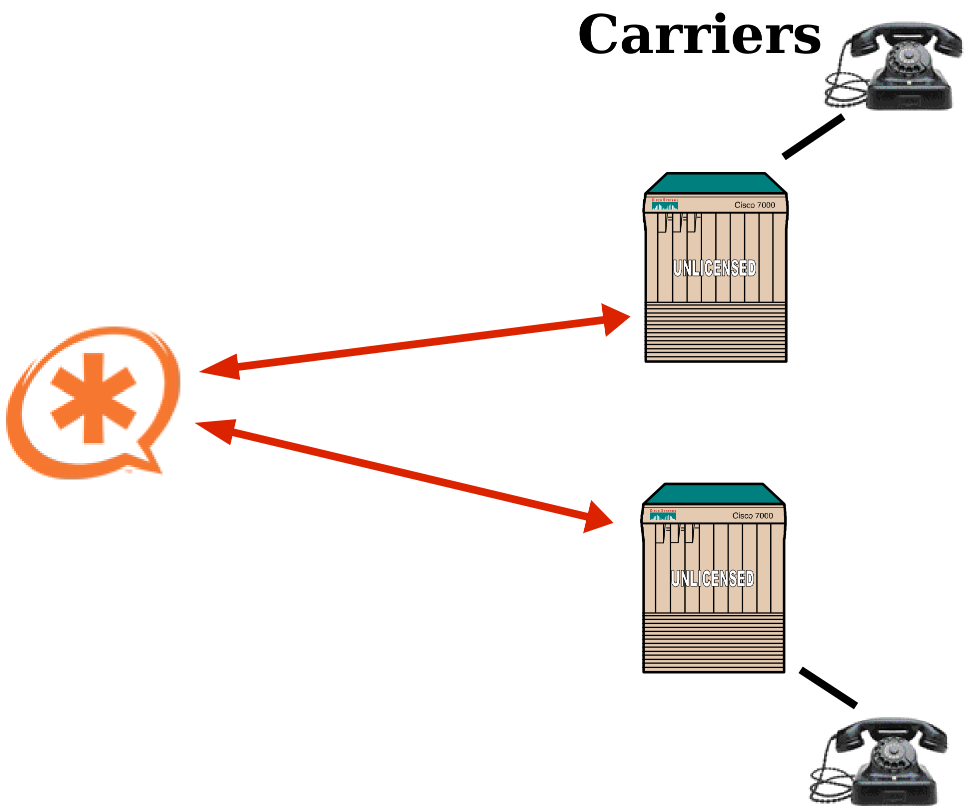


Asterisk Platform

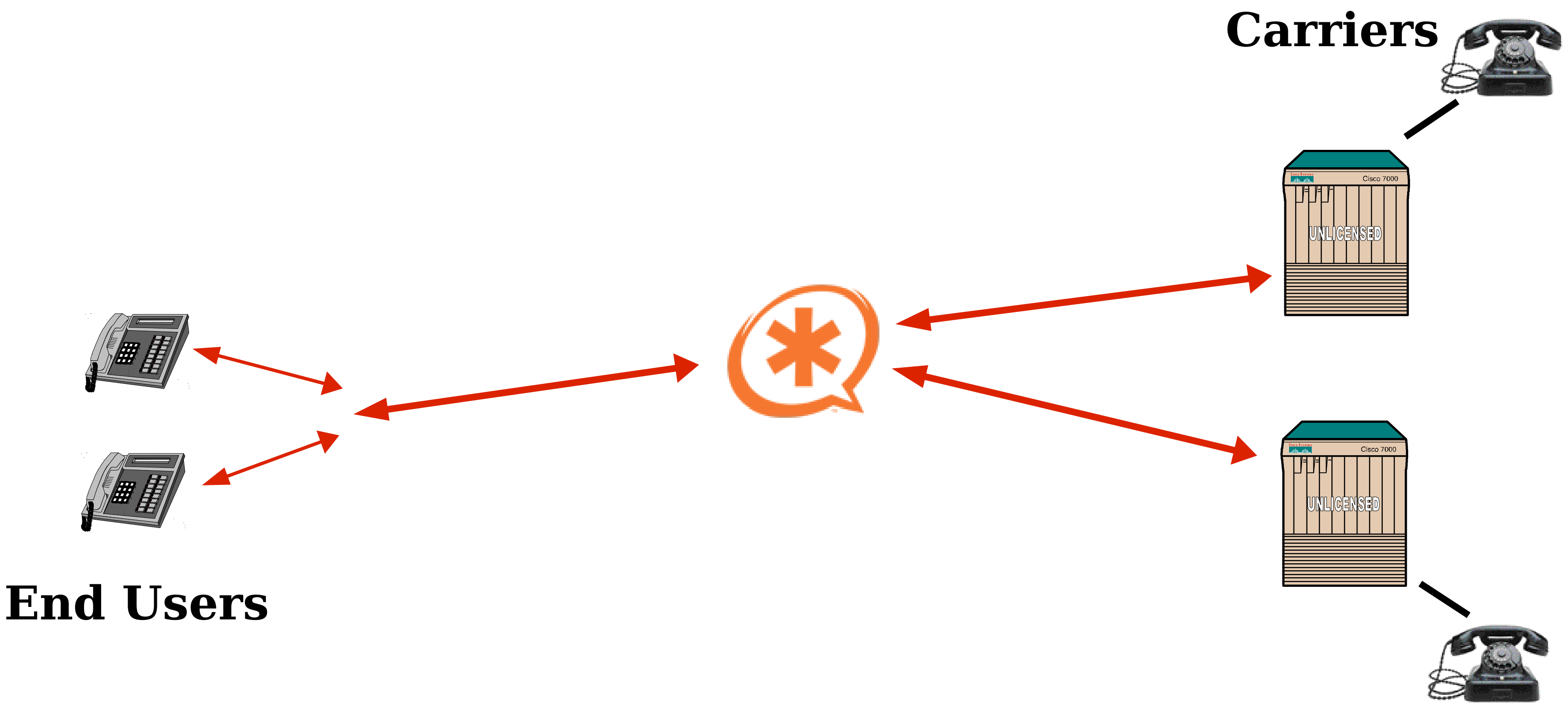


Asterisk Platform

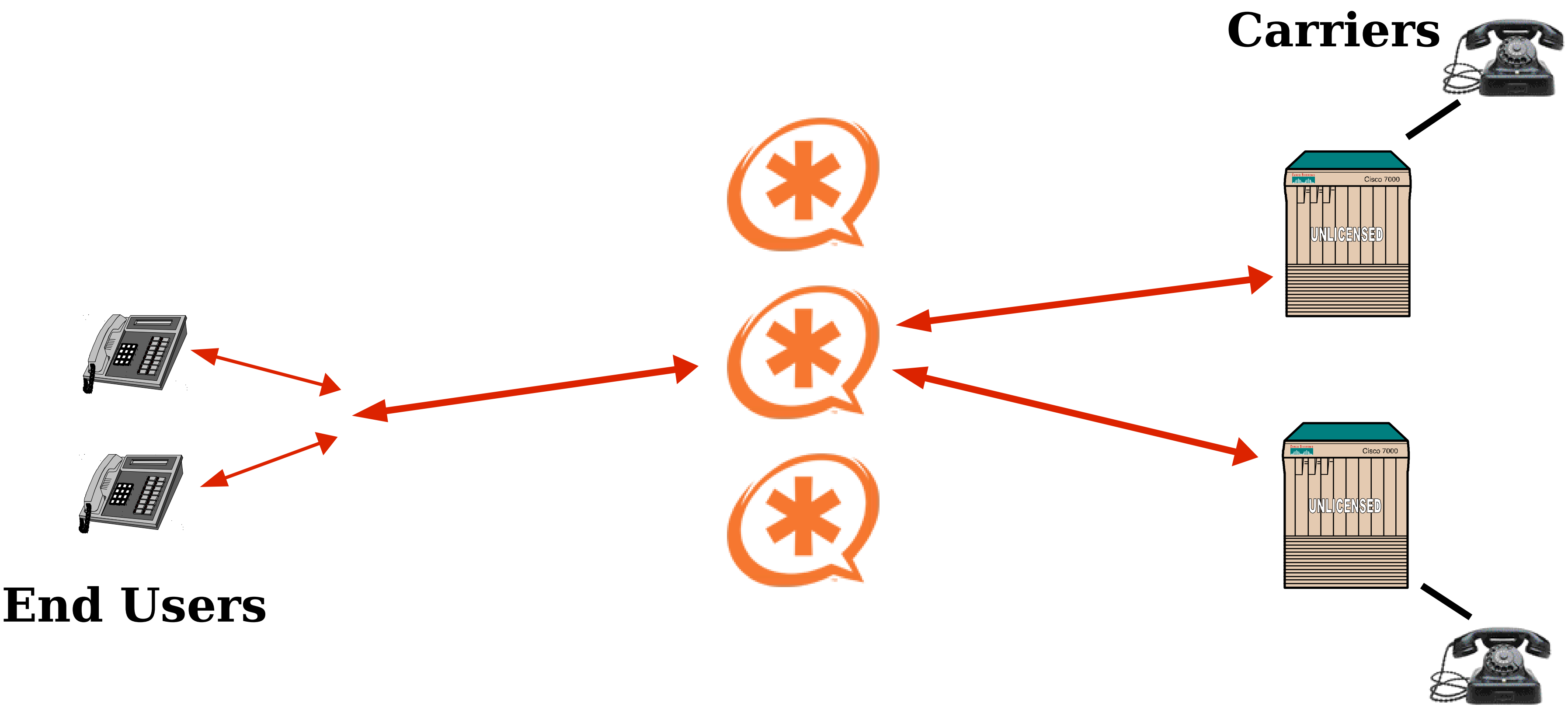
Asterisk Platform



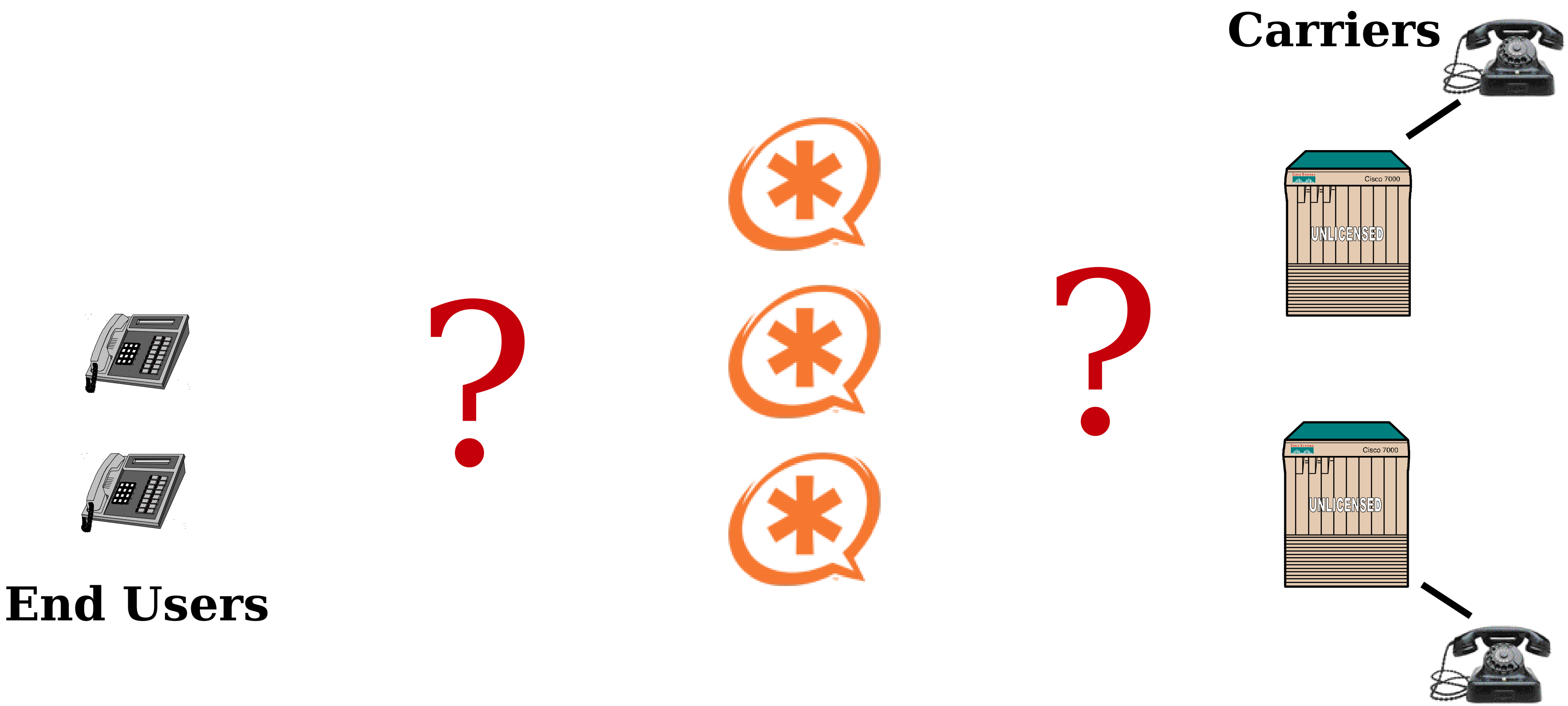
Asterisk Platform



Asterisk Platform



Asterisk Platform



Inbound questions

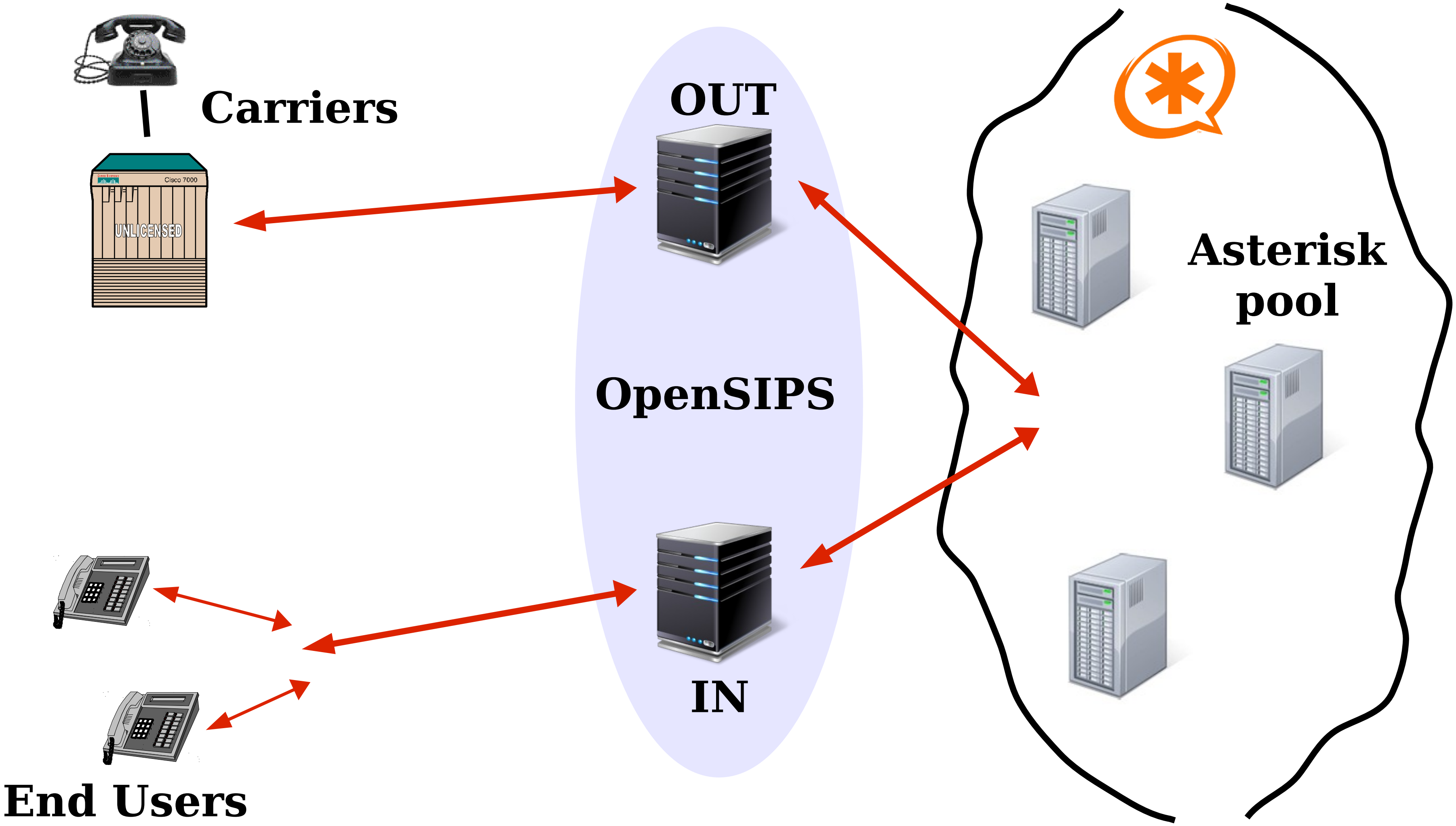
- How to balance clients
  - Static mappings, DNS based lb
- How do we provide high availability
- Who authenticates the clients
- Who validates the traffic

Outbound questions

- Where do we provision carriers
- Who handles LCR
- How do we map the carriers with the Asterisk instances
- What do carriers see and how do they authenticate

The answer is....

Diagram


Carriers
Cisco 7000
UNLICENSED
OUT
OpenSIPS
IN
Asterisk
pool
End Users

Inbound side

Inbound

- Provides a single entry point into the platform
- Offloads SIP messages processing
- Maintain a global view of the platform
- Efficient load balancing
- Single point of provisioning

Traffic

- Filtering (methods, UAs , other patterns)

```
if (!is_method(“INVITE|ACK|CANCEL|REGISTER|OPTIONS|BYE”)){
    sl_send_reply(“405”, “Method not allowed”);
    exit;
}

if ($ua =~ “friendly-scanner”) {
    xlog(“SECURITY: friendly scanner from $si\n”);
    drop();
}

if ($fd =~ "[0-9]{1,3}\.[0-9]{1,3}\.[0-9]{1,3}") {
    sl_send_reply("403", "Use domain name in FROM");
    exit;
}
```

Traffic

- Traffic validation (detect broken signaling)

```
# validate
sipmsg_validate("shm");
if ($retcode < 0){
    sl_send_reply(“400”, “Invalid SIP Message”);
    exit;
}

# check for preloaded routes
if (!has_totag() && loose_route()) {
    if (!is_method(“ACK”))
        sl_send_reply("403","Preload Route denied");
    exit;
}
```

Traffic

- Dialog validation

```
if (has_totag() && (loose_route() || match_dialog())) {
    if ($DLG_status != NULL) {
        if (!validate_dialog()) {
            # fix broken sequential
            fix_route_dialog();
        }
    }
}
```

Traffic

- Topology hiding

```
if (!create_dialog() || ! topology_hiding()) {
    sl_send_reply("500", "Service Unavailable");
    exit;
}
```

End-Points Authentication

- IP auth for SIP trunks & Digest auth

```
if (!check_source_address("0") {
    if (cache_fetch("local", "passwd_$fU",$avp(pass))) {
        if (!(pv_proxy_authorize(""))) {
            proxy_challenge("", "0");
            exit;
        }
    } else {
        if (!proxy_authorize("", "subscriber")) {
            proxy_challenge("", "0");
            exit;
        }
        cache_store("local", "passwd_$fU", "$avp(pass)",3600);
    }
    consume_credentials();
}
```

End-Points Authentication

- In memory caching support
- Digest with SQL, noSQL, LDAP, AAA

OpenSIPS can take care of the end-points authentication and authorization (using various mechanisms against various backends).

Caller/Callee services

- Simultaneous ringing (parallel registrations)
- Call hunting (serial forking)

OpenSIPS is the only one that has an entire view of the platform and knows where each client or service can be found.

Caller/Callee services

- DID mapping

```
if ($rU =~ “^\+[0-9]+$”) {
    if (alias_db_lookup(“dids”)) {
        # alias applied -> check if still in our domain
        if (!is_domain_local(“$rd”)) {
            sl_send_reply(“403”, “Domain not allowed”);
            exit;
        }
    }
}
```

Middle side
(managing the Asterisk pool)

• Select the proper Asterisk

- SIP and load wise balancing across the Asterisk pool
- Handle call transfer in a proper way
- Actively count ongoing calls for load estimation.
- Monitoring tools (in the * pool) may update in real time the balancing information in OpenSIPS

Select the proper Asterisk

- DID redirects

```
if ($rU =~ “^\+[0-9]+$”) {
    # find the proper * that handles this DID
    # using longest prefix matching
    if (!do_routing(“0”)) {
        sl_send_reply(“500”, “Service Error”);
        exit;
    }
    # enable failover
    t_on_failure(“dr_failover”);
}
```

Select the proper Asterisk

- Fair distribution

```
# Round Robin Asterisk selection
if (!ds_select_dst(“0”, “4”)) {
    sl_send_reply(“500”, “Service Error”);
    exit;
}
# enable failover
t_on_failure(“ds_failover”);
```

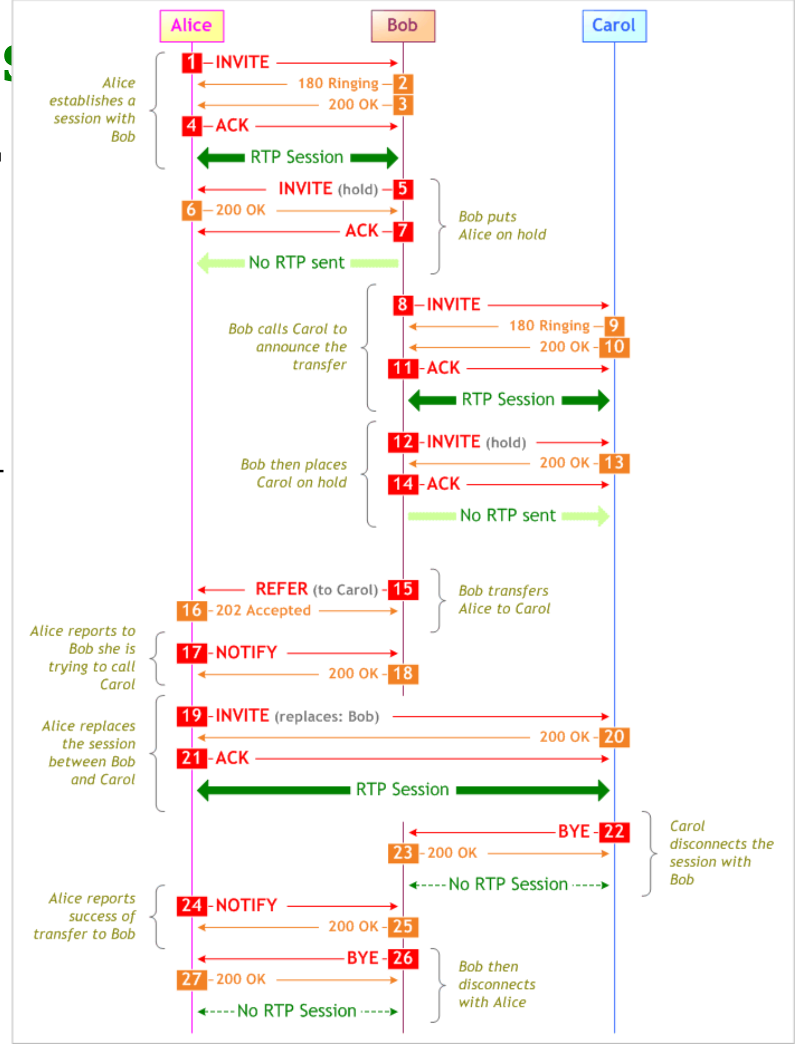
Select the proper Asterisk

- Load balancer

```
# balance the load based on limited resources
if (!load_balance("0", "pstn")) {
    sl_send_reply("500", "Service Error");
    exit;
}
# enable failover
t_on_failure("lb_failover");
```

Select the proper Asteris

- Attended call transfer
  1) Bob calls Alice using
  2) Bob puts Alice on hold
  3) Bob calls Carol using
  4) Bob puts Carol on hold
  5) Bob transfers Alice to Carol
  6) Alice calls Carol using
  7) Carol closes Bob
  8) Bob closes Alice
  9) Alice talks to Carol



Select the proper Asterisk

- Attended call transfer

The entire call flow has to use the same Asterisk instance!

Select the proper Asterisk

- Attended call transfer

```
if (get_dialog_info(“dst”, “$var(r)”, “caller”, “$fU”)||
    get_dialog_info(“dst”, “$var(r)”, “callee”, “$fU”)||
    get_dialog_info(“dst”, “$var(r)”, “caller”, “$rU”) ||
    get_dialog_info(“dst”, “$var(r)”, “callee”, “$rU”)) {
        # caller or callee are engaged in a different call
        # send the call to the same * instance
        $du = $var(r);
} else {
    # Choose the right Asterisk instance to be used
    ...
    create_dialog();
    $dlg_val(caller) = $fU;
    $dlg_val(callee) = $rU;
    $dlg_val(dst) = $du;
}
```

High Availability

- Detect faulty Asterisk boxes and re-route traffic
- Probing for auto re-enabling of Asterisk boxes
- The pool can be dynamically managed in terms of adding, removing or disabling boxes

Outbound side

PSTN Routing

- Prefix based routing / LCR
- Quality & capacity (CC or CPS) based routing
- Detect faulty carriers and re-route traffic

Single point of exit

- Aggregate traffic from all instances
- Single point of provisioning and control
- Provide a single IP to the carrier

Thank you for your attention
You can find out more at www.opensips.org
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www.opensips-solutions.com

Questions are welcome