



Asterisk, Instant Messaging and Presence, how?

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<http://www.saghul.net>



<http://www.sipdoc.net>



saghul

ironotec

Internet y Sistemas sobre GNU/Linux

<http://www.ironotec.com>

- saghul
 - VoIP enthusiast, playing around with Asterisk since 2k5
 - GNU/Linux lover likes everything “Software Libre”
 - Co-founder of <http://sipdoc.net>
 - Highly involved in spanish VoIP communities



- <http://www.saghul.net/blog/downloads/astricon2k9/>
- <http://www.slideshare.net/saghul/>
 - Slides
 - Complete configuration files
 - Database example data

-
1. Asterisk and presence status
 2. SIP SIMPLE or XMPP?
 3. The XMPP solution
 1. OpenFire setup
 4. The SIMPLE solution
 1. Kamailio + Asterisk setup
 5. Conclusions

- Asterisk SIP support (chan_sip)
 - In-dialog MESSAGE :-(
 - SUBSCRIBE and NOTIFY support
 - For Event: dialog
 - What about Event: Presence? :-(
 - No PUBLISH support :-(
- Asterisk XMPP support
 - res_jabber
 - JabberSend, JABBER_RECEIVE, JABBER_STATUS
 - chan_gtalk, chan_jingle
- Am I missing something?

Do we need presence and IM?

- “I want to talk to you, not to your phone”
- Are you **available**?
 - For an audio conference?
 - Just for IM?
 - For whom?
- **Where** are you?
 - Mobile
 - Office
 - Home
 - ...

We need to know if a user is **available**
and what his **status** is

- A **presence** server
- Users may **publish** their status
- Users may **subscribe** to other users status
- **Instant Messaging** between users

Is it possible *only* with Asterisk?

NO





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SIMPLE or XMPP?

- Did SIMPLE reinvent the wheel?
- Large companies started adopting SIMPLE (Microsoft, ...)
 - Proprietary extensions :-(
- XMPP does not provide voice capabilities
 - Well, there is Jingle...
- If SIP is the VoIP protocol: why not use it also for presence and IM?

The XMPP solution



XMPP

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- Integrate Asterisk with a XMPP server



- **Open Source**
- Java based
- Multiplatform
- **Asterisk integration plugin**
- SIP softphone plugin
- Gateways to multiple mi services: MSN, Yahoo, ...
- Easy installation!

OpenFire (II)

- Download deb package
- `dpkg -i openfire_3.6.4_all.deb`

Openfire Setup: Welcom... x

http://ast16.saghul.lan:9090/setup/index.jsp

openfire

Openfire 3.6.4

Setup

Setup Progress

Language Selection

Server Settings

Database Settings

Profile Settings

Admin Account

Welcome to Setup

Welcome to Openfire Setup. This tool will lead you through the initial setup of the server. Before you continue, choose your preferred language.

Choose Language

- Czech (cs_CZ)
- Deutsch (de)
- English (en)
- Español (es)
- Français (fr)
- Nederlands (nl)
- Polski (pl_PL)
- Português Brasileiro (pt_BR)
- Slovenčina (sk)
- 中文 (简体) Simplified Chinese (zh_CN)

Continue

- Web based configuration
- Clustering architecture
- Connection to the Asterisk Manager Interface
 - Multiple connections
- Mapping between Asterisk SIP users and OpenFire XMPP users
- Multiplatform Java client: Spark
- Flash based web client: SparkWeb



The screenshot shows the OpenFire Admin Console interface. The browser address bar displays the URL: `http://ast16.saghul.lan:9090/plugins/asterisk-im/phone-users.jsp?success=true&start=0&range=15`. The OpenFire logo and version (3.6.4) are visible in the top left, and the user is logged in as 'admin' with a 'Logout' link in the top right. The navigation menu includes 'Server', 'Users/Groups', 'Sessions', 'Group Chat', 'Plugins', and 'Asterisk-IM'. The 'Asterisk-IM' sub-menu is open, showing 'General Settings' and 'Phone Mappings'. The 'Phone Mappings' page displays a success message: 'Operation completed successfully.' Below this, it shows 'Total Users: 2 -- Sorted by Username - Users per Page: 15'. A table lists the users:

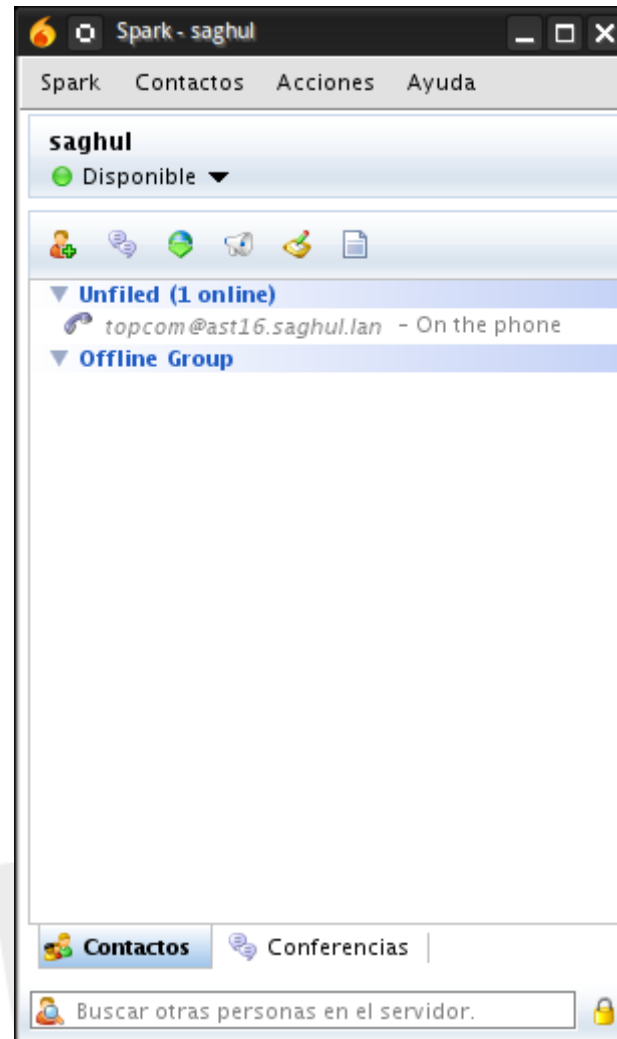
| Username | Device | Extension | Caller ID | Options |
|----------|---------------|-----------|-----------|---------|
| saghul | SIP/cisco7960 | 2001 | | |
| topcom | SIP/topcom | 2000 | | |

Below the table is a form titled 'Add User/Asterisk Phone mapping' with the following fields:

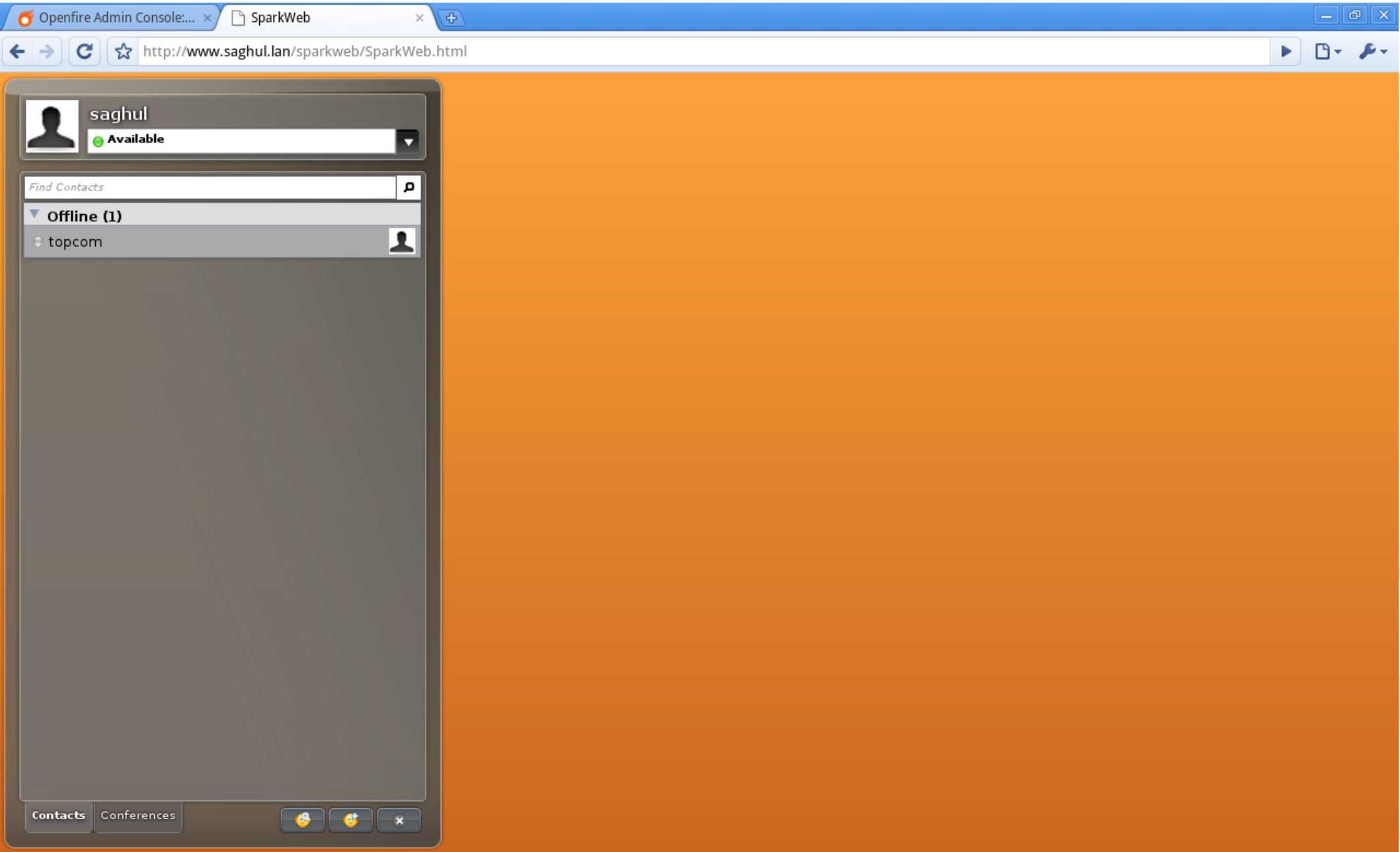
- * Username:
- * Device:
- * Extension:
- Caller ID:
- Primary:

Buttons for 'Add' and 'Cancel' are at the bottom of the form. A note indicates '* Required fields'.

- When a user is talking OpenFire puts it “On the phone”



OpenFire (VI)



- What we get
 - Instant Messaging
 - Presence
 - Gateways to other mi services
 - Text conferencing
- Problems
 - Duplicated users (we could partially fix it with LDAP)
 - Need to handle 2 protocols
 - Not many softphones support SIP and XMPP
 - Do any hardphones support XMPP?



The SIP solution



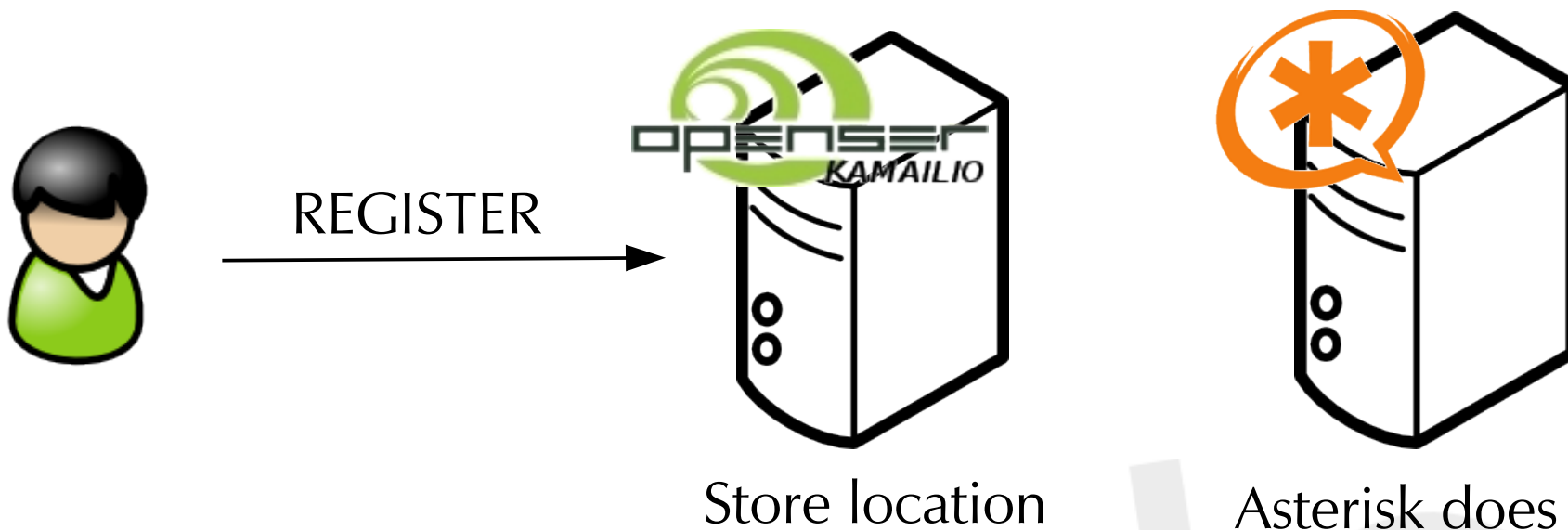
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- SIMPLE IETF working group
 - Presence RFCs
 - 3856, 3857, 3858, 3863, 4479, 4480, 4482, ...
 - XCAP
 - 4825, 4826, 4827, 5025, ...
 - Instant Messaging
 - 3428, 3994, 4975, ...

SIMPLE is NOT simple!



- Integrate Asterisk and Kamailio to provide IM and presence.
- Users are registered to Kamailio.
- INVITE requests are routed through the Asterisk server.
 - Asterisk RealTime user integration with Kamailio's subscriber table.
- PUBLISH, SUBSCRIBE and MESSAGE requests are handled by Kamailio.



Kamailio – Asterisk RealTime integration

- Asterisk peers are Kamailio's subscribers.
- MySQL view so that Asterisk 'sees' the users as his own.
- Peers IP → Kamailio IP.
- Calls between users go through Kamailio and Asterisk.
- We need to call to alphanumeric users → DB Alias

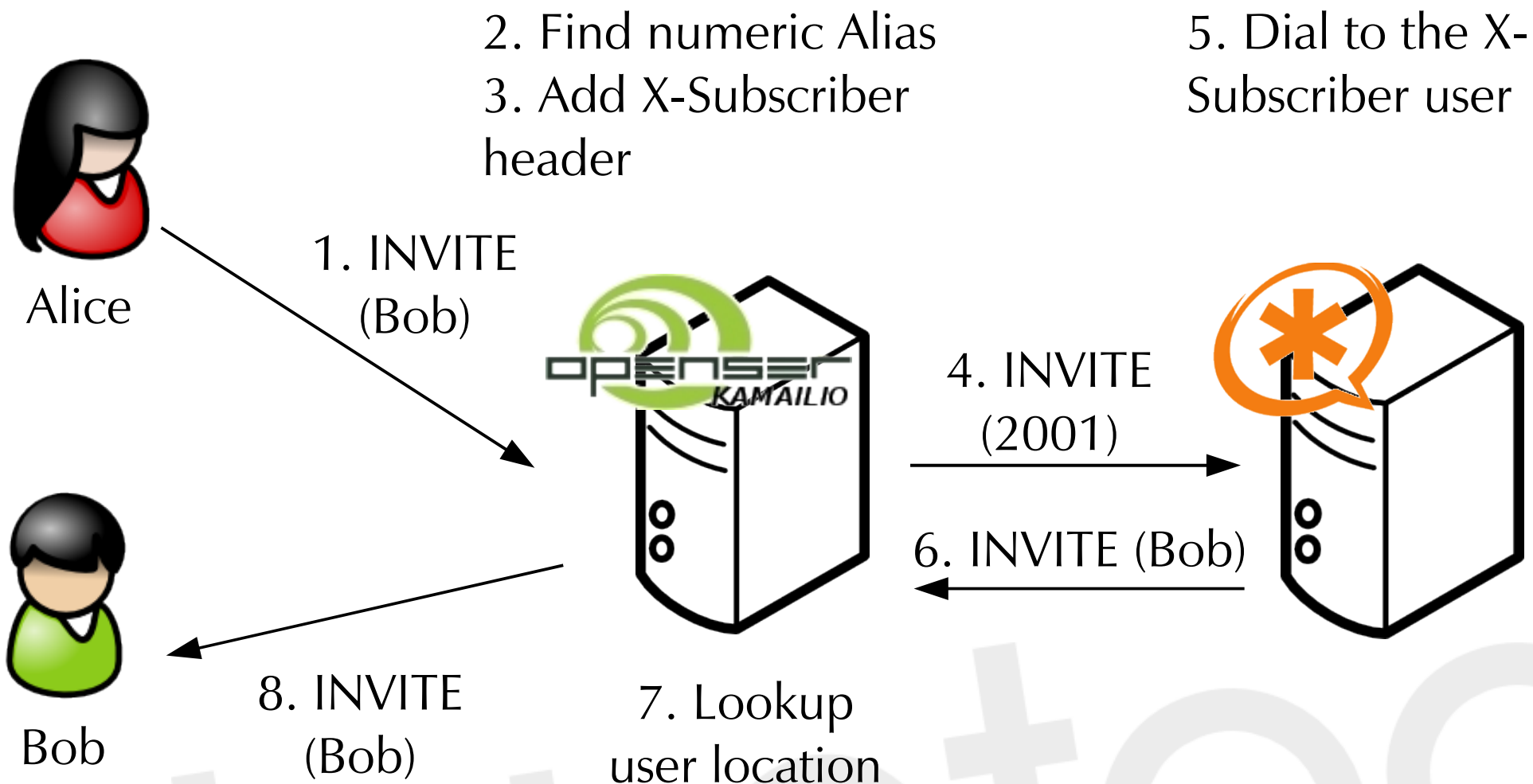
Kamailio – Asterisk RealTime integration (2)

```
CREATE VIEW sip_peers AS
SELECT subscriber.username AS name,
subscriber.username AS defaultuser,
'friend' AS type,
NULL AS secret,
subscriber.domain AS host,
concat(subscriber.rpid, ' ', '<', subscriber.username, '>') AS callerid,
'from-users' AS context,
subscriber.username AS mailbox,
'yes' AS nat,
'no' AS qualify,
'info' AS dtmfmode,
subscriber.username AS fromuser,
```

Kamailio – Asterisk RealTime integration (3)

```
NULL AS authuser,  
subscriber.domain AS fromdomain,  
NULL AS insecure,  
'no' AS canreinvite,  
NULL AS disallow,  
'all' AS allow,  
NULL AS restrictcid,  
subscriber.domain AS defaultip,  
subscriber.domain AS ipaddr,  
subscriber.domain AS outboundproxy,  
'5060' AS port,  
NULL AS regseconds  
FROM kamailio_1.subscriber;
```

Invitation



```
# Route all INVITE requests to Asterisk
if (is_method("INVITE")) {
    # Remove X-Subscriber header so that no one sees it...
    remove_hf("X-Subscriber");

    # We don't have to route the requests coming FROM Asterisk
    # back to Asterisk. We would make a loop!
    if (!( $si == "AST_IP" && $sp == "AST_PORT" )) {
        route(ASTERISK_USERS_ROUTE);
    }
}
```



```
# Send INVITE requests to the Asterisk server
route[ASTERISK_USERS_ROUTE] {
    # Call to the numeric alias
    avp_db_query("SELECT alias_username FROM dbaliases WHERE username
= '$rU' AND domain = '$avp(AVP_ORIGDOMAIN)'LIMIT 1",
"$avp(AVP_NUMALIAS)");
    if (is_avp_set("$avp(AVP_NUMALIAS)")) {
        # Save the subscriber in a header so we can use it in Asterisk
        append_hf("X-Subscriber: $rU\r\n");
        $rU = $avp(s:numalias);
    }

    $rd = "AST_IP";
    $rp = "AST_PORT";
    route(RELAY_ROUTE);
}
```

[from-users]

exten => _X.,1,NoOp()

exten => _X.,n,Set(SUBSCRIBER=\${SIP_HEADER(X-Subscriber)})

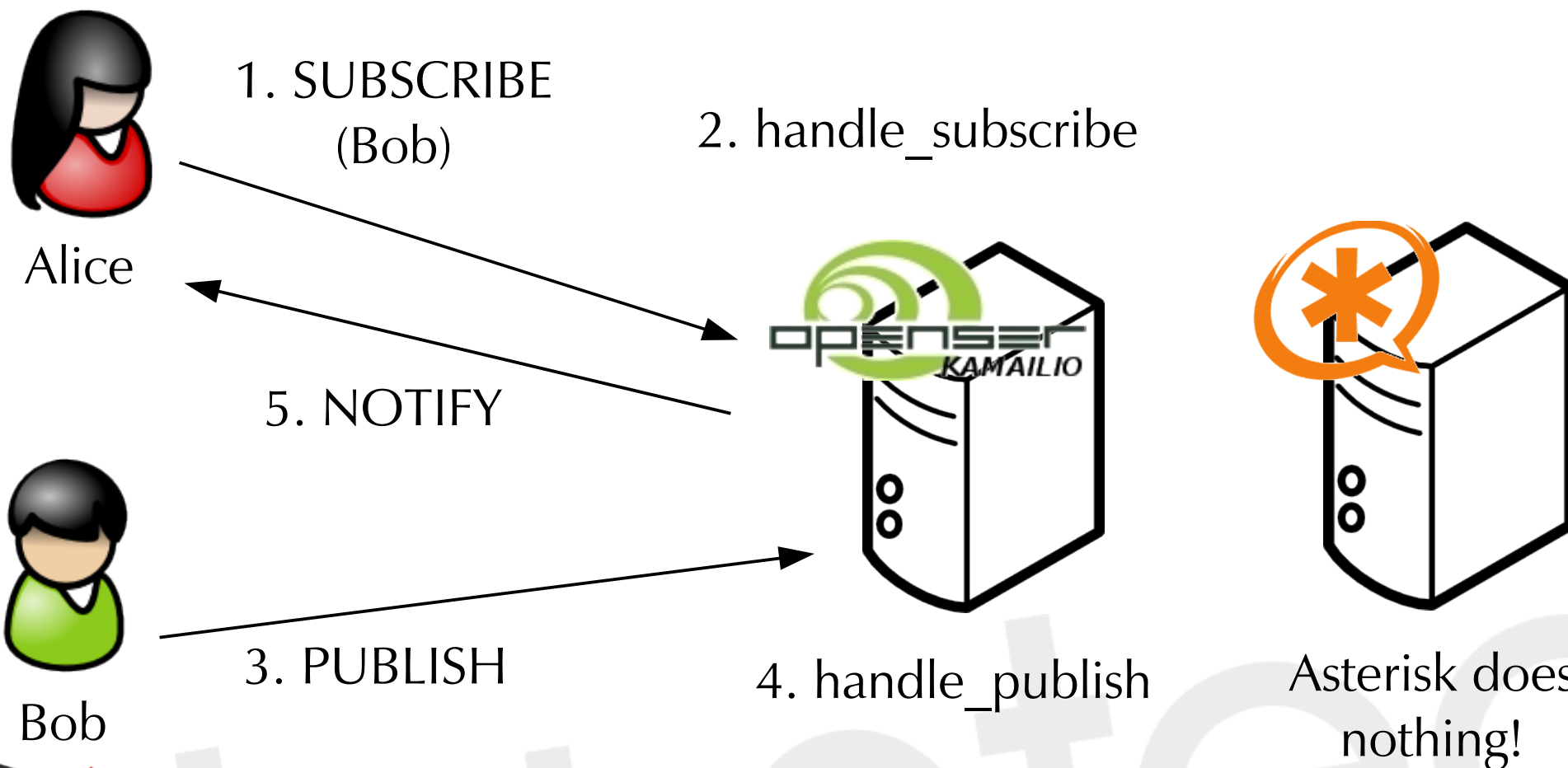
exten => _X.,n,GotoIf(\$[\${LEN}(\${SUBSCRIBER})] = 0]?hang)

exten => _X.,n,Dial(SIP/\${SUBSCRIBER})

exten => _X.,n(hang),Hangup



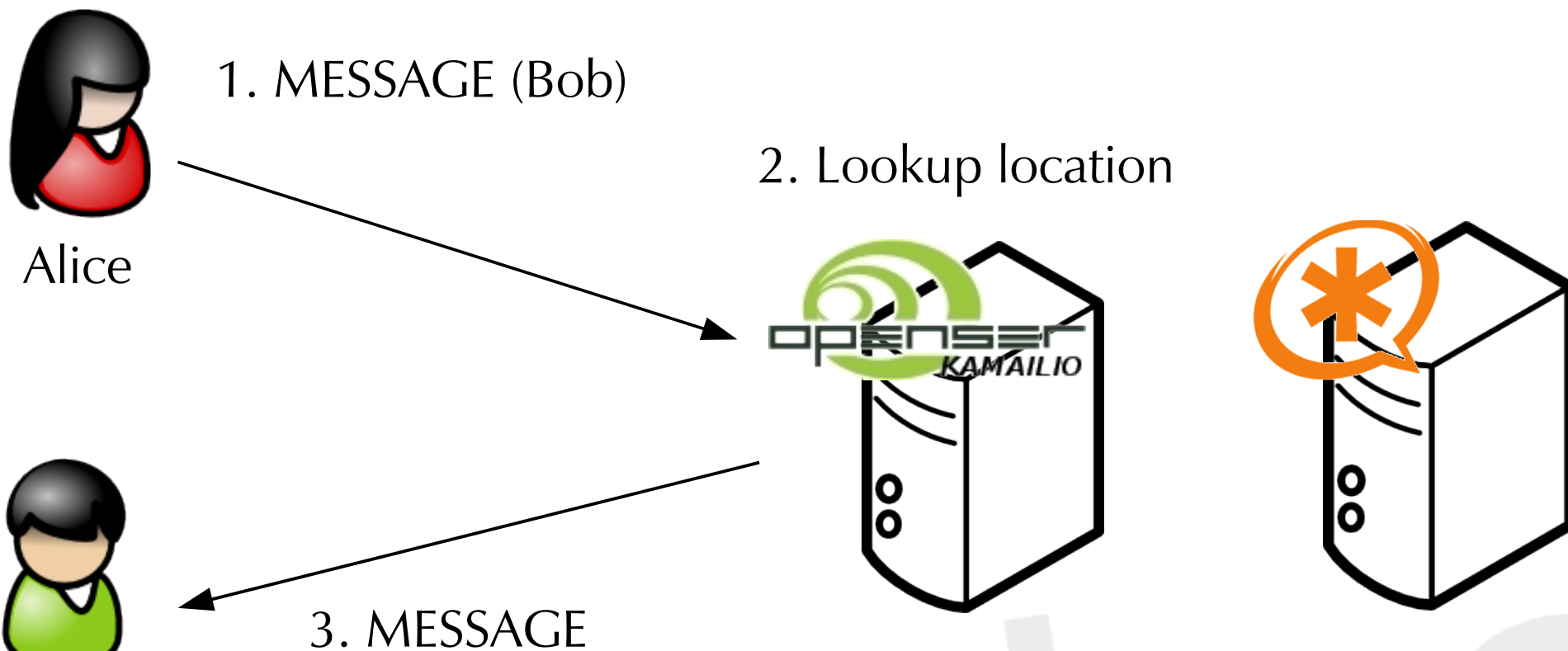
SIMPLE presence



SIMPLE presence (2)

```
# Handle presence requests
    if(is_method("PUBLISH|SUBSCRIBE")) {
        route(PRESENCE_ROUTE);
    }
```

```
# Handle presence
route[PRESENCE_ROUTE] {
    if (is_method("PUBLISH")) {
        handle_publish();
        t_release();
    } else if (is_method("SUBSCRIBE")) {
        handle_subscribe();
        t_release();
    }
    exit;
}
```

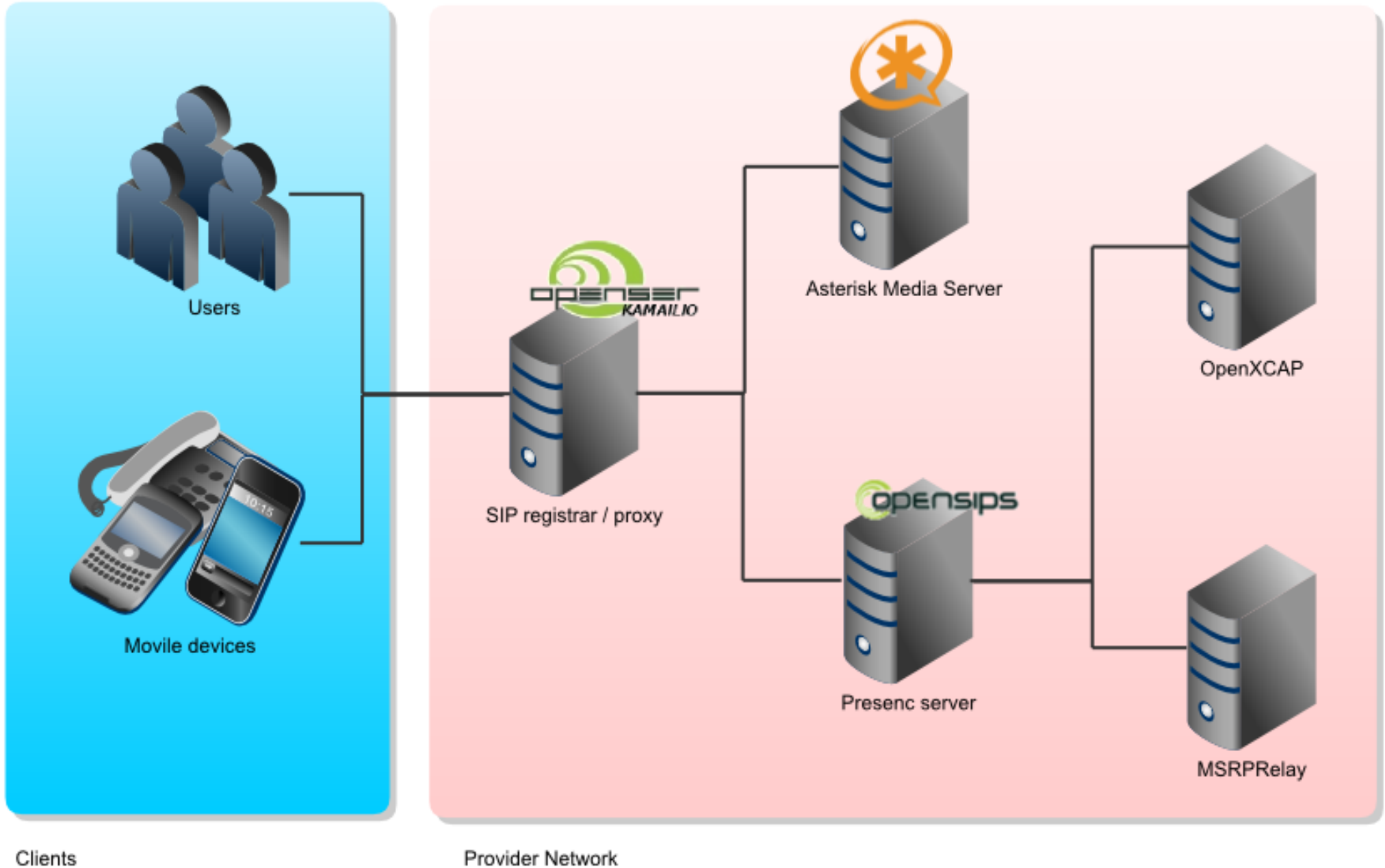


Asterisk does nothing!



- We just need to fix the NAT in signalling.
- Our Asterisk 'peers' are configured with nat=yes
 - COMEDIA mode
 - Audio will go through Asterisk

Further improvements...



What about mixing both?

- OpenFire's Asterisk plugin still works! (regardless of the integration with Kamailio)



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SIMPLE or XMPP?

Thanks!

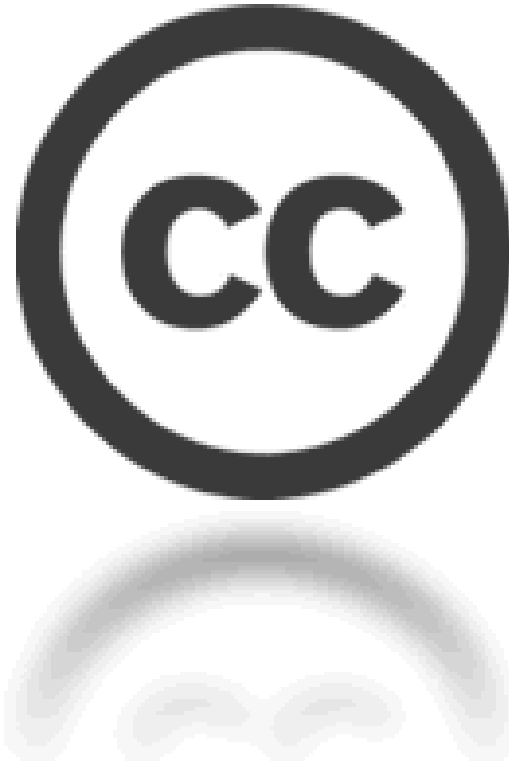
```
BYE sip:astricon@astricon.net SIP/2.0
Via: SIP/2.0/UDP guest.astricon.net:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
From: saghul <sip:saghul@sipdoc.net>;tag=8321234356
To: AstriCon <sip:astricon@astricon.net>;tag=9fxced76sl
Call-ID: 3848276298220188511@astricon.net
CSeq: 1 BYE
Content-Length: 0
```

Thanks for watching!



Any questions?

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