Asterisk, Instant Messaging and Presence, how?

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- 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
This presentation

- [http://www.saghul.net/blog/downloads/astricon2k9/](http://www.saghul.net/blog/downloads/astricon2k9/)
- [http://www.slideshare.net/saghul/](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
What we need

- 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**
SIMPLE or XMPP?
• Did SIMPLE reinvent the wheel?
• Large companies started adopting SIMPLE (Microsoft, …)
  – Propietary 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
• Integrate Asterisk with a XMPP server
OpenFire

- **Open Source**
- Java based
- Multiplatform
- **Asterisk integration plugin**
- SIP softphone plugin
- Gateways to multiple mi services: MSN, Yahoo, …
- Easy installation!
- Download deb package
- `dpkg -i openfire_3.6.4_all.deb`
OpenFire (III)

- 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
Phone Mappings

Total Users: 2 -- Sorted by Username - Users per Page: 15

<table>
<thead>
<tr>
<th>Username</th>
<th>Device</th>
<th>Extension</th>
<th>Caller ID</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>saghiu</td>
<td>SIP/cisco7960</td>
<td>2001</td>
<td></td>
<td></td>
</tr>
<tr>
<td>topcom</td>
<td>SIP/topcom</td>
<td>2000</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Add User/Asterisk Phone mapping

* Username: 
* Device: 
* Extension: 
  Caller ID:
  Primary: 
  Add  Cancel
  * Required fields
• When a user is talking OpenFire puts it “On the phone”
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OpenFire (VI)
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
A complex protocol

- 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!
The SIP solution

- 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.
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Registration

REGISTER → Store location

Asterisk does nothing!
• 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
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,
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;
2. Find numeric Alias
3. Add X-Subscriber header
5. Dial to the X-Subscriber user

1. INVITE (Bob)

Alice

Bob

4. INVITE (2001)

8. INVITE (Bob)

7. Lookup user location
# 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\n\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

1. SUBSCRIBE (Bob)

2. handle_subscribe

5. NOTIFY

3. PUBLISH

4. handle_publish

Asterisk does nothing!
# 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;
}
1. MESSAGE (Bob)

2. Lookup location

3. MESSAGE

Asterisk does nothing!
NAT handling

- 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)
SIMPLE or XMPP?
Thanks for watching!
Any questions?
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