Übung zu VoIP

Für die Übung zu VoIP wird der SIP-Proxy sipwitch als vermittelndes Element eingesetzt. Da der Proxy bereits beim Systemstart als Dienst (Daemon) mitgestartet wird, muss der Proxy zunächst gestoppt werden. Dies ist notwendig, um eine eigene Konfiguration (s.u.) verwenden zu können.

```
sudo sipwitch down
```

Der Proxy kann mit dem folgenden Befehl gestartet werden:

```
sipw -x9 -f
```

Als Basis für die Konfiguration kann die Datei `/etc/sipwitch.conf` [1] und für die Anlage der Benutzer `/etc/sipwitch.d/lab.xml-sample`[2].

Die folgende Beispieldatei basiert auf diesen beiden Dateien, muss aber noch an die örtlichen Gegebenheiten angepasst werden.

.sipwitchrc_sample

```
<?xml version="1.0"?>
<sipwitch>
    <!-- master config file.  The default config can be overriden with
a
  runtime one stored in /var/run/sipwitch which can be installed by
  a management system.  If one is using a server executed under "user"
  permissions, then this would be ~/.sipwitchrc.
 Erklärung:
 http://www.gnutelephony.org/index.php/GNU_SIP_Witch_configuration

 Konfig: .sipwitchrc in Home-Ordner kopieren.
 Aufruf: sipw -x9 -f
 -->
  <provision>
<!-- Allows provisioning to be in main config file as well as
scattered.
     This allows one to produce a single config file that represents
the
  complete phone system.

 <refer id="x"></refer>
 <alias id="test"><contact>sip:xxx@yyy</contact></alias>
 <user id="y"/>
 <gateway id="z"/>
-->
    <test id="testing">
      <secret>editme</secret>
      <extension>299</extension>
      <answer>12</answer>
      <duration>120</duration>
```

```
      <display>Testing</display>
    </test>

    <user id="editme1">
      <secret>editme</secret>
      <extension>201</extension>
      <display>editme 1</display>
    </user>

    <user id="editme2">
      <secret>1234</secret>
      <extension>202</extension>
      <display>editme 2</display>
    </user>

  </provision>

  <access>
    <!-- Access rules and cidr definitions.  By default 127.0.0.1/::1
are in
     a pre-generated "loopback" cidr.  Access rule entries are now
     automatically generated by scanning the network interface, so this
     is for special overrides or convenience naming.
 <local>172.16.59.0/24</local>

 <local>0.0.0.0/01</local>
 <local>128.0.0.0/01</local>
-->
    <local>editme</local>
  </access>

  <stack>
  <!-- The effective names this server processes requests for, and an
optional
     list of host or domain names this server will also respond to.
The
     default hostname is always accepted.
  <localnames>sip.gnutelephony.org, server.local, something
somewhere</localnames>
  -->

<!-- Stack configuration.  Here we restrict all access to the server
under
     the local subnet, and we specify the local subnet is "trusted".
Trusted
  means that challenge digests will be relaxed for devices that are
  already registered with     the server, and hence reduces the total
sip
  traffic needed.  We map for 200 calls, set 2 dispatch threads for
  sip events, and bind to all interfaces.
    <restricted>local</restricted>
```

```
    <trusted>local</trusted>

-->

    <mapped>200</mapped>
    <threading>2</threading>
    <interface>*</interface>
    <dumping>false</dumping>

<!-- peering entry used for setting "proxy" ip address for external
users
  when we are behind a NAT.  This is used for determining ip address
for
  media proxy in particular.  Example entry shown.  Can be ip address
or
     resolvable hostname.

  <peering>www.example.com</peering>
-->

<!-- special user id's.  The "system" id is used when the server
creates a
     sip message that is not on behalf of any registered "ua", but
rather
  from the server itself.  For example, when feeding a sms "message"
  through the control interface, this is generated as a "system"
message.
  Attempts to dial the "system" id will always return SIP FORBIDDEN.

  The "anon" id is used when anonymous messages are generated.  These
  always respond with SIP NOT FOUND if one wishes to contact anon.
-->
    <system>
      system
    </system>
    <anon>
      anonymous
    </anon>
  </stack>

  <timers>
    <!-- ring every 4 seconds -->xml
    <ring>4</ring>
    <!-- call forward no answer after x rings -->
    <cfna>4</cfna>
    <!-- call reset to clear cid in stack, 6 seconds -->
    <reset>6</reset>
  </timers>

<!-- we have 2xx numbers plus space for external users -->
  <registry>
```

```
    <prefix>200</prefix>
    <range>100</range>
    <keysize>77</keysize>
    <mapped>200</mapped>

    <!-- ACHTUNG: Der Tag <realm> muss mind. einen "." enthalten und
darf keine
 Leerzeichen beinhalten.!
 s. Hier:
http://lists.gnu.org/archive/html/sipwitch-devel/2011-01/msg00007.html
-->
    <realm>editme</realm>
  </registry>

  <routing>

  </routing>
</sipwitch>
```

[1] Basiskonfigurtation

[2] Beispiel für Benutzerkonfiguration -> <provision>