

Valet Parking Lot Example



Due to the complex nature of this scenario and all the REFERing involved sipXbridge is required to be in the signaling path for this to operate correctly.



The following scenarios either work poorly or not at all (there may be more that just don't work as well)

- Attended transfers to media services - Voicemail, Voicemail boxes, conferencing, auto attendant
 - Voicemail - Caller is disconnected
 - Voicemail boxes - Caller appears on hold to user while they leave a voicemail. Call is cleared once caller hangs up
 - Conferencing - Caller is disconnected
 - Auto Attendant - Caller is disconnected
- Blind Transfers to these services work just fine.

Add the following content to a new file, `cd /etc/sipxpbx/freeswitch/conf/dialplan/custom_context.xml`

`/etc/sipxpbx/freeswitch/conf/dialplan/custom_context.xml`

```
<context name="custom">
  <extension name="Park-Call">
    <condition field="destination_number" expression="^(3500)$">
      <action application="answer"/>
      <action application="sleep" data="300"/>
      <action application="valet_park" data="my_lot auto in 3501 3599"/>
    </condition>
  </extension>
  <extension name="Park-Retrieve">
    <condition field="destination_number" expression="^(35\d\d)$">
      <action application="answer"/>
      <action application="valet_park" data="my_lot $1"/>
    </condition>
  </extension>
</context>
```

This dialplan has a valet parking number of 3500 with parking slots 3501-3599

Now we need to set up sophia

Add the following content to a new file, `/etc/sipxpbx/freeswitch/conf/sip_profiles/custom_profile.xml`

MAKE SURE YOU REPLACE THE DOMAIN NAMES AT THE BOTTOM OF THIS CONFIG FILE WITH WHATEVER YOUR sipXecs DOMAIN NAME IS

`/etc/sipxpbx/freeswitch/conf/sip_profiles/custom_profile.xml`

```
<profile name="custom_dialplan">
  <aliases>
    <!--
    <alias name="outbound"/>
    <alias name="nat"/>
    -->
  </aliases>

  <domains>
    <domain name="all" alias="false" parse="true"/>
  </domains>

  <settings>
    <param name="debug" value="1"/>
    <param name="sip-trace" value="no"/>
    <param name="rfc2833-pt" value="101"/>
    <param name="sip-port" value="9061"/>
    <param name="dialplan" value="XML"/>
    <param name="dtmf-duration" value="100"/>
    <param name="codec-prefs" value="G722,PCMU@20i,PCMA@20i,speex,L16"/>
    <param name="hold-music" value="$$${hold_music}"/>
    <param name="rtp-timer-name" value="soft"/>
  </settings>
</profile>
```

```

<!--<param name="enable-100rel" value="true"/>-->
<!-- This could be set to "passive" -->
<param name="local-network-acl" value="localnet.auto"/>
<param name="manage-presence" value="false"/>
<param name="context" value="custom"/>

<!-- used to share presence info across sofia profiles
      manage-presence needs to be set to passive on this profile
      if you want it to behave as if it were the internal profile
      for presence.
-->
<!-- Name of the db to use for this profile -->
<!--<param name="dbname" value="share_presence"/>-->
<!--<param name="presence-hosts" value="$org.sipfoundry.sipxconfig.domain.Domain@1"/>-->
<!--<param name="force-register-domain" value="$org.sipfoundry.sipxconfig.domain.Domain@1"/>-->
<!--all inbound reg will stored in the db using this domain -->
<!--<param name="force-register-db-domain" value="$org.sipfoundry.sipxconfig.domain.Domain@1"/>-->
<!-- ***** -->

<!--<param name="aggressive-nat-detection" value="true"/>-->
<param name="inbound-codec-negotiation" value="scrooge"/>
<param name="nonce-ttl" value="60"/>
<param name="auth-calls" value="false"/>
<param name="accept-blind-auth" value="true"/>
<!--
      DO NOT USE HOSTNAMES, ONLY IP ADDRESSES IN THESE SETTINGS!
-->
<param name="rtp-ip" value="${local_ip_v4}"/>
<param name="sip-ip" value="${local_ip_v4}"/>
<param name="ext-rtp-ip" value="auto-nat"/>
<param name="ext-sip-ip" value="auto-nat"/>
<param name="rtp-timeout-sec" value="300"/>
<param name="rtp-hold-timeout-sec" value="1800"/>
<param name="enable-3pcc" value="true"/>

<!-- TLS: disabled by default, set to "true" to enable -->
<param name="tls" value="${external_ssl_enable}"/>
<!-- additional bind parameters for TLS -->
<param name="tls-bind-params" value="transport=tls"/>
<!-- Port to listen on for TLS requests. (5081 will be used if unspecified) -->
<param name="tls-sip-port" value="${external_tls_port}"/>
<!-- Location of the agent.pem and cafile.pem ssl certificates (needed for TLS server) -->
<param name="tls-cert-dir" value="${external_ssl_dir}"/>
<!-- TLS version ("ssl23" (default), "tls1"). NOTE: Phones may not work with TLSv1 -->
<param name="tls-version" value="${sip_tls_version}"/>

</settings>
  <gateways>
    <gateway name="pbx.example.com">
      <param name="proxy" value="pbx.example.com"/>
      <param name="realm" value="pbx.example.com"/>
      <param name="username" value="~~id-media"/>
      <param name="password" value="1kJLRiF9YS"/>
      <param name="register" value="false"/>
    </gateway>
  </gateways>
</profile>

```

Grab the FreeSWITCH sounds from <http://files.freeswitch.org/freeswitch-sounds-en-us-callie-8000-1.0.13.tar.gz> and place them in **/usr/local/freeswitch/sounds** so FreeSWITCH can tell you what slot the call is in.

In sipXecs create a SIP trunk to the hostname of your sipXecs server on port **9061**. Make sure to select the internal SBC or you will experience weird issues.

Next, add a custom dial plan entry for **35 and 2 digits** and append **entire dialed number** and use your new SIP trunk as the gateway.

After you've done all that you need to make sure **/etc/sipxpbx/freeswitch** is owned by **sipxchange:sipxchange**
Run the following command to do this:

```
chown -R sipxchange:sipxchange /etc/sipxpbx/freeswitch
```

Restart all necessary services as well as **Media Services**

Test It Out

Try it out by doing an attended transfer to 3500, then wait for FreeSWITCH to say which slot the call is parked in. Once you hear the slot number, complete the attended transfer.

To retrieve the call simply dial the slot number that was played to you in the transfer.