

Provider voip.ms Swiftvox (USA)

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[voip.ms aka Swiftvox](#)

Termination

- 2012-04-07

After a year of running fine with below, I had to add from domain param set to [voip.ms](#) for termination to work. I'm not sure what happened, other than it worked one day (for a year) and then stopped working.

```
<param name="from-domain" value="voip.ms" />
```

Config

- If you have a linksys device (spa2102 spa5xx series), they will reject calls if ptime is not set to 20. Make sure you change that in the phone's configuration (rtp packet size 0.020 [from 0.030]). You should do this anyway.
- [\[Voip.ms\]](#)
- Based on ical config above
- Only tested with IP auth
- Place the following in *conf/sip_profiles/external/voipms.xml*.

```
<include>
  <gateway name="voipms">
    <!-- Replace the values below with your Voip.ms username and password. Even if you use an IP-based sub-
    account, FreeSWITCH likes values for the username and password -->
    <param name="username" value="100XXX" />
    <param name="password" value="PASSWORD" />
    <!-- This gateway could be different depending on which switch you are on -->
    <param name="proxy" value="sip.us4.voip.ms" />
    <param name="realm" value="sip.us4.voip.ms" />
    <!-- This should be set to "false" for IP-based accounts, or "true" for registration based -->
    <param name="register" value="false" />
  </gateway>
</include>
```

- Tested with registration auth (2010-10-19)
- Updated so Caller ID works properly

```

<include>
  <gateway name="voipms">
    <!-- Replace the values below with your Voip.ms username and password. -->
    <param name="username" value="your_username" />
    <param name="password" value="your_password" />
    <!-- This gateway could be different depending on which switch you are on -->
    <param name="proxy" value="montreal|houston|newyork|etc.voip.ms" />
    <param name="realm" value="voip.ms" />
    <!-- This should be set to "true" for registration based -->
    <param name="register" value="true" />
    <!-- Voip.ms requires the Remote-Party-Identity Header to be set in the Sip invite for Caller-ID to work
right
DON'T FORGET TO REMOVE ANY CALLER ID INFO IN http://voip.ms->Main Menu->Account Settings->CallerID
Number
-->
    <param name="sip_cid_type" value="rpd" />
    <!--Setting in one place is much easier than everywhere you may bridge. You can do this since 2010 Sept 27
http://jira.freemswitch.org/browse/FS-2722
-->
  </gateway>
</include>

```

Caller ID

[Voip.ms](#) requires the Remote-Party-Identity Header to be set in the Sip invite. Use:

```
<param name="sip_cid_type" value="rpd" />
```

Errors

1. destination_number (inbound public) is getting set as my Sip_profile username.
 - if you use [voip.ms](#) and have a subaccount set up, make sure "Device Type" is set to "Asterisk, ip pbx, gateway or voip switch" and not "ata device, ip phone or soft phone". Once this is fixed, destination_number will be the correct DID number.