Shared Line Appearance

About

Setting up SLA/SCA on various model handsets. SCA allows you to have multiple phones register to the same extension and then use the line key on the handset to barge into a live call at any point. This is a huge feature for small companies and workgroups who work within earshot and like to say "Call on line 1 for you Bob". SCA also has the more familiar feature (for key phone system users) of the line key lighting up (or flashing depending on the handset vendor) when the hook is lifted on another handset rather than just when you have a call up. As mentioned in the announcement[1] the FreeSWITCH developers have implemented the Broadsoft method of handling so multi-vendor support is possible!

FreeSWITCH Configuration

In the SIP profile you wish to run SCA on enable this option:

```xml
<param name="manage-shared-appearance" value="true"/>
```

And rescan your profile.

Handset Configuration

Note that currently SCA works reliably only on Polycom and Linksys per a 2010-06-18 mailing list post.

Snom

**NOTE:** that support for SCA/SLA is broken in all Snom firmware releases[2]. This appears to be due to "the latest Snom firmware that causes this, Its missing the appearance-index on the call-info header on just about every packet involved in the dialog." This is tracked in Snom internal reference SCPP-1633[3]. Please note, that this ticket has been resolved according to Snom (reference contained in the snom forum thread mentioned here), and using the new Beta series (8.7.x) appears to resolve the issues from my admittedly minor testing. Note that the changelogs throughout the 8.7.x series have a large number of resolved issues around the Broadsoft SCA/SLA support.

- Set an identity up to register as you would normally and then under the same identity do the following:
  - Set Shared Line to 'On'
  - Set Server Type Support to 'Broadsoft'
- Then under Function Keys set up a line to work as your line appearance by selecting the identity you are sharing, setting the type to 'Shared Line' and entering a SIP URI e.g. 1008@x.x.x.x where x.x.x.x would be the IP of your FreeSWITCH server and 1008 the extension you are sharing.

Linksys/SPA

**Tested on SPA-942 firmware 6.1.5(a)**

- Setup the handset as per normal to register to the extension you are going to share.
- In addition under 'Share Line Appearance' in Ext 1 (or which ever line you are using) set 'Share Ext:' to 'Shared'
- Under Phone, Line Key 1 (Or whichever you are using), set 'Share Call Appearance:' to Shared

Polycom

**Tested on firmware 3.1.3.0439**

- Setup the handset as per normal to register to the extension you are going to share, in addition under the line you need to set Type to Shared rather than Private.
- To enable 'Barge In' to work you need setup provisioning and enable this option under your registration:

  ```xml
  reg.1.bargeInEnabled="1"
  ```

Aastra

Only the latest 2.6 firmware with 675Xi phones can support SCA partially(The other party can barge in calls initiated from Aastra phone, but Aastra phone will drop any existing SCA calls when picking up), any older firmware SCA just not working. The setup for SCA is simple

Under "Global SIP Settings" or any Line 1/2/3/4 you want to setup as SCA line, set "BLA Number" to the shared line number, say 101, set "Line Mode" to "Broadsoft SCA"
Grandstream

Can someone with a Grandstream handset please fill in what you need to do

Yealink

Tested on a Yealink T28 Firmware 2.43.0.50

- Configure an account as normal on the phone. This will be the account or line you want to be shared
- Click Advanced at the bottom of the page
- Then change Shared Line to be Broadsoft SCA, and click confirm
- Click Phone, then DSS Key
- Set the type to be Shared Line and the Line to be the Account you configured. In the expansion field enter extension@ip or extension@sipdomain. Click confirm to save changes.

- This isn’t working at all. If you get it working see NormT on the IRC.

Troubleshooting

If SLA works for outgoing calls and SLA does not work for inbound calls to the SLA phones, you may have some presence problem related to mixed IP and domain names. When using ODBC you may issue the following SQL statement

```
select sip_to_host,sip_from_user,sip_from_host,hostname,presence_id,call_info from sip_dialogs;
```

and see your server’s IP address in sip_to_host and sip_from_host. If this is the case: Try to modify the dialstring when calling a phone. Example

```
application="bridge" data="{sip_invite_domain=${domain_name}}user/200@mydomain.org"
```

Also take care about similar settings in the sofia profiles. You may try to add one or some of the following settings in the sofia profiles to force the needed behaviour:

```
<param name="manage-shared-appearance" value="true"/>
<param name="dbname" value="share_presence"/>
<param name="presence-hosts" value="mydomain.org"/>
<param name="multiple-registrations" value="true"/>
<param name="force-register-domain" value="mydomain.org"/>
<param name="force-register-db-domain" value="mydomain.org"/>
```

In the internal profile you should add the following parameter

```
<param name="manage-presence" value="true"/>
```

In all other profiles you may add

```
<param name="manage-presence" value="passive"/>
```