Integrating Microsoft Lync 2010 and 3CX Phonesystem using Freeswitch

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Introduction

This guide is the product of a [discussion](#) we had on the Technet forums, which addresses the need of some users to deploy Lync 2010 while keeping their legacy 3CX system working for current desk phones.

It wouldn't have been possible for me to get to this result without the brilliant help of Drago Totev, who provided the initial idea of using Freeswitch and a [useful guide](#) which helped me set up the basic system with Lync 2010. This guide re-uses portions of his tutorial with permission.

Please note that this solution is neither supported or endorsed by Microsoft or 3CX, and shouldn't be used in critical production systems. If you want a fully supported solution I suggest you to take a look at [SmartSip](#) which provides a suitable gateway for Lync, and will let you interface it with… well, just about anything.

This guide will always refer to using a Skype SIP Trunk, since they're easy and cheap to obtain, and can get you up and running in no time, but hopefully you already have a proper trunk provisioned by some provider you can hook your system up to. Some of the configuration parameters will be Skype-specific, but I will always point it out in those cases.

By the end of the tutorial you should be able to receive incoming calls and make both Lync and 3CX ring at the same time, and perform outbound calls as well from both systems.

Ready to start? Let's go!

Target scenario
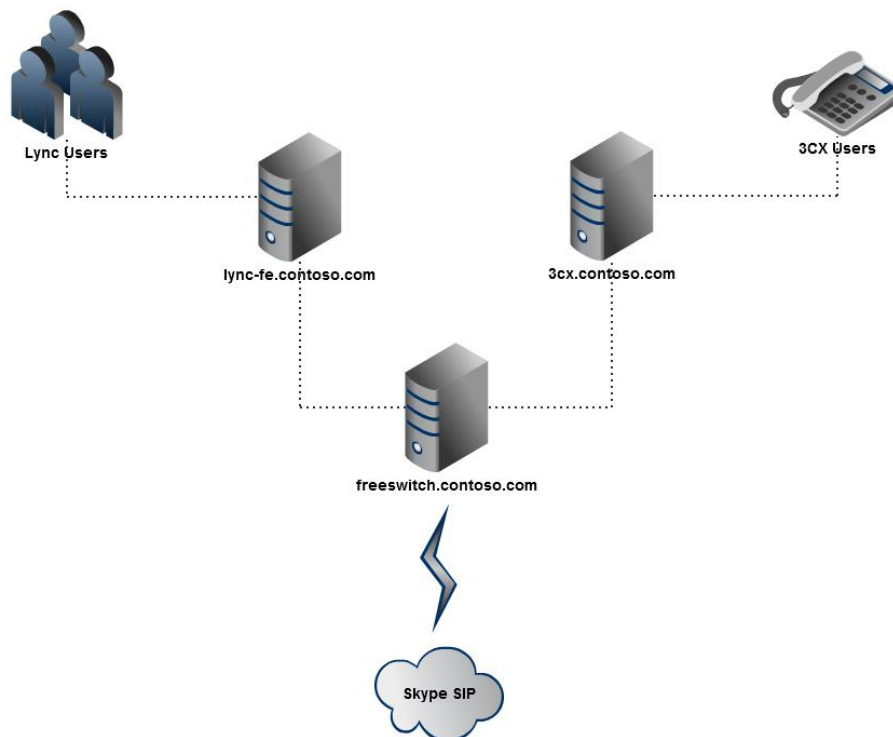
What is needed?

1. Microsoft Lync 2010 FE or EE with Mediation role installed and configured properly;
2. 3CX PhoneSystem 9.x with at least one extension configured with a public DID routable from your trunk provider;
3. At least one Lync user enabled for Enterprise Voice with Line URI provisioned with public routable DID from your trunk provider;
4. A Skype SIP Trunk or your choice of trunk (could be a single channel with at least one DID);
5. A standalone Microsoft Windows Server 2008 R2 for Freeswitch with single NIC, IP address on the same or fully routable subnet with Lync server and 3CX PhoneSystem.

Setting up Freeswitch

Make sure your W2K8 R2 is fully patched before proceed further. On my knowledge, there are no prerequisites prior to installing FS. I would advise to install Wireshark: http://www.wireshark.org/download.html because most definitely something will not work out of the box.

Once Wireshark is installed, download FS (64 bit installer) from this link: http://www.freeswitch.org/node/295

Run the installer and select "Complete" installation. The files will be placed in "C:\Program Files\FreeSWITCH". I would advise you to create shortcuts on your desktop to the installation folder because we will be visiting it at least 1000 times.

1. Navigate to "C:\Program Files\FreeSWITCH\conf" and open vars.xml with text or XML editor.
2. Locate this line –

   ```
   <X-PRE-PROCESS cmd="set" data="console_loglevel=INFO"/>
   ```

   and change "INFO" to "DEBUG". We need raised logging level until get everything going.
3. With the file still open, scroll down and delete the following lines, as they're not required and will only introduce confusion later:

   ```
   <X-PRE-PROCESS cmd="set" data="default_provider=example.com"/>
   <X-PRE-PROCESS cmd="set" data="default_provider_username=joeuser"/>
   <X-PRE-PROCESS cmd="set" data="default_provider_password=password"/>
   <X-PRE-PROCESS cmd="set" data="default_provider_from_domain=example.com"/>
   <!—true or false -->
   <X-PRE-PROCESS cmd="set" data="default_provider_register=false"/>
   <X-PRE-PROCESS cmd="set" data="default_provider_contact=5000"/>
   ```

4. Now we have to set the trunk with our provider. Navigate to "C:\Program Files\FreeSWITCH\conf\sip_profiles\external". There you will see one file – example.xml
5. If you want to set up your own trunk, visit this link: http://wiki.freeswitch.org/wiki/SIP_Provider_Examples and select your provider (hopefully, it will be in the list). Read the notes and copy the portion that exemplifies the provider's registration. Please note that some settings in the dial plans later will be specific to Skype SIP.
6. In this guide, we will use a Skype SIP Trunk, so rename the file to skype.xml, delete its contents and paste the following (note that we're talking about Skype SIP Trunks, the ones you buy using Skype Manager, not a normal Skype account):

   ```
   <include>
   <gateway name="sip.skype.com">
   <param name="username" value="your_skype_username"/>  (looks like 99091209129120)
   <param name="realm" value="sip.skype.com"/>
   <param name="from-user" value=" your_skype_username "/>
   <param name="from-domain" value="sip.skype.com"/>
   <param name="password" value=" your_skype_password "/>
   <param name="extension" value=" your_skype_username "/>
   <param name="proxy" value="sip.skype.com"/>
   <param name="retry-seconds" value="30"/>
   <param name="caller-id-in-from" value="true"/>
   <param name="contact-params" value=" your_skype_username@sip.skype.com"/>
   <param name="expire-seconds" value="45"/>
   </gateway>
   </include>
   ```

7. Start Freeswitch (Start -> Freeswitch). You will see a command prompt window scrolling with a lot of info that may not make sense right now, and finally stop. I suggest you to increase the window buffer to 5000 (You'll find it in the properties of the command prompt window) so you can scroll up enough to see all the debug messages.
8. Type sofia status and press Enter. If you are lucky enough, your provider status will be shown as REGED (registered).

```
freeswitch@skhq-sip-router> sofia status
                Name           Type                          Data         State
==============================================================================================
         internal-ipv6         profile    sip:mod_sofia@[2001:0:5ef5:79fd:2c13:998:3f57:f5a1]:5060  RUNNING (0)
              external         profile             sip:mod_sofia@192.168.10.94:5080        RUNNING (0)
    external::sip.skype.com    gateway            sip:99051000113816@sip.skype.com         REGED
              internal         profile             sip:mod_sofia@192.168.10.94:5060        RUNNING (0)
        internal::lync         gateway                   sip:none@10.10.3.33:5068          NOREG
        internal::3cx          gateway               sip:none@192.168.10.31:5060           NOREG
         192.168.10.94         alias                             internal                 ALIASED
==============================================================================================
3 profiles 1 alias
```

First, note that the gateway is listed as sip.skype.com. This is because above we have "<gateway name="sip.skype.com">". The gateway name is important, because later we'll have to refer to it to place outbound calls. Second, if the registration does not succeed, start Wireshark and see what's going on. Lastly, Google is your friend. Search strings such as freeswitch and provider_name might give you a hint or solution. Do not proceed further until the trunk registration is successful.

9. Now that the trunk is successfully registered, the next step is to set up Lync as a gateway. Navigate to "C:\Program Files\FreeSWITCH\conf\sip_profiles\internal" and create a new file named "lync.xml" with the following content (xxx.xxx.xxx.xxx is the IP address of your Lync (Mediation Role) server. Note the port: 5068. This is the default Lync port for SIP over TCP, but if you changed yours, adjust it accordingly:

```xml
<include>
<gateway name="lync">
<param name="username" value="none"/>
<param name="password" value="none"/>
<param name="caller-id-in-from" value="true"/>
<param name="proxy" value="xxx.xxx.xxx.xxx:5068"/>
<param name="register" value="false"/>
<param name="register-transport" value="tcp"/>
</gateway>
</include>
```

10. Now let's set up 3CX as a gateway too! In the same folder as above create a file named "3cx.xml" and paste the following contents into it (again, replace xxx.xxx.xxx.xxx with the IP of your 3CX server, we'll use UDP for it, since it's more tested than TCP – although it supports it):

```xml
<include>
<gateway name="3cx">
<param name="username" value="none"/>
<param name="password" value="none"/>
<param name="caller-id-in-from" value="true"/>
<param name="proxy" value="xxx.xxx.xxx.xxx:5060"/>
<param name="register" value="false"/>
<param name="register-transport" value="udp"/>
</gateway>
</include>
```
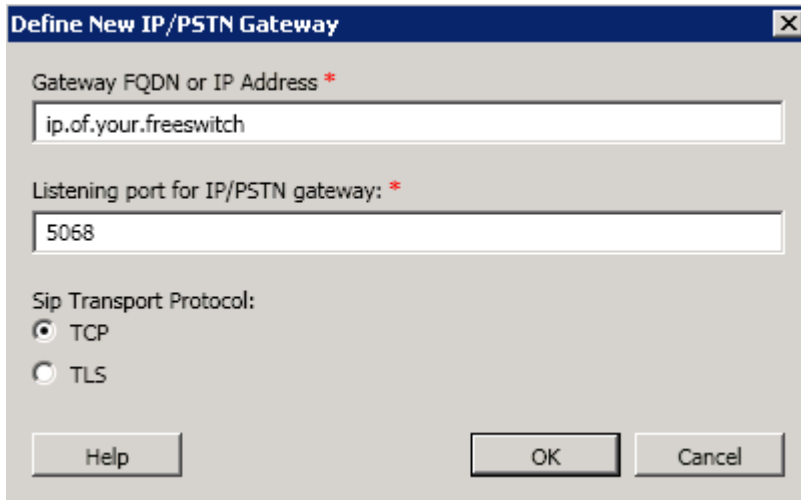
11. The next step is to configure the inbound routing. Navigate to "C:\Program Files\FreeSWITCH\conf\dialplan\public", and create a new file called 01_inbound_did.xml, and replace the content with the following. This file will route any inbound call to both systems:

```xml
<include>
<extension name="INBOUND">
<condition field="${sip_to_user}" expression="^(\d{12})$">
<action application="bridge" data="sofia/gateway/lync/+$1,sofia/gateway/3cx/+$1"/>
</condition>
</extension>
</include>
```

First, the Regex "^(\d{12})$" means "Match numbers that are exactly 12 digits long (e.g.

44207123456). If your provider sends the number with a + prepended in front of it, you may want to change it to 13 digits. In any case, we have to normalize the number to E.164, which is done here:

```
<action application="bridge" data="sofia/gateway/lync/+$1,sofia/gateway/3cx/+$1"/>
```

So, for a 13 digits INVITE, we should use:

```
<condition field="${sip_to_user}" expression="^(\d{13})$">
```

and

```
<action application="bridge" data="sofia/gateway/lync/+$1,sofia/gateway/3cx/+$1"/>
```

12. Now it's time to set up outbound routing! Navigate to "C:\Program Files\FreeSWITCH\conf\dialplan\public" and create a file named 00_outboud_did.xml with the following content:

```
<include>
<extension name="ALL">
<condition field="destination_number" expression="^\+(\d*)$">
<action application="ring_ready" />
<!--<action application="set" data="ignore_early_media=true"/> -->
<action application="set" data="proxy_media=true"/>
<action application="bridge" data="sofia/gateway/sip.skype.com/$1"/>
</condition>
</extension>
</include>
```

Make sure your Lync will be set up properly to send the number in an expected format to Freeswitch, Drago for example prefers to remove the + before sending INVITE to the gateway. You may need to uncomment the ignore_early_media line, depends on the provider, in some cases it can help with the ring back.

One last note: during the initial phase of configuration and testing, disable Windows Firewall on the Freeswitch server. Once you get it running, go back and adjust accordingly.

Configuring Lync

1. Remote desktop into your Lync Frontend server
2. Start Lync Topology Builder
3. Right click on PSTN Gateways, and create a new gateway as follows:

4. Right click on your Mediation server inside Mediation pools
5. Assign the newly created PSTN gateway to it
6. Publish the topology
7. Important: Restart the Lync Mediation Server service or these changes won't take effect
8. Start Lync Control Panel
9. Go to Voice Routing -> Dialplan and right click on the Global dialplan
10. Scroll to Normalization rules and create a new one

11. Set up the parameters as follows. This simple dial plan means that if we start a number with 0, exactly 11 digits long (which is the way you dial a national number in the UK), e.g. 02071234567, it will automatically convert it in international format. This is specific to Skype, as it wants to receive the number with an international prefix in front of it. If you have another provider it may allow you to dial national numbers without having to prepend the +44. Please note the test conversion I did on the bottom of the screenshot:

12. Now go to Route and create a new route, to redirect external traffic towards Freeswitch. This route will match anything starting with + or with 0... (part 1):

13. Scroll down in the interface and fill in the second part (we need to suppress the caller ID as Skype wants the username, but your provider will probably be fine with the caller ID being transmitted). And add the associated gateway:
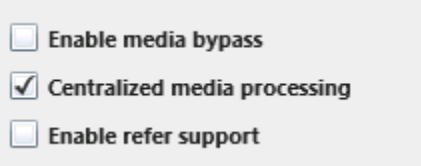
14. Open Trunk Configuration and edit Global, make sure the three magic tickboxes in the middle of the screen are set as follows if you want to get traffic through:

15. Edit an extension, and enable it for Enterprise Voice, and set up the line URI accordingly. For example, if your SkypeIn number you bought is +442071234567, in that field type tel:+442071234567;ext=567 – the second part is for when you'll want to set up internal phone calls to the extension directly.

Now, if you dial your DID from any phone, you should see a lot of text scrolling in Freeswitch and then Lync will start ringing. If it doesn't please check the debug info thoroughly, as it will show where the call is hanging, it could be simply a regex failing because you had the wrong number of digits in your inbound dialplan.

Configuring 3CX

The process of configuring 3CX is pretty straightforward, once you know a few gotchas that made me spend a lot of time in the effort of fixing the configuration.

1. Login into your 3CX admin interface
2. Click on VOIP Providers
3. Create a new Generic SIP Trunk
4. Fill in the parameters as follow, to complete the wizard:

VOIP Providers

Add VOIP Provider Wizard

VOIP Provider Details:

Enter the hostname and port for your VOIP Provider's SIP Server

SIP server hostname or IP	your.freeswitch.ip
SIP Server port	5080
Outbound proxy hostname or IP	your.freeswitch.ip
Outbound proxy port (default is 5060)	5080

< Back   Next >

Enter a random external number, 0 as user id and 0 as password, it's not important:

VOIP Providers

Add VOIP Provider Wizard

Account Details

Enter the Authentication ID, Password and number of your account

External Number	10020
Authentication ID	0
Authentication Password	•

Simultaneous Calls

Maximum simultaneous calls	100

< Back   Next >

Don't bother with the routing right now and just set it to end the call:

In the next screen just skip the Outbound call rule so we can get out from that wizard.

5. Now edit the VOIP provider we just created and in the advanced tab make sure the settings look like this:

6. Leave outbound and inbound parameters as they are and move on to the DID tab. This is very important, you need to add ALL the DID numbers that come with your trunk so we can later route them towards our internal extensions:

7. Now, most important step of all, 3CX will keep on rejecting your incoming calls as unauthorised until you do this. Open the Source ID tab, then tick the Source identification by DID box, then press Add DID and select all the DIDs that come with your trunk as follows:

8. Now make sure that in the outbound rules you have Freeswitch selected as primary route for all the external phone calls, and let's set it to remove the 0 and prepend +44 (this is Skype specific, other providers won't require you to provide an international phone number):

9. Now the boring bit: Go to inbound rules, you'll see a list of all the new DIDs you configured for your Freeswitch provider. You'll basically have to set each one up to connect to an extension (or to just end the call if it's an unallocated number). That's pretty straightforward:

How to verify if everything works

Ok, now that we've set everything up test the following thing in order:

- From a mobile, call your DID number (e.g. our imaginary +442071234567) and check the Freeswitch window: you should see lots of debug info scrolling through (please mind that if you use a Skype SIP Trunk, the phone will ring 3 times before actually seeing anything happening in Freeswitch, that's due to how Skype works, they take their time to route calls through). Once you see all the debug info, your Lync and your 3CX extension should ring at the same time. Make sure you can hear audio on both systems by doing a couple of phone calls.
- From Lync, try to dial an external number, in our case it could be a national call starting with 0 (e.g. 07791234567), Lync should convert the number into +447791234567 and your external phone should ring after 3-4 pages of debug info in Freeswitch.
- From 3CX try to dial an external number, the same as above, should ring fine.

If you have problems, as I said before read the logs in Freeswitch, and in 3CX's Server activity log.

A useful debugging tool is to turn sip tracing on in Freeswitch, type the following in the console: sofia profile external siptrace on (and off to turn it off)

If you need to change your configuration files (and trust me, you will), instead of shutting down Freeswitch, the following commands will reload the whole configuration:

reloadxml
reload –f mod_sofia

Hope you enjoyed this guide, and if you got so far without problems, you probably deserve to get yourself a good beer!

Max Sanna