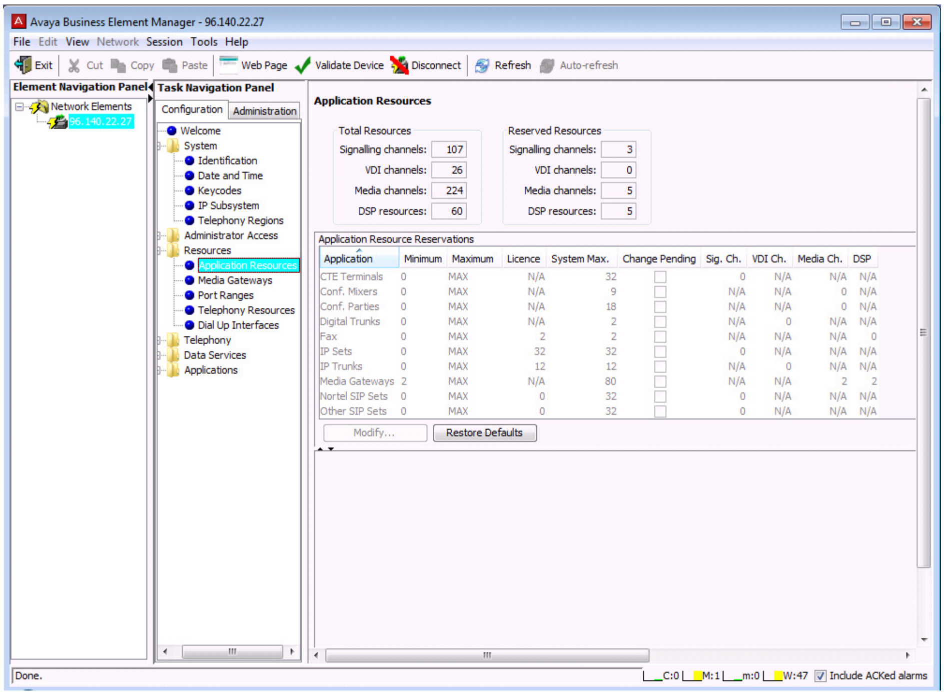
Avaya BCM 50 and 450 unofficial SIP phone feature provided at version 6.0

This document has been written to explain how to configure and set up SIP phones onto the BCM 50/450 V6.0 system. This process is totally unsupported, not documented or released by AVAYA and this feature will probably only be available on unofficially modified systems that allows keycode changes to be made by using the “Dark Art” method of SSH access which won’t be covered here!.

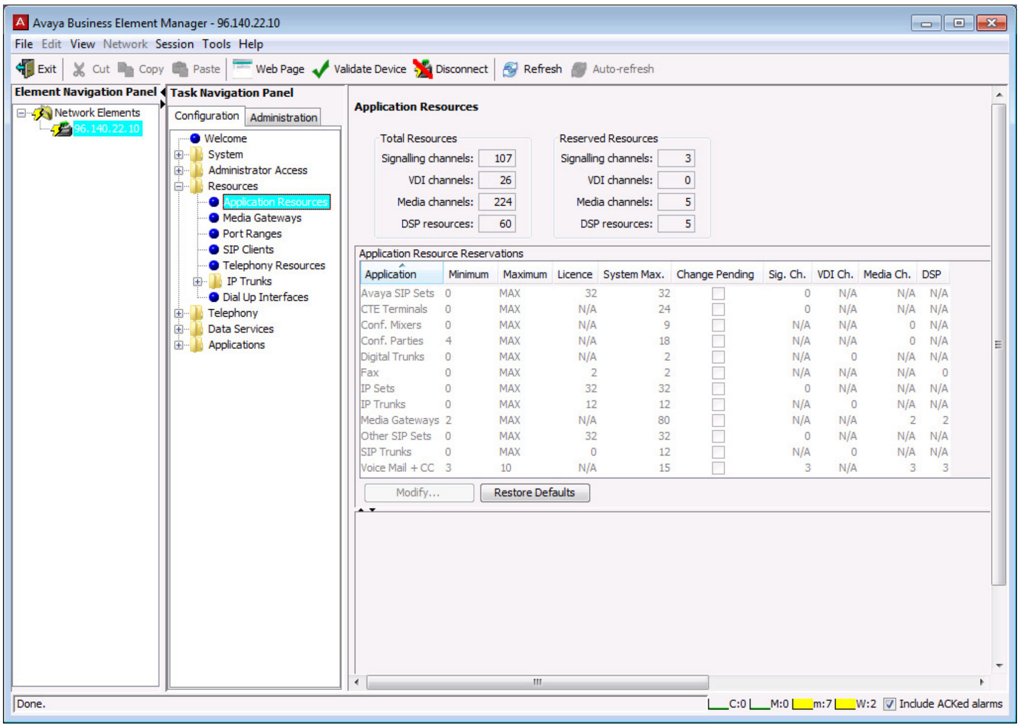
It looks as if the development work for SIP phone was started at release 5.0. However the only reference to this can be found in the Application Resources section as shown below. Note the nice “Nortel SIP Sets” and “Other SIP Sets” at the bottom. This is all that exists of the SIP phone feature and therefore is of little value to you, even after the extra packages have been added in.



The two packages needed in the BCM keycode to enable the SIP Phone configuration are 282 and 283. The maximum allowable ports available are 32 for the BCM 50 and 300 for the BCM 450 system. This is the same values that are used for the standard Nortel / Avaya IP sets.
Your modified keycode would need extra lines of data like this below. (BCM 50 example)

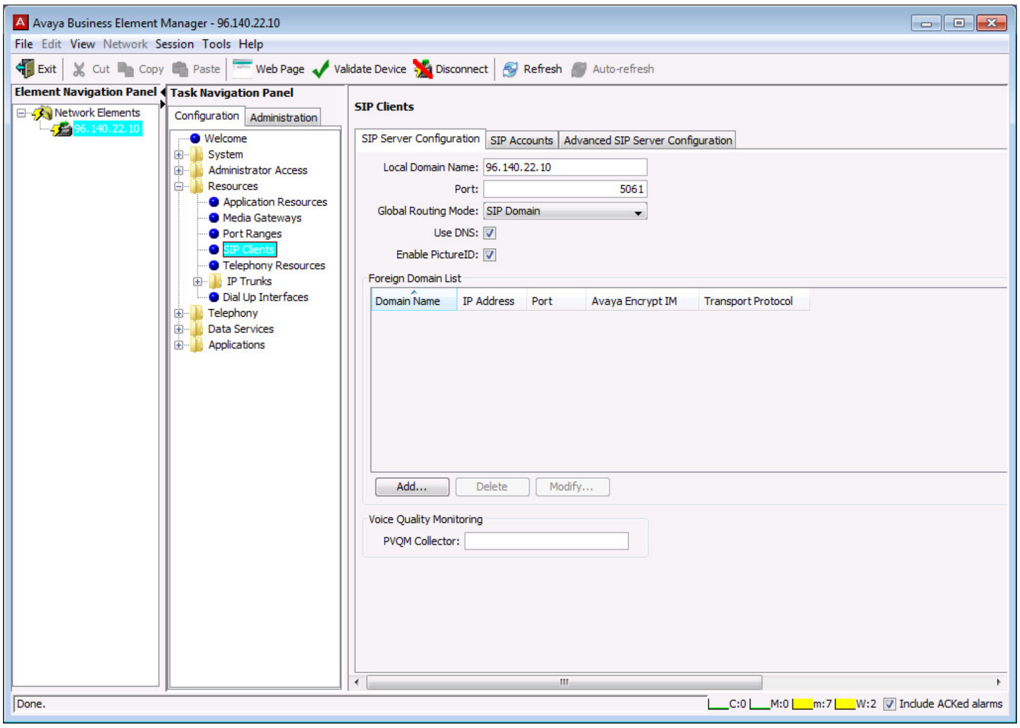
```
<feature>
  <code>282</code>
  <data>32</data>
  <name>Avaya SIP Sets</name>
  <comp></comp>
  <expiry></expiry>
</feature>
<feature>
  <code>283</code>
  <data>32</data>
  <name>Other SIP Sets</name>
  <comp></comp>
  <expiry></expiry>
</feature>
```

At release 6.0, a new configuration area opened up under “Resources” as shown below.



The Nortel SIP Sets line was replaced with Avaya SIP Sets. A new “SIP Client” area appears at 6.0.

SIP Clients First tab called SIP Server Configuration



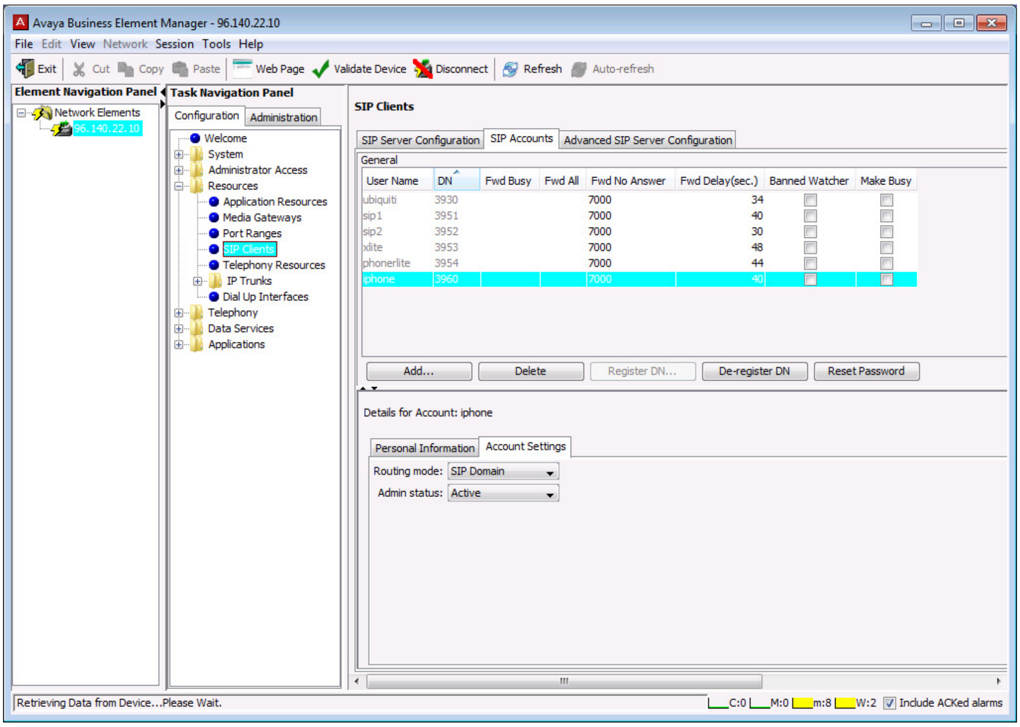
This is the area where new SIP phones can be configured. The examples I have shown will be explained, giving as much information as possible to assist you in setting them up.
Please be aware that there might be other ways on configuring SIP phones.

My BCM 50 internal LAN1 IP address is 96.140.22.10 and I've added this in the "Local Domain Name:" box on the above screen. I've also selected the "Global Routing Mode:" to be **SIP Domain**. Also "Port:" is left as **5061**. I'm not sure if the other areas on this screen are important?.

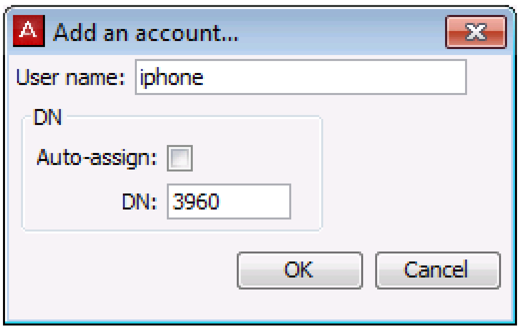
The "Use DNS:" tick box works, however I couldn't initially get the "Enable PictureID:" tick box to function without an error of "true is not a valid Enable PictureID" being display after I had ticked it, unticked it and then tried to tick the box again. Please leave it ticked when used for the first time!. There is now a workaround for this by making changes via SSH to enable the feature which wasn't originally finished by Avaya due to it being dropped in favor of their IP Office product.

SIP Clients. Second tab called SIP Accounts

This tab is where you configure up the actual SIP DN's. I'm going to show you how to add a SIP phone for extension 3960 to be used on an IPhone via the Zoiper App that can be downloaded.



Click on the "Add" tab and enter in the name (which must match the name entered on the SIP device!) without any spaces. Add in the DN or tick the Auto assign box. Click OK to add it in.



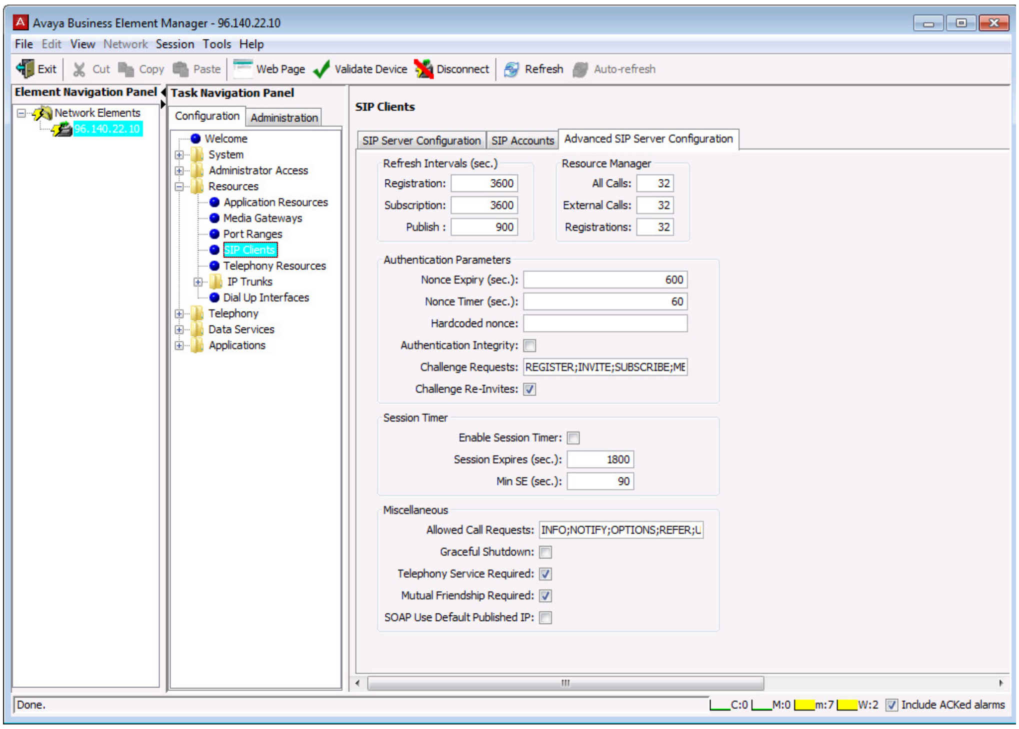
Against the Account Settings tab, select "Routing mode:" **SIP Domain** and for the "Admin status:" box, select **Active**.

There is another tab for Personal Information. You can add in details such as name and contact details. It doesn't appear to have any effect on the operation of the SIP phone and the details entered are found in the users browser at https://"BCM IP address"/mailboxmanager.

Against the SIP phone DN line are a couple of columns where you can add in any forwarding on busy and no answer etc. This area has been left untouched in case any problems occur. However, changes made in Mailbox Manager are shown in these columns.

SIP Clients. Third tab called Advanced SIP Server Configuration

The only one area I altered here was the box called "Authentication Integrity:" this was **unticked**!.



SIP Monitor

There is another new screen under "Administration" and "System Metrics" called SIP Monitor. More information is available on page 6 in the guide.

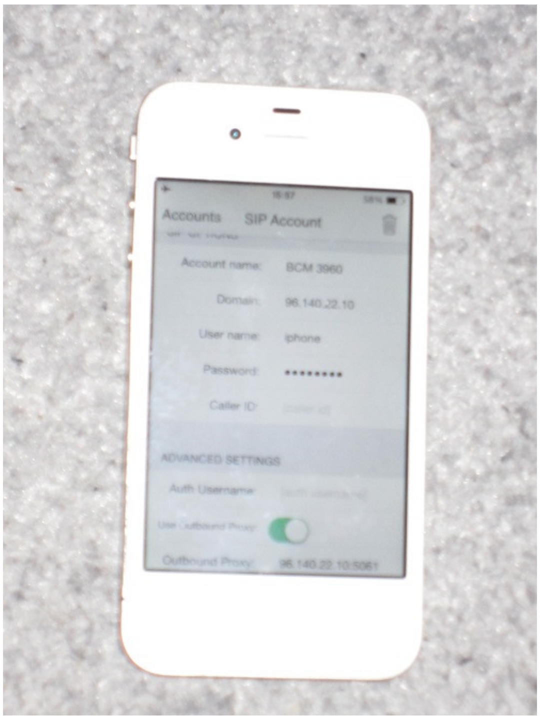
Configuring up a SIP Phone

I’m going to show you a couple of programs that have worked for me. The first one is an app called Zoiper that is freely available for Smart cell phones and computers etc. In my case, the example was used on an IPhone 4. I’m going to attach photos of the areas to configure and check.

Install the Zoiper App on your Smart phone and then load up the app. Find the area for Accounts and click on the “**Add**” or “**+**” icon to provide one. If you get a message asking if you have an account with a username and password, then please select the “**Yes**” option.

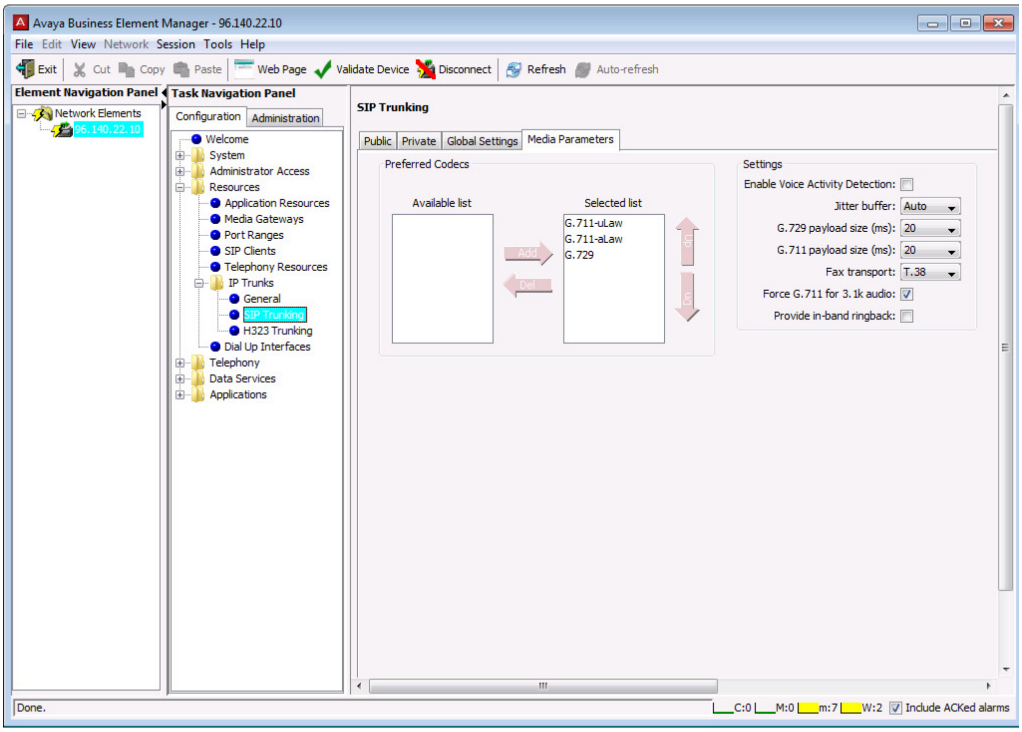
Then select the “**Manual Configuration**” option. The account type will be a **SIP account**.
This area for SIP Options is very important. The “Account name:” can be the same as on the BCM or another name!. The “Domain:” is the BCM LAN1 IP address. The “User name:” has the be the exact name used in the BCM SIP account configuration area. The “Password:” is the **default nnadmin password** and no other password will work, even if you change the nnadmin password account on the BCM. The “Caller ID:” field is left blank.

An example is shown here

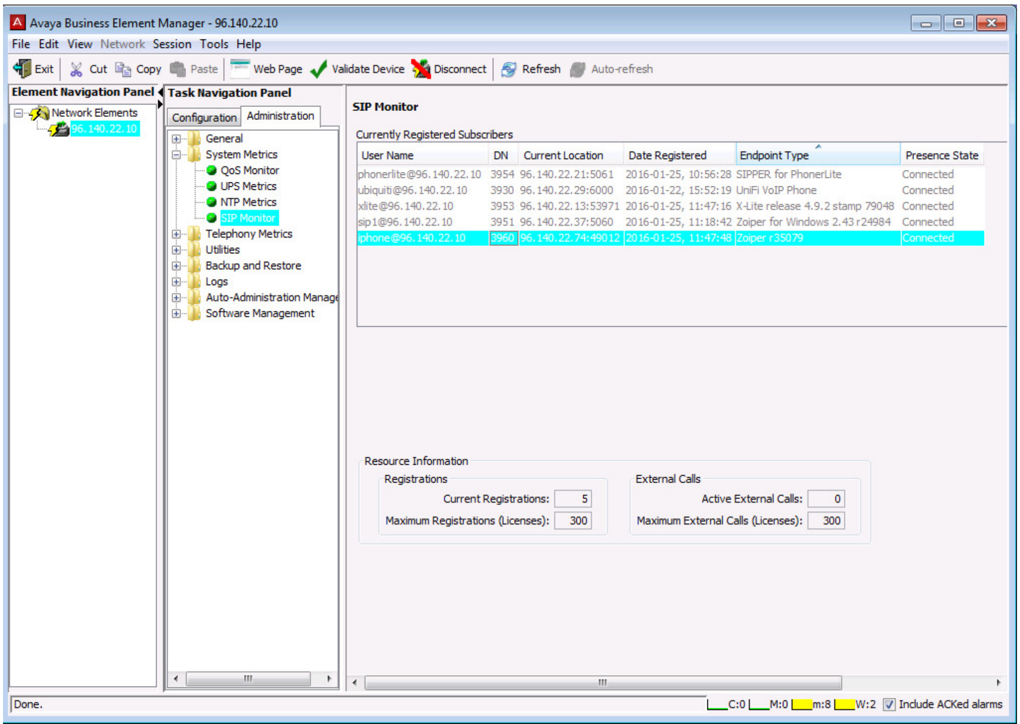


Now go to Advanced Settings and the “Auth Username:” field is left blank. The “Use Outbound Proxy:” is flicked to green (On) and thenset the “Outbound Proxy:” to be the BCM LAN 1 IP address with **:5061** at the end.
Once that is done, then go to Advanced settings. The Features area isn’t used. The Network Settings area has by default Enable STUN: turned on (green) and I’ve disabled this by **switching it off**. All other settings in this area are untouched.

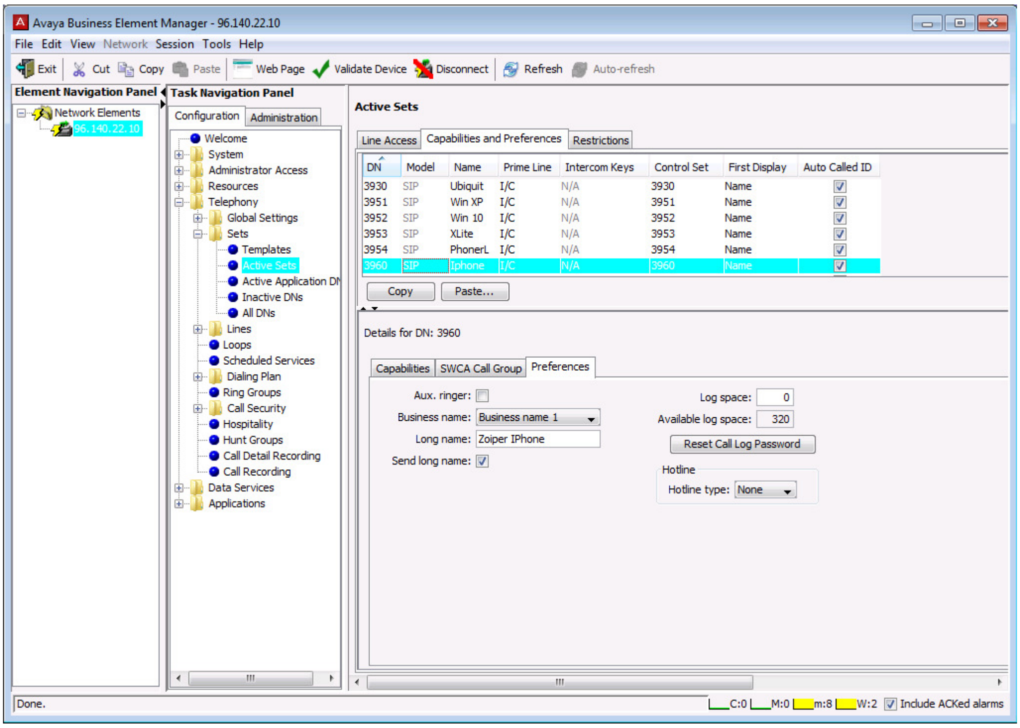
The Audio Codecs section has many protocals and most of these aren’t used. I’ve turned them all off except for the **a-law**, **u-law**, G729 (I paid extra for this one, but I don’t think it is needed!). The reason for this is that the BCM only suports three of them and these can be found under Configuration and Resources, IP Trunks, SIP Trunking and finally Media Parameters. You can play with adding the other Audio Codecs on the phone as mine seems to be OK.



Once all the values have been entered in, click on the registration tab and hopefully the phone will register OK. If it doesn't, then please check your settings and also open up the SIP Monitor tool, you will see that the phone I've done has registered OK as shown below.



Also you can look under active sets for a few more details as Model **SIP** is now listed.



The example above has been tested with a standard IP phone that has been connected onto the BCM via the remote worker NAT Tranversal license code without any problems. Because some SIP phones and softphones are non Avaya versions, then be aware that all the typical BCM features and codes such as speed dial, and voicemail etc might not be available as you have no “Feature” button!. I’m not sure if there is a workaround for this.

Line assignment

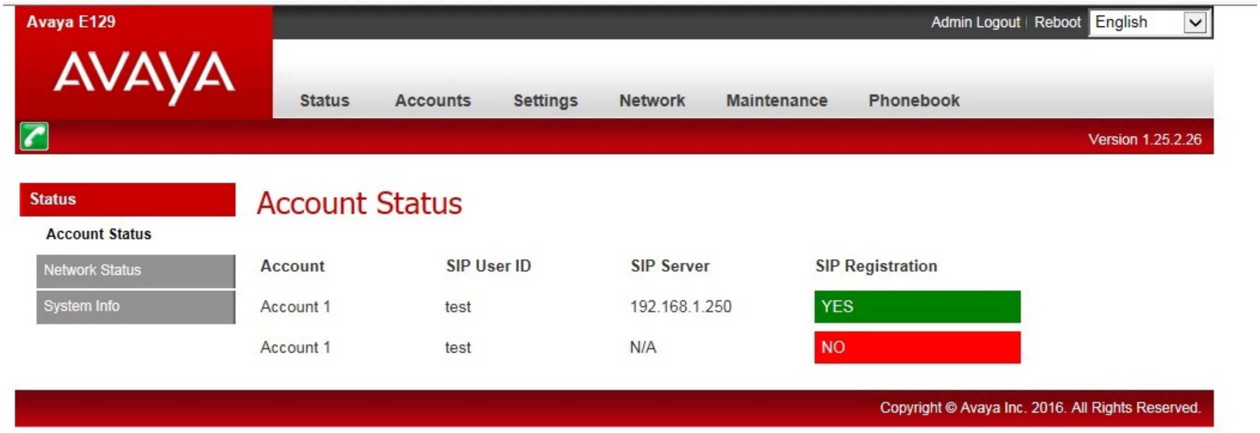
One thing to consider is that most SIP phones will probably be non Avaya softphones or phones etc. It’s been notices that any SIP Phones that have any outside lines assigned to them will cause problems with calls going to the general mailbox after one ring.
Target lines don’t appear to be affected, but it’s best **not to assign any lines**. This might be due to the Non Avaya phone only having limited features available. This issue was noticed on a BCM 50 with Analogue lines.

Initial testing

When calling the SIP phone for the first time that has a mailbox configured, you might get the phone’s voicemail greeting message. This will be an indication that the SIP phone hasn’t been configured correctly.

Avaya SIP phones and softphones

The Avaya SIP phone E129 registered OK. A mailbox was created and tested by leaving a message and it was retrieved. The message lamp lit up fine upon leaving a message on the Avaya phone. Other Call Pilot functions were carried out without any issues.



Router changes

As you might want to try out calls via the phones to external destinations, you may have to configure your router as one way transmission has resulted on some SIP calls.

My router has been set up with the following changes.

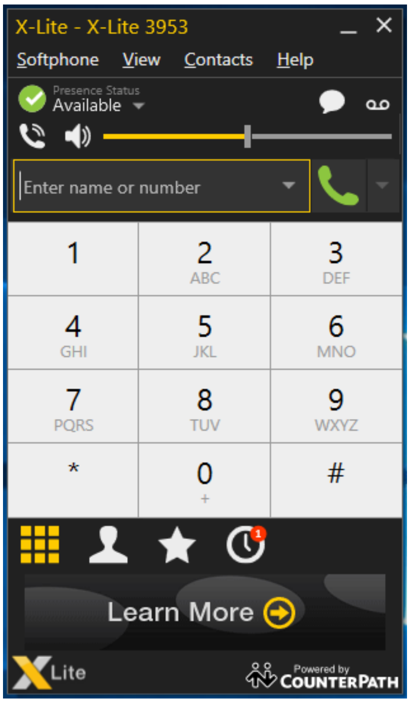
"BCM RTP over UDP" protocol=UDP port range=30000-30099
"BCM RTP " protocol=UDP port range=28000-28249
"BCM IP Phone Sig " protocol=UDP port range=7000-7002

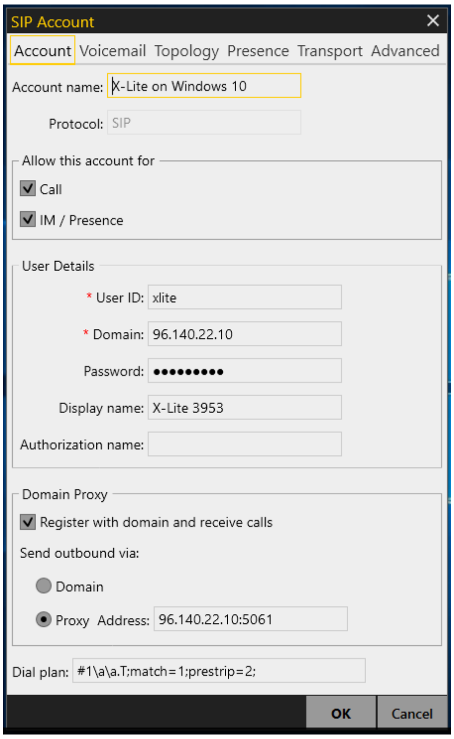
Using non Avaya SIP software programs (Many programs are available other than those below)

X-Lite Softphone

This program is freely available from the internet for use on MAC and Window computers. The software was from the same developer “CounterPath” that Nortel had used before.
The software is available at http://www.counterpath.com/x-lite-download/

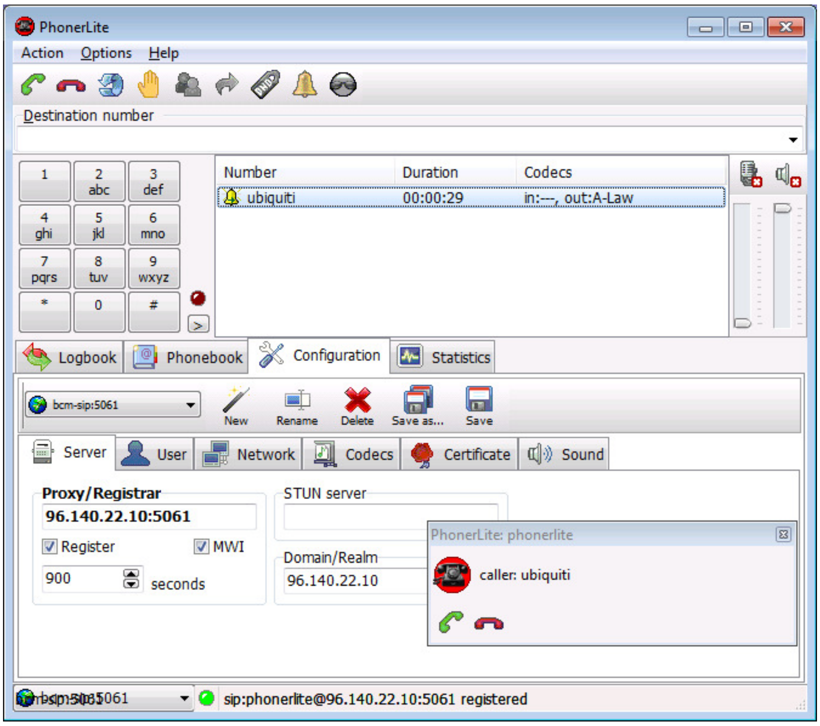
It was easy to install, Voice calls were fine but I couldn’t get the Video to work. This might be a problem on my computer.

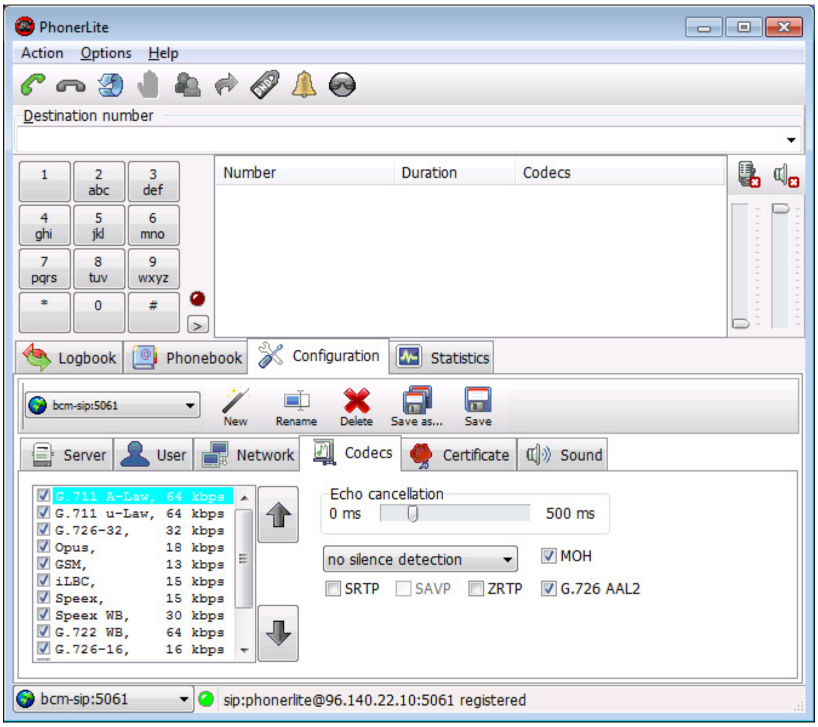




PhonerLite Softphone

PhonerLite was tried out on Windows 10 as shown below with an incoming call in progress and another screen shot with the typical codec settings that have G 711 U and A law as first choice. It's available for download at http://www.phonerlite.de/download_de.htm

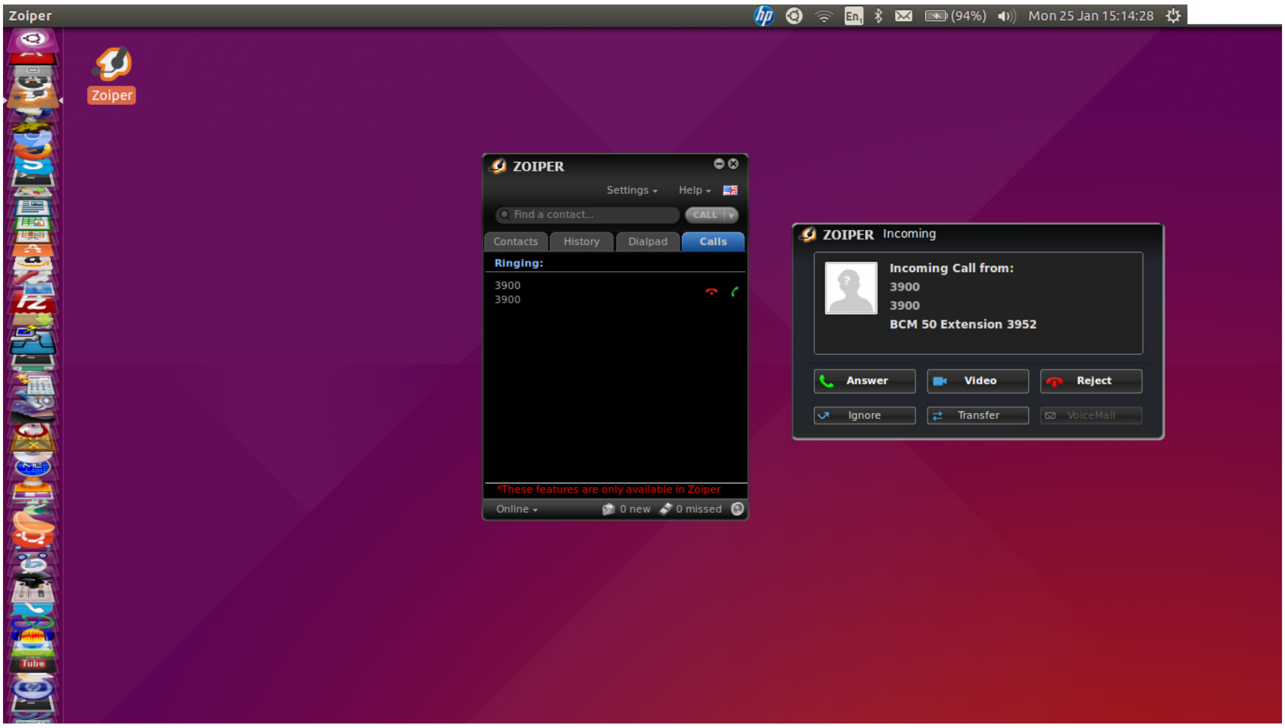




Zoiper Softphone

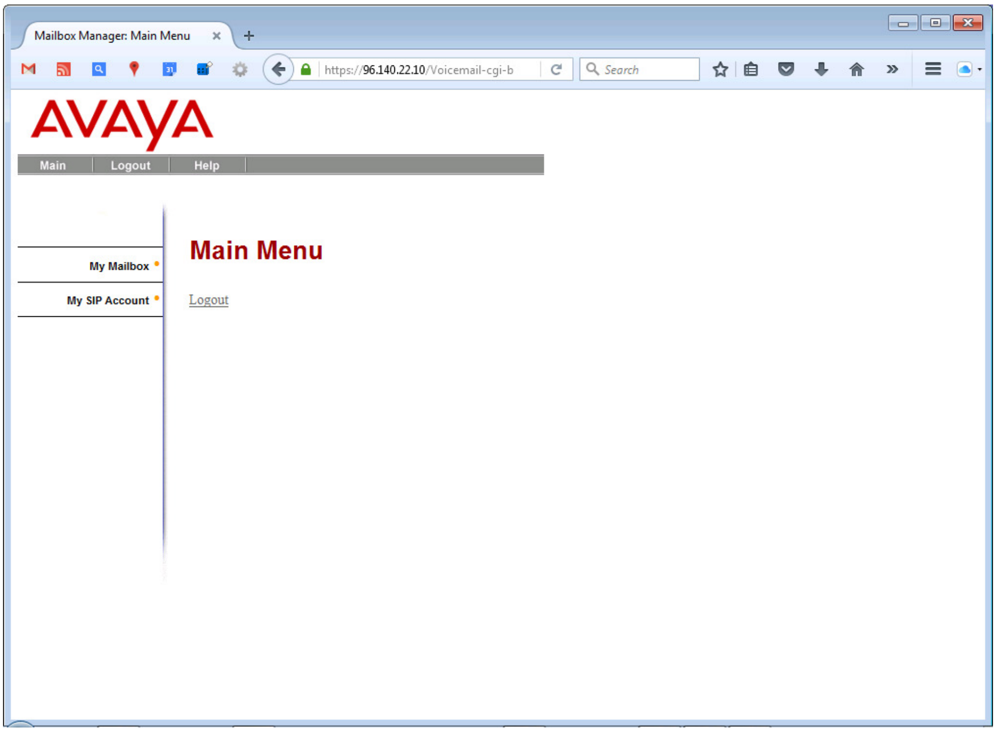
I have used a copy of Zoiper Free successfully on my Windows XP and it is available for download at http://www.zoiper.com/en for almost all types of operating systems.

I have tested it on a Linux Ubuntu 64 Bit laptop as shown below.

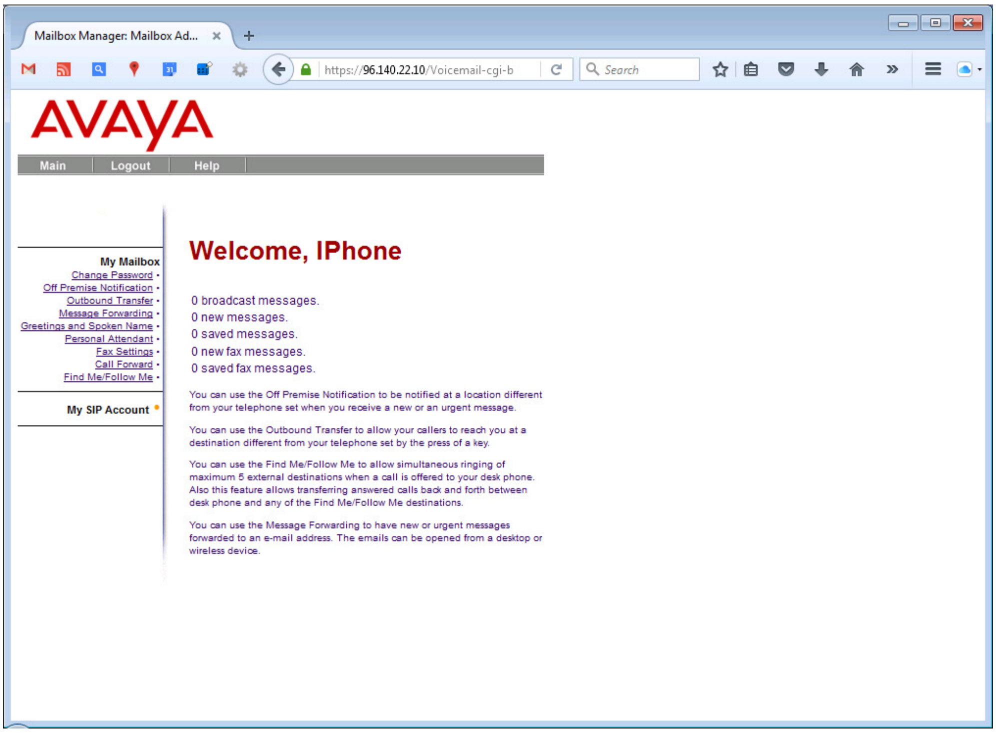


Extra SIP features found in the Mailbox Manager Web browser

Create a voicemailbox for your sip userin Call Pilot first, and then login via Mailbox Manager. You will probably have to log in with the deaful of "0000" and then change the passcode to something else.

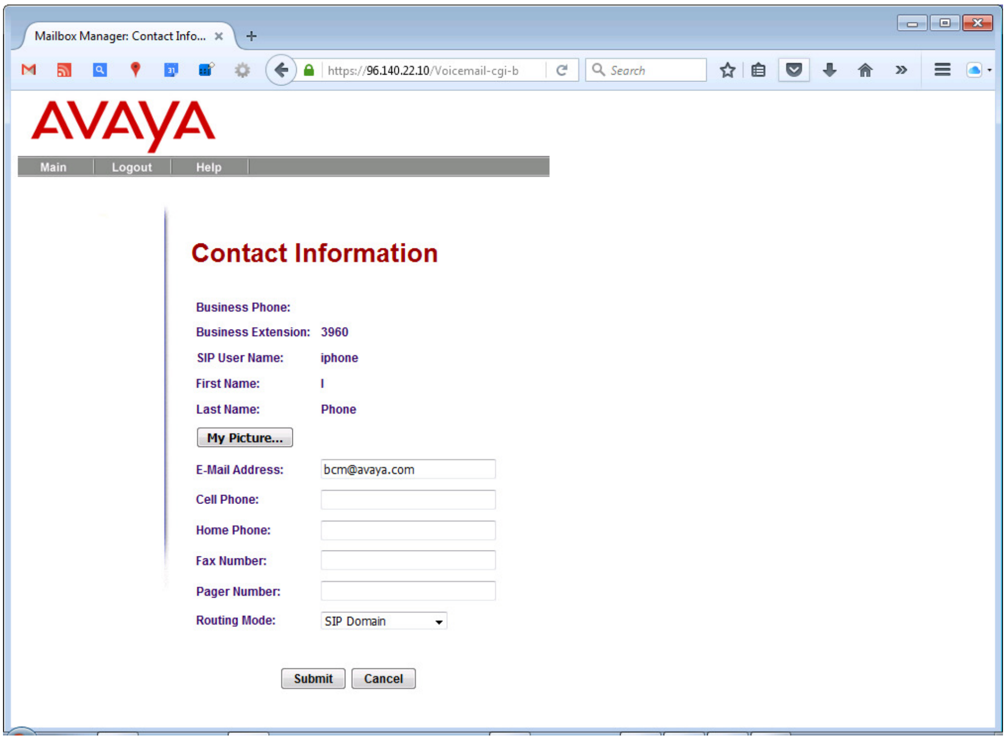


Once logged in, you can access the full voicemail section. Notice the new "**My SIP Account**" area!

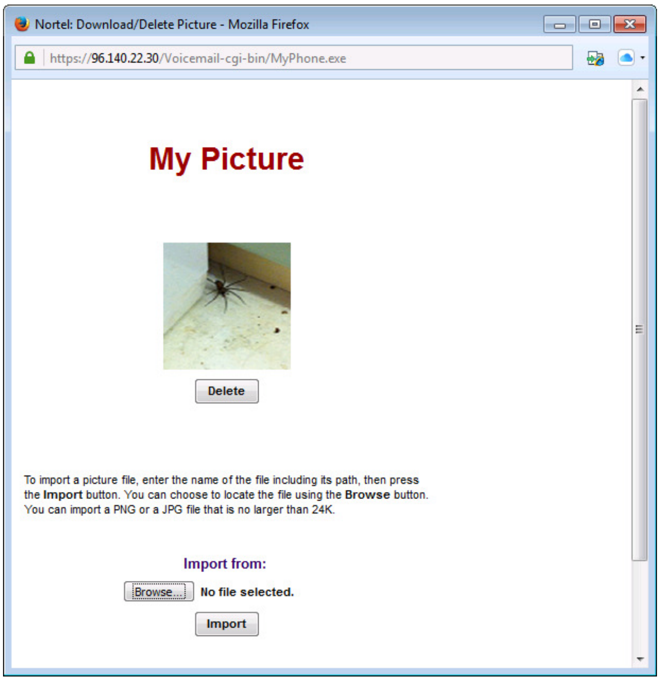


In “**My SIP Account**”, there is a field where you can change your contact information. Also you can change SIP parameters like the routing mode, which might be “**SIP Domain**”.

A picture can be inserted under the “My Picture” button but the file has to be in a PNG or a JPG format that **isn’t larger than 24K**. I couldn’t get this to import correctly until a patch was applied via SSH to enable it as this feature appeared to have never been finished by Avaya.

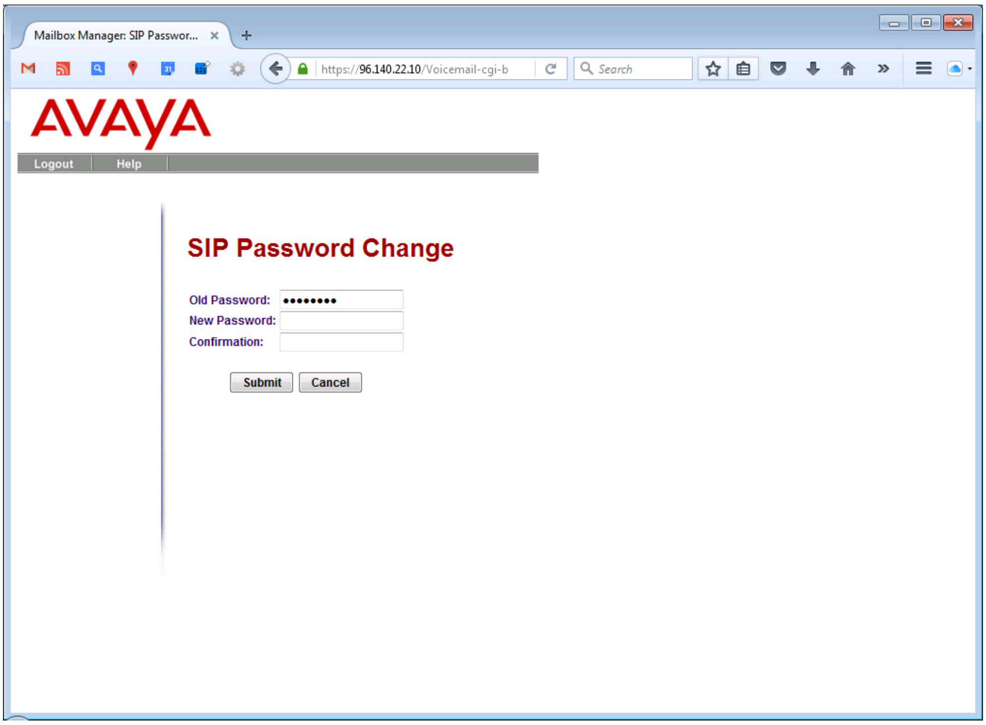


Here is a typical photo uploaded after the patch was applied. It looks as if this feature was developed to allow certain SIP phones to send out a photo to be displayed on the called SIP phone only, as it can’t be done with standard sets. I’ve not been able to make use of this feature on my test phones!.



Finally there is the option to change the SIP registration "nnadmin" password.
It's best to change it to something else other than the default nnadmin password and you should see your SIP set gets unregistered. Go to the web interface of the SIP set and change the password to the new one and the SIP set will register again. The voicemail lamp should be on again because a voice message was left for it. While the set was not registered it was dialed from another BCM digital set and then the voicemail prompt came up allowing a message to be left.

This means that we can change the registration password for each SIP account and if there is a need to reset it to default, then this can be done from the "SIP Client" area under "SIP Accounts" and the Reset Password box in Element manager.



Note!

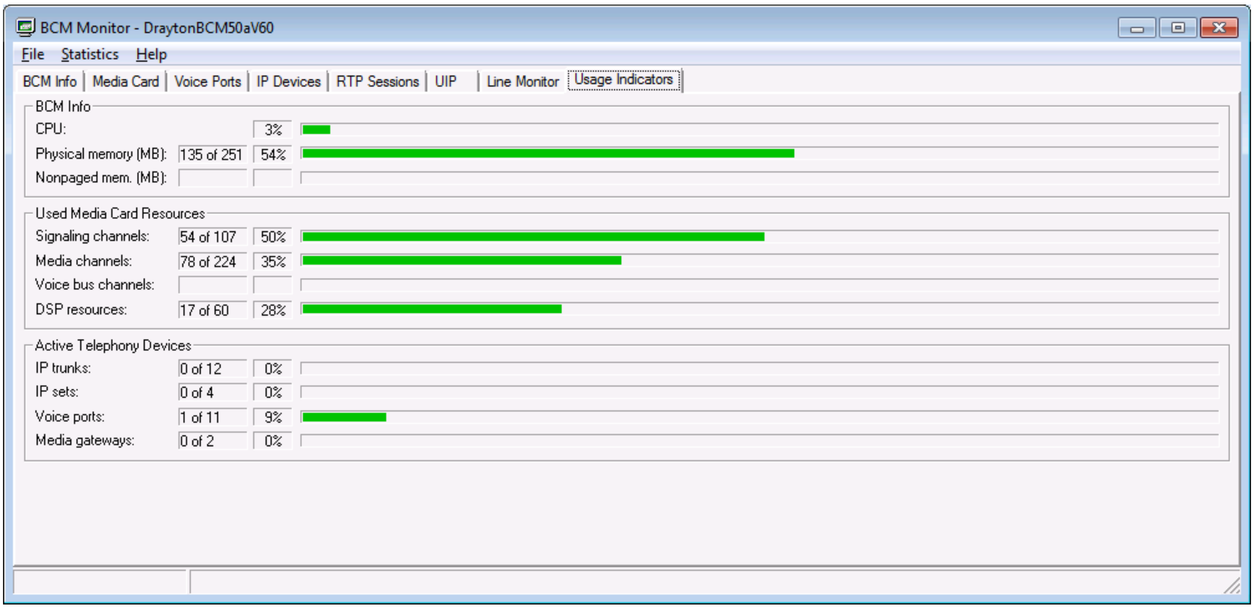
Unlike standard IP phones whereby only one set can register at a time as it uses the phones MAC address for verification etc, SIP Phones can register on multiple devices at the same time and this can lead to a security issue. I'd strongly advise the BCM administrator to create the set first and then the mailbox in Call Pilot. Once that is done to then go into the new mailbox and **change the SIP registration password** and inform the new user prior to the phone being used as this will prevent any sets retaining the default password.

Troubleshooting and tips

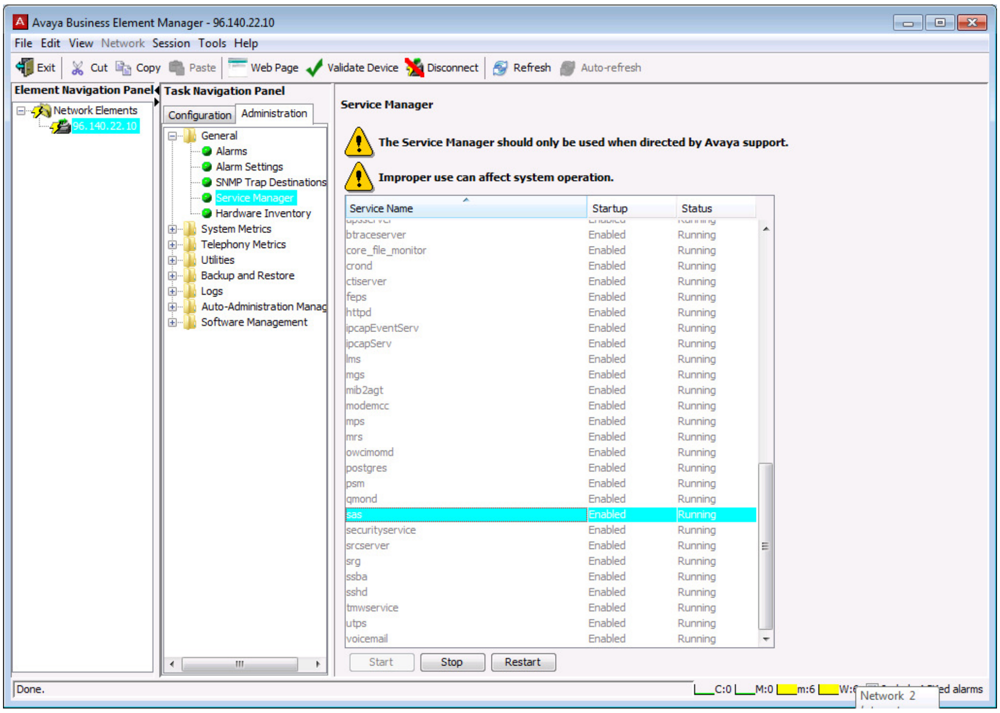
One problem that has appeared when using SIP phones is that the BCM Monitor tool becomes unusuable if any SIP phones have been registered onto a BCM 50 system. The BCM 450 might not be affected as it would have a bigger processor. Here are the main points below.

Sip phone and BCM 50 Monitoring with the "sas" service problem

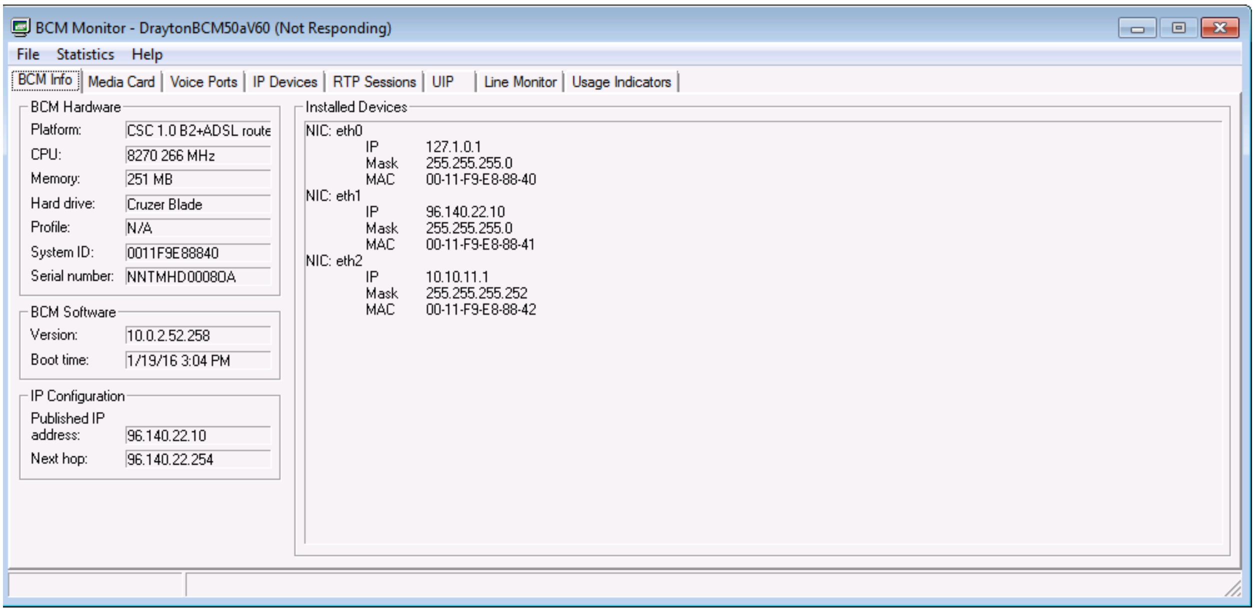
Typical BCM 50 running at a normal operating pace with CPU physical memory at a low level.



In Service Manager, the sas service (SIP Authentication Service) is enabled and running OK.



As soon as a SIP phone has been registered, the BCM monitor becomes unusable



If the BCM monitor tool is closed down and then reopened again, the problem is still the same which suggests that enabling a SIP phone causes a lot of CPU usage on the phone system?.

If all the SIP phone applications are closed down, then the BCM monitor tool is still showing as (not responding)!. The only way to fix this is to **stop the sas service** as shown on the previous page via the BCM Service Manager tool and then start the BCM monitor program. I briefly see a high CPU physical memory usage when the BCM monitor tool is run again.

Nortel / Avaya phone to SIP phone

It seems that one audio is present. More testing to be done on this.