

## Callcentric settings used on a BCM phone system to provide a new SIP account

Here are a few screen shots taken showing the settings that I used to set it up. I registered via the Callcentric service at [www.callcentric.com](http://www.callcentric.com) and all they require from you is a valid email address. You are given a USA number which can be used anywhere to make and receive calls.

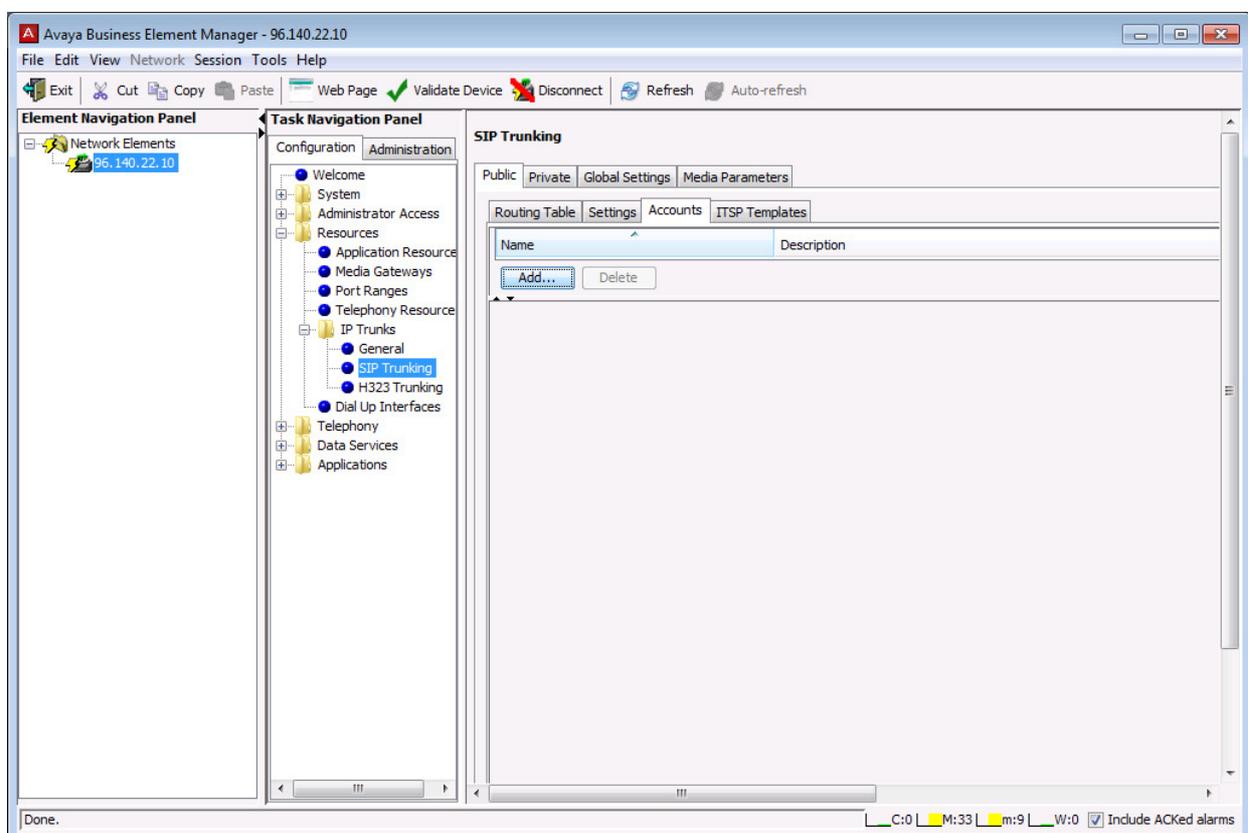
The good thing about this is that the service will receive CLID and send out CLID with the call, but you can turn this on or off via the web browser. There aren't any restrictions and of course the good thing here is that Callcentric to Callcentric calls are free. This means that I can make and receive calls from anyone else who has a Callcentric number connected and this can also be a PC Softphone, Mobile phone apps such as Zoiper,, SIP to analogue devices ( e.g. Cisco Linksys 2 port adapter) and SIP trunks etc.

This is very handy for customers who want a USA presence to appear on their phone system that has SIP trunks and features enabled.

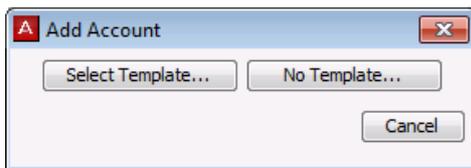
In the example below, I've explained how I added an account on a BCM 50 at 6.0 and also that I gave it a two digit access code (79) that was set up in the routing table to allow me to dial out to other Callcentric users. If you are outside of the UK, you can still have this on your BCM system and configure it in almost the same way as I did as shown in the instructions below.

### SIP Trunking (Public) and Accounts

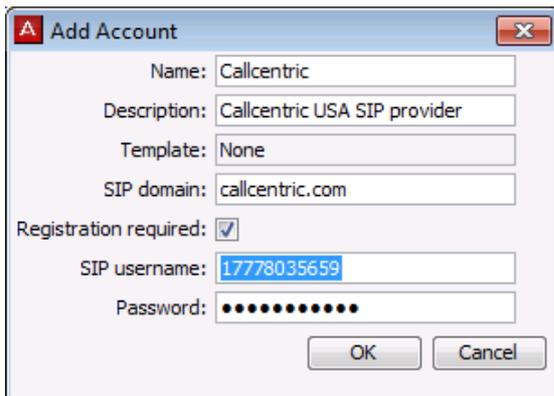
Log in with Element Manager and navigate to "Configuration", "Resources", "IP Trunks", "SIP Trunking", "Public" and "Accounts". Click on the tab option to "Add".



Select the “No Template...” box on the right.

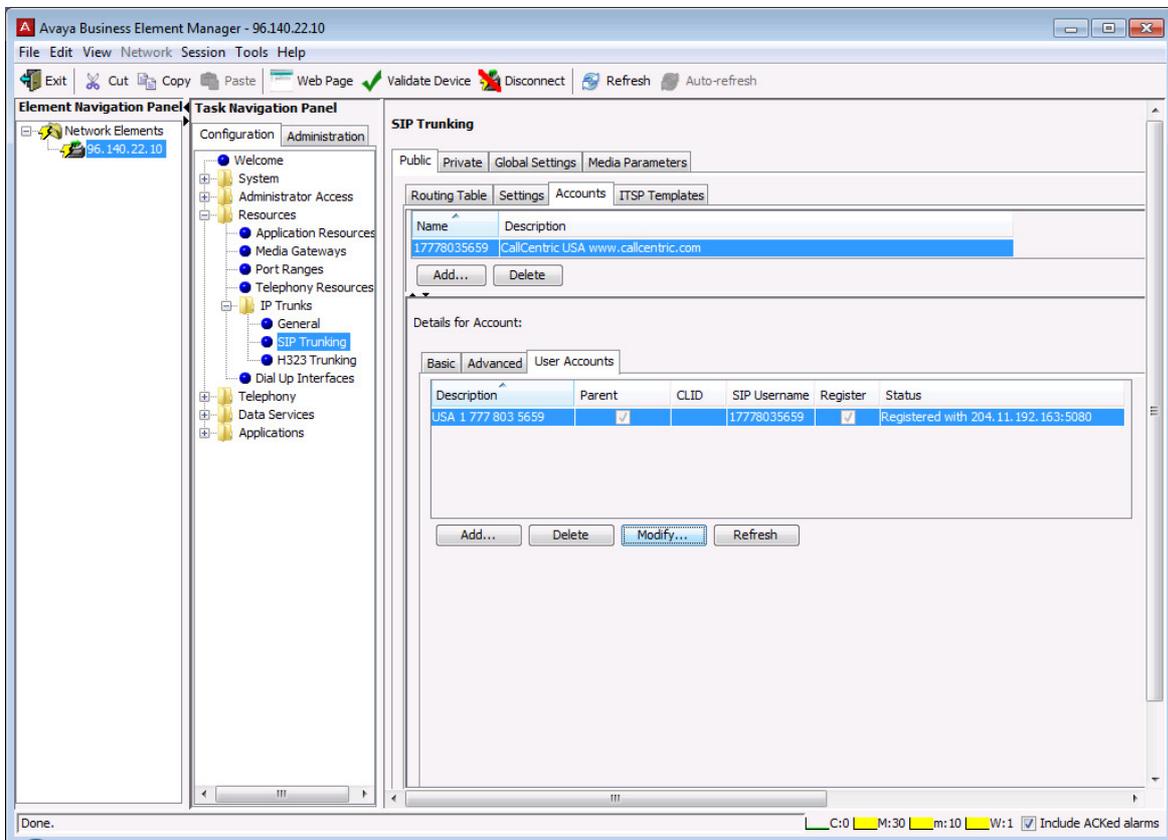


Fill in the details given to you via Sipgate as shown in the example below. The Name: and Description: entries can be whatever you want to use. Click OK, once completed.



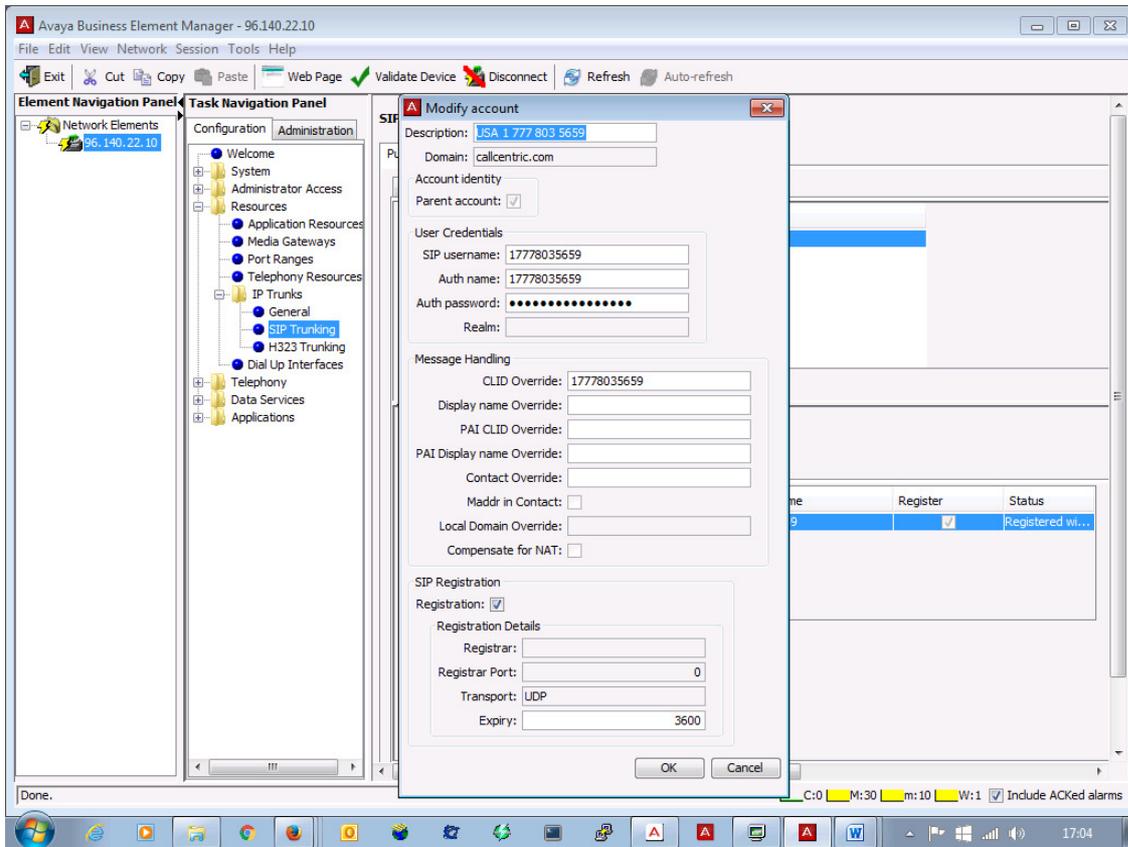
### Making changes to the SIP account.

Now select the new SIP account entry by highlighting the line and click the sub tab “User Accounts”.

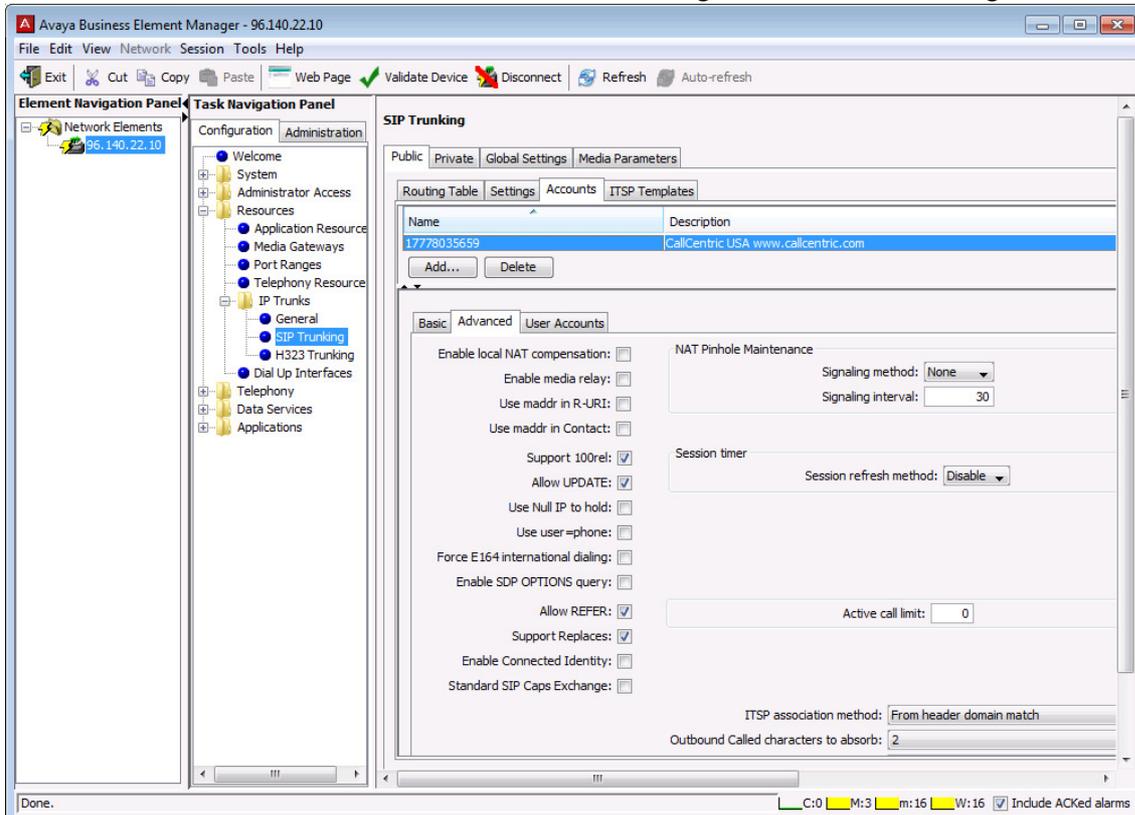


If the details were entered correctly, then you will see that it is now registered with Callcentric.

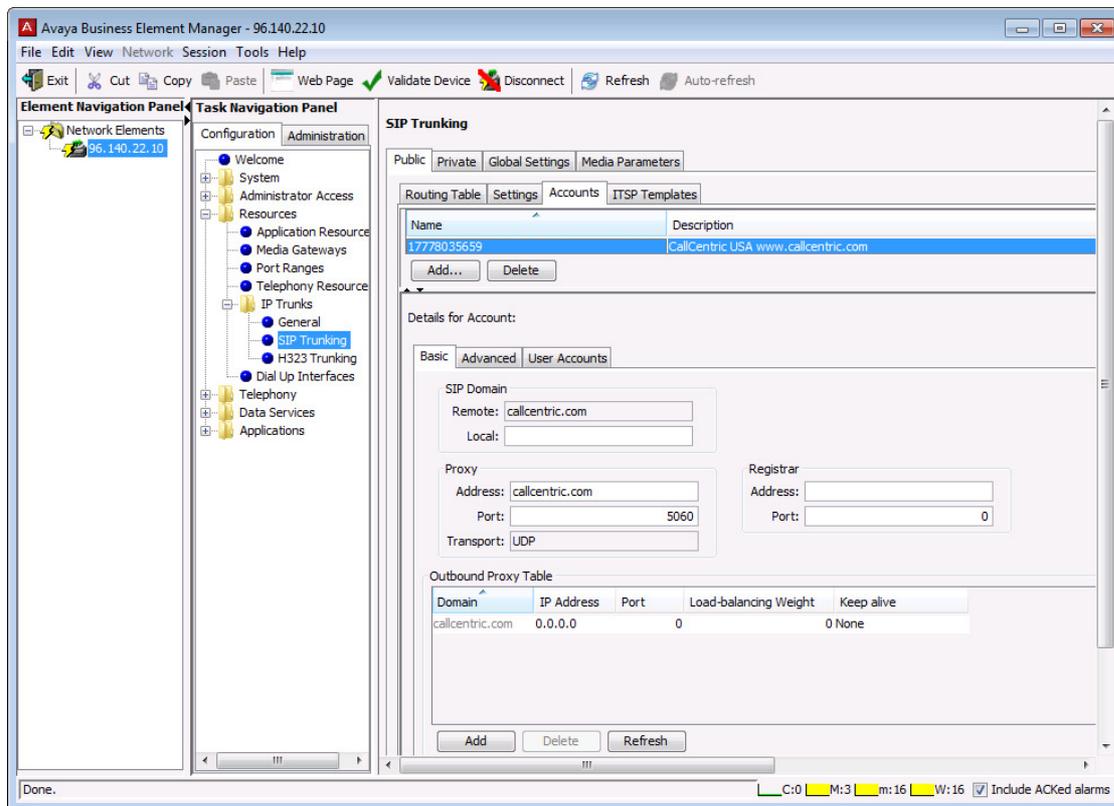
Whilst in this area, click on the “Modify...” tab. The SIP account number is the username, Auth name and CLID Override which is also used in the target line.



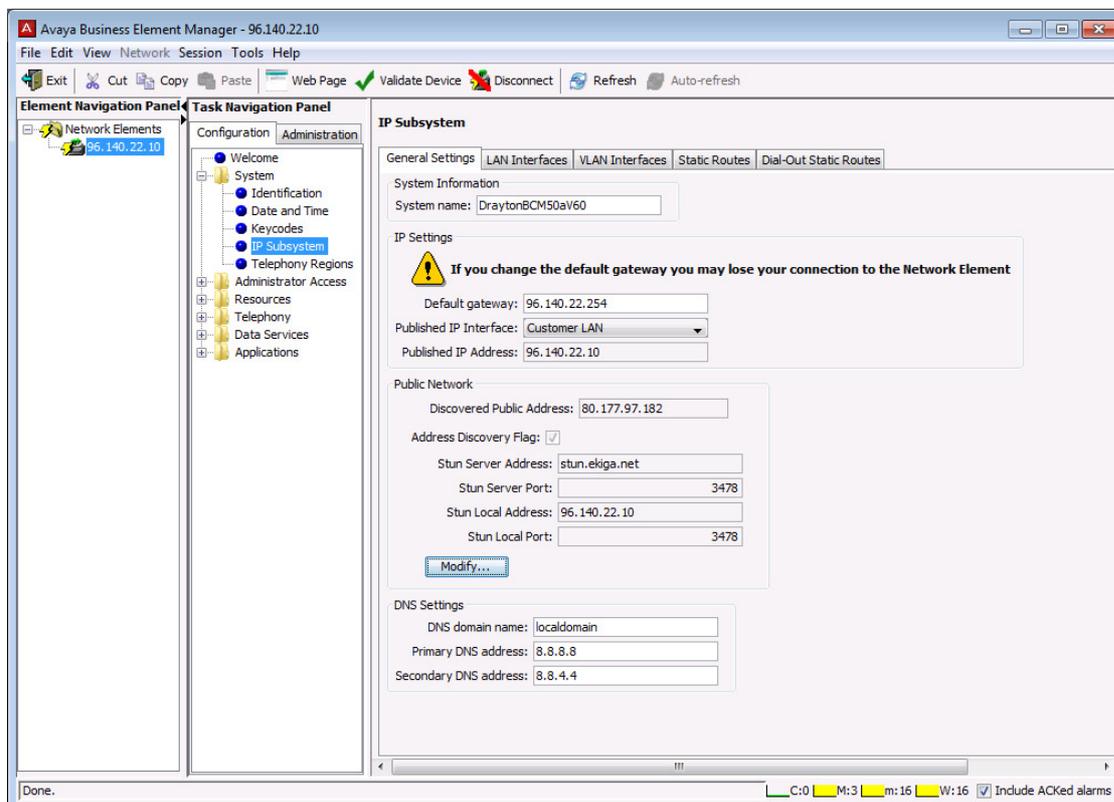
Now click left, on the “Advanced” tab to look at the settings where some areas might need changing.



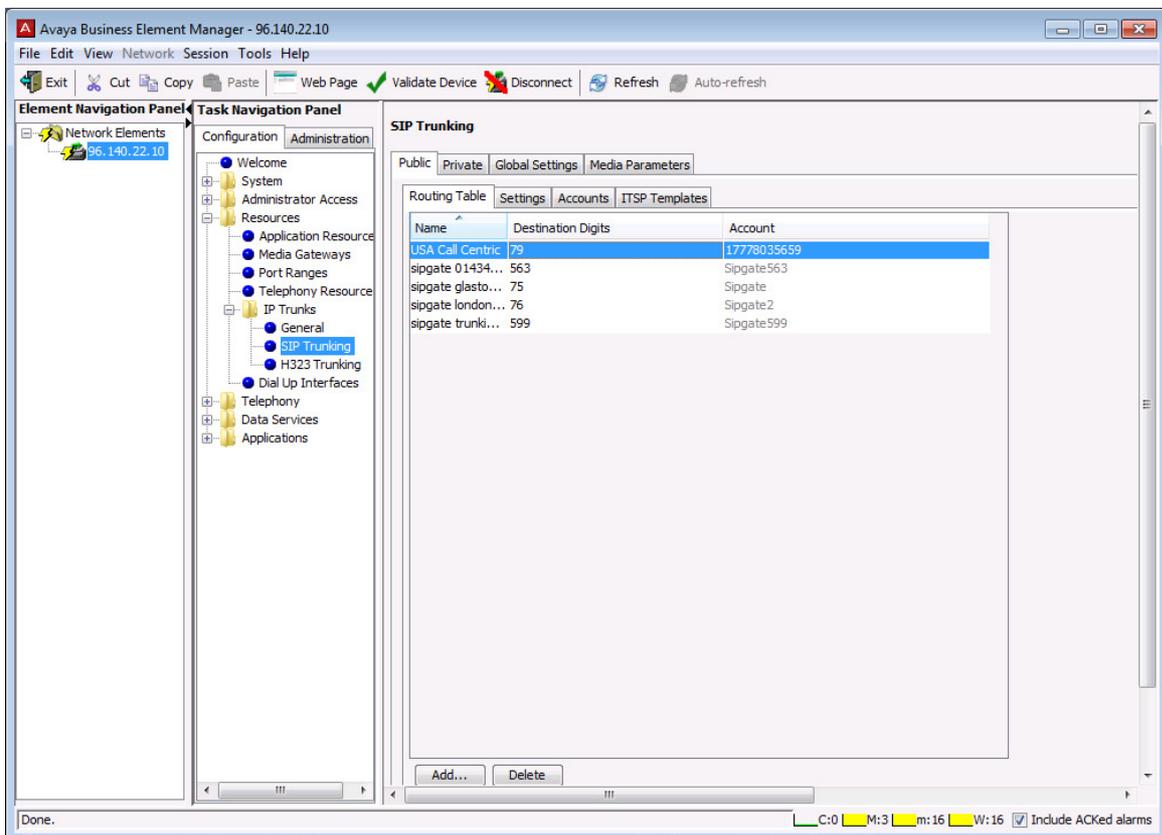
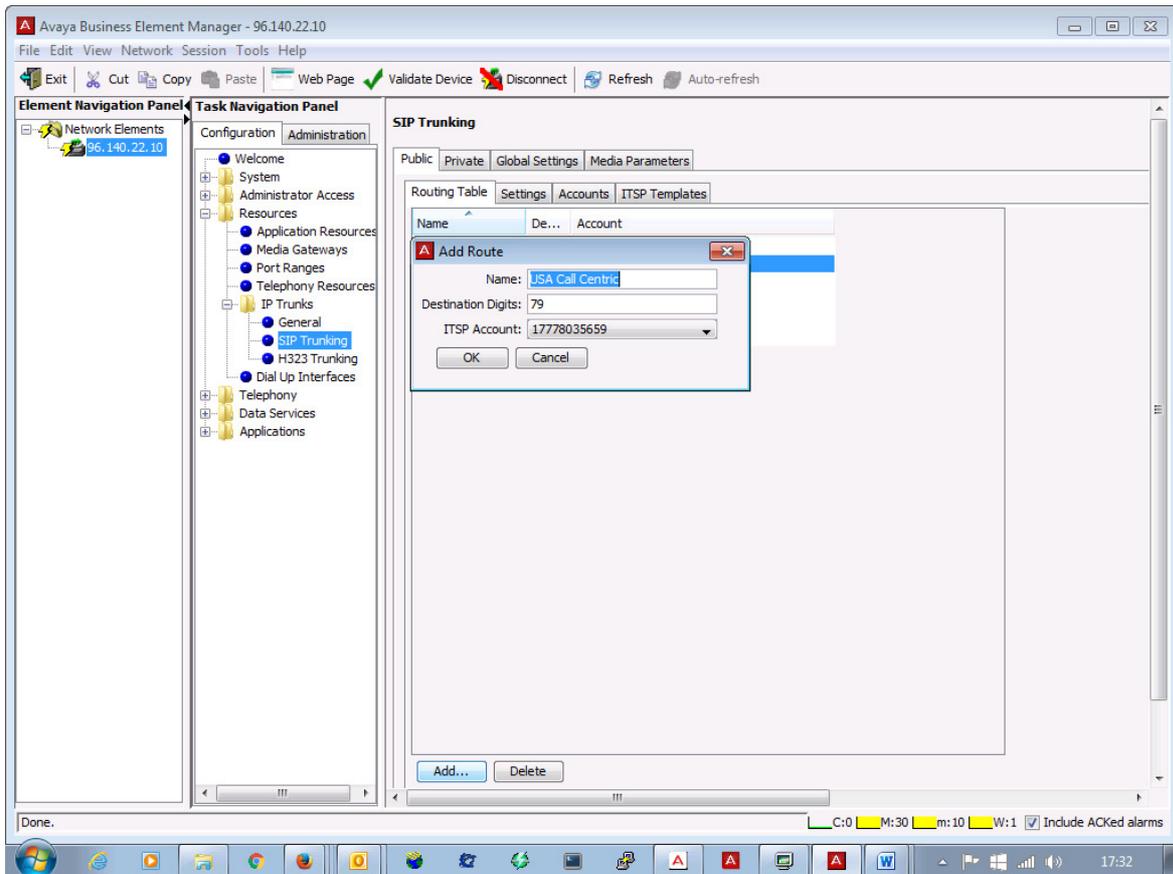
Note above that you need to set the “Outbound Called characters to absorb” (2) = 2 digs. What this means is that we will be using a 2 digit access code that is not sending out these digits to line. Click left, for the “Basic” tab to look at the settings where some changes might need making.



Network settings which work fine for me via a Stun Server. You might not actually need to use it!.



Navigate to “Configuration”, “Resources”, “IP Trunks”, “SIP Trunking”, “Public” and “Routing Table”. Click on the tab option to “Add”. This is where you add in a route to dial for the SIP trunk.



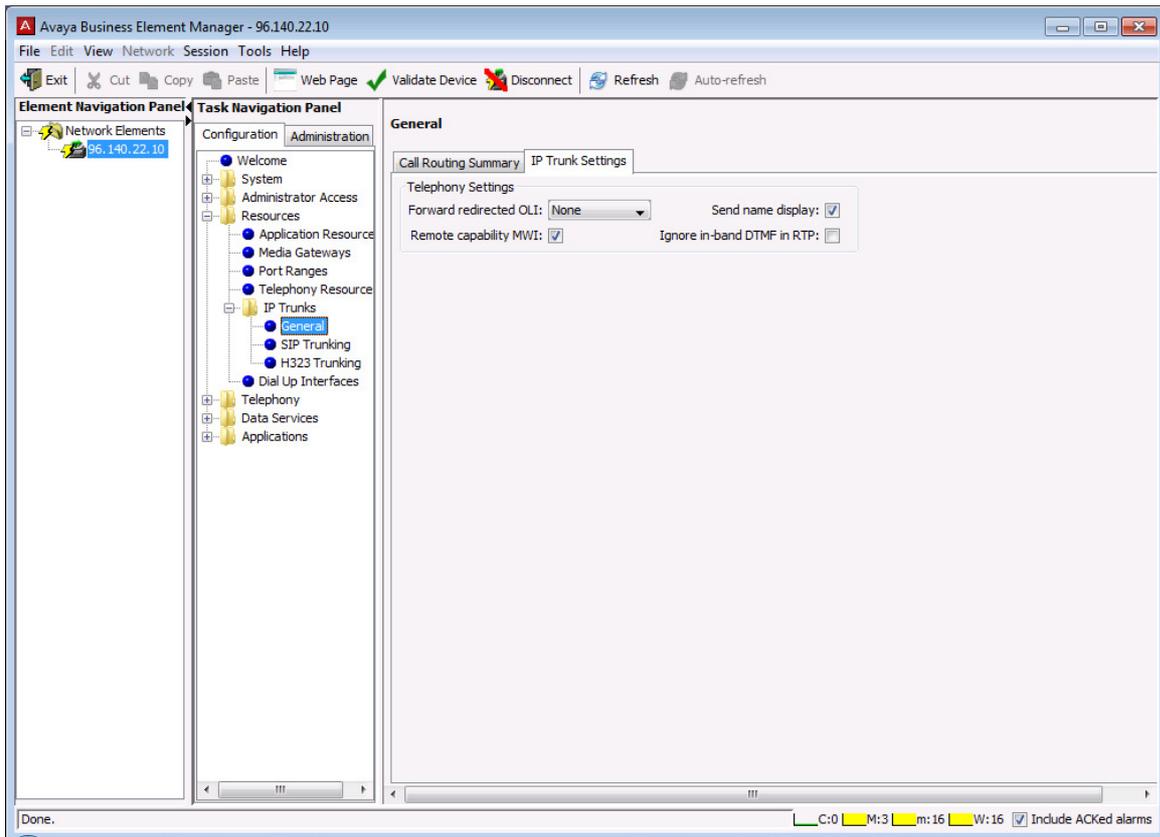
## Global Settings

The screenshot shows the Avaya Business Element Manager interface. The left pane displays a tree view with 'SIP Trunking' selected under 'IP Trunks'. The main area shows the 'SIP Trunking' configuration page with tabs for 'Public', 'Private', 'Global Settings', and 'Media Parameters'. The 'Global Settings' tab is active, showing fields for 'Local Domain', 'Call signaling port' (5060), 'RTP Keepalives' (Scope: None), 'Fallback to circuit-switched' (Disabled), and 'RFC2833' (Dynamic Payload: 101). The status at the bottom indicates 'Gateway is running'.

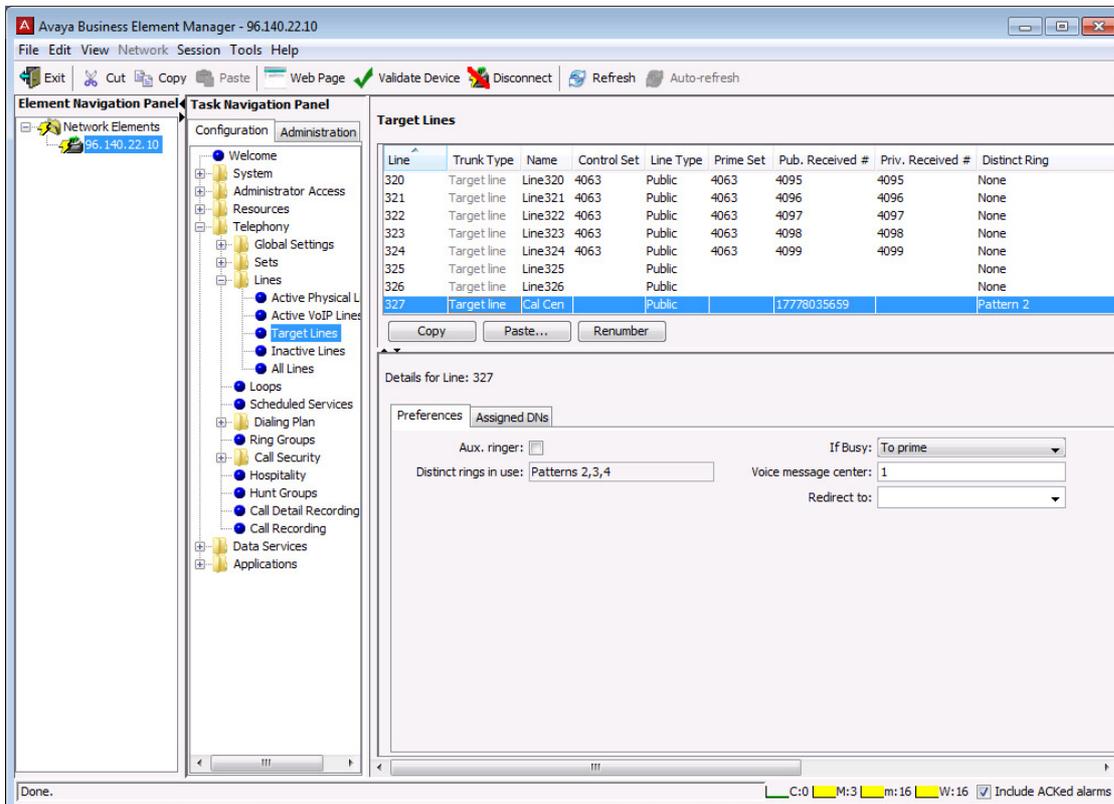
**Media Parameters** Some of these might need to be amended below.

The screenshot shows the Avaya Business Element Manager interface with the 'Media Parameters' tab selected for 'SIP Trunking'. The 'Preferred Codecs' section features two lists: an empty 'Available list' and a 'Selected list' containing 'G.729', 'G.711-uLaw', and 'G.711-aLaw'. 'Add' and 'Del' buttons are positioned between the lists, and 'Up' and 'Down' buttons are on the right of the 'Selected list'. To the right, the 'Settings' section includes checkboxes for 'Enable Voice Activity Detection' and 'Provide in-band ringback', and dropdown menus for 'Jitter buffer' (Auto), 'G.729 payload size (ms)' (20), 'G.711 payload size (ms)' (20), and 'Fax transport' (T.38). The 'Force G.711 for 3.1k audio' checkbox is checked.

## IP Trunks and General settings



You now need to set up the target line and I used target line 327 to add in my Callcentric account in the Pub. Received column as shown below. Users in the USA don't need the initial "1" in there.



Because I used the "2" within the SIP programming to absorb two access code digits when wanting to make an outgoing call, I needed to go into the BCM routing table to configure access code 79 for my Callcentric account. I used route 79 with BlocA and Public (UnKnown).

The screenshot shows the Avaya Business Element Manager interface. The 'Task Navigation Panel' on the left shows the 'Routing' option selected under 'Dialing Plan'. The main window displays the 'Dialing Plan - Routing' configuration page with the 'Routes' tab active. The following table represents the data shown in the 'Routes' table:

Route	External Number	Use Pool	DN Type	Service Type	Service II
000		A	N/A	N/A	N/A
001		A	Public (Unknown)	N/A	N/A
002	141	A	Public (Unknown)	N/A	N/A
011	79	BlocA	Public (Unknown)	N/A	N/A
073		N	Public (Unknown)	N/A	N/A
074		O	Public (Unknown)	N/A	N/A
075		BlocA	Public (Unknown)	N/A	N/A
076		BlocA	Public (Unknown)	N/A	N/A
077		BlocA	Public (Unknown)	N/A	N/A
078		BlocA	Public (Unknown)	N/A	N/A
079		BlocA	Public (Unknown)	N/A	N/A
099		C	Public (Unknown)	N/A	N/A
507		BlocA	Public (Unknown)	N/A	N/A
563		BlocA	Public (Unknown)	N/A	N/A
599		BlocA	Public (Unknown)	N/A	N/A

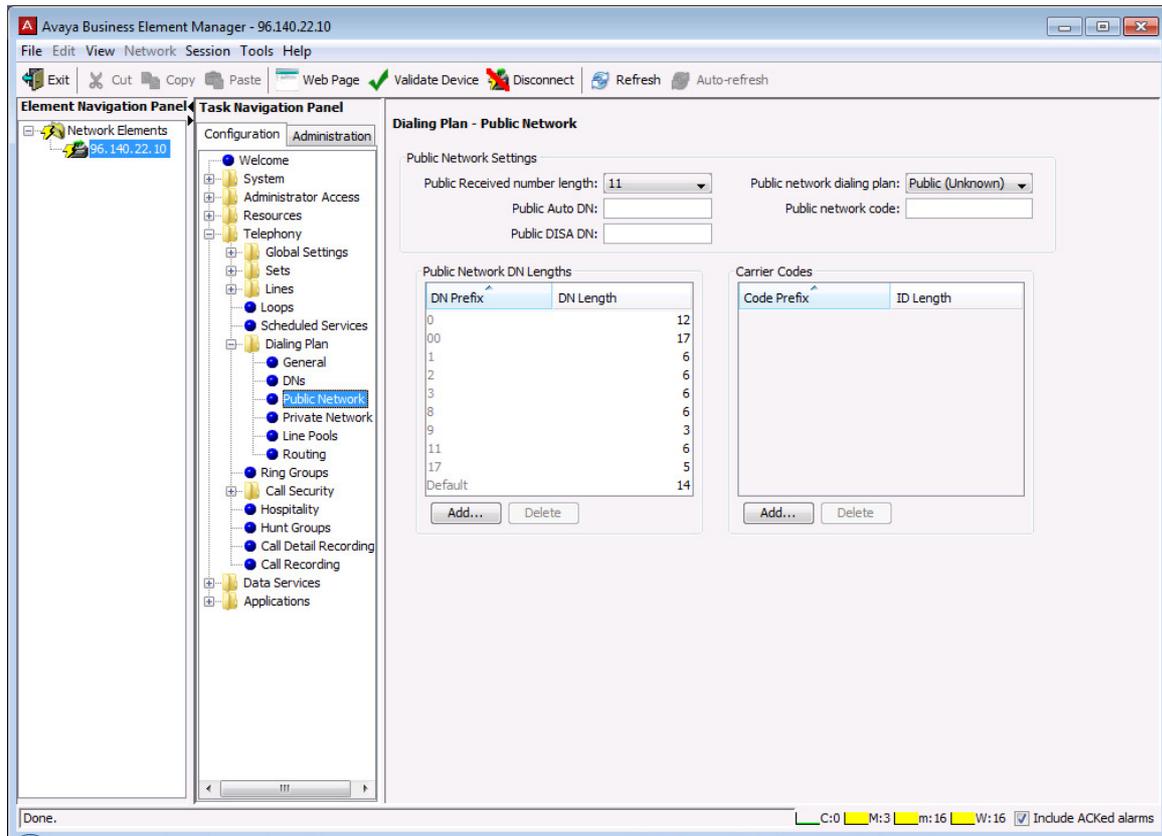
The screenshot shows the Avaya Business Element Manager interface. The 'Task Navigation Panel' on the left shows the 'Routing' option selected under 'Dialing Plan'. The main window displays the 'Dialing Plan - Routing' configuration page with the 'Destination Codes' tab active. The following table represents the data shown in the 'Destination Codes' table:

Destination Code	Normal Route	Absorbed Length	Wild Card: 0
74	074	All	<input type="checkbox"/>
75	075	0	<input type="checkbox"/>
76	076	0	<input type="checkbox"/>
77	000	0	<input type="checkbox"/>
78	078	0	<input type="checkbox"/>
79	079	0	<input type="checkbox"/>
81	002	0	<input type="checkbox"/>
82	000	0	<input type="checkbox"/>
83	000	0	<input type="checkbox"/>
84	000	0	<input type="checkbox"/>
loc	000	0	<input type="checkbox"/>

Below the 'Destination Codes' table, the 'Alternate Routes for Destination Code: 79' section is visible. The following table represents the data shown in the 'Alternate Routes' table:

Schedule	First Route	Absorbed Length	Second Route	Absorbed Length
Evening		All		All
Lunch		All		All
Night		All		All
Sched 4		All		All
Sched 5		All		All
Sched 6		All		All

Finally, the Dialing Public Network plan was changed when I had problems dialling out by increasing the default digits to 13 or 14. Also the Public Received number length was increased to 11.



The information listed in this guide should provide the basic information need to configure public SIP trunk accounts.

### Trouble shooting and additional non-essential information

When setting up the BCM for SIP trunking, there are other areas that need to be checked. Some of these aren't in the Avaya documentation.

### Router configuration

Initially, I had UDP port 5060 configured to port forward onto the BCM system. This has been removed and only the following are used.

```

BCM Unistim Signalling ALL    UDP    7000 - 7001
UFTP Server             ALL    UDP    7002
BCM IP Phone Signalling ALL    UDP    20000 - 20249
  
```

Even these ports aren't really required for SIP trunks, but will be needed if you plan to connect up IP phones across the internet using the BCM NAT Traversal license.

### Application DN's

There is in my opinion a major flaw in security as my BCM system was recently hacked by dial through fraud a few months after installing a couple of SIP trunks onto it. I carried out an extensive

search to try and find out when it started and how it was done. Although I had incoming trunks go to voicemail with a menu option, I didn't think that was the cause which is usually the case!.

What I discovered was that I had **UDP port 5060** set to my BCM, which I think gave the SIP hackers the ability to probe my system for any weaknesses and I had spotted some odd behaviour some time before it happened. It was quite trivial on the BCM Monitor tool that I saw incoming calls but none of my phones rang and no indication of who was calling me. This kept repeating around every 15 to 20 minutes.

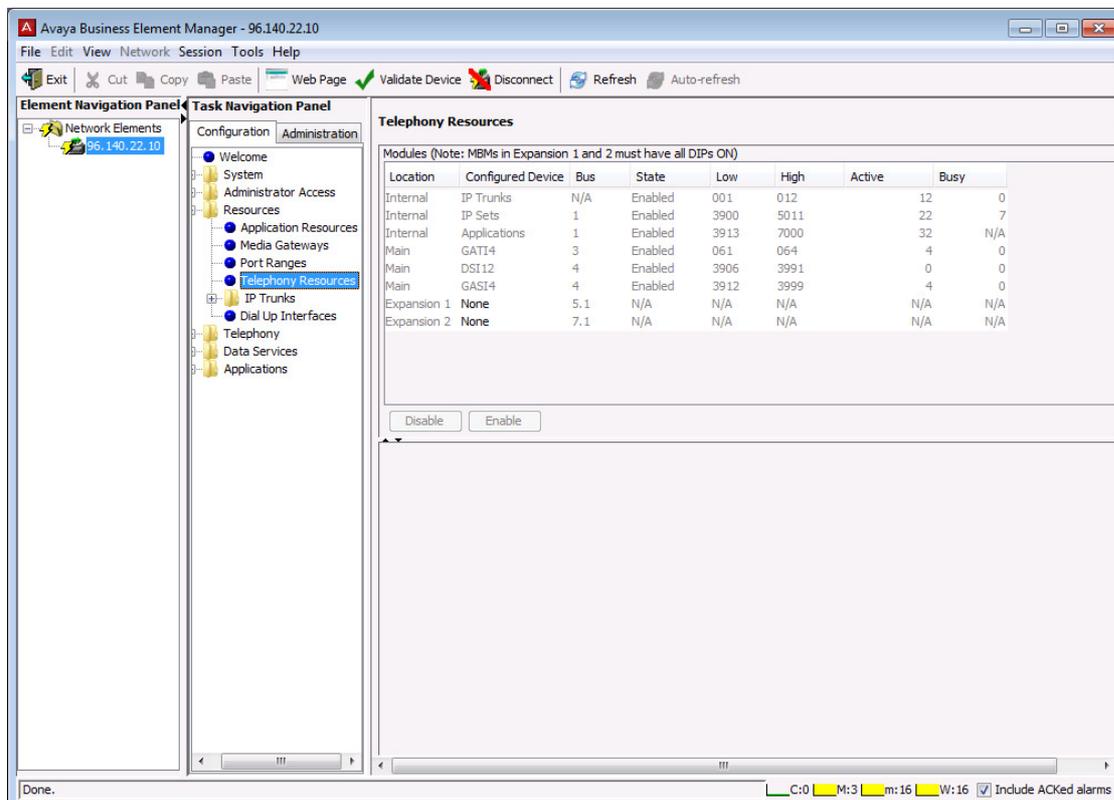
I had one analogue trunk that was assigned in programming to **Pool A**. On just about all BCM systems, it looks as if Pool A was assigned to all Application DN's along with a **separate target line**.

The hackers broke out via the analogue line. The weird thing was that the CDR records didn't show any incoming traffic but only outgoing traffic via an Application DN, hence the reason for investigating this area.

Since the outbreak of calls, I've tightened up everything which includes removal of any line pool and target lines from the Application DN's and made international calls restricted and changed their access to public or a different line pool. As stated above, the **UDP port 5060** was removed from my router and only the minimum remained. I also checked that **port 443** was removed from my router.

My BCM monitor tool now only shows active or previous calls which indicates that my problem has now been blocked to the hackers.

## Telephony Resources



The screenshot shows the Avaya Business Element Manager interface. The main window displays the 'Telephony Resources' configuration page. The left sidebar shows a tree view of network elements, with 'Telephony Resources' selected. The main content area shows a table of modules with columns for Location, Configured Device, Bus, State, Low, High, Active, and Busy. Below the table are 'Disable' and 'Enable' buttons.

Location	Configured Device	Bus	State	Low	High	Active	Busy
Internal	IP Trunks	N/A	Enabled	001	012	12	0
Internal	IP Sets	1	Enabled	3900	5011	22	7
Internal	Applications	1	Enabled	3913	7000	32	N/A
Main	GATI4	3	Enabled	061	064	4	0
Main	DSI12	4	Enabled	3906	3991	0	0
Main	GASI4	4	Enabled	3912	3999	4	0
Expansion 1	None	5.1	N/A	N/A	N/A	N/A	N/A
Expansion 2	None	7.1	N/A	N/A	N/A	N/A	N/A

## BCM Port Ranges

Here is a screen shot showing the ranges used on my BCM system.

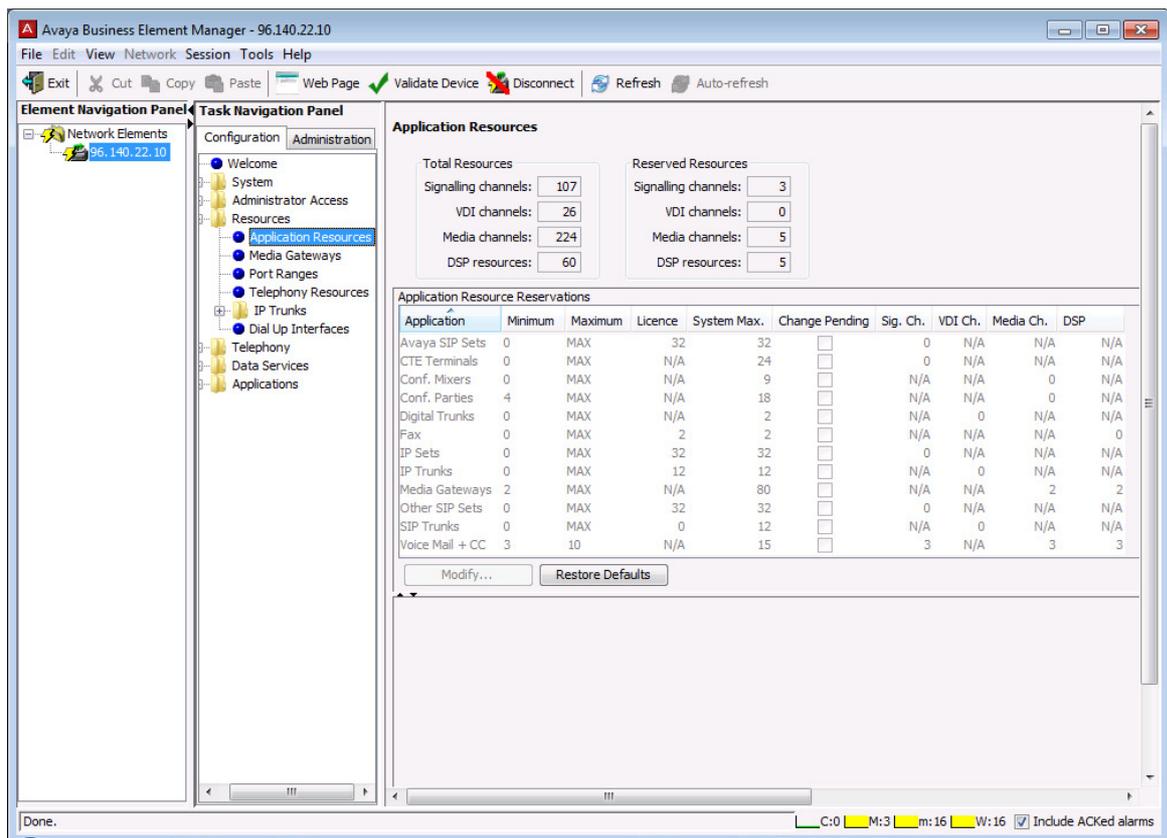
The screenshot displays the Avaya Business Element Manager interface. The main window title is "Avaya Business Element Manager - 96.140.22.10". The interface includes a menu bar (File, Edit, View, Network, Session, Tools, Help) and a toolbar with icons for Exit, Cut, Copy, Paste, Web Page, Validate Device, Disconnect, Refresh, and Auto-refresh.

The left sidebar contains the "Element Navigation Panel" with a tree view showing "Network Elements" and "96.140.22.10". The "Task Navigation Panel" is active, showing "Configuration" and "Administration" tabs. The "Port Ranges" section is selected, displaying three tables:

RTP over UDP		UDP		Signalling	
Begin	End	Begin	End	Begin	End
28000	28249	5060	5060	0	1023
30000	30099	7002	7002	1718	1719
		20000	20249	2216	2227
		51000	51010	5000	5000
				7000	7000
				60000	60240

Below each table are "Add..." and "Delete" buttons. The status bar at the bottom shows "Done." and "C:0 M:3 m:16 W:16 Include ACKed alarms".

## Application Resources



When dialling out, I use an access code of 79 that is deleted and then any North American number (USA or Canada etc) such as 1777 1234567 is dialled out to line. Incoming calls will ring into the target line against any phones that have been given incoming access.

Once registered on the Callcentric website, payment can be made with PayPal as well as normal electronic methods and the lowest amount starts at \$5. It's worth remembering that Callcentric to Callcentric calls are free!.