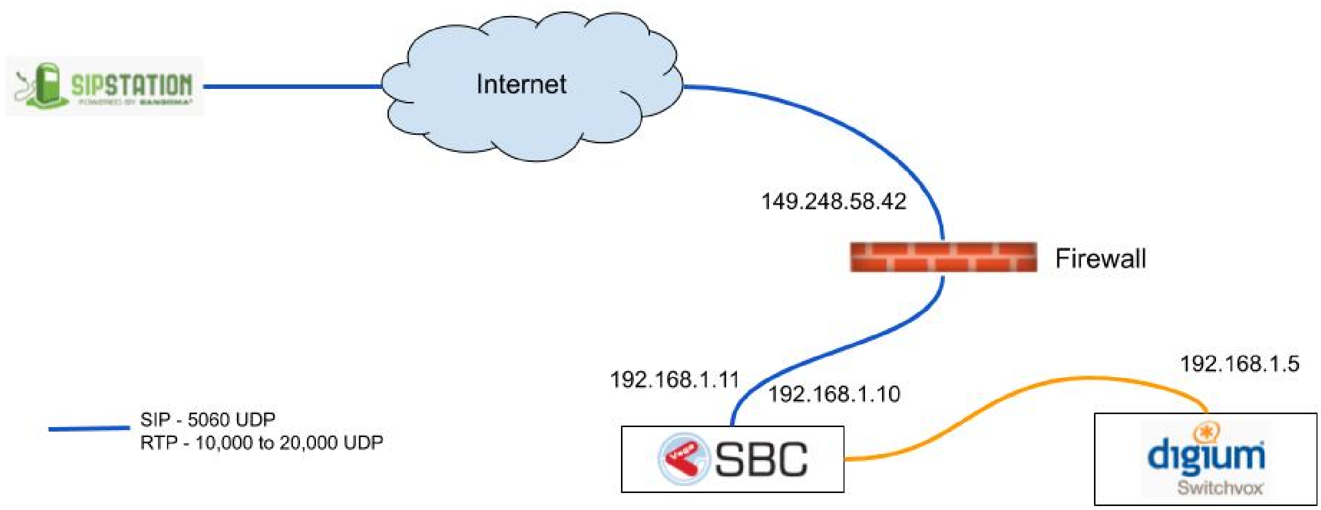
Switchvox - SIP Trunking

Switchvox Private IP: 192.168.1.5

SBC Public IP: 149.248.58.42
SBC Private IP #1: 192.168.1.11 (Connection to ITSP - Public IP Ports Forwarded to this IP)
SBC Private IP #2: 192.168.1.10 (Connection to Switchvox)



Router Configuration

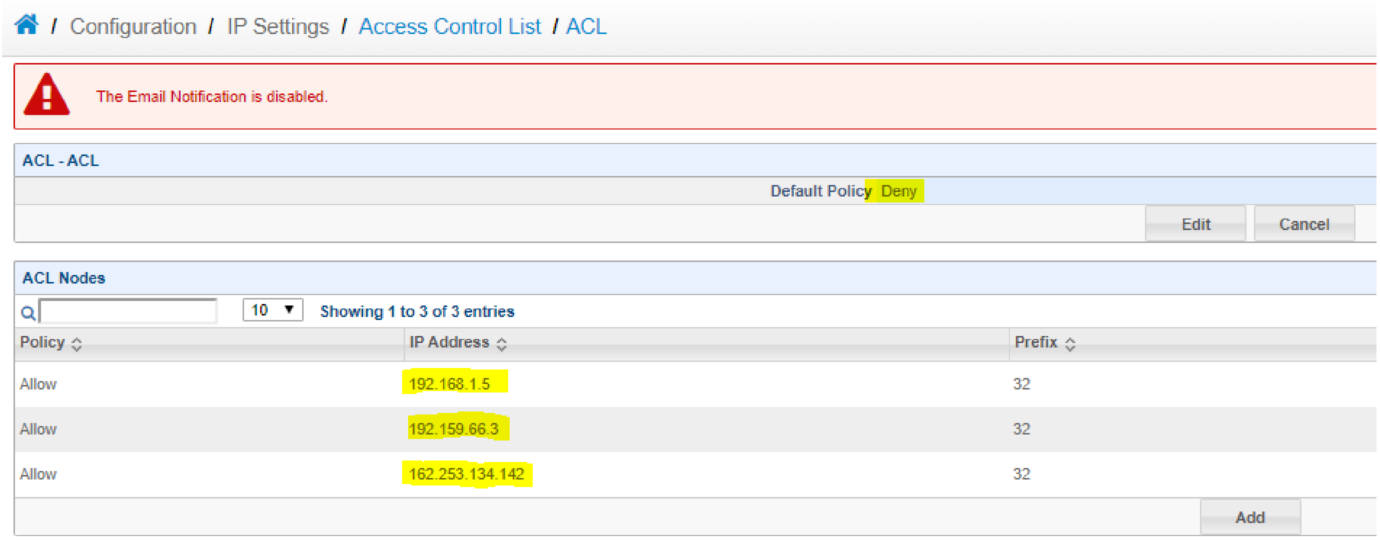
Ensure the following ports are open or forwarded to the public IP of the SBC.

SBC Public IP Ports

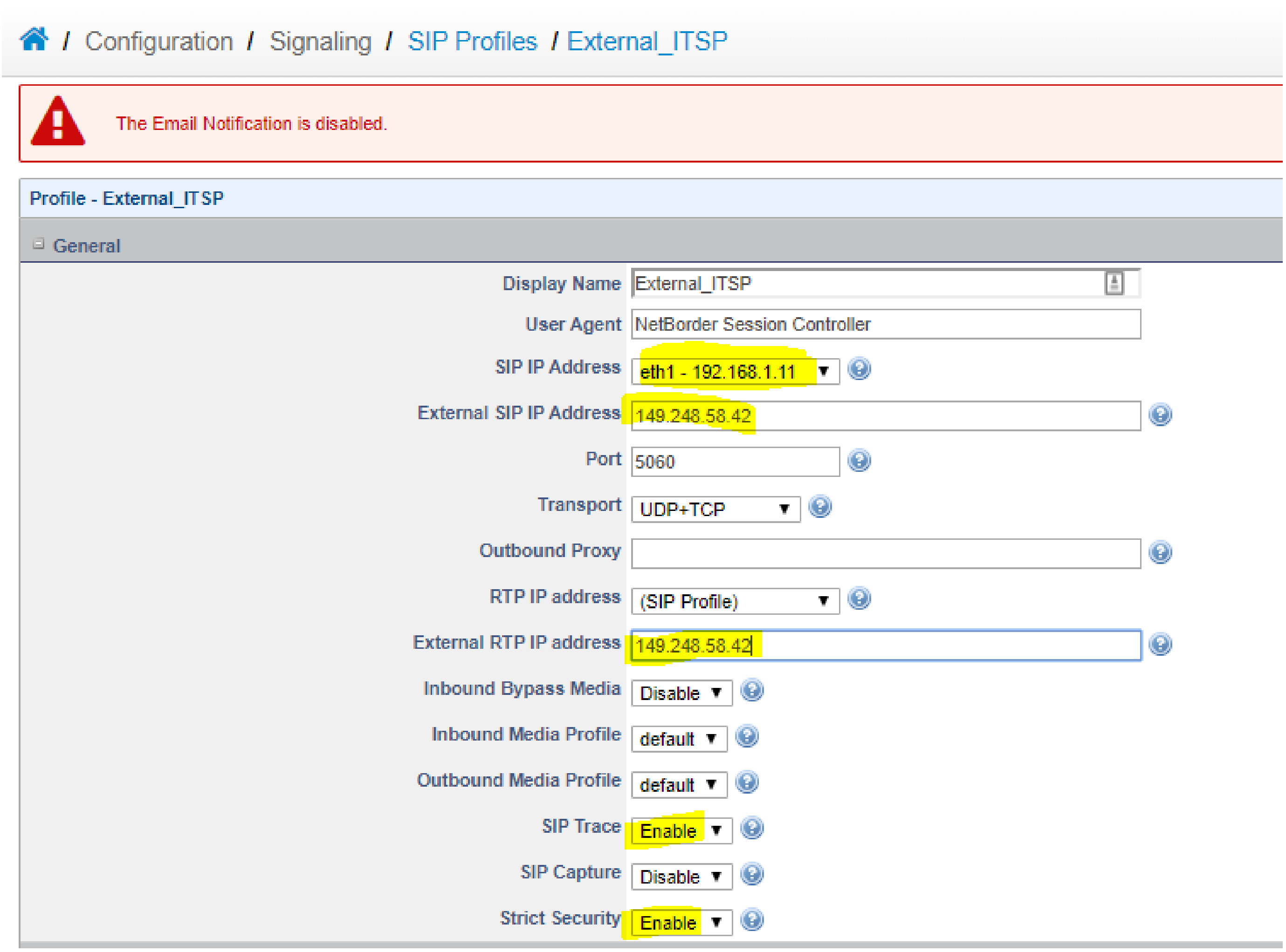
- **5060 UDP**
- **10,000 to 20,000 UDP**

SBC Configuration

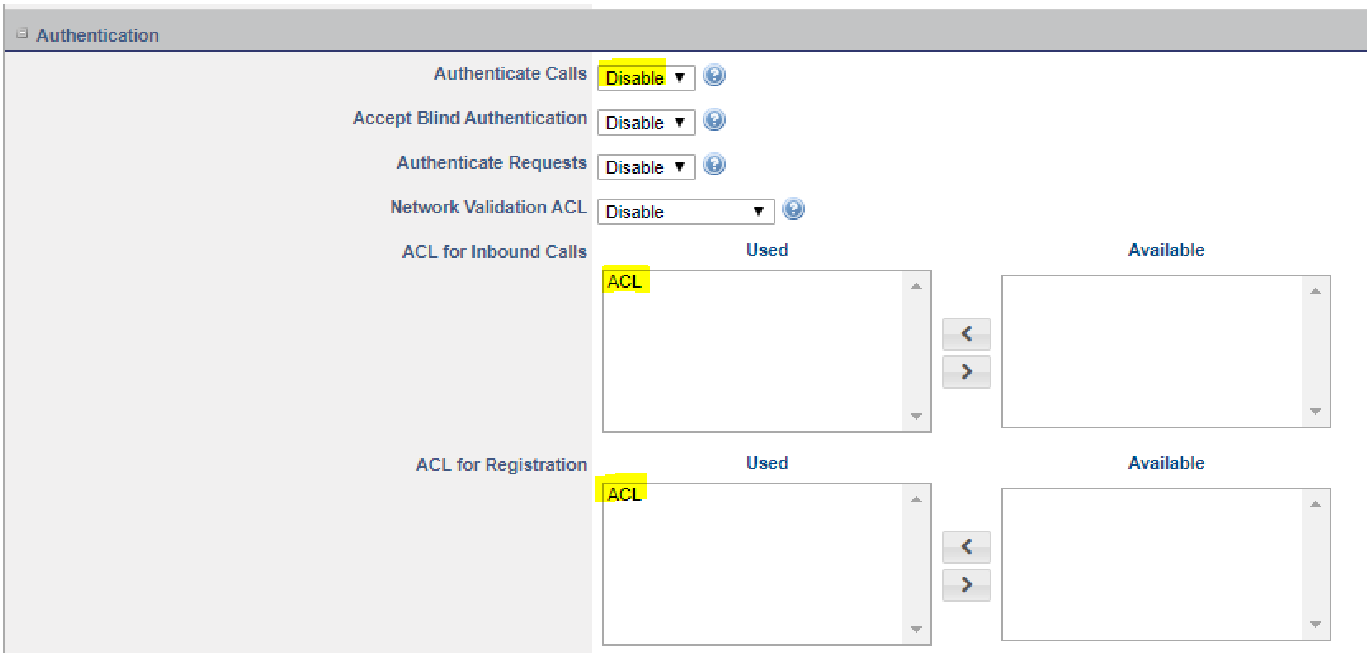
1) Go to **Configuration IP Settings Access Control List** and add a new list called ACL. Ensure the default policy is Deny, and then add both the IP of Switcvox, and the IP(s) of your ITSP. Ensure the prefix is /32 to only allow the single IP.
Note: In this case the ITSP is Sangoma's SIP Station. The FQDN's are trunk1.freepbx.com and trunk2.freepbx.com. Check with your ITSP if you need the IPs.



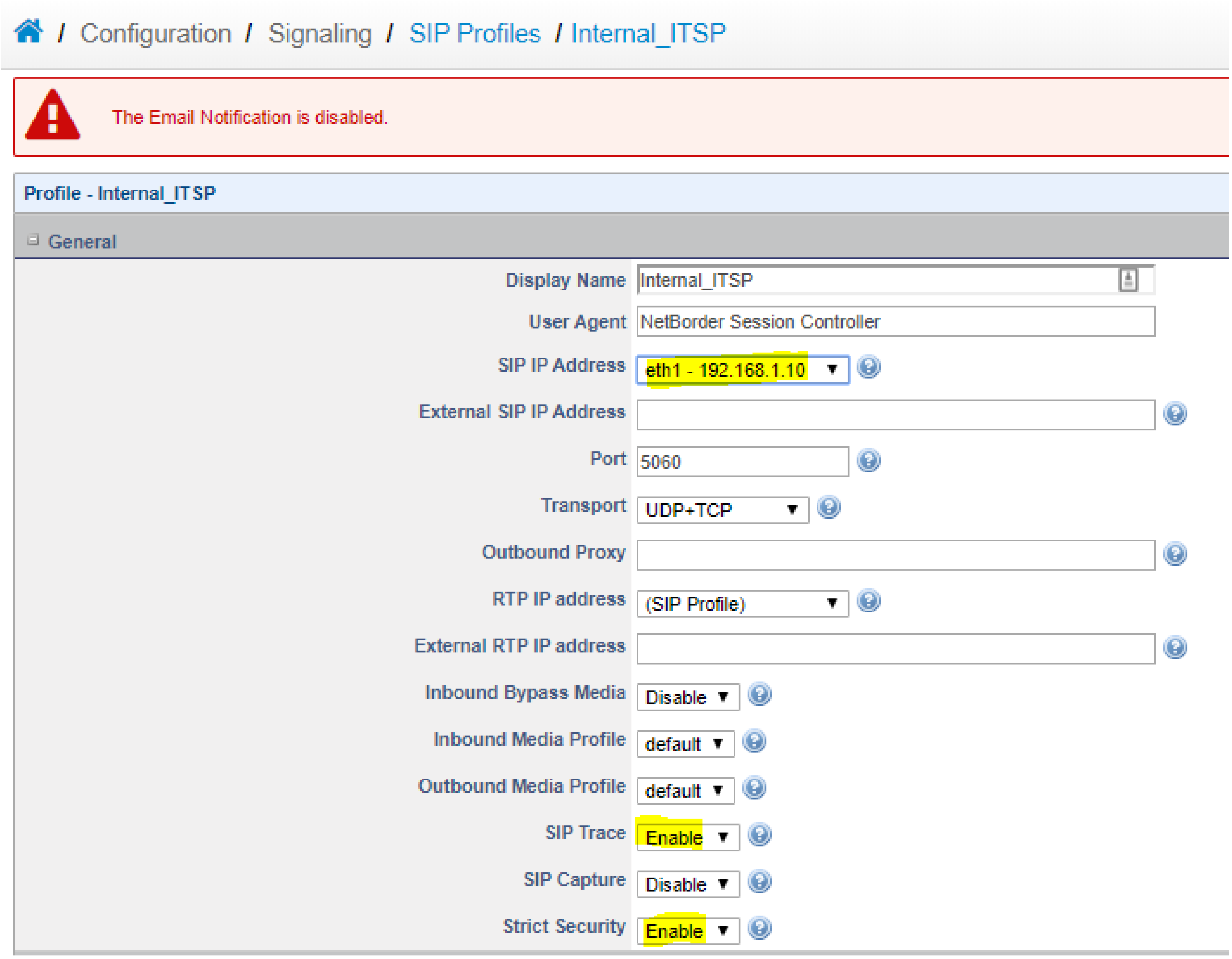
2) Go to **Configuration Signalling SIP Profiles** and add a new SIP profile called External_ITSP. Select the private IP that the public IP ports are forwarded to. In this example 149.248.58.42 is forwarded to 192.168.1.11. Then put the public IP of the SBC in **External SIP IP Address** and **External RTP IP Address** as shown below. Then ensure SIP Trace is enabled, as well as Strict Security as shown below.



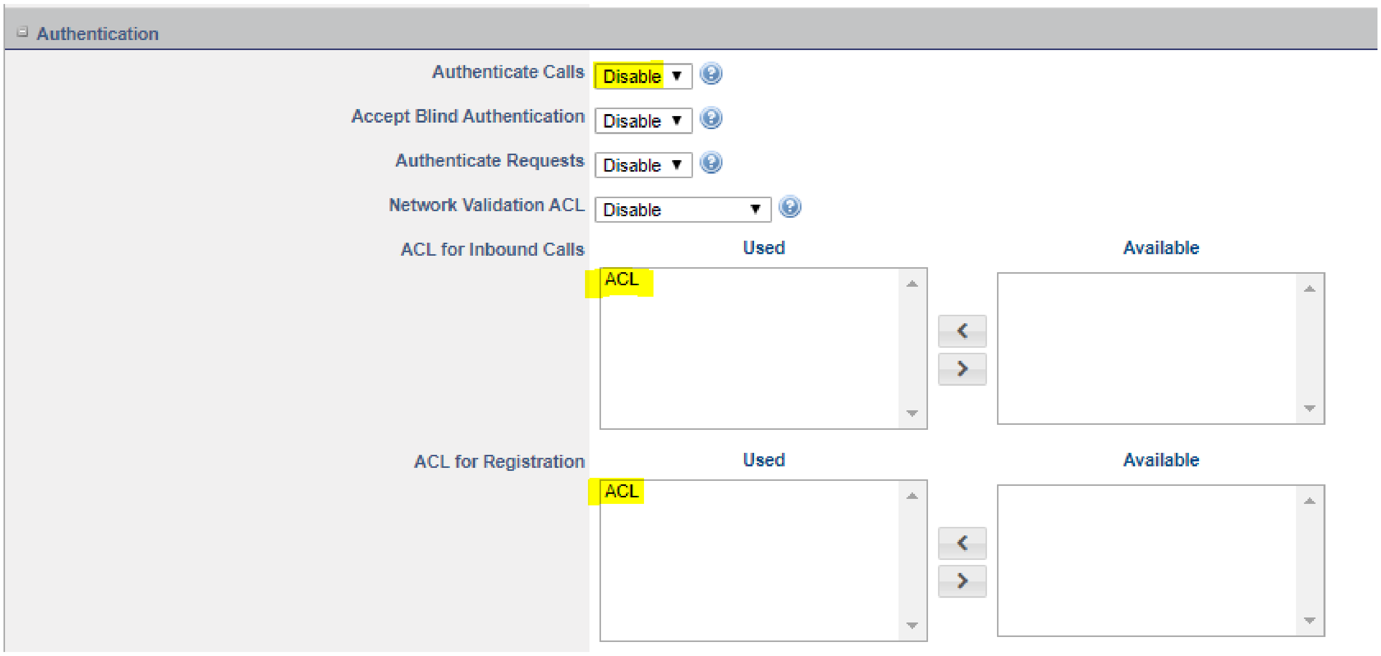
3) In the Authentication section Disable authenticate calls, and add the ACL list created previously to both inbound calls and registrations as shown below.



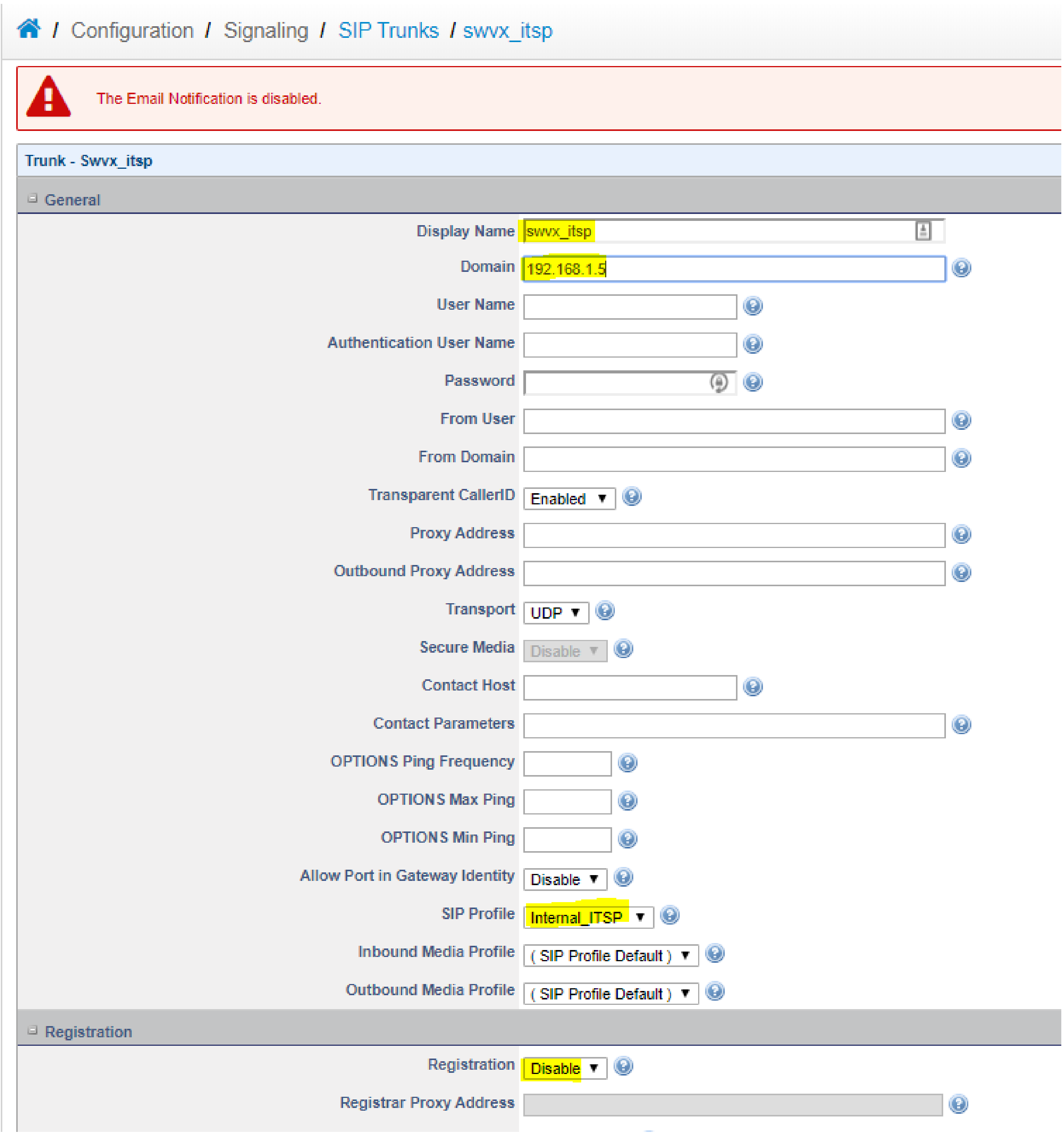
4) Go to **Configuration Signalling SIP Profiles** and add a new SIP profile called Internal_ITSP. Selecting the private IP, enabling SIP trace and enabling Strict Security.



5) In the Authentication section Disable authenticate calls, and add the ACL list created previously to both inbound calls and registrations as shown below.



6) Go to **Configuration Signalling SIP Trunks** and create a new SIP trunk called swvx_itsp. This trunk will point to your Switchvox PBX. Put the IP of the Switchvox in the domain, and ensure the SIP profile is set to Internal_ITSP. As well ensure Registration is Disabled.

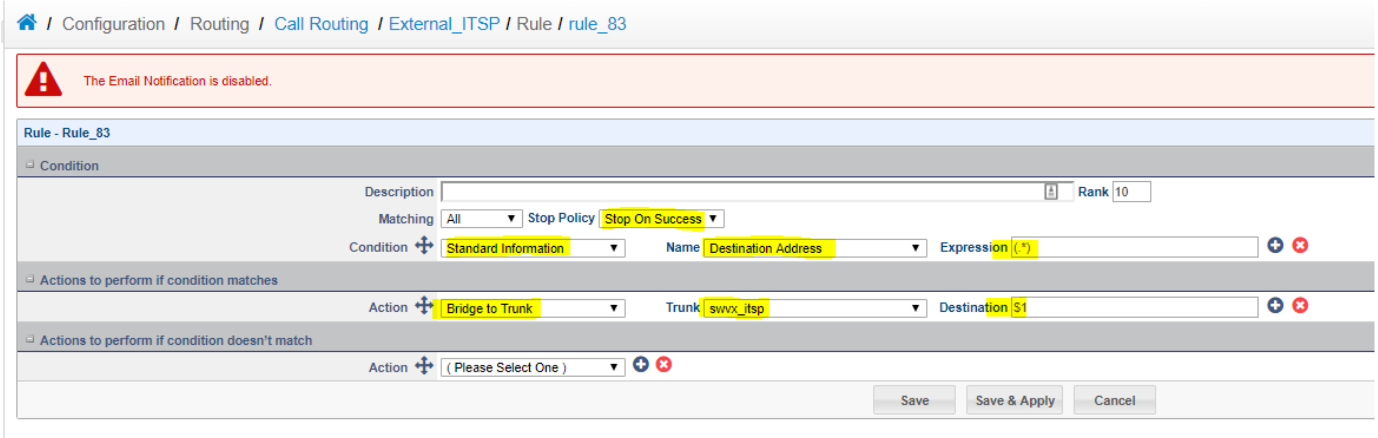


7) Create another SIP trunk called Trunk1. This will go to trunk1.freepbx.com. Enter the username and password. Set the SIP profile to External_ITSP and enable Registration.

8) Create another SIP trunk called Trunk2. This will go to trunk2.freepbx.com. Enter the username and password. Set the SIP profile to External_ITSP and enable Registration.
Note: Some ITSP's may only have 1 SIP Trunk. If this is the case skip this step.

9) Go to **Configuration Routing Call Routing** and create a new routing plan called External_ITSP. Then make a new rule as shown below. Ensure the Stop policy is set as shown below, and the trunk is set to swvx_itsp.

Expression: (.*)
Destination: $1



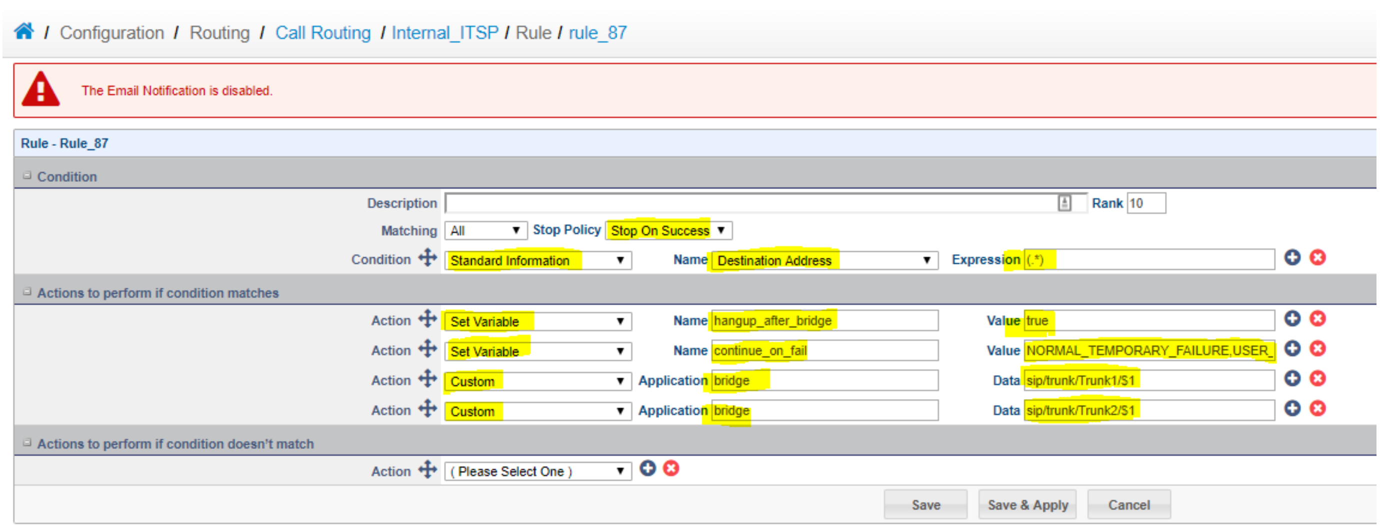
10) Go to **Configuration Routing Call Routing** and create a another new routing plan called Internal_ITSP. If your provider only has a single trunk, then you can use the same rule as in step #9, but select your providers trunk. If you are using SIP Station or any other provider with two trunks, then use the rule below. This will allow fail over to work; where the call will go to trunk1, and if that is down, then it will go to trunk2.

Action 1: hangup_after_bridge
Value 1: true

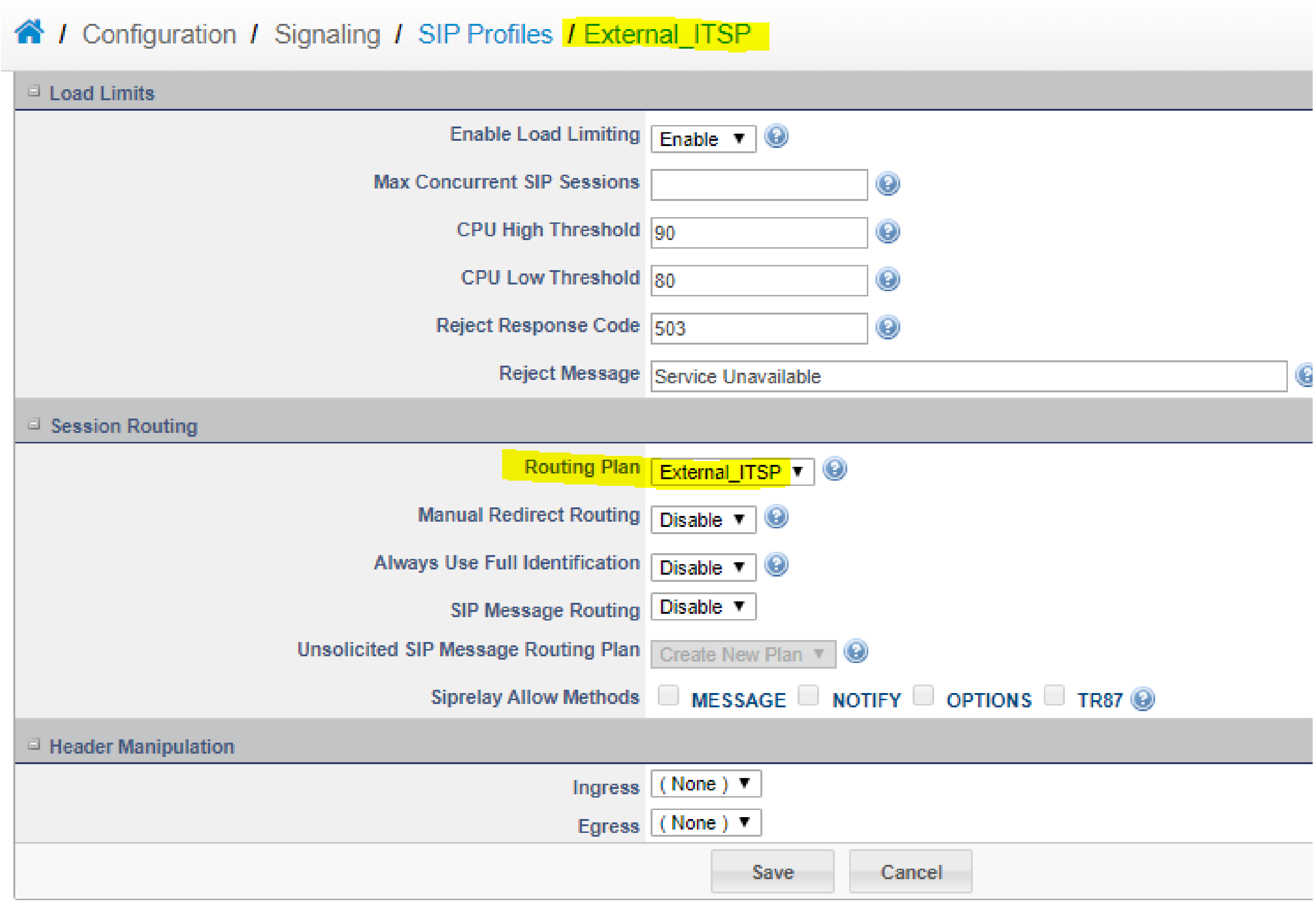
Action 2: continue_on_fail
Value 2: NORMAL_TEMPORARY_FAILURE,USER_BUSY,NO_ANSWER,NO_USER_RESPONSE,NO_ROUTE_DESTINATION, NETWORK_OUT_OF_ORDER,CALL_REJECTED,DESTINATION_OUT_OF_ORDER,NORMAL_CIRCUIT_CONGESTION

Action 3: bridge
Value 3: sip/trunk/Trunk1/$1

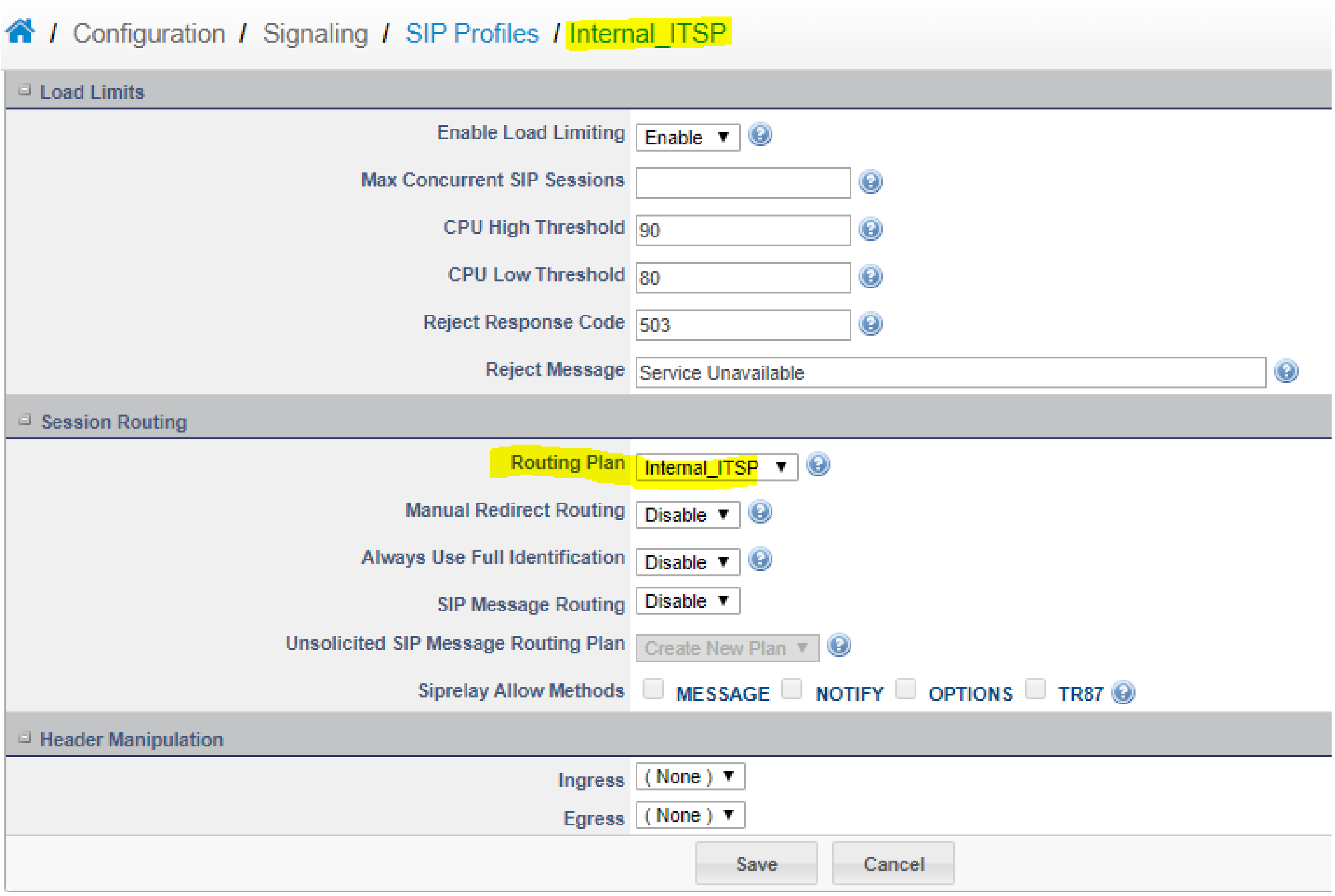
Action 4: bridge
Value 4: sip/trunk/Trunk2/$1



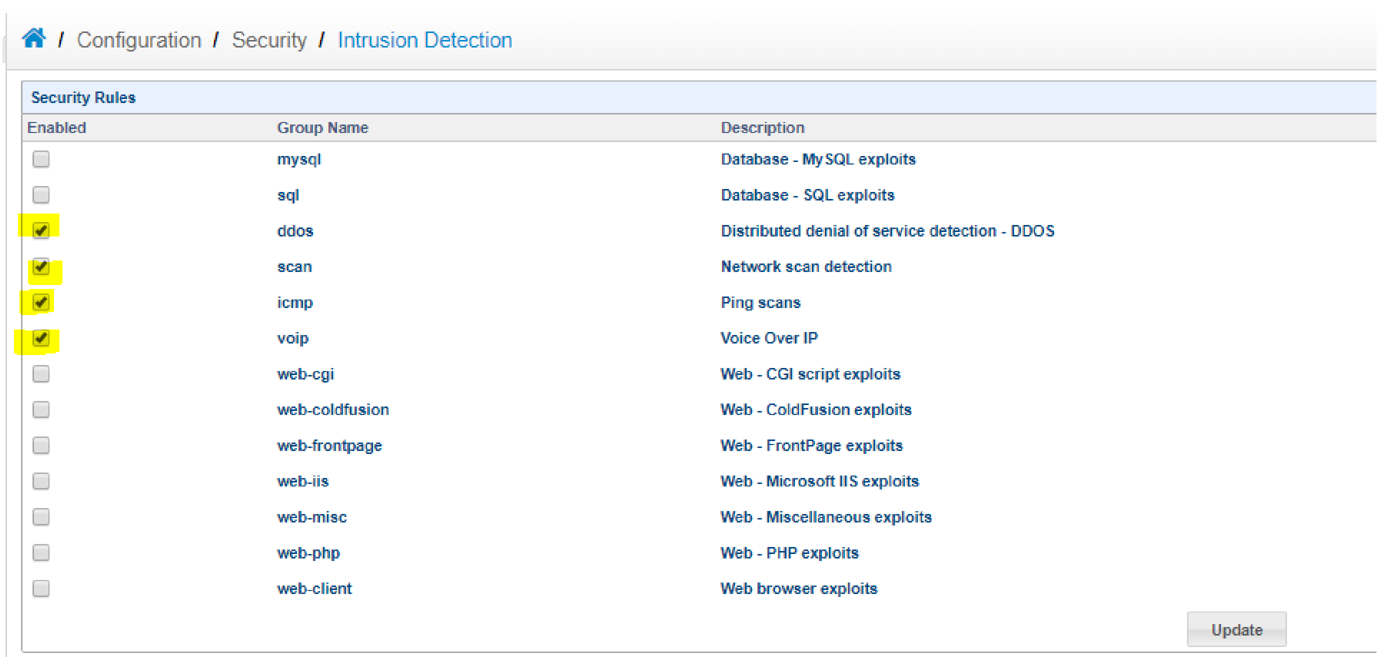
10) Go to **Configuration Signalling SIP Profiles External_ITSP** and modify and then edit the profile. Scroll to the bottom and set the Routing plan to External_ITSP.



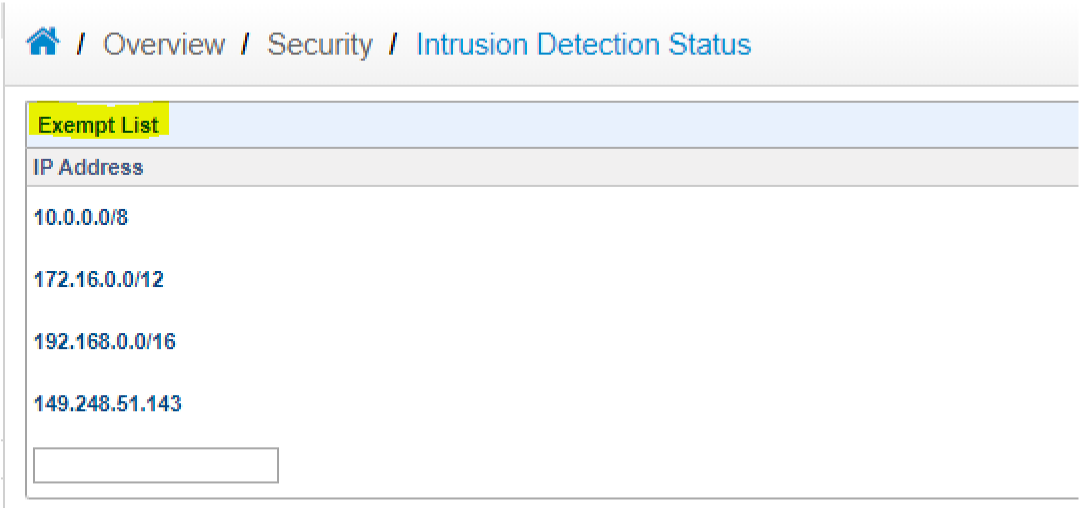
11) Go to **Configuration Signalling SIP Profiles Internal_ITSP** and modify and then edit the profile. Scroll to the bottom and set the Routing plan to Internal_ITSP.



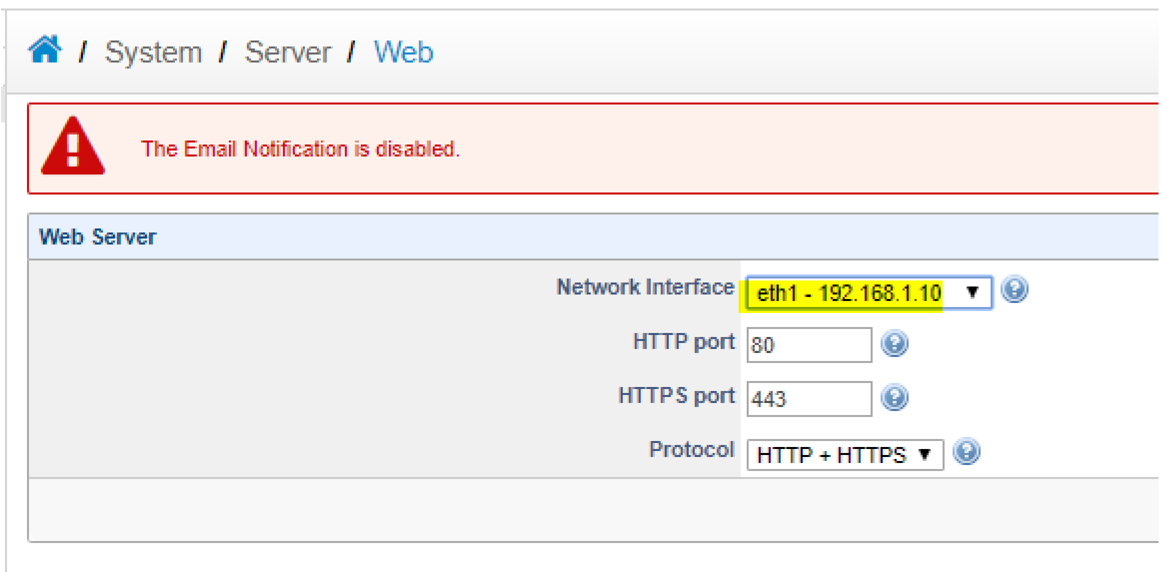
12) To configure the Intrusion Detection or IDS simply go to **Configuration Security Intrusion Detection** and select the following 4 rule groups as shown below. We will be isolating the webUI from the internet, so there is no need for the other rules. Once done click the update button at the bottom to save changes.



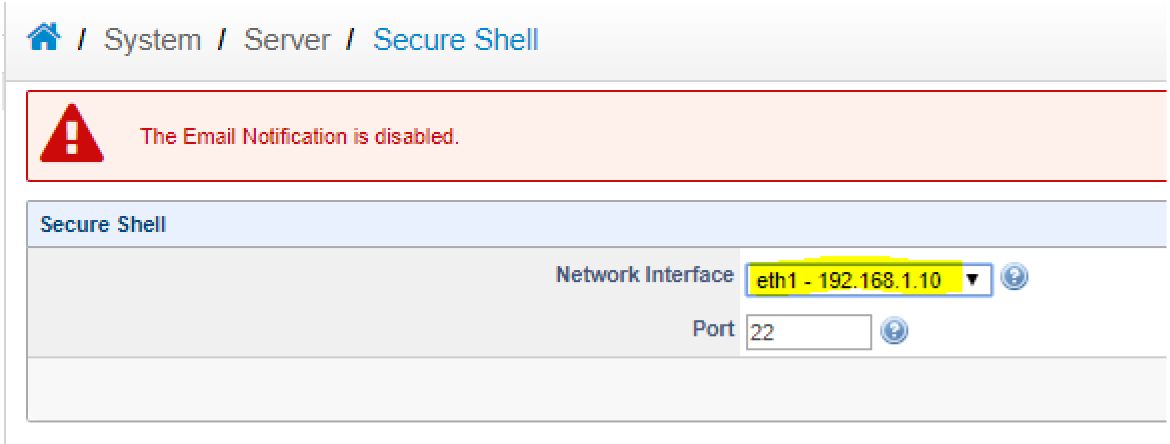
13) Go to **Overview Security Intrusion Detection Status** and then ensure the Switchvox IP is in the list. In this case the Switchvox is 192.168.1.5, which falls in the 192.168.0.0/16 range, which is part of the default config. In most cases this step can be skipped, as all private addresses are included here by default.



14) Last step to security is configuring both the webUI and SSH to only listen on the internal network. To do this go to **System Server Web** and set the Network Interface to the private network, then save changes.

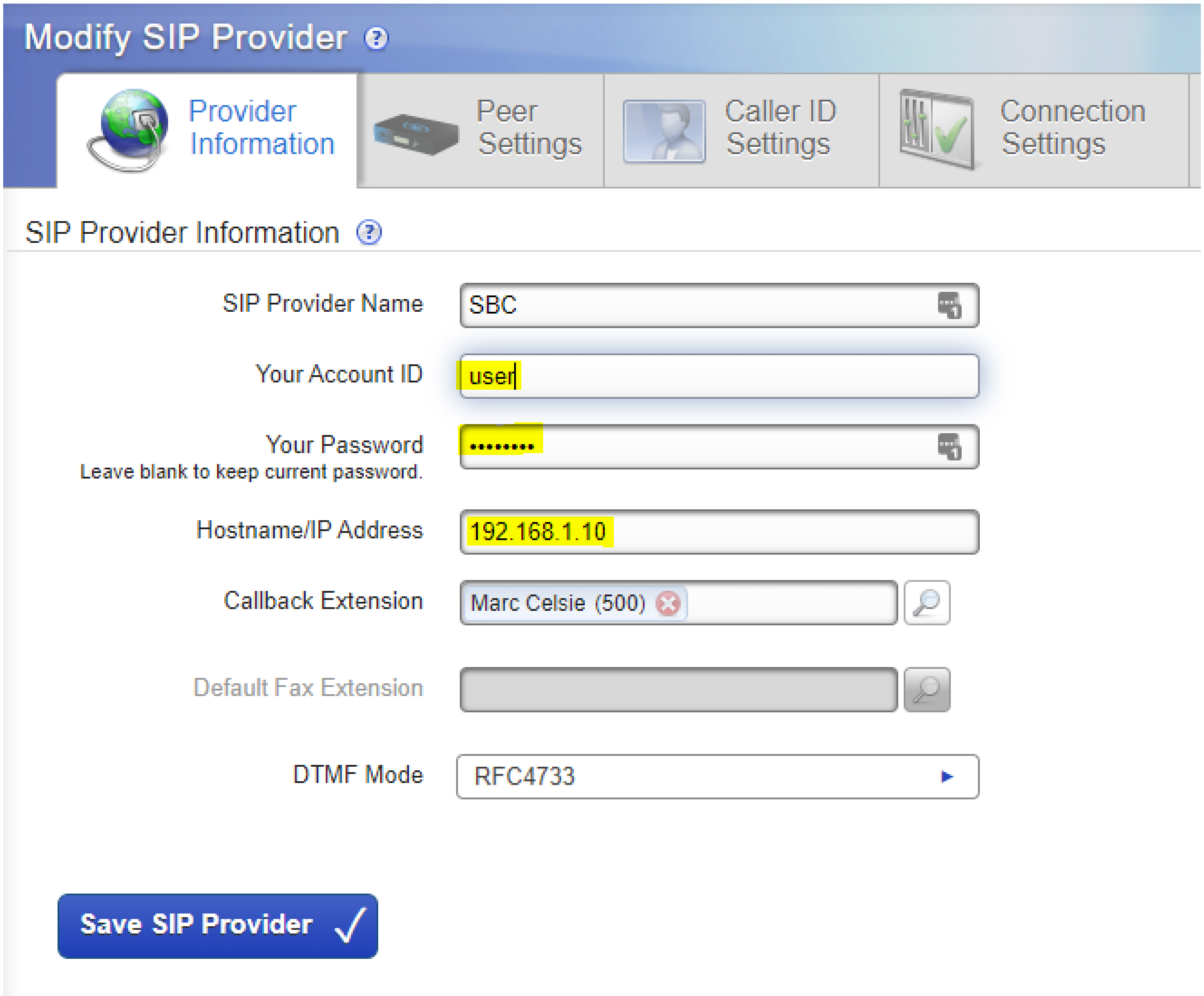


15) Go to **System Server Secure Shell** to do the same for SSH. Setting the Network Interface to the private IP.

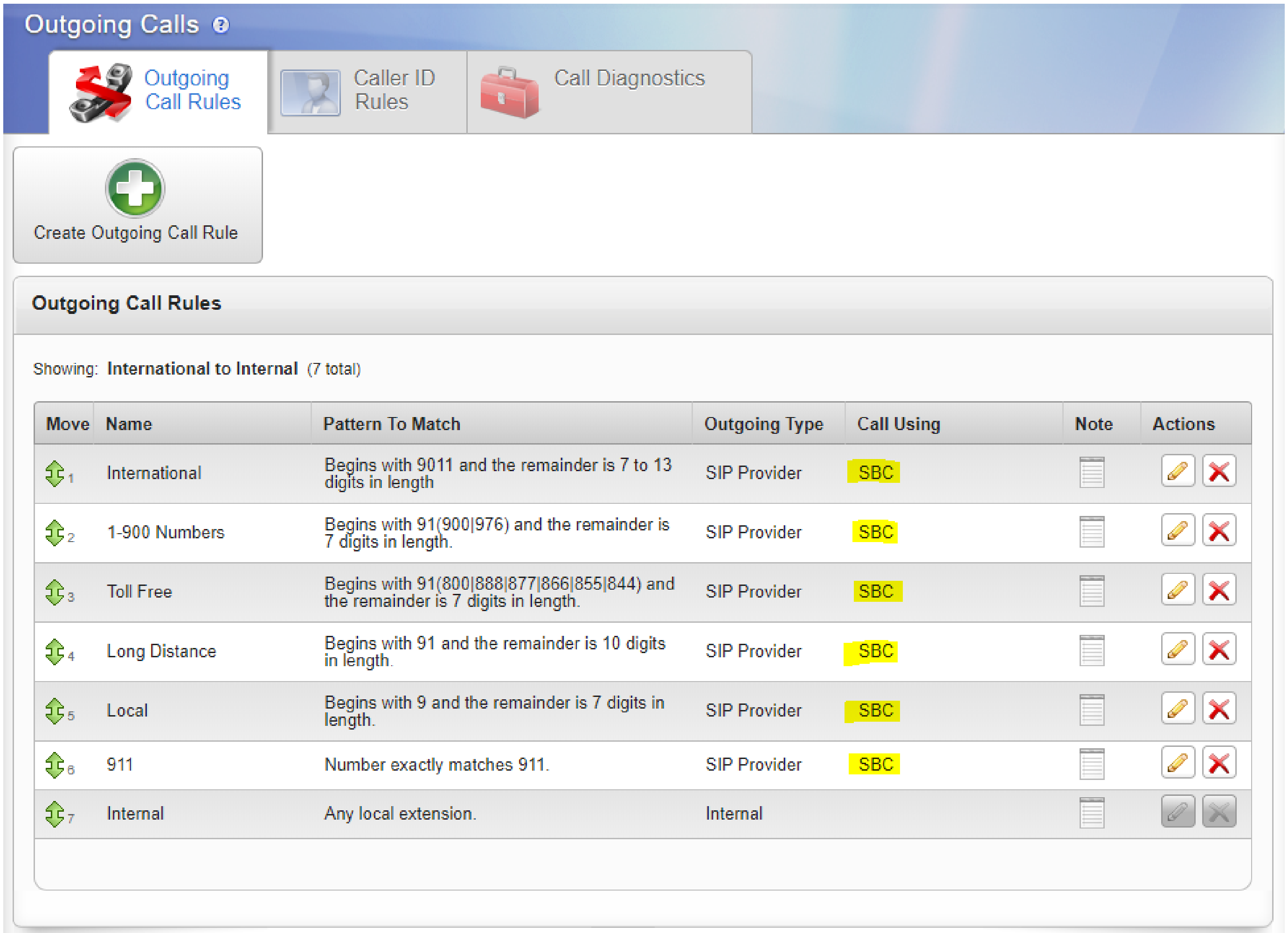


Switchvox Configuration

1) Go to **Setup Call Routing VoIP Providers** and create a new Provider called SBC. Put anything into the Your Account ID and Your Password as a place holder. The put the IP of the SBC into the Hostname/IP Address field. Use the second IP assigned to the SBC. The one that doesn't have the public IP forwarded to it.

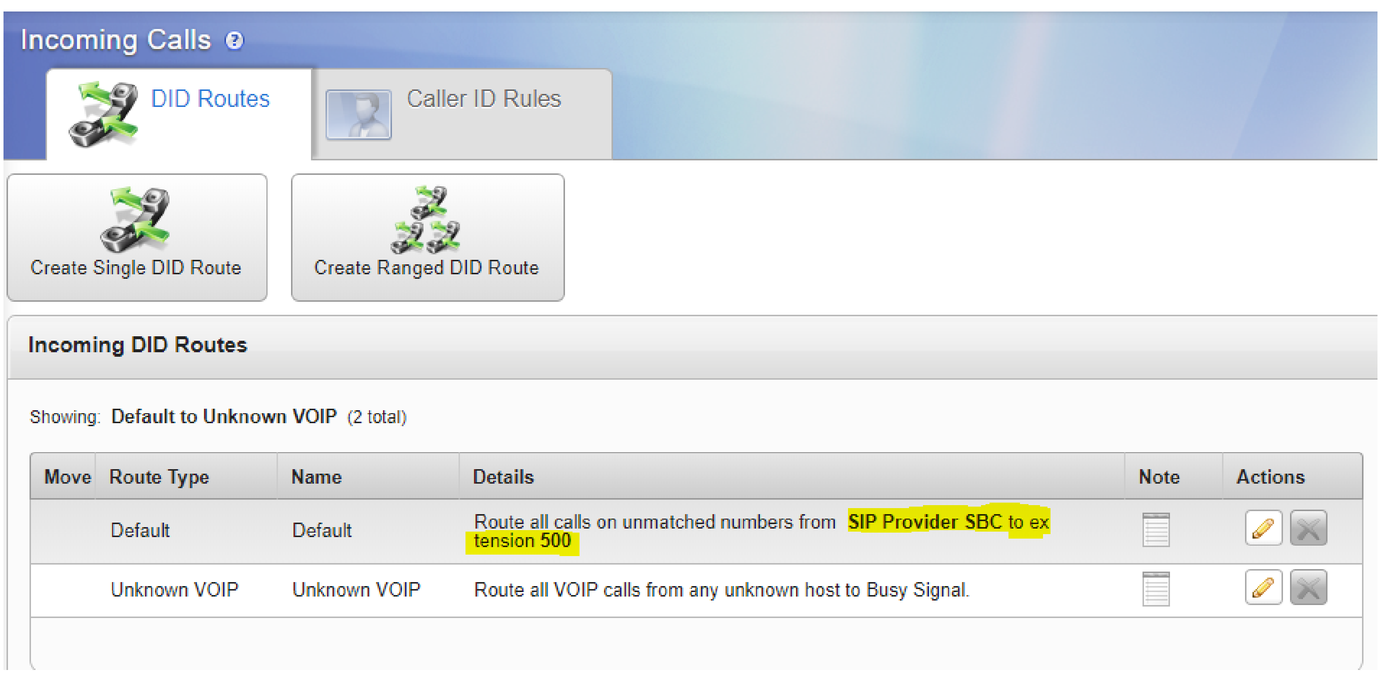


2) Go to **Setup Call Routing Outgoing Calls** and ensure the new Provider called SBC is assigned to the correct routes as shown below.



Move	Name	Pattern To Match	Outgoing Type	Call Using	Note	Actions
1	International	Begins with 9011 and the remainder is 7 to 13 digits in length	SIP Provider	SBC		
2	1-900 Numbers	Begins with 91(900|976) and the remainder is 7 digits in length.	SIP Provider	SBC		
3	Toll Free	Begins with 91(800|888|877|866|855|844) and the remainder is 7 digits in length.	SIP Provider	SBC		
4	Long Distance	Begins with 91 and the remainder is 10 digits in length.	SIP Provider	SBC		
5	Local	Begins with 9 and the remainder is 7 digits in length.	SIP Provider	SBC		
6	911	Number exactly matches 911.	SIP Provider	SBC		
7	Internal	Any local extension.	Internal			

3) Go to **Setup Call Routing Incoming Calls** and set the the destination. Here we have simply just sent the calls to an extension as an example.



Move	Route Type	Name	Details	Note	Actions
	Default	Default	Route all calls on unmatched numbers from SIP Provider SBC to extension 500		
	Unknown VOIP	Unknown VOIP	Route all VOIP calls from any unknown host to Busy Signal.		

4) If there is any issues support will need the info up at https://wiki.sangoma.com/display/SBC/How+To+Capture+Logs when reporting an issue related to the SBC.