THINKTEL COMMUNICATIONS

DIGIUM G100/G200

PRI OVER IP – SIP TRUNKING

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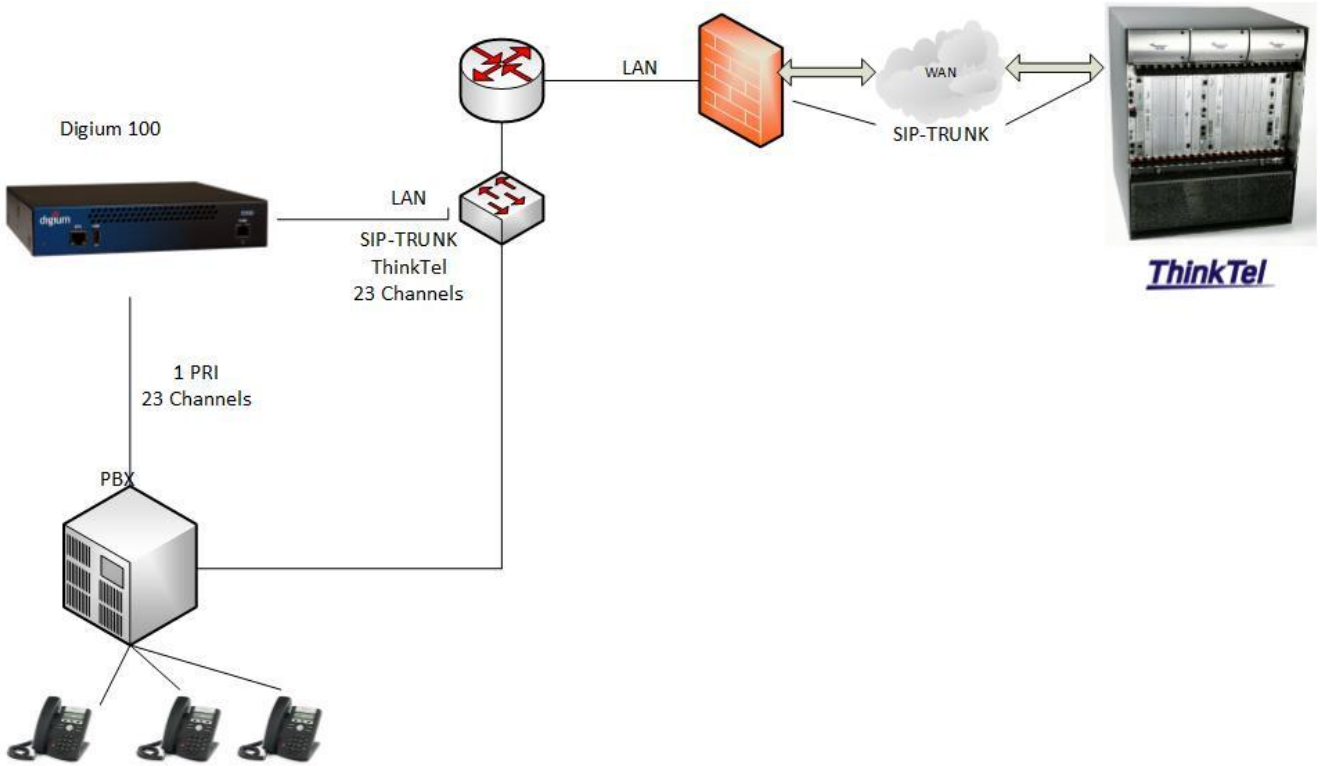
1.1 NETWORK DIAGRAM



-For this case we are going to use only one PRI interface on a G100, 23 concurrent calls

1.2 COLLABORATION OF MONARQUE TELECOM

- All these tests were carried out and implemented in collaboration with MONARQUE TELECOM Elite Digium Switchvox IP-Systems and DIGIUM G100/G200 distributor, a company specializing in business VoIP and hosted IP telephony
- http://www.monarque.ca

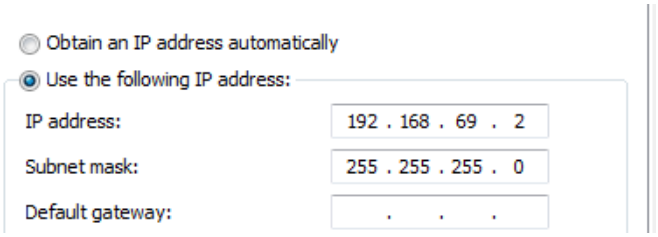
1.3 CONNECTING TO THE DIGIUM G100

- First the DIGIUM G100 must be connected to the mains with the power cable.

- It is recommended to connect the DIGIUM G100/G200 Via Statically assigned IP address, Connect one end of an Ethernet cable to the G100 Ethernet port, and the other end to an Ethernet connection on a computer. That computer will need to use the network configuration listed below

  IP: 192.168.69.2

  Netmask: 255.255.255.0



- Open a browser window and enter the default IP address for the Gateway using HTTPS

  https://192.168.69.1

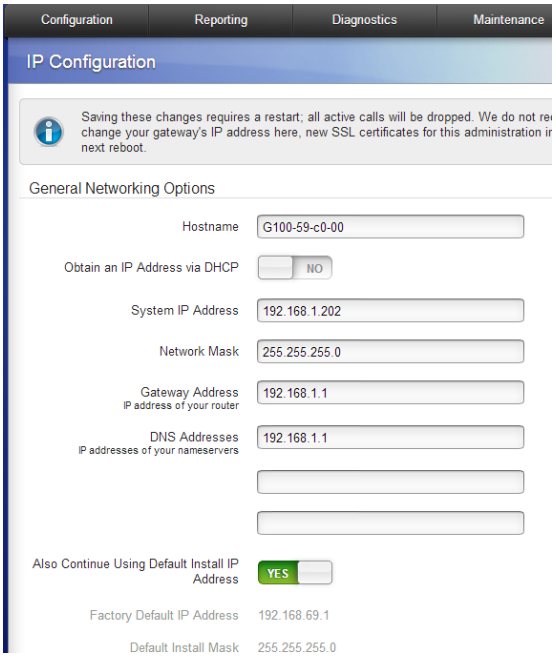
  username: admin

  password: admin

- It is highly recommended that the “admin” password be changed using the “System Administrators” menu on the web GUI

1.4 IP CONFIGURATION

- Click on “Configuration” >> “ IP Configuration”
- Don’t enable the parameter “Obtain an IP address via DHCP”
- Specify the static IP address for the G100 by entering a valid local IP to “System IP Address”
- Enter the Network Mask
- Enter the Gateway IP Address
- Enter the DNS IP Address
- Leave “ Also Continue Using Default IP Address” to “YES”



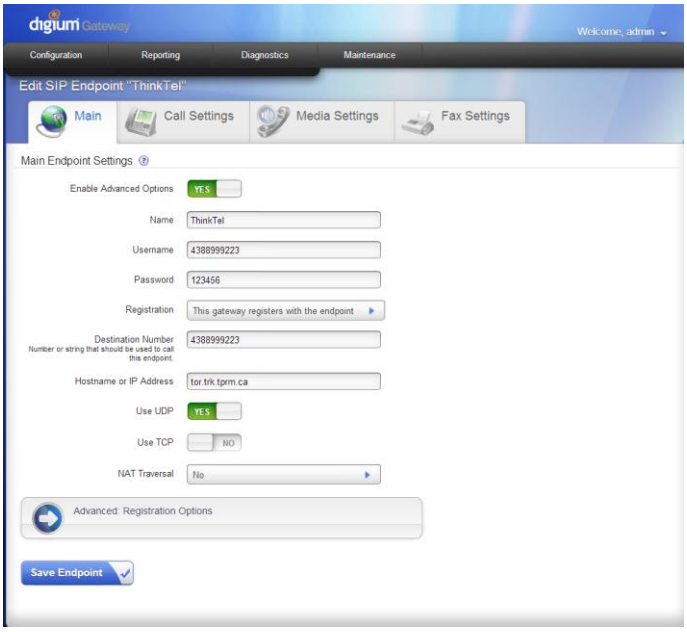
- Note : When you enter the Static IP Address the Gateway will immediately be unresponsive from its old

  IP address. It may take up to 15 seconds for the new IP address to be assigned and functional.
  The new IP address of the Gateway (For our case 192.168.1.202) must be entered into the Browser and use the following link https://192.168.1.202

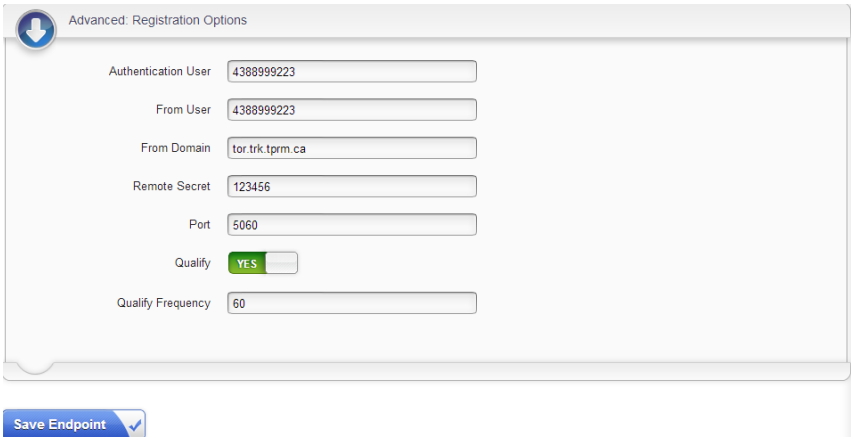
1.5 SIP-TRUNK CONFIGURATION

- Click on “Configuration” >> “SIP Endpoints”
- Click on “Create SIP Endpoint”
- On the Tab “MAIN” set “Enable Advanced Options” to “YES”
- On “Name” enter “ThinkTel”
- On “Username” enter the pilot number provided by ThinkTel
- On “Password” enter the password provided by ThinkTel
- On “Registration” choose “This gateway registers with the endpoint”

- On " Destination Number" enter the same number as the pilot number provided by ThinkTel
- On "Hostname or IP Address" enter the FDQN or the IP Address of the SIP Proxy server provided by ThinkTel
- On "Use UDP" choose "YES"
- On "Use TCP" choose "NO"

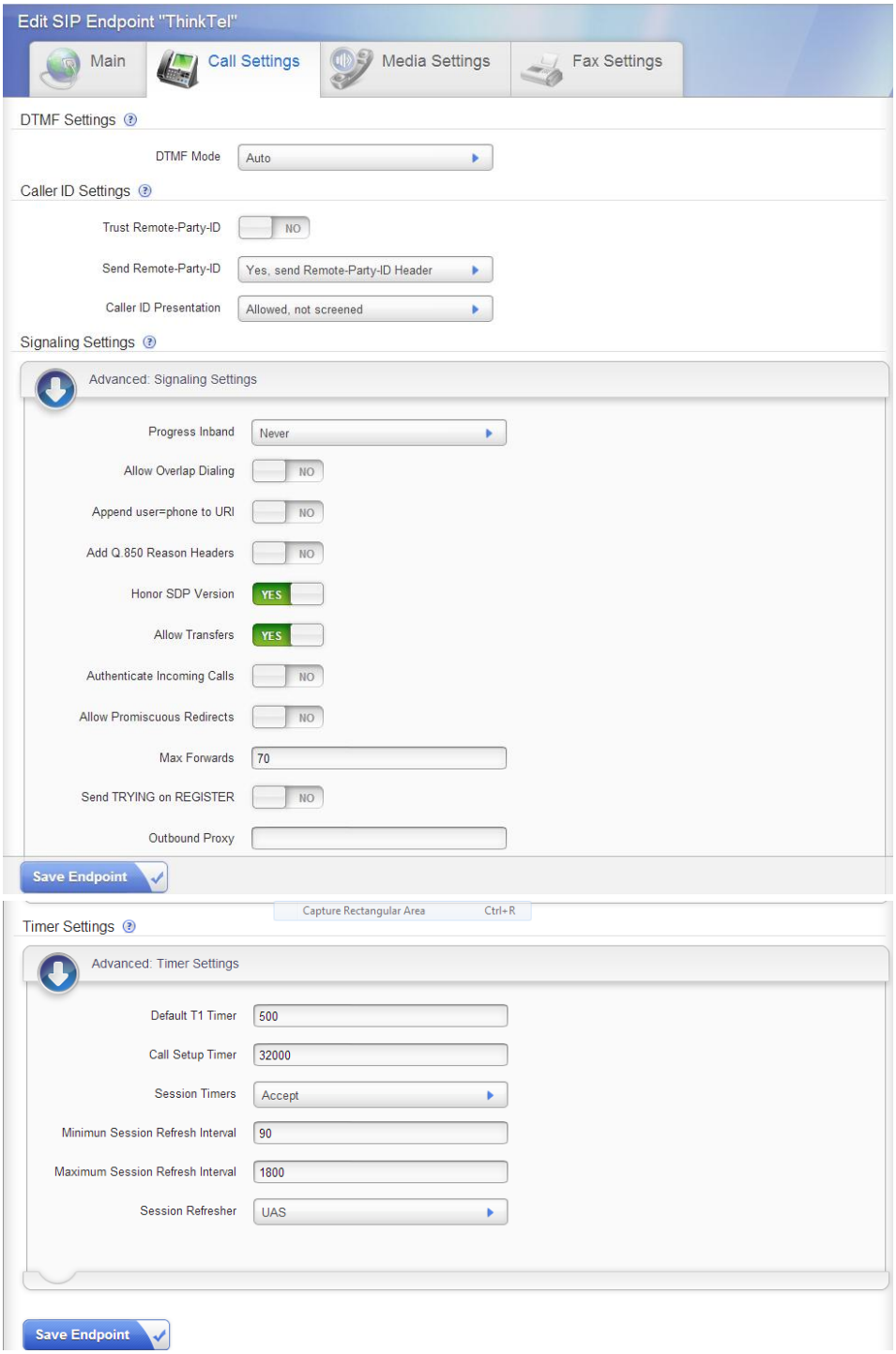


- On "Authentication User" same as pilot number
- On "From User" same as pilot number
- On "From Domain" same as "Hostname"
- On "Remote Secret" same as "Password"
- On "Port" 5060
- On "Qualify" enabled "YES"
- On "Qualify Frequency" 60
- Click on "Save Endpoint"



1.6 CALL SETTINGS

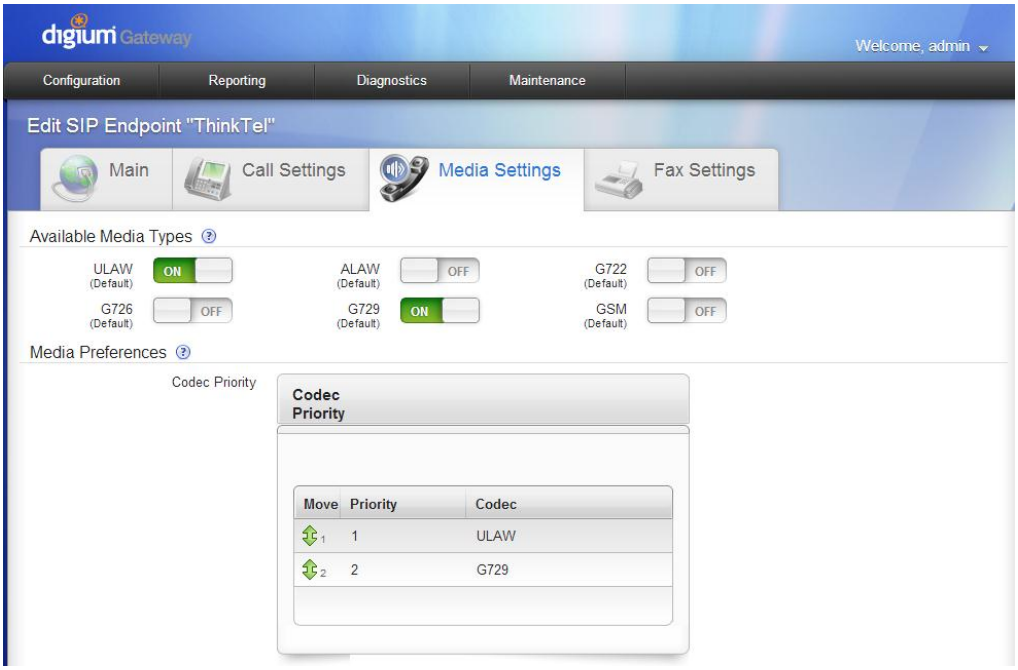
- Click on TAB “Call Settings”
- Configure all Parameters as shown below

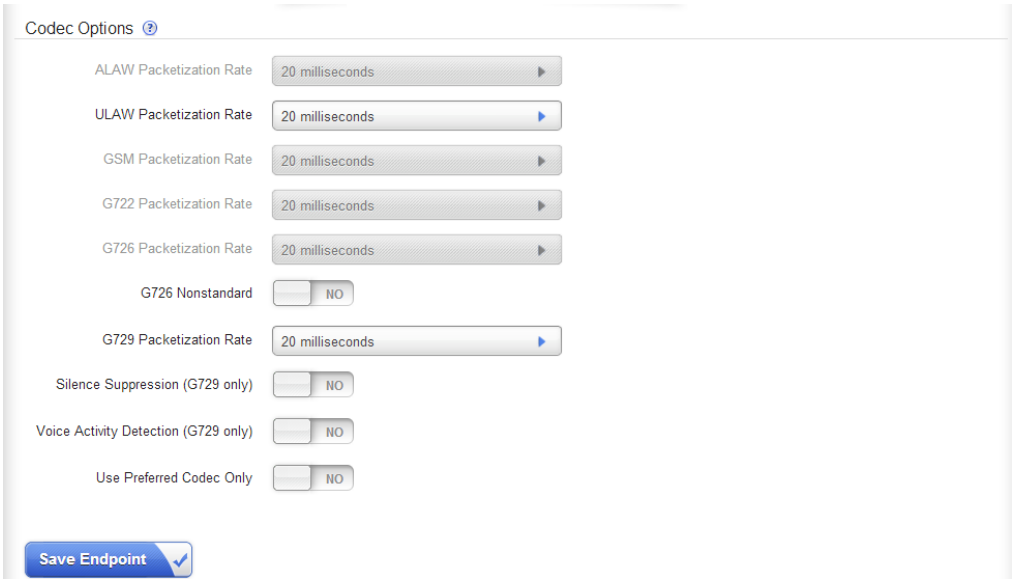


- Click on “Save Endpoint”

1.7 MEDIA SETTINGS

- Click on Tab “Media Settings”
- Enable “ULAW” and “G729”
- Disable “ALAW” , “G722” , “G726” , “GSM”

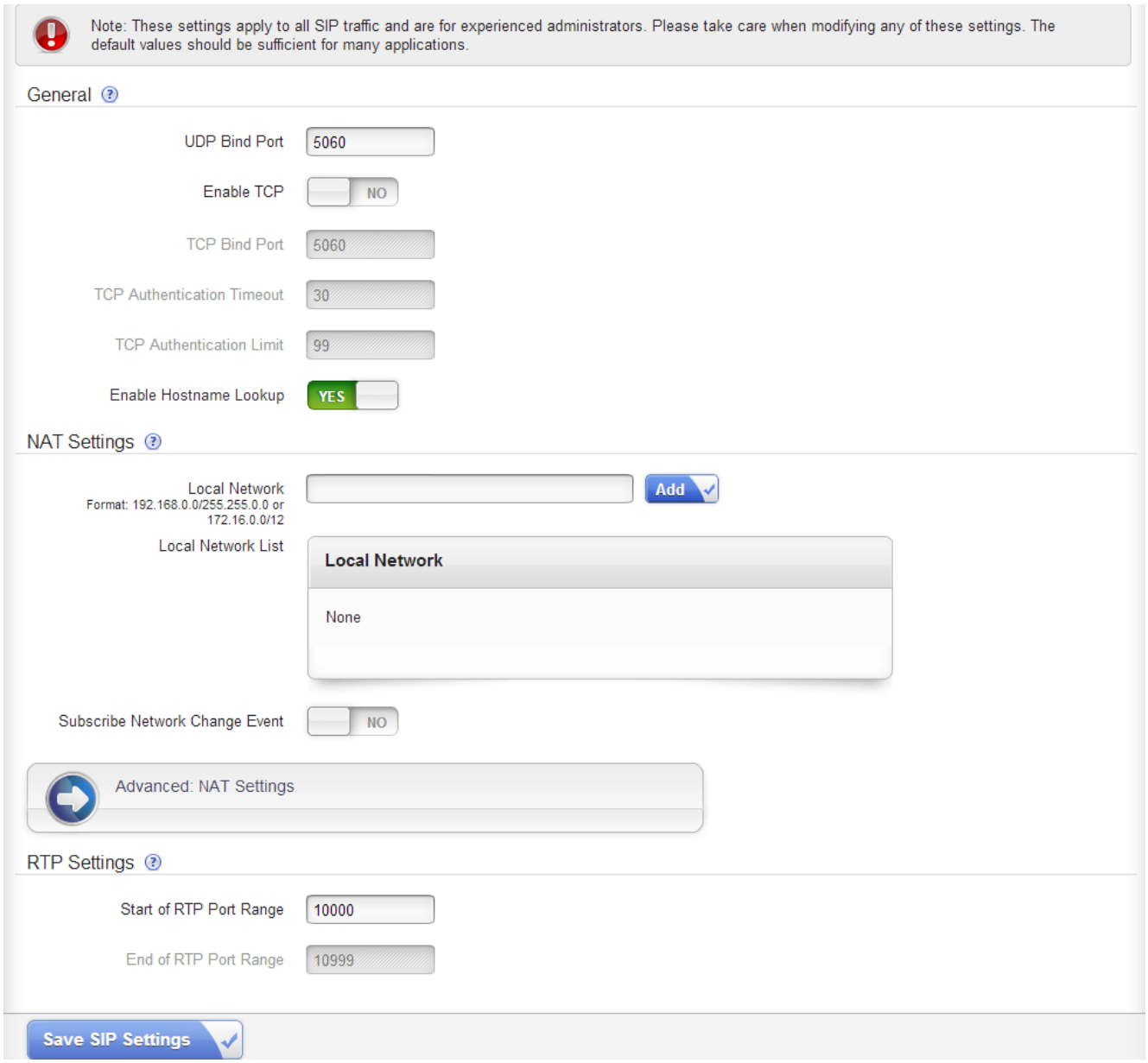




- Click on “Save Endpoint”

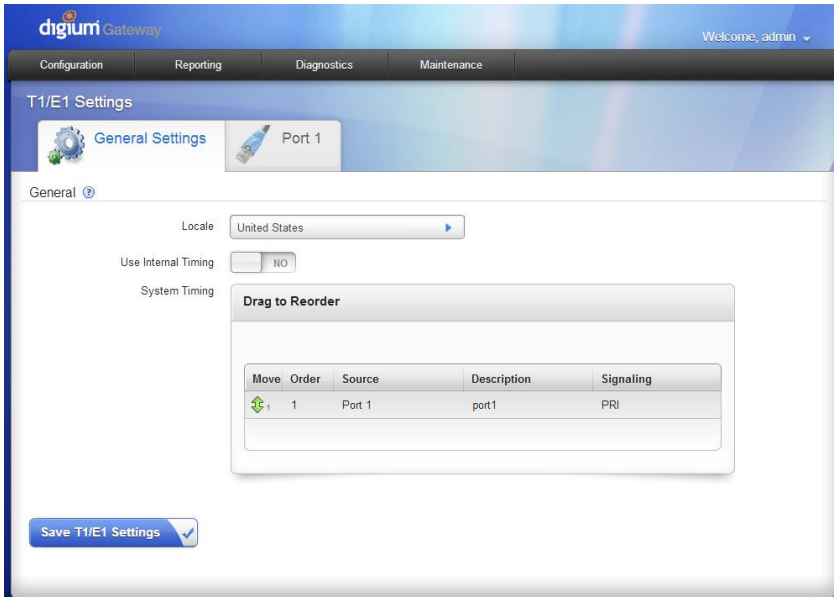
1.8 GLOBAL SIP SETTINGS

- Click on “Configuration” >> “Global SIP Settings”
- Configure all Parameters as shown below

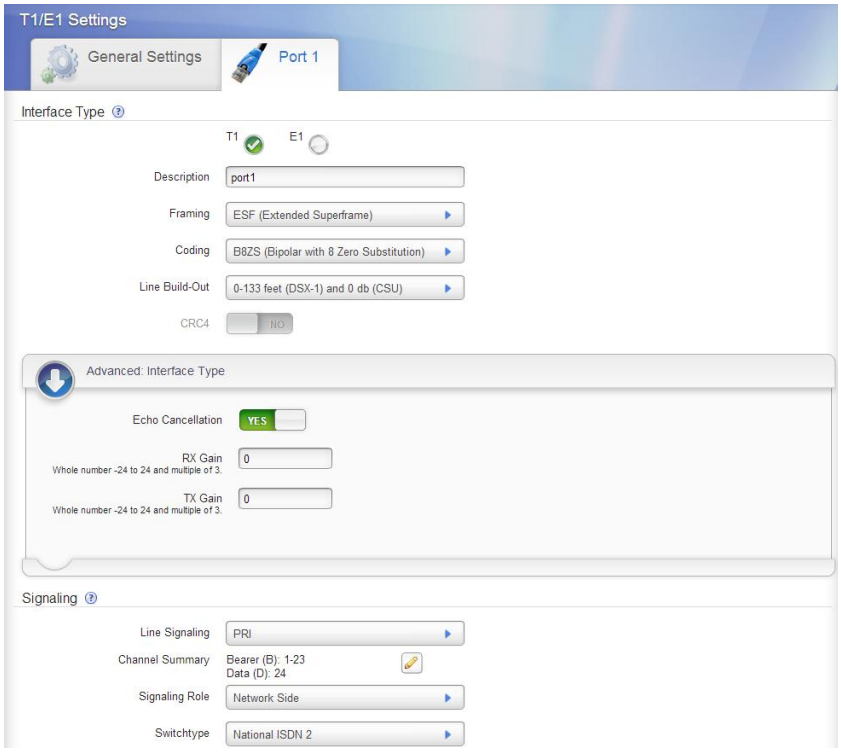


1.9 T1/E1 CONFIGURATION

- Click on “Configuration” >> “T1/E1”



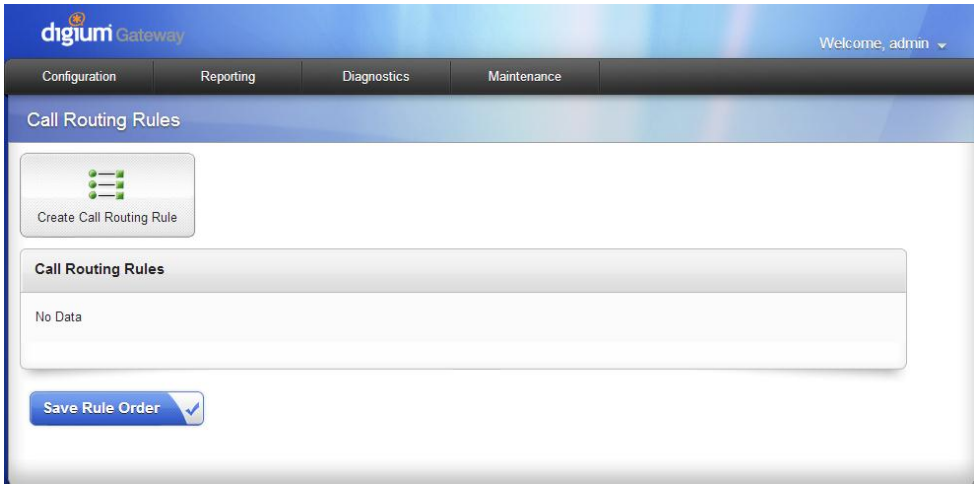
- Click on Tab “Port 1”
- Configure all Parameters as shown below



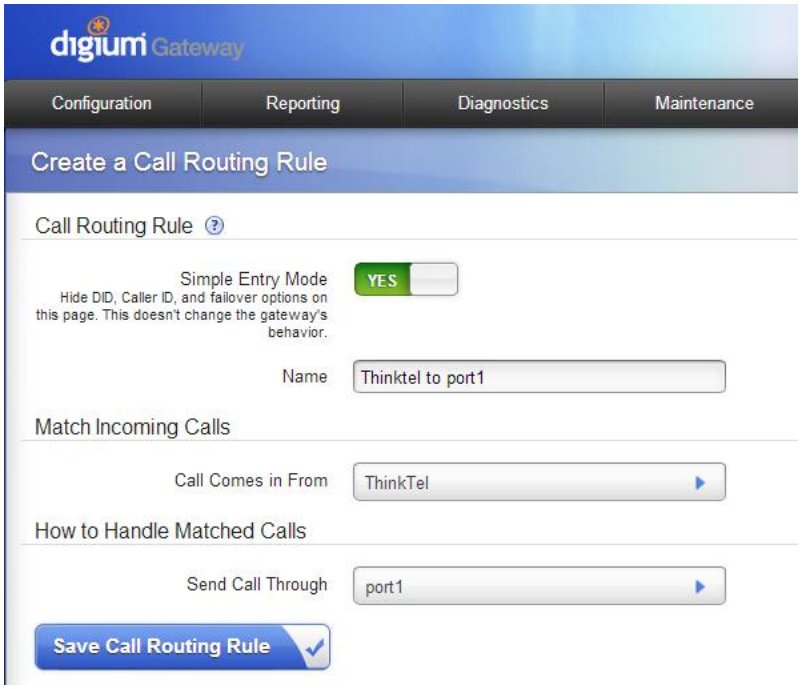
- Click on “Save T1/E1 Settings”

1.10 CALL ROUTING RULES

- Click on “Configuration” >> “Call Routing Rules”

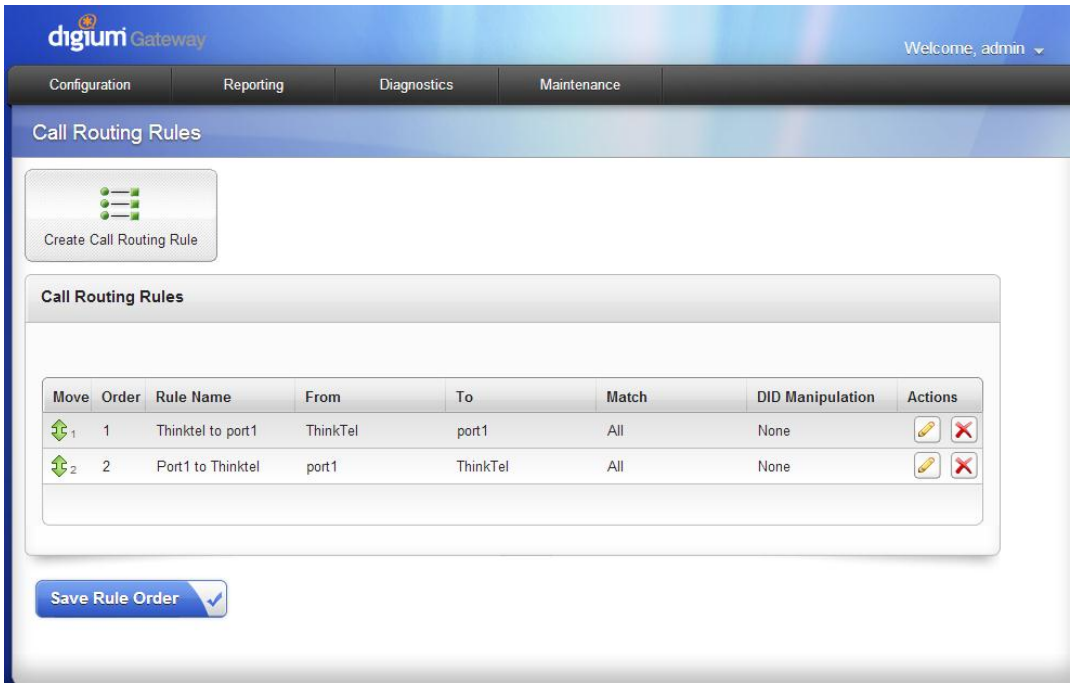


- Click on “Create Call Routing Rule”



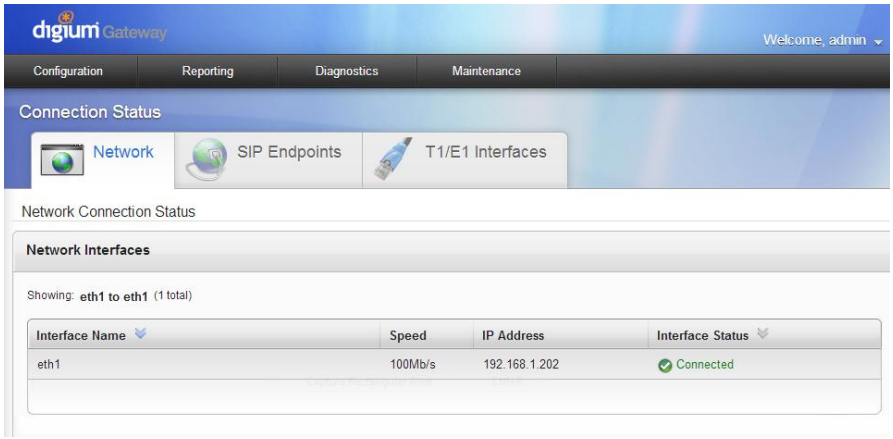
- Enable “Simple Entry Mode”
- On “Name” enter a name for this incoming rule example “ThinkTel to Port1”
- On “Call Comes in From” enter the name of the SIP Endpoint already created “ThinkTel”
- On “Send Call Through” enter “Port1”
- Click on “Save Call Routing Rule”
- Click again on “Create Call Routing Rule”

- Enable “Simple Entry Mode”
- On “Name” enter a name for this outgoing rule example “Port1 to ThinkTel”
- On “Call Comes in From” enter “port1”
- On “Send Call Through” enter the name of the SIP Endpoint already created “ThinkTel”
- Click on “Save Call Routing Rule”

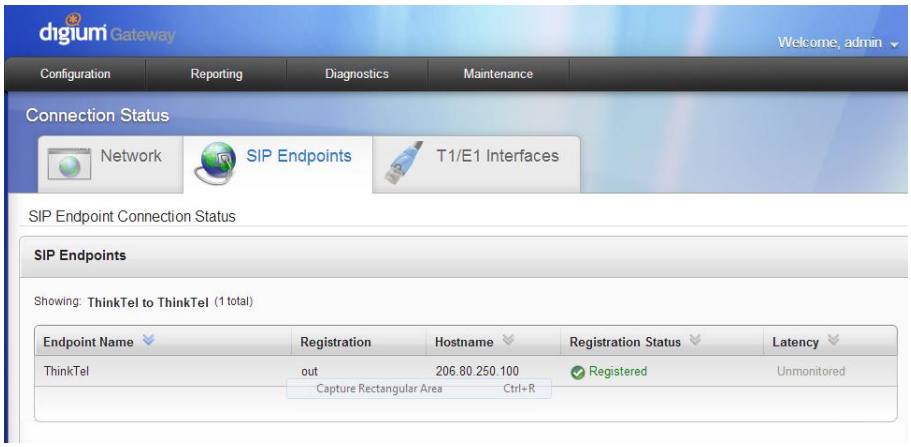


1.11 CONNECTION STATUS

- Click on “Diagnostics” >> “Connection Status”
- On the Tab “Network” verify the connection status of the interface ETH1 (Connected)



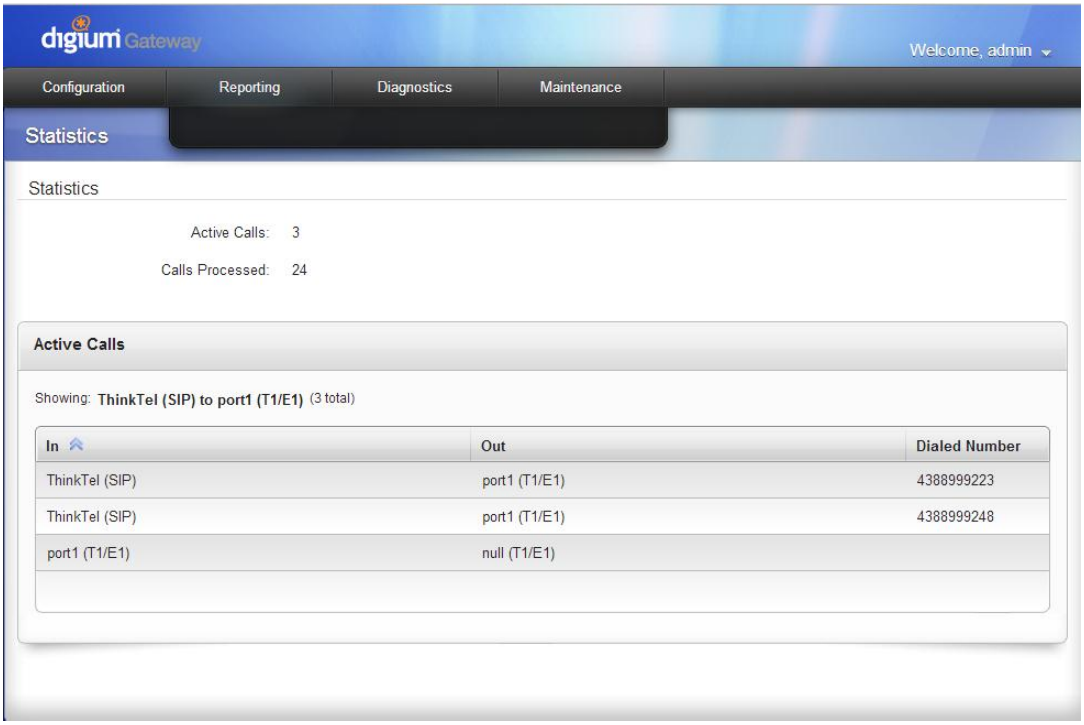
- On the Tab “SIP Endpoints” verify the connection status of the SIP-TRUNK (registered)



- On the Tab “T1/E1 Interfaces” verify the connection status of the PRI (UP,Active)

1.12 OUTGOING AND INCOMING CALLS

- Make outgoing and incoming calls
- Click on “Reporting” >> “Statistics”
- Verify the Active Calls
- Verify the Calls Processed



1.13 BACK UP THE CONFIGURATION

- Click on “Maintenance” >> “Backups”
- Click on “Download Backup” , Save the zipped File

1.14 FIREWALL CONFIGURATION

- If you are connected to the proxy edm.trk.tprm.ca (208.68.17.52) Primarily assigned to Western Canada base PBX trunks you will need to allow :

  1- UDP port 5060 for SIP signaling

  2- UDP ports 10000-65500 for 208.68.17.32/27 , 206.80.250.96/27 & 209.197.133.0/26

- If you are connected to the proxy tor.trk.tprm.ca (206.80.250.100) Primarily assigned to Eastern Canada base PBX trunks you will need to allow :

  1-UDP port 5060 for SIP signaling

  2-UDP ports 10000-65500 for 208.68.17.32/27, 206.80.250.96/27 & 209.197.133.0/26