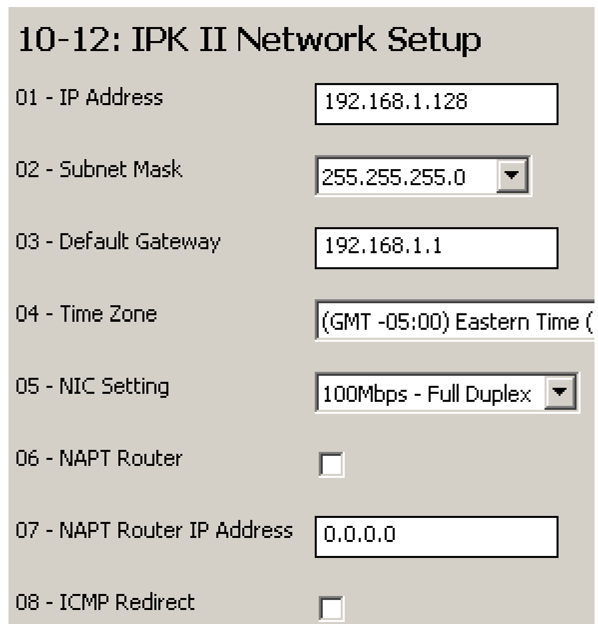
SIP Trunk Connections

Some Basic Requirements

- CPU must have a minimum of four VoIP Clients (SIP) licenses to support SIP trunks. License can be purchased in blocks of 1 (N E C part# 750896), 4 (N E C part number 750893), and 8 (N E C part number 750894).

- A PVA card with the minimum four channel license is required and must be loaded with either the MG16 or Combo application to support SIP trunks. Combo firmware requires the CPU be at 2.X release. PVA licenses can be purchased in blocks of 4 (N E C part# 750874) or 24 (N E C part# 750895)

CPU Configuration



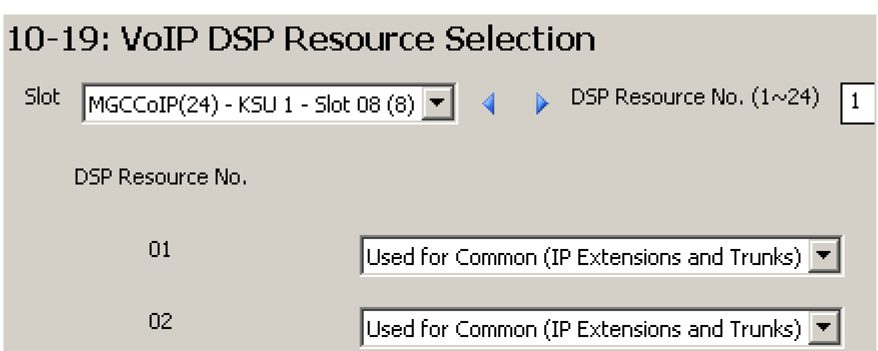
1. ***10-12-XX***: assign the CPU IP Address, Subnet and Gateway.
   Note: ***A CPU reset is required after changing any of these three assignments.***

2. ***10-40-01***: Enable the ability for the MG 16/Combo PVA to support SIP trunks.

3. ***10-40-02***: Assign the number of SIP trunks required. These are assigned in groups of 4.
   Note1: ***Number of trunks assigned must be equal to or less that the number licensed PVA channels.***

   Note2: ***10-40-01/02 above should be done through the phone. Once set the PVA card can be reset and the database will assign the trunks.***

4. ***10-19-01***: This allows you to assign DSP resources (channels on the PVA card) to Trunks only, Stations only, or Both Trunks and Stations.
   Note: ***Leave at default Common unless you have specific requirements to limit the number of connections for one of the two types.***



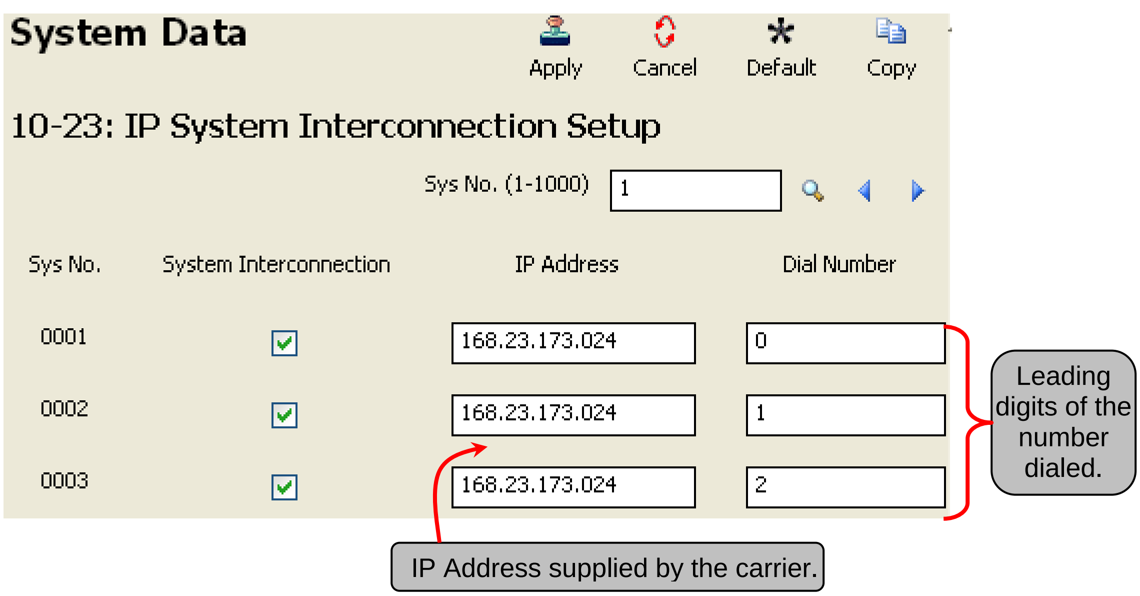
SIP Connection Configuration

There are three connection types when using SIP trunks.

- **Non-Registration Mode**- This is where the trunk simply sends the digits dialed, in an invite message (call setup message), to an IP-Address given to you by the SIP trunk provider. This is also the method used for SIP Tie lines between two phone systems.
- **Registration Mode**- This mode requires the SIP trunk to register with the SIP provider's Registration Server before being allowed to place calls over the provider's network. The system will register with the carrier when first connected and then at programmable intervals. It is a simple registration message notifying the SIP carrier of the IPKII's location/contact information.
- **Registration with Authentication Mode**- Authentication is an additional step to the registration mode. The IPKII will attempt to register to the SIP providers Registration Server and the carrier will respond with a request for authentication which is simply a password. The IPKII forwards on the password and the carrier gives the ok allowing the IPKII access to start making calls on the trunks. As with Registration Mode the Registration Mode with Authentication is also performed upon initial service connection and at programmable intervals there after.

Non-Registration Mode

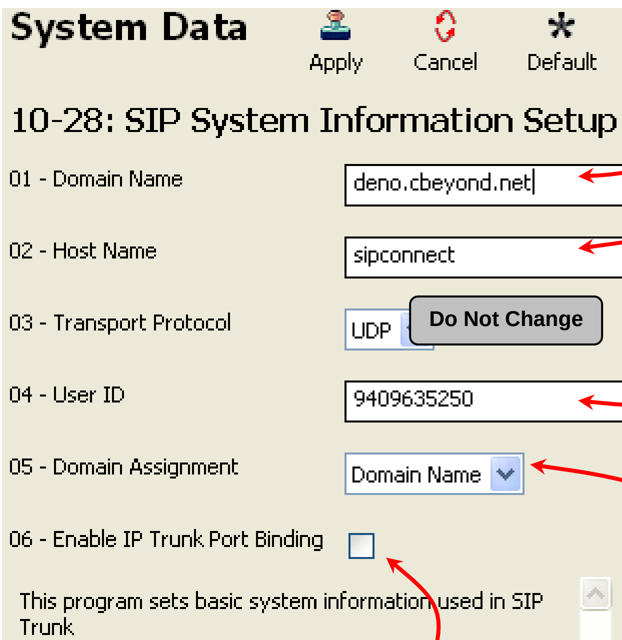
If you are doing Non-Registration Mode (to SIP carrier or SIP Tie lines) you must go to CM 10-23 to set up the dialing options. Typically with this type of setup you need to enter the providers IP Address for each leading digit that will be sent to the network. ***Note:*** ***If you are connecting with Registration Mode via Domain Name or Registration via IP Address, these assignments are not necessary.***



Registration Mode

To make calls the SIP provider will give you the connection information to be assigned in the IPKII. This will be either a domain name or an IP address. Similar to below…..

sipconnect.deno.cbeyond.net (www.cbeyond.com)
Or **168.23.173.024**



Domain Name. This will be the Domain Name of the SIP Trunk provider. If connection is via Domain Name. **Note:** This may not be required if connecting via IP Address in 10-28-05.

Host Name. This is the Host Name also provided from the SIP Provider.

User ID. This is provided by the SIP Trunk provider and usually is the main billing number. Also used for outgoing Caller ID if nothing is assigned in the stations and trunks in 21-17/21-19. ***See page 6.*** This field ***MUST*** be assigned for calls to complete.

Domain Assignment. This is the connection to the SIP provider and will be either via a Domain Name (most common) or an IP address. ***Note: If you are using SIP Tie lines to another Key system or PBX this MUST be set to IP Address.***

Trunk Port Binding. With this enabled the trunks will act like regular ring down trunks where the incoming number will be binded to the trunk it came in on. As a result a second call to that number would provide a busy back to the caller unless trunk hunting is set up with 14-12-02. If left at default the incoming calls will route just like regular DID calls (multiple calls per DID).

Registration Settings

Must be set

10-29: SIP Server Information Setup

01 - Outbound Proxy (Transmit)	☑
02 - Default Proxy	☐
03 - Default Proxy IP Address	0.0.0.0
04 - Default Proxy Port	5060
05 - Registrar Mode	Manual
06 - Registrar IP Address	0.0.0.0
07 - Registrar Port	5060
08 - DNS Mode	☑
09 - DNS IP Address	168.23.171.007
10 - DNS Port	53
11 - Registrar Domain Name	sipconnect.deno.cbeyond.net
12 - Proxy Domain Name	
13 - Proxy Host Name	
14 - SIP Carrier Choice	Carrier B
15 - Registration Expiry Time	3600

The Proxy address is the IP address the IPKII will send the calls to. Sometimes this address will be different from the IP address the IPKII will register to. If registration is via IP Address this must be checked.

IP Address supplied by Sip Trunk provider. When set to Registration via IP Address this is the address the calls will be sent to. If the Proxy and Register addresses are the same both must be entered.

Port 5060 is the popular standard for SIP Trunking and should not be changed unless instructed to by the Sip Trunk Provider.

Must be set to "Manual" when using Registration Mode.

IP Address supplied by Sip Trunk provider. When set to Registration via IP Address this is the address the IPKII will register to. If Register and Proxy addresses are the same both must be entered.

Must be checked when registration is via Domain Name.

DNS: Domain Name Service is a system of servers located throughout the Internet that translate domain names into IP Addresses. The SIP Trunk provider may supply this server address for you otherwise any internet DNS will comply.

Port 53 is the standard for DNS. Do not change unless instructed to by the Sip Trunk provider.

If registering via Domian Name enter here. If the Register and Proxy domain are the same name, entry is only required in the "Register Domain Name" field.

Set to Carrier B

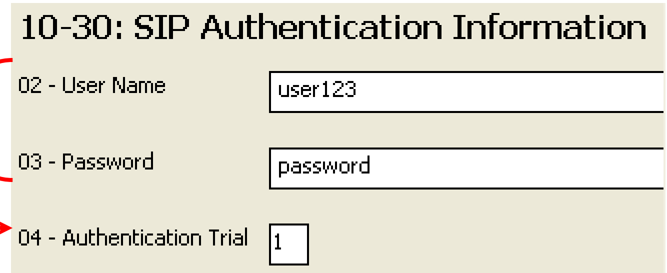
This is he interval between registrations sent from the IPKII to the SIP Trunk provider. The re-registration occurs at half of the value set in this CM. Eg. 3600 = a re-register message sent to the provider every 1800 seconds (30min).

Registration With Authentication Mode

Registration with Authentication requires the registration steps in the previous pages with the addition of CM 10-30

The Authentication User Name and password will be supplied by the SIP Trunk Provider if this feature is to be used.

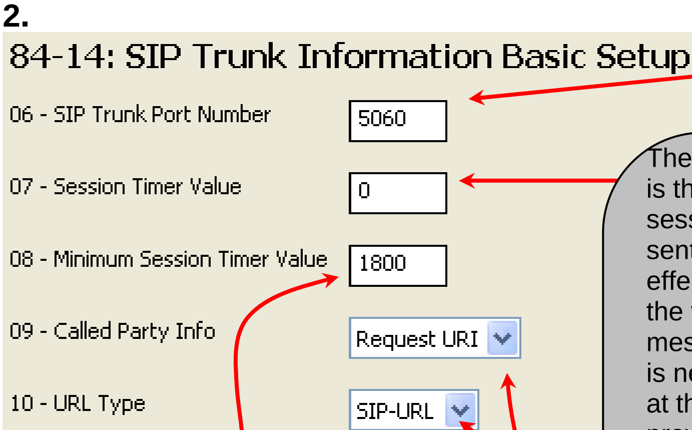
The amount of times the CPU will attempt to register to the SIP Trunk provider. E.g. If an attempt is made to register with the carrier and a "401 Unorthorized" message is received the CPU will re-attempt to register for the amount of times set in 10-30-04



PVA Configuration

1. ***84-05-XX*** assign the PVA IP Address, Subnet, and Gateway. ***Note:*** ***PVA must be in the same subnet (network) as the CPU.***

2.



The SIP Trunk UDP source port (***NOT*** destination port). Should not be changed unless instructed to by the SIP Trunk provider.

The timer the ***UA*** (***User Agent*** which in this case is the IPKII) uses to determine the length the call session is active. If set to ***0*** the "Session" field is not sent in the initial invite message and ***84-14-08*** has no effect. During an active call, at intervals equal to half the value set in *this CM*, the IPKII will send a re-invite message to confirm the call session is still active. This is necessary in some cases as the ***BYE*** message (sent at the end of a call by the IPKII or the SIP Trunk provider) can sometimes get lost due to network issues causing the SIP trunk to stay off hook.
The re-invites ensure that active calls stay active and completed calls are terminated. On an outbound call the IPKII will send the timer value in the initial invite (setup) message. If the SIP Trunk provider supports this feature they will reply with a 200 OK message containing a timer field. If no timer field is present in the 200 OK message the feature is not supported and the IPKII will not send the re-invites.

This field is only valid if a value is entered in ***84-14-07***.
If a received incoming SIP call has a Session field in the invite message containing a value less than half of the value in this CM, the IPKII will reject the call. E.g. Incoming Invite message has a session field set to 720 seconds and **84-14-08** is set to 1800 the call will fail. Setting of 1800 is actually only 900 seconds.

There are 2 fields in the invite message that can contain the called party information (DID info).
In most cases the Request URI field will be used for this information to route the call. The other filed is the "TO Header" which is contained in every SIP message. Leave at URI unless instructed to change by Sip Trunk provider.

This is the format of the invite message from the SIP Trunk provider. This should only be changed if directed to by the provider.

Caller ID Configuration

In the SIP Invite message there are two parts in the "**From**" header that contain information that can be used for the Caller-ID. The following is an example of the "**From**" field contained in the incoming and outgoing SIP Trunk invite message…..

From: "2142622000"<sip:necntacdallas@necntac.com

Display Field **URI Field**

Receiving Caller ID

When a call is received with the above format the **Display Field** shows in the left side of the display while the **URI Field** shows in the right side of the display.
20-09-02 should be enabled for the COS of the station to allow the received caller ID info to the display of the phone.

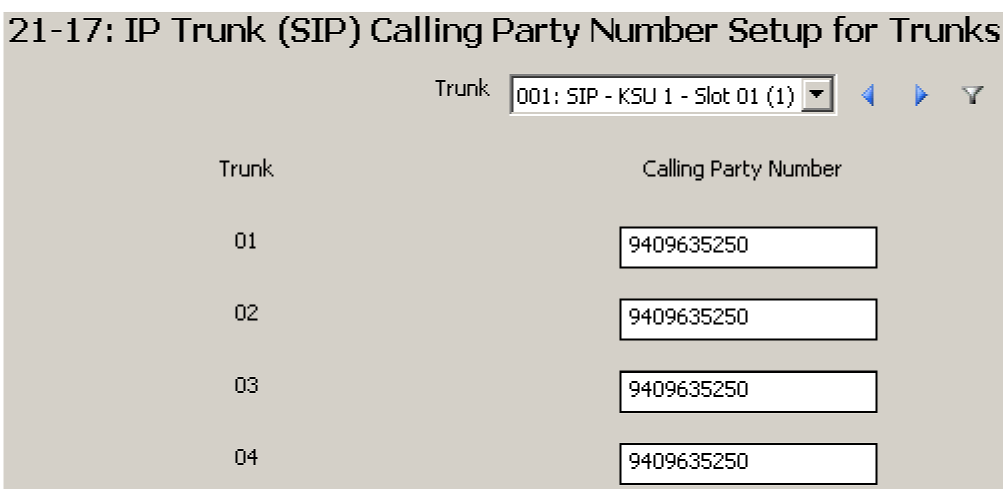
Sending Caller ID

When a call is sent the receiving party will typically look to the URI Field for the caller ID. Not all SIP devices have the ability to show the information in both the URI and Display Field's, if received. Make sure the IPKII is at minimum **1.6BE** or **2.1BE**.

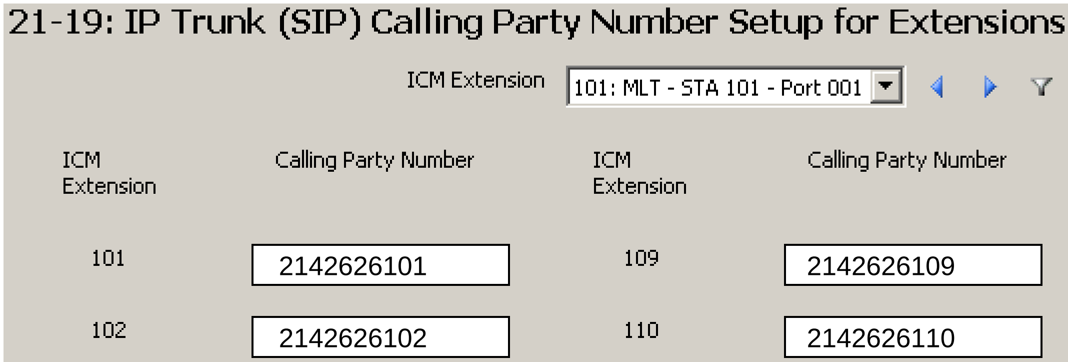
1. **10-28-04** This is the default location for outgoing caller ID. The entry in this CM will populate the Display and URI field if no other Caller ID programming is assigned. This CM MUST be completed to allow SIP Trunks to function.

2. **20-08-13** The ability to send caller ID must be enabled in the COS for the station.

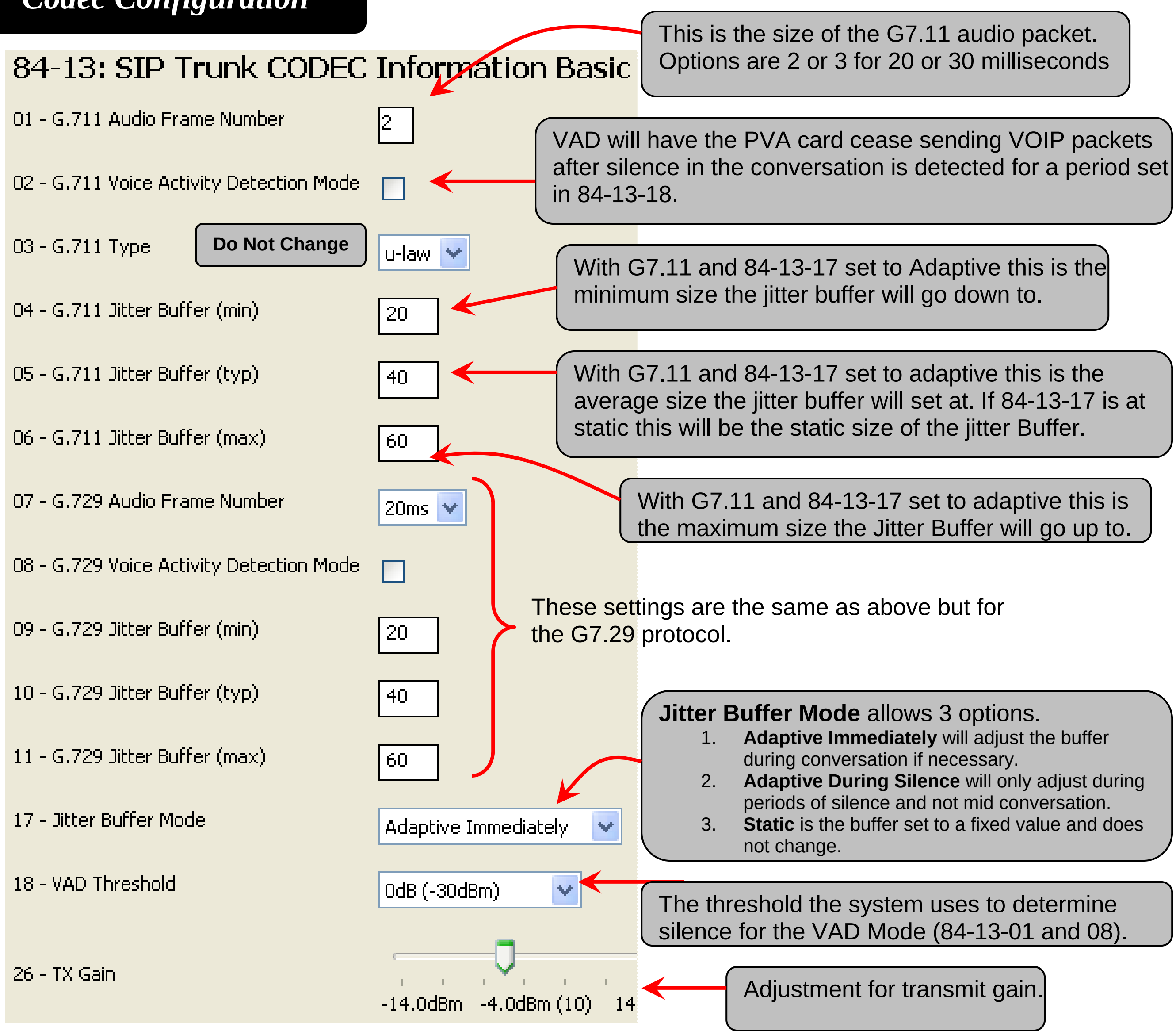
3. Assign the outgoing Calling Number on a trunk basis in 21-17. Enter 10 digits only for each trunk.

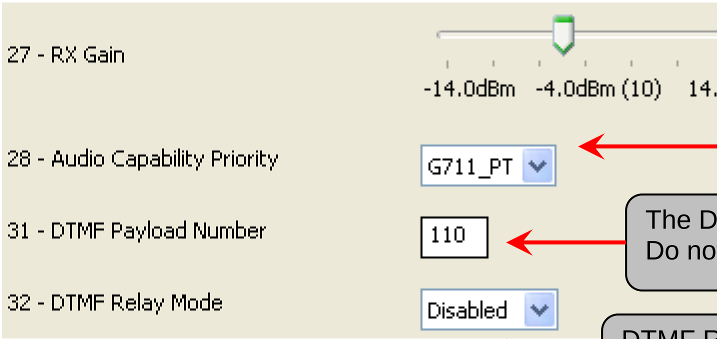


4. Assign outgoing Calling Number on a station basis in 21-19. If this is assigned along with 21-17 (trunk basis) 21-19 will take priority and be sent on the outgoing call.



Codec Configuration





Adjustment for Receive gain.

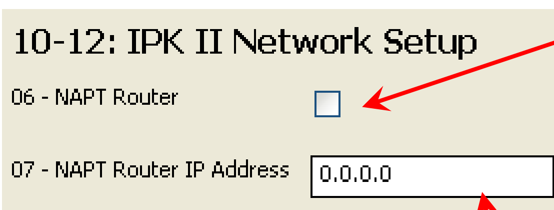
The Voice codec used. G711 or G729

The DTMF payload is part of the initial setup for a call. Do not change this or you will not be able to pass DTMF.

DTMF Relay Mode is another way of passing DTMF to the carrier. Do not enable unless instructed to by the carrier.

NAT Configuration

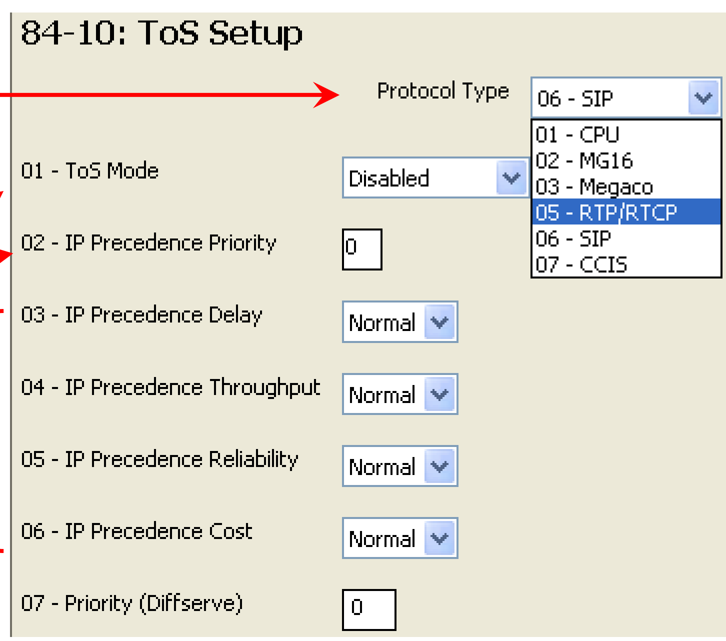
First the customer's router, configured for NAT, must have UDP port 5060 forwarded to the LAN IP address of the CPU (10-12-01) for the SIP call signaling packets.
UDP ports 10020 and higher must be forwarded to the LAN IP address of the PVA card (84-05-01) for the SIP voice packets. The number of ports to forward is dependant on the number of SIP trunks assigned. Each SIP trunk requires two ports. E.g. Eight SIP trunks would require UDP ports 10020 ~10035 forwarded to the LAN address of the PVA card.



Check to enable

Assign the Public (internet) IP address of the router performing the NAT translation. This address will be added to the SDP portion of the SIP signaling and voice packets instead of the default private LAN IP addresses from 10-12-01 and 84-05-01.

QOS



QOS/TOS setup for SIP trunks must be applied to Protocol Type **05 RTP/RTCP** for the SIP voice packets and **06** for the SIP control packets.

Set the ToS Mode to either **IP Presedence** or **Diffserve**

If **IP Presedence** was selected choose the priority 1~7

Not used.

If **Diffserve** was selected choose the priority 0~63.
Note: Expedited Forwarding =46