

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 (NE C part# 750896), 4 (NE C part number 750893), and 8 (NE 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 (NE C part# 750874) or 24 (NE 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.

10-12: IPK II Network Setup

01 - IP Address	192.168.1.128
02 - Subnet Mask	255.255.255.0
03 - Default Gateway	192.168.1.1
04 - Time Zone	(GMT -05:00) Eastern Time (
05 - NIC Setting	100Mbps - Full Duplex
06 - NAPT Router	<input type="checkbox"/>
07 - NAPT Router IP Address	0.0.0.0
08 - ICMP Redirect	<input type="checkbox"/>

10-19: VoIP DSP Resource Selection

Slot: MGCCoIP(24) - KSU 1 - Slot 08 (8) DSP Resource No. (1~24) 1

DSP Resource No.	Assignment
01	Used for Common (IP Extensions and Trunks)
02	Used for Common (IP Extensions and Trunks)

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.**

System Data

Apply Cancel Default Copy

10-23: IP System Interconnection Setup

Sys No. (1-1000)

Sys No.	System Interconnection	IP Address	Dial Number
0001	<input checked="" type="checkbox"/>	<input type="text" value="168.23.173.024"/>	<input type="text" value="0"/>
0002	<input checked="" type="checkbox"/>	<input type="text" value="168.23.173.024"/>	<input type="text" value="1"/>
0003	<input checked="" type="checkbox"/>	<input type="text" value="168.23.173.024"/>	<input type="text" value="2"/>

IP Address supplied by the carrier.

Leading digits of the number dialed.

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**

System Data Apply Cancel Default

10-28: SIP System Information Setup

01 - Domain Name

02 - Host Name

03 - Transport Protocol Do Not Change

04 - User ID

05 - Domain Assignment

06 - Enable IP Trunk Port Binding

This program sets basic system information used in SIP Trunk

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

10-29: SIP Server Information Setup

01 - Outbound Proxy (Transmit)	<input checked="" type="checkbox"/>	Must be set
02 - Default Proxy	<input type="checkbox"/>	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.
03 - Default Proxy IP Address	0.0.0.0	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.
04 - Default Proxy Port	5060	Port 5060 is the popular standard for SIP Trunking and should not be changed unless instructed to by the Sip Trunk Provider.
05 - Registrar Mode	Manual	Must be set to "Manual" when using Registration Mode.
06 - Registrar IP Address	0.0.0.0	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.
07 - Registrar Port	5060	
08 - DNS Mode	<input checked="" type="checkbox"/>	Must be checked when registration is via Domain Name.
09 - DNS IP Address	168.23.171.007	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.
10 - DNS Port	53	Port 53 is the standard for DNS. Do not change unless instructed to by the Sip Trunk provider.
11 - Registrar Domain Name	sipconnect.deno.cbeyond.net	If registering via Domain Name enter here. If the Register and Proxy domain are the same name, entry is only required in the "Register Domain Name" field.
12 - Proxy Domain Name		
13 - Proxy Host Name		
14 - SIP Carrier Choice	Carrier B	Set to Carrier B
15 - Registration Expiry Time	3600	

This is the 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 Unauthorized" message is received the CPU will re-attempt to register for the amount of times set in 10-30-04

10-30: SIP Authentication Information	
02 - User Name	<input type="text" value="user123"/>
03 - Password	<input type="text" value="password"/>
04 - Authentication Trial	<input type="text" value="1"/>

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. **84-14: SIP Trunk Information Basic Setup**

06 - SIP Trunk Port Number	<input type="text" value="5060"/>
07 - Session Timer Value	<input type="text" value="0"/>
08 - Minimum Session Timer Value	<input type="text" value="1800"/>
09 - Called Party Info	<input type="text" value="Request URI"/>
10 - URL Type	<input type="text" value="SIP-URL"/>

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.

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

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 field is the "TO Header" which is contained in every SIP message. Leave at URI unless instructed to change by Sip Trunk 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

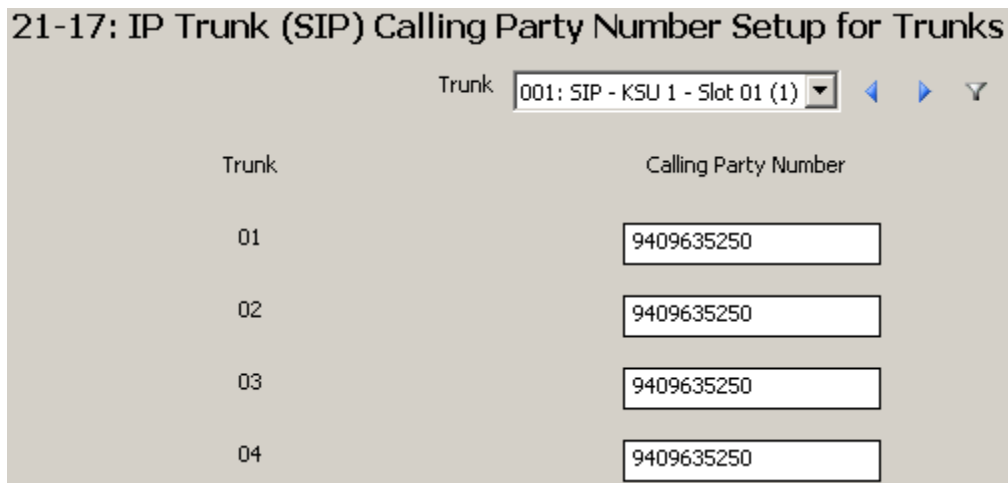
Receiving Caller ID

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.



- 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.

21-19: IP Trunk (SIP) Calling Party Number Setup for Extensions

ICM Extension: 101: MLT - STA 101 - Port 001

ICM Extension	Calling Party Number	ICM Extension	Calling Party Number
101	2142626101	109	2142626109
102	2142626102	110	2142626110

Codec Configuration

84-13: SIP Trunk CODEC Information Basic

01 - G.711 Audio Frame Number: 2

02 - G.711 Voice Activity Detection Mode:

03 - G.711 Type: Do Not Change u-law

04 - G.711 Jitter Buffer (min): 20

05 - G.711 Jitter Buffer (typ): 40

06 - G.711 Jitter Buffer (max): 60

07 - G.729 Audio Frame Number: 20ms

08 - G.729 Voice Activity Detection Mode:

09 - G.729 Jitter Buffer (min): 20

10 - G.729 Jitter Buffer (typ): 40

11 - G.729 Jitter Buffer (max): 60

17 - Jitter Buffer Mode: Adaptive Immediately

18 - VAD Threshold: 0dB (-30dBm)

26 - TX Gain: -4.0dBm (10)

This is the size of the G.711 audio packet. Options are 2 or 3 for 20 or 30 milliseconds

VAD will have the PVA card cease sending VOIP packets after silence in the conversation is detected for a period set in 84-13-18.

With G7.11 and 84-13-17 set to Adaptive this is the minimum size the jitter buffer will go down to.

With G7.11 and 84-13-17 set to adaptive this is the average size the jitter buffer will set at. If 84-13-17 is at static this will be the static size of the jitter Buffer.

With G7.11 and 84-13-17 set to adaptive this is the maximum size the Jitter Buffer will go up to.

These settings are the same as above but for the G7.29 protocol.

Jitter Buffer Mode allows 3 options.

- Adaptive Immediately** will adjust the buffer during conversation if necessary.
- Adaptive During Silence** will only adjust during periods of silence and not mid conversation.
- Static** is the buffer set to a fixed value and does not change.

The threshold the system uses to determine silence for the VAD Mode (84-13-01 and 08).

Adjustment for transmit gain.

27 - RX Gain: Slider set to -4.0dBm (10). Callout: Adjustment for Receive gain.

28 - Audio Capability Priority: G711_PT. Callout: The Voice codec used. G711 or G729

31 - DTMF Payload Number: 110. Callout: 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.

32 - DTMF Relay Mode: Disabled. Callout: 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.

10-12: IPK II Network Setup

06 - NAPT Router: . Callout: Check to enable

07 - NAPT Router IP Address: 0.0.0.0. Callout: 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 Precedence** or **Diffserve**

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

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

Not used.

84-10: ToS Setup

Protocol Type: 06 - SIP (dropdown menu shows 01 - CPU, 02 - MG16, 03 - Megaco, 05 - RTP/RTCP, 06 - SIP, 07 - CCIS)

01 - ToS Mode: Disabled

02 - IP Precedence Priority: 0

03 - IP Precedence Delay: Normal

04 - IP Precedence Throughput: Normal

05 - IP Precedence Reliability: Normal

06 - IP Precedence Cost: Normal

07 - Priority (Diffserve): 0