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SIP Setup

This page defines global SIP configuration parameters.

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SIP Networking Settings Section

Sections/Fields	Description
Network Address Included	The PBX automatically includes its own localnet if the setting below is set to yes.
Local Network & Subnet Masks	This is where you can add additional IP address to the local network and associated subnet masks.
Delete Selected	Allows you to delete the item selected from list of networks.
Add Local Network	Allows you to add local network information. This information will appear in the list of networks.
Include LAN network as Localnet	When set to Yes, the PBX will automatically include its own localnet.
Remote Clients Access PBX by	Defines how Remote Clients (remote phones or SIP providers) access the PBX. Set to No Remote if neither is being used. Set to IP address if the site has a static IP or set to Hostname if the site uses a dynamic IP. To work, they will need a Dyndns.com domain, and enter that as the External Host.
External IP	Used if Remote Clients Access PBX by is set to IP Address. Click the Get IP button to populate this field with the public IP of the gateway the PBX uses to access the Internet
External Host	Used if Remote Clients Acccess PBX By is set to Hostname. Enter the dyndns.com domain that resolves to the current dynamic IP of the system here.
External Host Refresh	Used if Remote Clients Access PBX By is set to Hostname. This is the internval which the PBX will check for an up to date IP. Set this to a value within the internval set at dyndns, typically between 300 and 3000.

Add SIP Networking Settings

STEPS:

1. From the **PBX Setup->SIP** page, locate the **SIP Networking Settings** section.
2. Enter the IP Address and Subnet Mask for the network the PBX is being installed on.
3. Click the Add button.
4. Click the . button.
5. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.

Delete SIP Networking Settings

STEPS:

1. From the **PBX Setup->SIP** page, locate the **SIP Networking Settings** section.
2. Highlight the listing you wish to delete. You can use Shift/Ctrl click functionality to select multiple listings.
3. Click the Delete Selected button.
4. Click the . button.
5. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database

SIP Advanced Settings Section

Advanced SIP settings define in more detail the management of network traffic. These settings are automatically provisioned when the system registers with the router. In most business implementations it is not necessary to make changes to these defaulted settings

IMPORTANT: The default settings for the SIP configuration should not require any changes. If it is necessary for you to do so to meet your customer?s business requirements, we recommend that you contact IPitomy?s Technical Support for assistance.

Sections/Fields	Description/Default Parameters
Call Context	Default: **INCOMING**
Allow Guest Calls	Default: **YES**
Host/Domain Name	Default: **BLANK**
UDP Port	Default: **5060**
Bind Address	Default: **0.0.0.0**
Enable DNS SRV Lookup	Default: **NO**
Domains	Default: **BLANK**
Allow External Invites	Default: **YES**
Auto Domain	Default: **N/A**
Enable Pedantic Checking	Default: **N/A**
SIP TOS	

	Default is **CS3**. To configure QOS on your LAN, set your managed switches to prioritize packets flagged with CS3
RTP TOS	Default is **CS3**. To configure QOS on your LAN, set your managed switches to prioritize packets flagged with CS3
Video TOS	Default is **CS3**. To configure QOS on your LAN, set your managed switches to prioritize packets flagged with CS3
Max Length of Registration	Default: **7200**
Default Length of Registration	Default: **3600**
Notify Mime Type	Default: **BLANK**
Time Between Mailbox Checks	Default: **BLANK**
Voicemail Extension	Default: **BLANK**
SIP Video Support	Default: **YES**
Record History of Default	Default: **N/A**
First disallow all Codecs	Default: **ALL**
Allow Codecs	Default: **G.711 ulaw, G.711 alaw, GSM**
Default Music on Hold	This will display whatever playlist is set to default on the PBX Setup=>Music On Hold page
Relax DTMF Handling	Default: **YES**
RTP Keep-Alive	Sends RTP packet when none received on active call for X seconds, 0 for disabled, which is the default.
RTP Timeout	Default is **BLANK**. Set to a value, in seconds, if you wish the PBX to end a call when no RTP traffic is detected for that long. Typically used in regards to lines that are not disconnecting correctly.
RTP Timeout on Hold	Default: **BLANK**
Trust Remote Party ID	Default: **N/A**
Send Remote Party ID	Default: **N/A**
Progress in Band	Default: **BLANK**
User Agent	Default: **BLANK**
Allow Redirect to Non-local SIP Address	Default: **N/A**
User = Phone	Default: **N/A**
DTMF Mode	Default: **AUTO**
Compact SIP Headers	Default: **N/A**
SIP Debug	Default: **N/A**
Subscriber Context	Default: **BLANK**
Notify Ringing	Default: **YES**
Qualify	Default: 8000
Generate Manager Events	Default: **YES**

NAT	Default: **YES**
Insecure	Default: **VERY**
Can Reinvite	Default: **N/A**
Cache Realtime Friends	Default: **YES**
Real Time Update	Default: **N/A**
Auto-Expire Friends	Default: **N/A**
Ignore Registration Expiration	Default: **N/A**
Allow External Domains	Default: **YES**

Edit Advanced SIP Networking Settings

STEPS:

1. From the **PBX Setup->SIP** page, click on the **Advanced** link.
2. The **Advanced SIP Networking Settings** page is displayed.
3. Set the **SIP Network** parameters base on your business requirements or what is recommended by IPitomy.

The default settings should not require any changes. If it is necessary for you to do so to meet your customer?s business requirements, we recommend that you contact IPitomy?s Technical Support for assistance..

1. Click the **.** button.
2. Click on the **Apply Changes** link at the top of the page to save the information and commit the changes to the database.