



VCOM and SIP

SIP (Session Initiation Protocol) is a widely adopted signaling protocol for Internet conferencing, telephony, presence, events notification, and instant messaging.

The VCOM Virtual Matrix includes an integrated SIP server that allows connection of SIP-enabled, VoIP peripherals directly into the Virtual Matrix from any network connection without the need for a PC. IP phones/phone systems, analog telephone adapters (ATAs), or IP gateways can be seamlessly connected into a network. The VCOM System Administration application can map SIP peripherals to Point-to-Points, Group Calls, and/or Party Lines in any complexity.

Typically SIP devices are used with Telephone IP PBX systems to implement a traditional Telephone system whereby one user dials the number of another user and waits for the recipient to answer. However, as a non-blocking intercom system VCOM is always on and the SIP implementation has been tailored to best compliment this functionality. As such each configured SIP client becomes an extension of a VCOM port by connecting the SIP device to a dedicated virtual SIP client associated with a VCOM port.

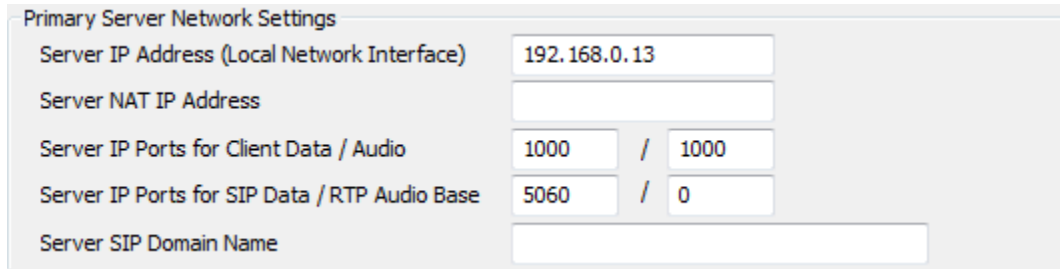
Below are a number of common applications in which SIP devices are used with VCOM:

- **Interoperating ProComm and IP Phone Systems:** ProComm Administrators can seamlessly bridge intercom with widely deployed IP Phone Systems that are prevalent in the government and defense markets;
- **Interfacing Phone Lines:** Analog and digital phone lines can be bridged into VCOM from any network connection using SIP-ready VoIP gateways;
- **Party Line Systems:** Users with simple intercommunication requirements that only need to monitor or talk on one or two channels have a wide array of user interface options that do not require a PC. Options include WiFi-enabled IP phones, iPhones (running softphone clients), and desktop speaker phones;
- **Hoot n' Holler:** VCOM is widely deployed as a next generation Hoot n' Holler system replacing expensive, private circuit leased lines, which can have a low quality of service. In this application most users have the simple requirement of monitoring and talking on a single party line and use an analog telephone attached to an ATA or IP phone, requiring no client-side PCs.

Configuring SIP Devices for VCOM

Softphones

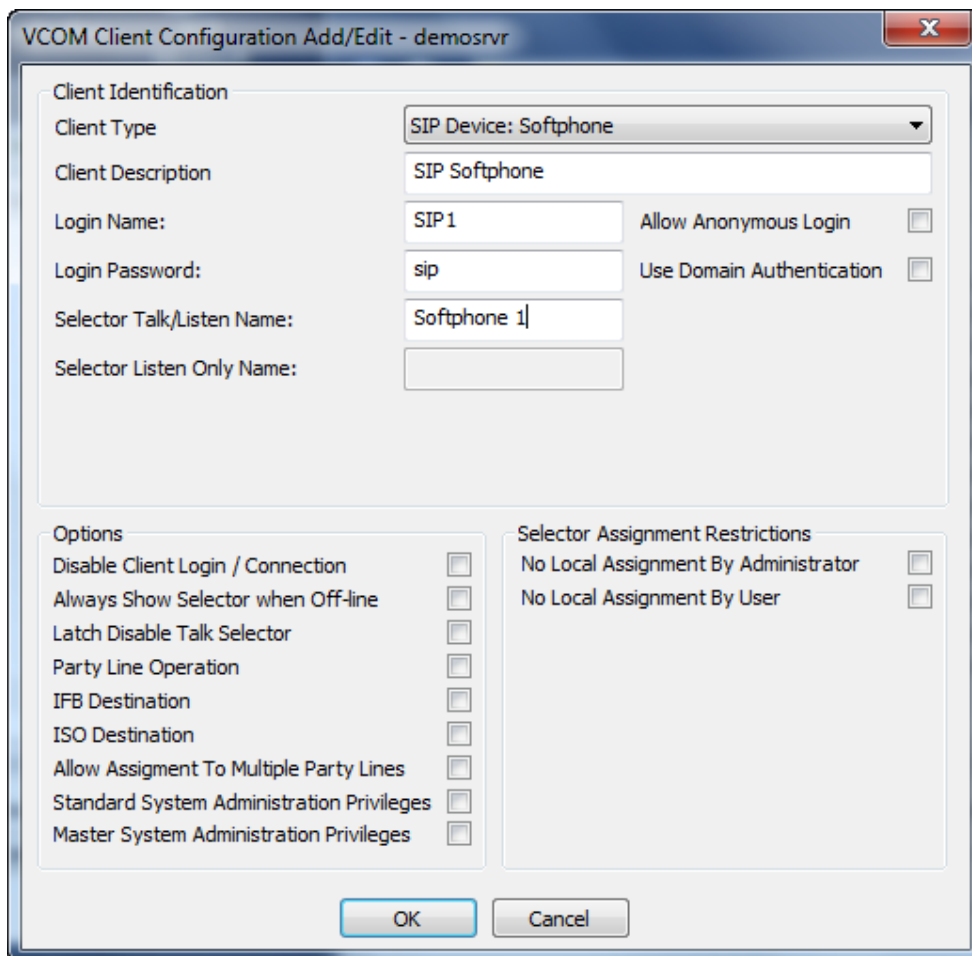
Typically softphones will have an account setting or configuration area and a properties tab under which you can point your softphone to a Domain. Find this area and use your VCOM Server IP Address as the Domain, Proxy, and the Registrar address. Unless changed in the VCOM System Administration the standard SIP default port number is 5060.



Primary Server Network Settings

| | | |
|---|--------------|--------|
| Server IP Address (Local Network Interface) | 192.168.0.13 | |
| Server NAT IP Address | | |
| Server IP Ports for Client Data / Audio | 1000 | / 1000 |
| Server IP Ports for SIP Data / RTP Audio Base | 5060 | / 0 |
| Server SIP Domain Name | | |

Set the Display Name(Selector Talk/Listen Name), User Name, and Password to match the settings programmed in the 'Client Configuration' section of the VCOM System Administration for any given softphone.



VCOM Client Configuration Add/Edit - demosrvr

Client Identification

Client Type: SIP Device: Softphone

Client Description: SIP Softphone

Login Name: SIP1 Allow Anonymous Login

Login Password: sip Use Domain Authentication

Selector Talk/Listen Name: Softphone 1

Selector Listen Only Name:

Options

- Disable Client Login / Connection
- Always Show Selector when Off-line
- Latch Disable Talk Selector
- Party Line Operation
- IFB Destination
- ISO Destination
- Allow Assignment To Multiple Party Lines
- Standard System Administration Privileges
- Master System Administration Privileges

Selector Assignment Restrictions

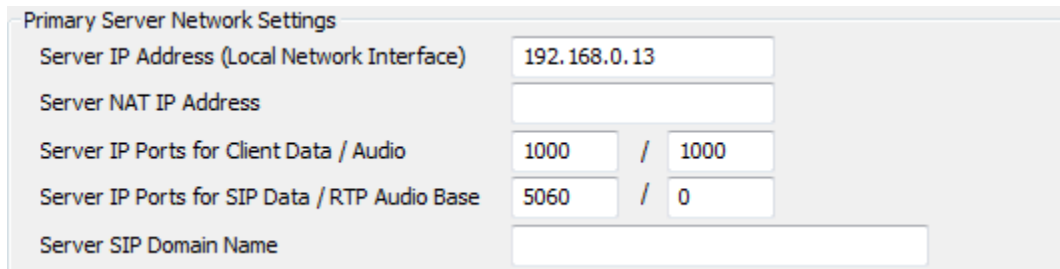
- No Local Assignment By Administrator
- No Local Assignment By User

OK Cancel

Hardphones and ATAs

SIP hardphones and ATAs are configured via a web interface. You need to begin by determining the device IP Address typically by using Internet Explorer to access your router's web interface and view the list of attached devices.

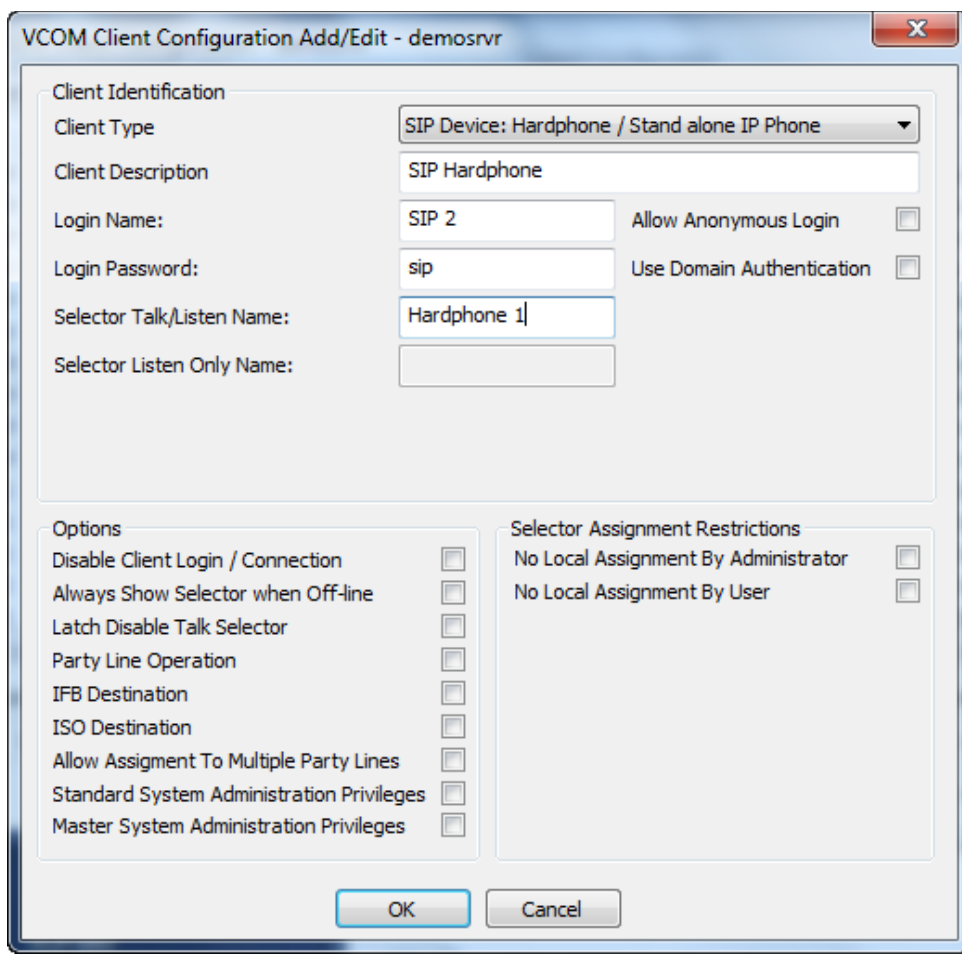
Once you determine your device's IP Address input it into a web browser to access the device's configuration page. Use your VCOM Server IP Address as the Domain, Proxy, and the Registrar address. Unless changed in the VCOM System Administration the standard SIP default port number is 5060.



Primary Server Network Settings

| | |
|---|--------------|
| Server IP Address (Local Network Interface) | 192.168.0.13 |
| Server NAT IP Address | |
| Server IP Ports for Client Data / Audio | 1000 / 1000 |
| Server IP Ports for SIP Data / RTP Audio Base | 5060 / 0 |
| Server SIP Domain Name | |

Set the Display Name(Selector Talk/Listen Name), User Name, and Password to match the settings programmed in the 'Client Configuration' section of the VCOM System Administration for any given hardphone or ATA.



VCOM Client Configuration Add/Edit - demosrvr

Client Identification

Client Type: SIP Device: Hardphone / Stand alone IP Phone

Client Description: SIP Hardphone

Login Name: SIP 2 Allow Anonymous Login

Login Password: sip Use Domain Authentication

Selector Talk/Listen Name: Hardphone 1

Selector Listen Only Name:

Options

- Disable Client Login / Connection
- Always Show Selector when Off-line
- Latch Disable Talk Selector
- Party Line Operation
- IFB Destination
- ISO Destination
- Allow Assignment To Multiple Party Lines
- Standard System Administration Privileges
- Master System Administration Privileges

Selector Assignment Restrictions

- No Local Assignment By Administrator
- No Local Assignment By User

OK Cancel