SIPSTATION

User Guide

Schmooze®

Schmooze Com Inc.

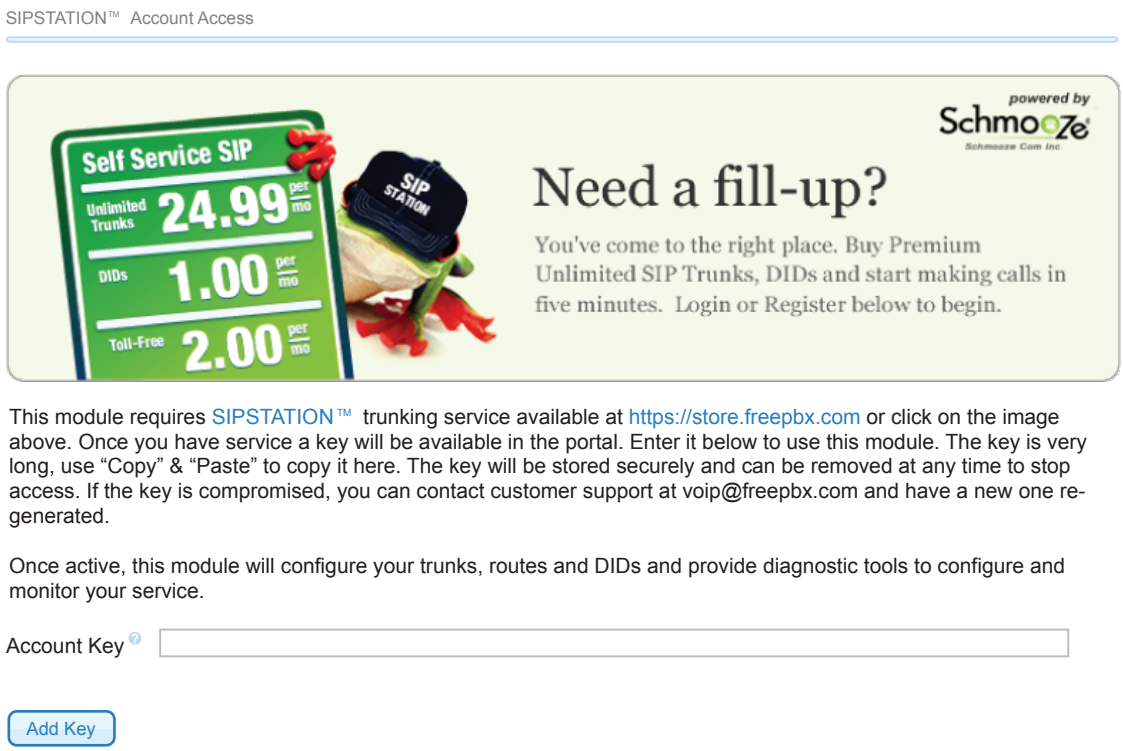
Chapters

Overview

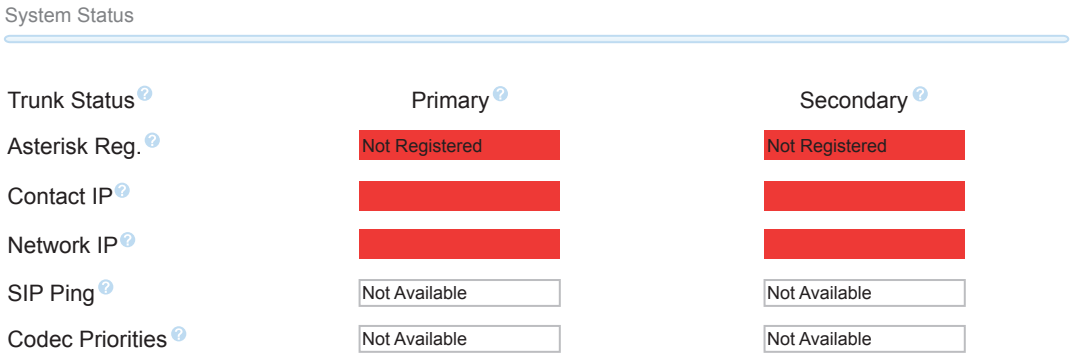
The SIPSTATION module, when combined with a SIPSTATION SIP Trunk account, provides a quick and easy method for getting a SIP Trunk online and ready to go fast. In general, there is not much to adjust here besides pointing your new DIDs or adding an area code for 7-digit dialing.

Logging In & Adding a Key

- Log into the SIPSTATION module and you should see a screen like this. Here you would enter the **Account Key** (provided when you create a SIP Station account, get a DID and assign E911 data). Simply copy and paste your key into the **Account Key** field and press "Add Key."



- Once you have entered your key and before you have pressed the "Apply Config" button, your screen may look like this.



- Once you press the “Apply Config” button, your screen should look like this. Note that you would normally see the entire external IP Address.

System Status

Trunk Status	Primary	Secondary
Asterisk Reg.	Registered	Registered
Contact IP	63.229.xx.xx	63.229.xx.xx
Network IP	63.229.xx.xx	63.229.xx.xx
SIP Ping	OK (99 ms)	OK (156 ms)
Codec Priorities	ulaw | g729	ulaw | g729

- You will now also see several new buttons.
  - **Remove Key**- Pressing this button will remove the account key, but leave the trunks intact.
  - **Remove Key & Delete Trunks**- Pressing this button will remove the account key and remove the trunks.
  - **Update Account Info**- If for instance, you were to log in to the portal and add a DID, you would want to come back to the SIPSTATION module and update the account information by pressing this button and then the “Apply Config” button.
  - **Run Firewall Test**- Running the firewall test will tell you if you have port-forwarding setup correctly on your firewall. Note that it is recommended that you do setup port forwarding, as it will provide more reliable SIP Trunking.

SIPSTATION™ Account Access

To disable account access, click *Remove Key*. To update account information, click *Update Account Info*. If port forwarding is configured on your firewall/router, you can test it with the *Run Firewall Test* button. Port forwarding can provide more reliable service and better quality and we recommend setting it up. The test sends a packet to an unused Asterisk RTP port at your WAN address and results in a PASS if the packet is properly received.

Remove Key | Remove Key & Delete Trunks | Update Account Info | Run Firewall Test

  - Your firewall test may result in the following being displayed.

FIREWALL TEST WARNING

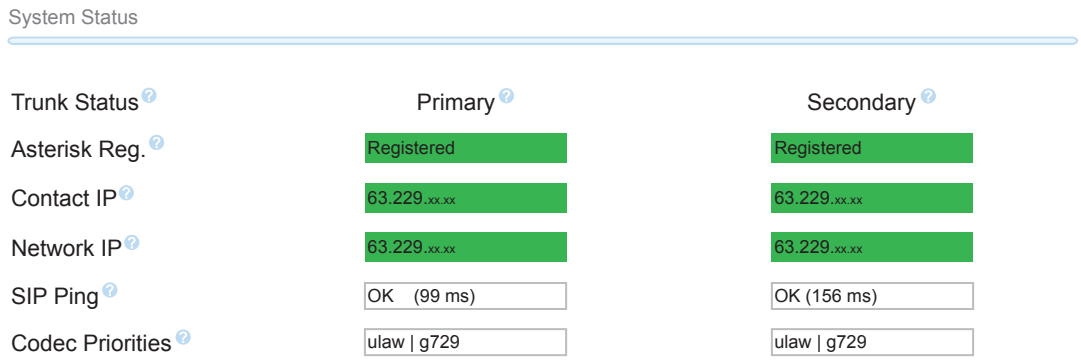
- The test timed out which means your firewall is probably configured wrong. If subsequent tests fail, check your port forwarding on the firewall.

  - After you correct your firewall issues and rerun the firewall test, you should see the following.

Firewall Test Status: PASS External IP: 63.229.xx.xx

System Status

- **Asterisk Reg**- Displays the Asterisk Registration status.
- **Contact IP**- This is the contact IP as seen on the gateway and provides warnings if errors are detected. These should be your external IP as seen on the WAN side of your router. If they are not, or if they do not match your Network IP, you should configure your NAT settings in the Asterisk SIP Settings module or in sip_nat.conf (if not using that module).
- **Network IP**- This is the network IP as seen on the gateway and provides warnings if errors are detected. These should be your external IP as seen on the WAN side of your router. If they are not, or if they do not match your Network IP, you should configure your NAT settings in the Asterisk SIP Settings module or in sip_nat.conf (if not using that module).
- **SIP Ping**- Round-trip signaling delay to SIP server as determined by the Asterisk "qualify" command. This is signaling delay only. The voice connections (RTP media streams) are routed from your system to the closest POP (Point Of Presence) where the call enters the PSTN. This assures the optimal minimum latency, but can't be reported because it is dependent on each call.
- **Codec Priorities**- Codec Priority Asterisk reports for these trunks. This is filtered to show codecs supported by the gateways. The codecs can be edited on the trunk page to make changes to priority or available codecs.

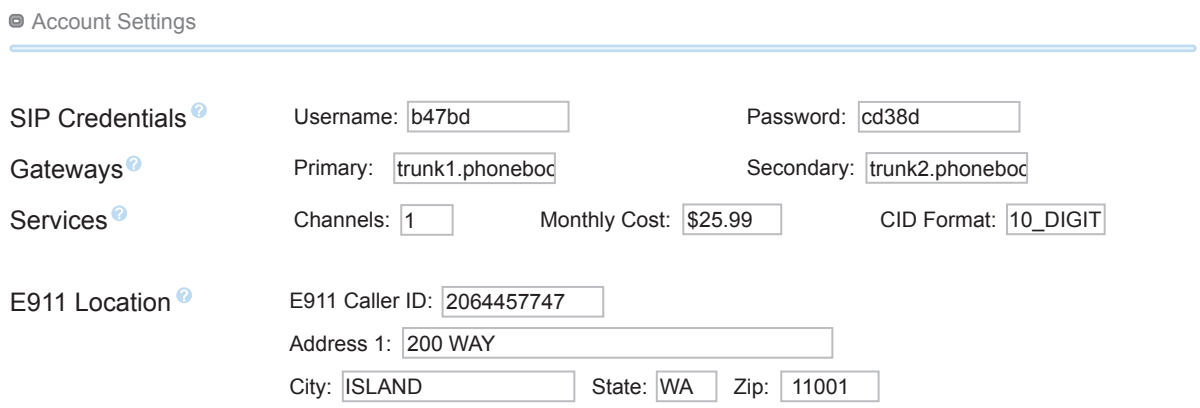


Account Settings

- **SIP Credentials**- SIP Trunk username and password.
- **Gateways**- Primary and Secondary trunks for SIP traffic. These are automatically configured.
- **Services**- The number of concurrent calls that have been purchased and are configured for your service. Also referred to as trunks and are similar to the number of PRI lines or POTS lines in a traditional telco environment. Your monthly charges, including all trunks and unlimited trunks. The caller ID number can be configured in the portal (**https://store.freepbx.com**) to send either

standard 10 digit NPA (for North American Numbers) or the E164 standard, which is +1NXXNXXXXXX for NPA numbers and +NNXXXX.. for countries where +NN is the county code.

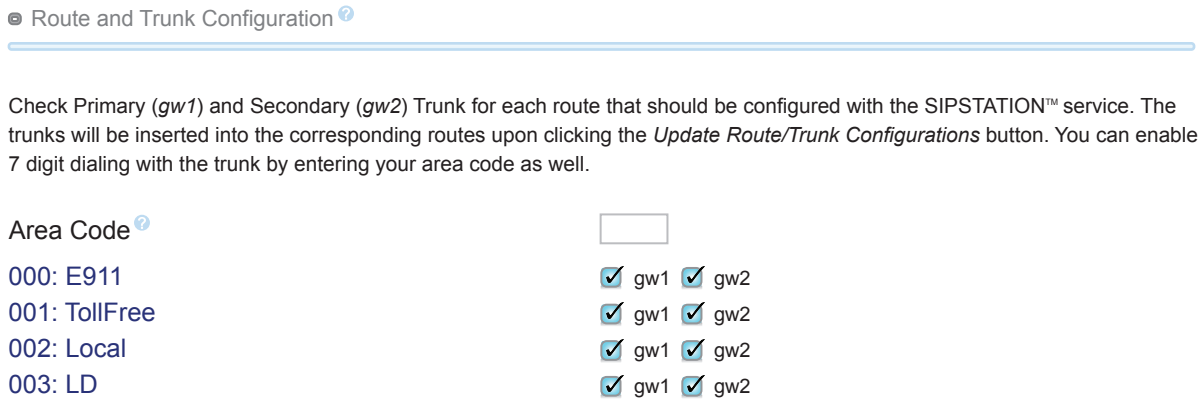
- **E911 Location**- This displays the E911 information as set in the portal (**https://store.freepbx.com**). It is vitally important that this information is correct.



Route & Trunk Configuration

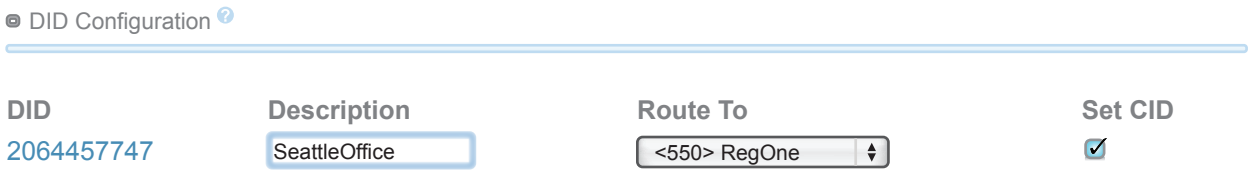
- **Area Code**- You may enter an area code if you want your trunks to allow 7-digit dialing. You must ensure your outbound routes are set up for 7-digit dialing to use.

- **Routes**- By using the gw1 and gw2 toggles, you may set which outbound routes use these trunks. Press the "Update Route/Trunk Configurations" button after making any changes. Press the "Apply Config" button when all changes are completed.



DID Configuration

- Here you will see a list of your DIDs, a description (if desired,) where the DID is routed to and whether or not you want the Extension CID to be set as the DID (if to an extension). Press the "Update DID Configurations" button after making any changes. Press the "Apply Config" button when all changes are completed.



Recap

- The SIPSTATION module provides an easy and almost completely automatic method of setting up premium SIP Trunks on your PBX. While most settings are set in the portal, (**https://store.freepbx.com**) you may change the outbound routing and where DIDs are routed.

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