SYMETRIX SOLUTIONS: TECH TIP

June 2019

Symetrix

## Tech Tip: How to Configure the Symetrix VoIP Card to be Used with a RingCentral PBX

Ring Central is a provider of VoIP services that is increasingly popular with business customers for telephone conference systems. As a cloud-based provider, choosing RingCentral for VoIP service can reduce the cost of a customer's VoIP infrastructure by moving the Private Branch Exchange (PBX) off-site requiring less physical hardware at the customer's location as well as less administration by the customer's IT department. In the interest of this increased demand, and because of some subtle nuances that are involved with integrating Symetrix VoIP cards into a RingCentral environment, this Tech Tip is written to assist with that process.
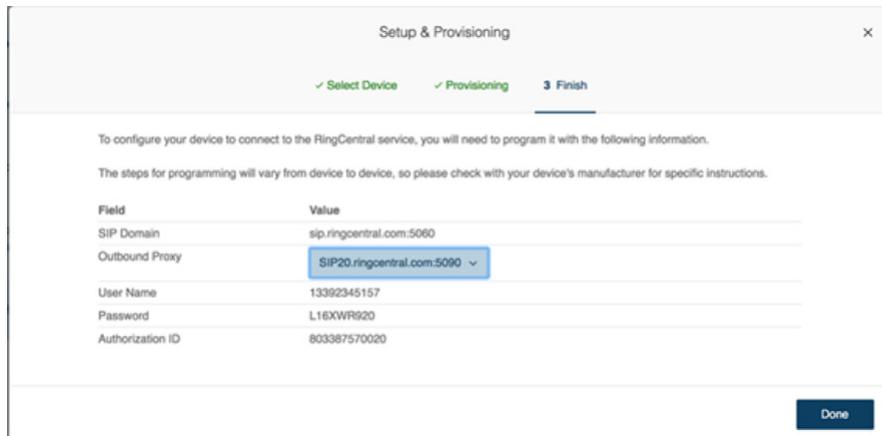

This configuration is an example of what Ring Central sends to customers. As with all VoIP providers, information like this is necessary to ensure proper communication between the Symetrix VoIP card and the server that handles all registration and calls. This information is used to fill out the VoIP web administration for the Symetrix system.
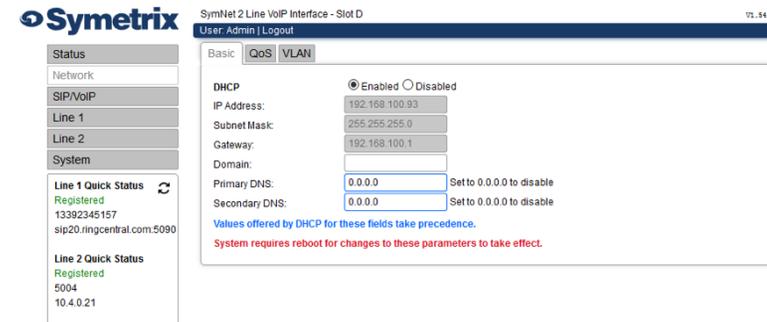

1. The Network tab is dependent on the physical network the VoIP card is connected to. With a DHCP server in place, this tab will auto populate as necessary. If the IT department has a different network configuration, this is where that information is placed. This should not impact the interaction with the RingCentral system.


2. Registration begins with the Line tab. The minimum requirements for this tab are the user name and domain name. The user name will match the user name tab in the RingCentral information. The domain should be the actual SIP domain without the port extension (:5060). This domain is where the call audio passes through.
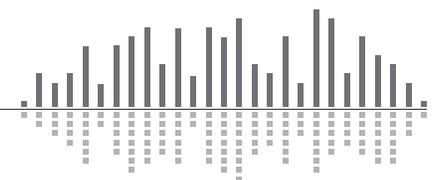
Tech Tip: How to Configure the Symetrix VoIP Card to be Used with a RingCentral PBX
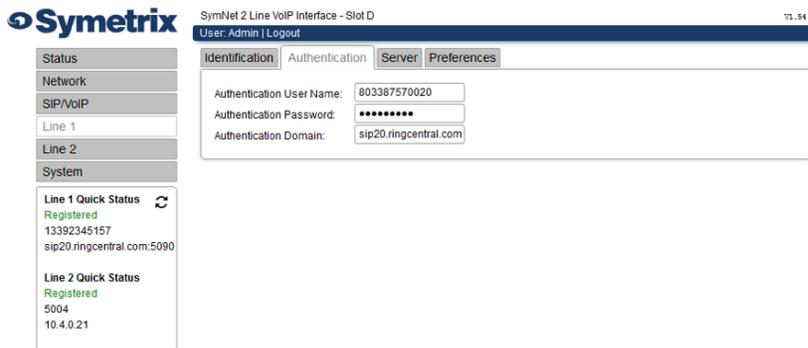
3. The authentication tab is next. The authentication user name should match the authorization ID. The password will be the included password. The authentication domain will be the outbound proxy domain including the port extension.
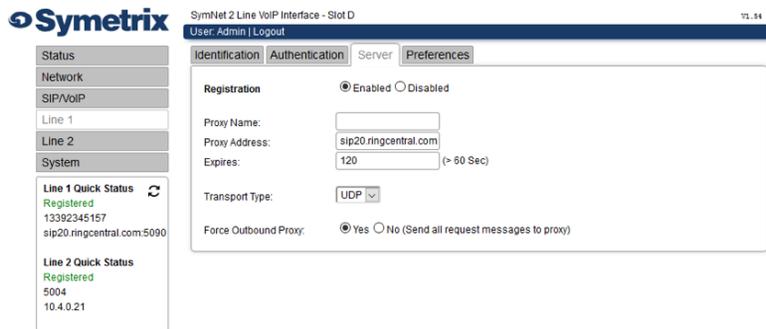
4. In the server tab, force outbound proxy must be turned on. The proxy address should match the proxy that RingCentral provides. This differs from many VoIP providers because the authorization domain is different than the audio transport domain. A different server is used to handle registration than calls.

5. Once this information is filled in, the line should register and remain registered. From this point, the VoIP system should be fully integrated and working as expected.

One final important note:

The SIP Application Layer Gateway (SIP ALG) is a setting that can be enabled on outbound network traffic that will travel across the internet. For certain providers, SIP ALG can improve call quality. With other providers, this setting can do the opposite to call quality. With RingCentral, enabling SIP ALG will cause important packets to drop, and this will end up resulting in actual calls being disconnected. It is important to disable SIP ALG on outbound traffic. This is a setting that must be applied locally on the network's internet traffic. Neither Symetrix nor RingCentral can disable SIP ALG, so it is important to be in clear communication with the local IT administrator about keeping SIP ALG disabled.