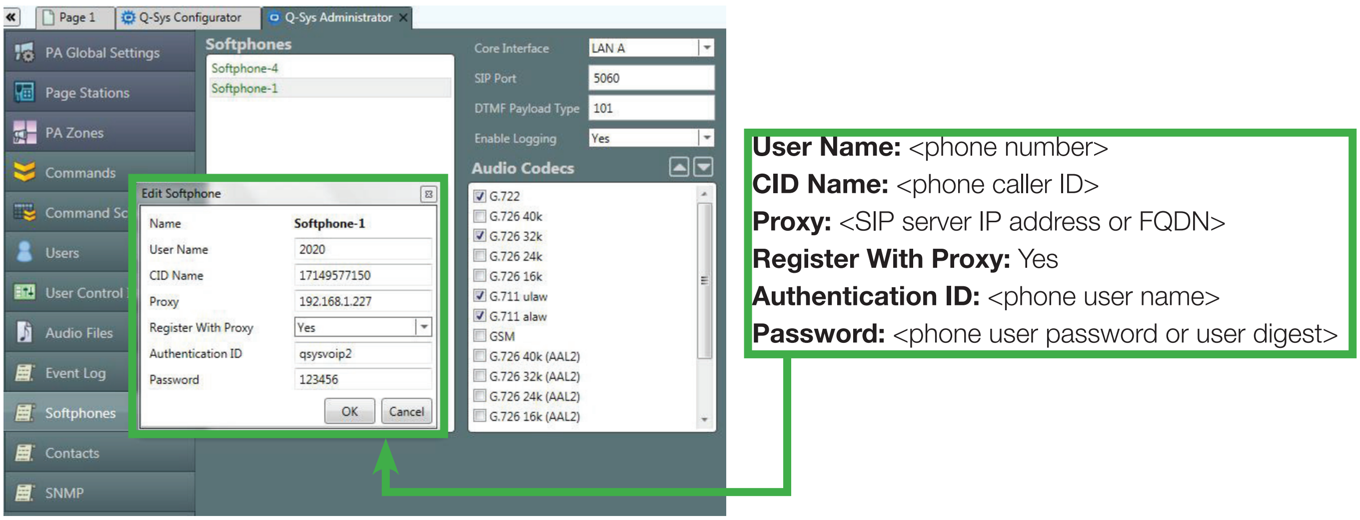
Technical **Notes**

Q-SYS Platform



Best practices for setting up VoIP

This technical note explains the workflow for setting up VoIP on a Q-SYS network. To complete this procedure, you should have this information in advance:

- The phone number and Caller ID name of the softphone.
- The IP address of the proxy (i.e., the SIP server or the fully qualified domain name, or FQDN).
- The authentication ID and password for logging in to the proxy.

Please note:

- The Q-SYS softphone requires SIPv2 early offer.
- The domain name lookup requires that the Q-SYS Core unit be configured with a domain name server (DNS).

The Q-SYS softphone downloads no configuration information from the call controller (the SIP/proxy server); therefore in a clustered environment the softphone must be configured to the correct server.

Procedure

Repeat this procedure for each softphone to be configured.

1. In Q-SYS Administrator, select **Softphones** and then double click the softphone to be configured.
2. In the **Edit Softphone** box, enter the parameters described above:
   - **User Name** (phone number)
   - **CID Name** (Caller ID)
   - **Proxy** (the SIP server's IP address or FQDN)
   - **Register With Proxy** (the **Authentication ID** and **Password** fields will not appear until you select **Yes**)
   - **Authentication ID** (the user name for the phone to register with the proxy)
   - **Password** (the user password or user digest for the phone to register with the proxy)
3. Click **OK**.

Continued on next page →

Verifying with Wireshark

Below is a Wireshark capture of a successful incoming call, from start to finish. The flow starts with an incoming SIP INVITE and end with an acknowledgement of a BYE request. The SIP Request Codes and the three-digit SIP Response Codes allow you to verify correct operation or troubleshoot any problems.

1. SIP Request INVITE—Notification from an incoming call; invites the client to participate in a call session.
2. SIP Response 100—Trying.
3. SIP Response 180—Ringing. The destination User Agent has received the INVITE and is alerting the user.
4. SIP Response 200—OK. The request was successful.
5. SIP Request INVITE ACK—Confirms that the client has received a response to the INVITE request.
6. The RTR packets are the data streams of the actual phone call.
7. SIP Request BYE—Either the caller or the recipient can terminate the call.
8. SIP Response 200—OK. The request to terminate the call was successful.

SIP Response Codes in the ranges 1xx or 2xx are normal. Codes of 4xx would indicate a client failure; 5xx indicates a server failure; and 6xx indicates a global failure.