USER GUIDE

# 3CX Technical Application
# (For Fusion Static Configuration)

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When configuring the 3CX PBX (Version 15) for Fusion Connect's Static SIP Trunk, you will receive a technical welcome letter with the following information:

**Image 1: Welcome Letter Account Information**

| Traffic Type | IP Addresses | Domain Name | Protocol | Port Range |
|---|---|---|---|---|
| SIP Static | 216.86.41.167 | peer3.thevoicemanager.com | UDP and TCP | 5060 |

For your static configured SIP Trunk, this will only pertain to the domain and IP address (highlighted in yellow above). This also shows the ports that need to be opened via an ACL, Whitelist, or Port Forwarding statement on your firewall. The list shown below shows the default installation environment:

| Protocol | Port | Description |
|---|---|---|
| TCP | 5000 or 80 | HTTP port of Web Server (only accessible from LAN). This port can be configured. |
| TCP | 5001 or 443 | HTTPs port of Web Server. This port can be configured. |
| TCP | 5015 | This port is used for the online Web-Based installer wizard (NOT 3CX configuration command line tool) only during the installation process. |
| UDP & TCP | 5060 | Default SIP Port |
| TCP | 5061 | Default SIP TLS Port |
| UDP | 9000-9500 | Default 3CX Media Server |

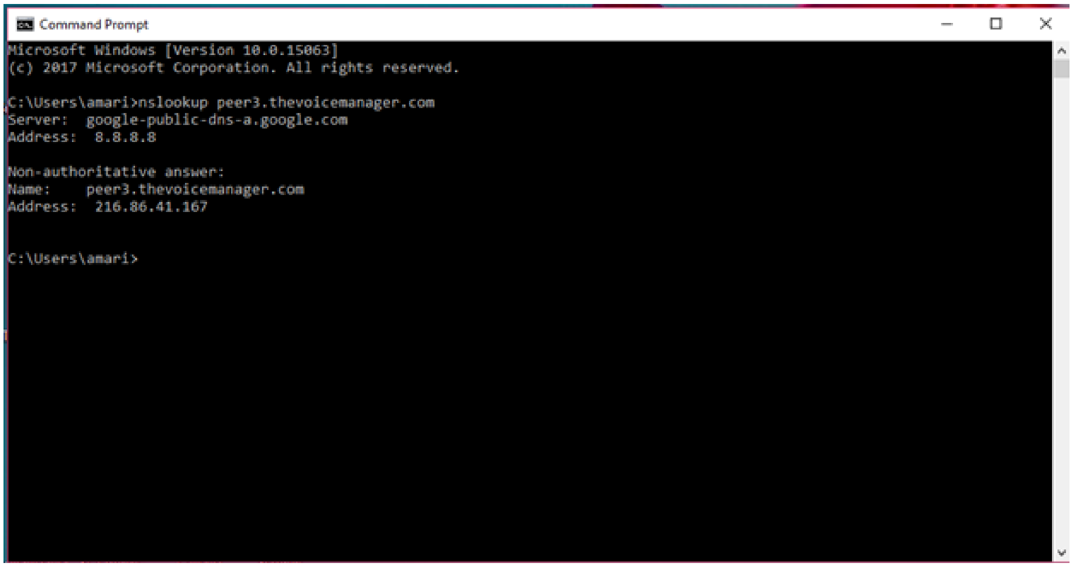
## Service Records

Service Records (SRV) are a form of the Domain Name System (DNS) record. SRV's hold information of where to submit requests for services offered by the domain itself. In the case of your Fusion Connect Static SIP Trunk, the IP address, port number, and preferences for sending SIP calls over UDP, TCP, and TLS to the appropriate SIP Server are shown on the previous page.

Most customers with the Fusion Connect Static SIP Trunk would like to ensure that the DNS entries are functioning properly by verifying the IP address. This is done in Windows OS by using the *nslookup* command and following instructions based on the Windows 7 terminal:

1. Click the Start button.
2. Within search, type "cmd".
3. Select cmd from the list of programs.
4. Type "nslookup peer3.thevoicemanager.com" (or any domain name provided by Fusion Connect in the welcome letter). See the example image below to obtain the IP address:

**Image 2: Domain Lookup Example**




This can also be done in an OS X environment with the "Terminal.app" application and performing the same command. You can also accomplish this in a UNIX or LINUX environment by using the "dig" command.

## Preferred Codecs

With your Fusion Connect Static Trunk, you are allowed to use the following codecs to meet your bandwidth requirements and limitations:

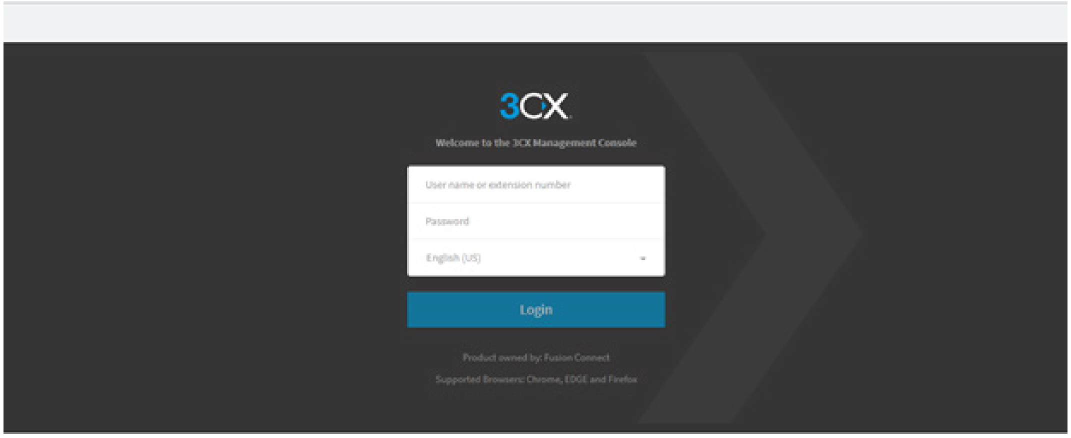
- G711u (or labeled as PCMU) - Which consumes approximately 87.2 Kbps of bandwidth per call over SIP
- G729 - Which consumes approximately 31.2 Kbps of bandwidth per call over SIP

Please be advised that the G711 codec is a higher quality codec than G729. If you have limited bandwidth for your VOIP environment, we advise using the G729 codec to prevent any quality issues.

## Configuring the 3CX Phone System

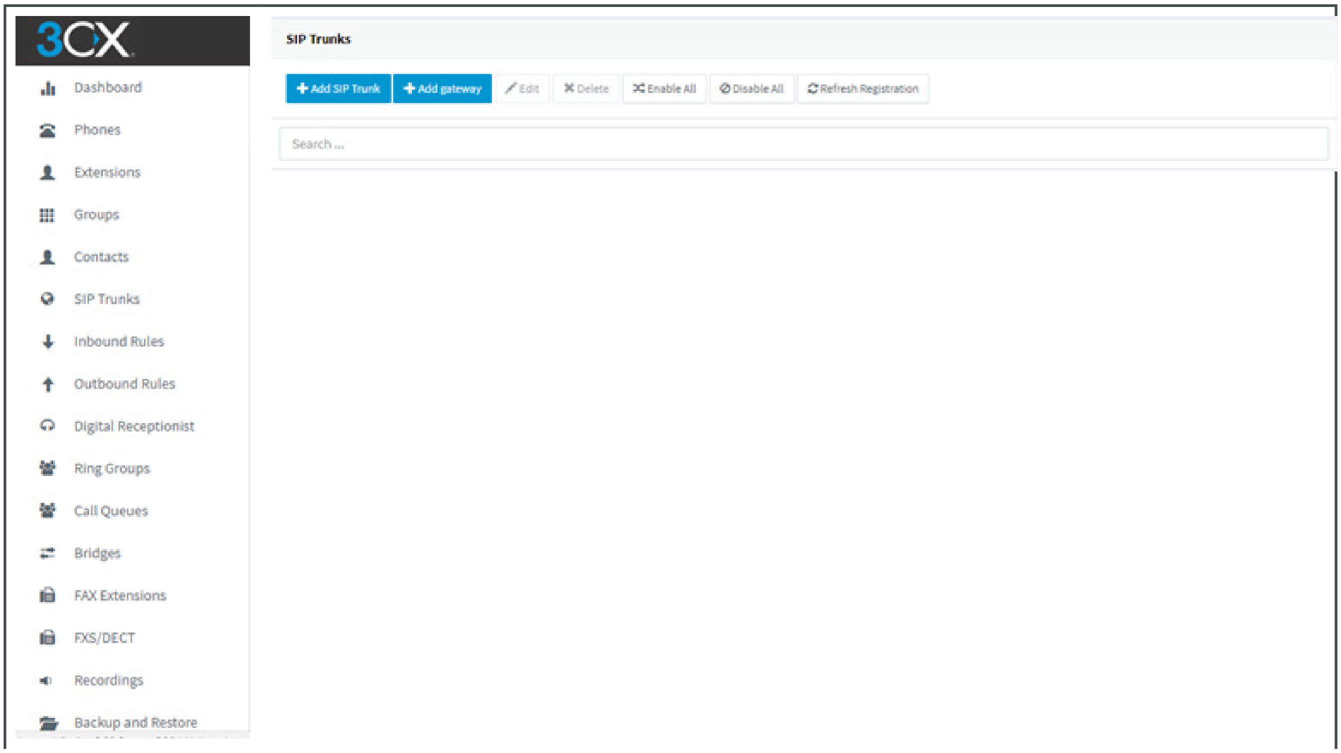
The section below explains how to configure the Fusion Connect SIP Trunk with a new installation of the 3CX Phone System. Begin with the installation and complete the setup wizard. As shown in the image below, chose your doman for the management console as https://your.domain.3cx.us:5001 or port 5000 if you are only using “http”.

**Image 3: Management Console**



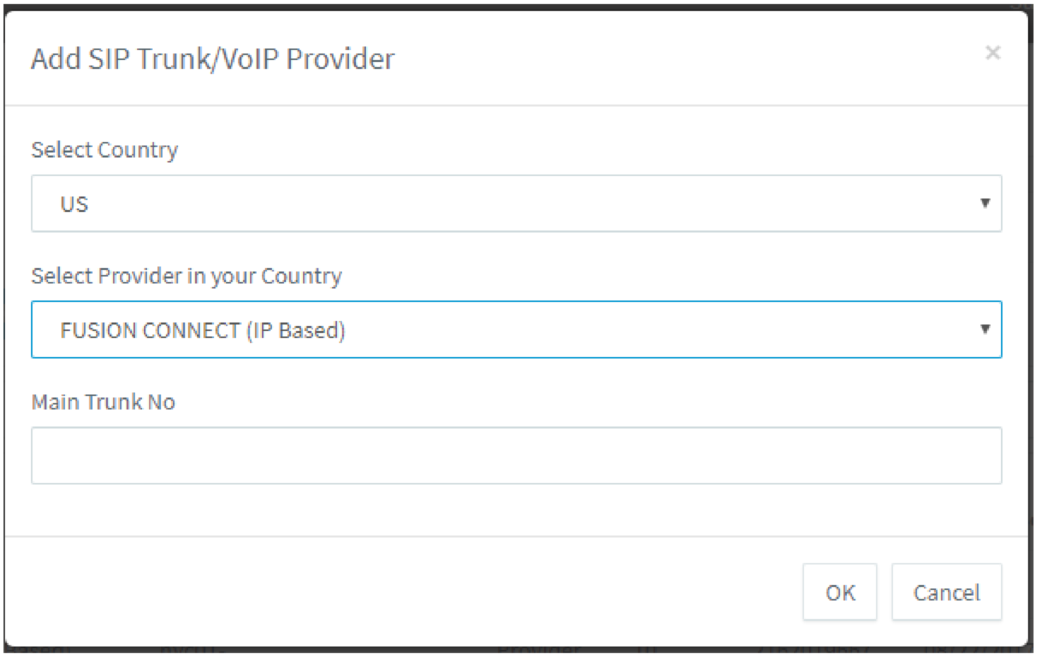
• After you log in to the management console with the credentials you've created, navigate to the SIP Trunks section (in the side panel) and add SIP Trunks as shown below:

**Images 4 & 5: 3CX Side Panel and SIP Trunk Addition**



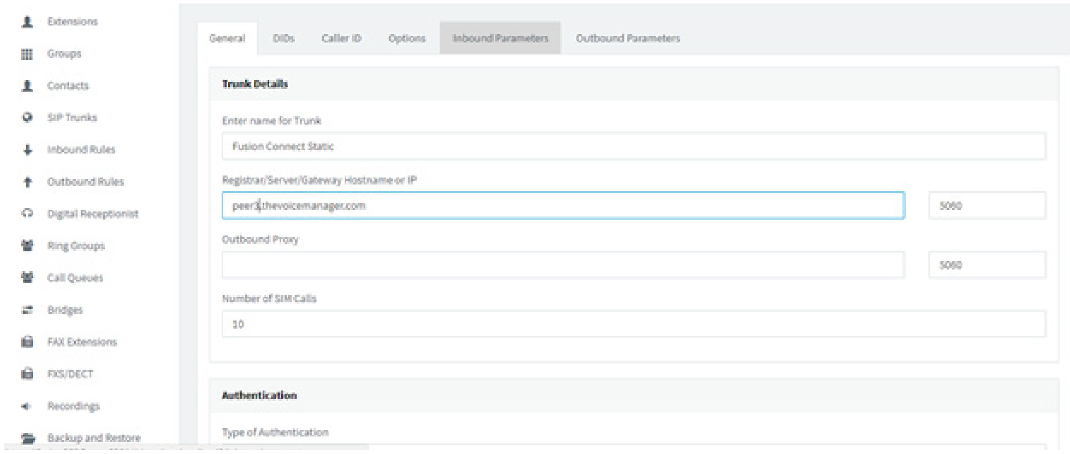
• After selecting the “Add SIP Trunk” option, you will then see the following screen (see image below). For the Country, select “US” and for “Provider” select “FUSION CONNECT (IP Based)”.

**Image 6: 3CX Provider Selection**




• In the “Main Trunk No” field enter the Main Number/BTN (Billing Telephone Number) provided in your technical welcome letter. After selecting ‘OK’, you will then be brought to the screen shown below.

**Image 7: General Trunk Configuration**



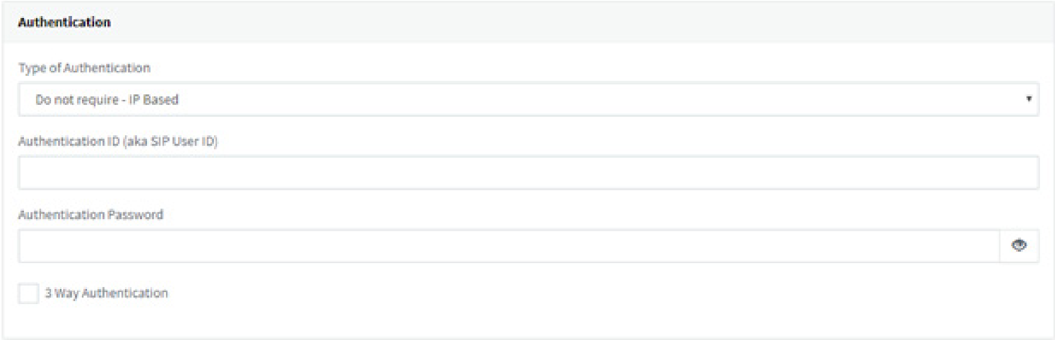
• After naming the trunk, the "Registrar/Server/Gateway Hostname or IP" entry will be the hostname or IP address as outlined in the technical welcome letter: "peer3.thevoicemanager.com" or the IP address 216.86.41.167.

• The "Outbound Proxy" field does not require an entry of a Static Trunk Configuration.

• The number simultaneous (SIM in the image) will be the Concurrent Call Cessions (CCS). This number is referenced in the technical welcome letter. For example, our trunk is capable of making 10 calls.

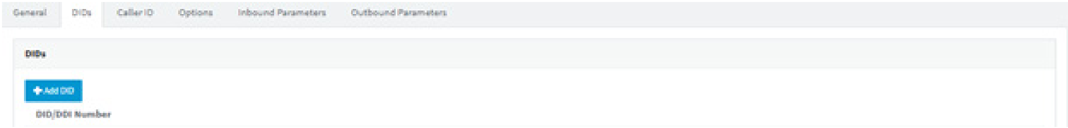
• The SIP port does not need to be changed uness it is required by your network configuration.

**Image 8: Authentication**



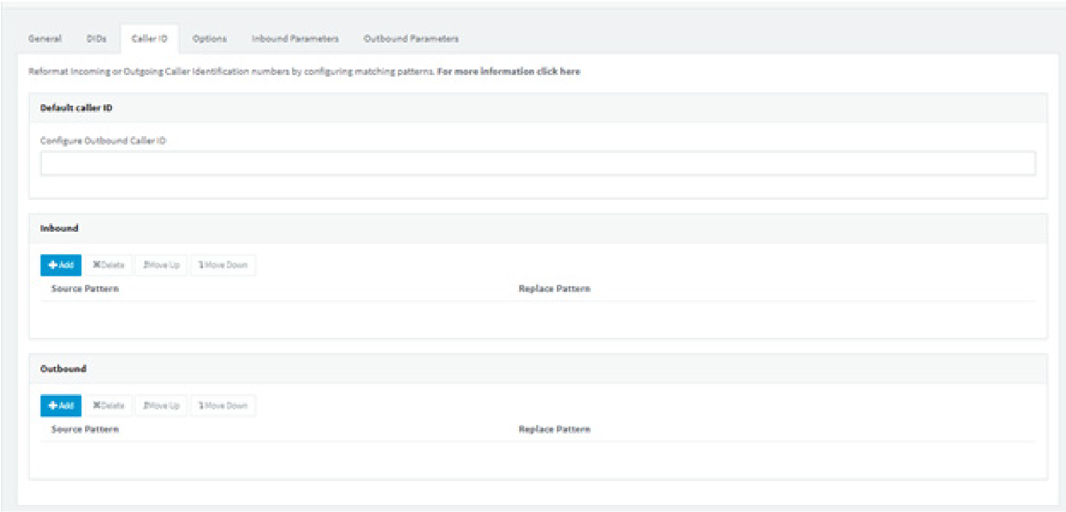

• The SIP Trunk is not required and does not need to be changed as shown in the image above.

**Image 9: DIDs**



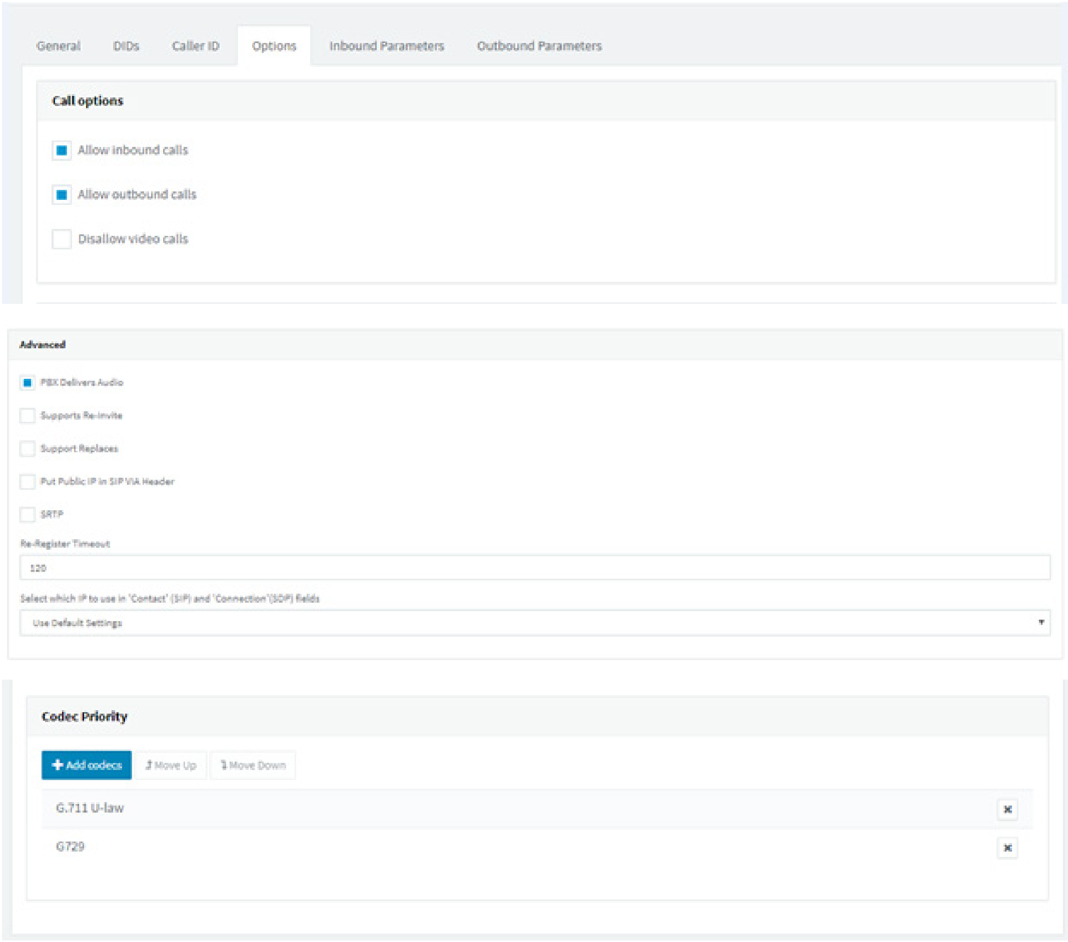

• To add a DID to your 3CX, select "Add DID" and enter the 10 digit DIDs located in the technical welcome letter. Direct the DID to the desired endpoint or extension.

**Image 10: Caller ID**




• As the image shows, the dialing pattern for incoming and outgoing calls can be configured for the selected destination for its outbound caller ID (number) and incoming caller ID format when reaching its endpoint or extension.

**Images 11, 12, and 13: Options and Codecs**



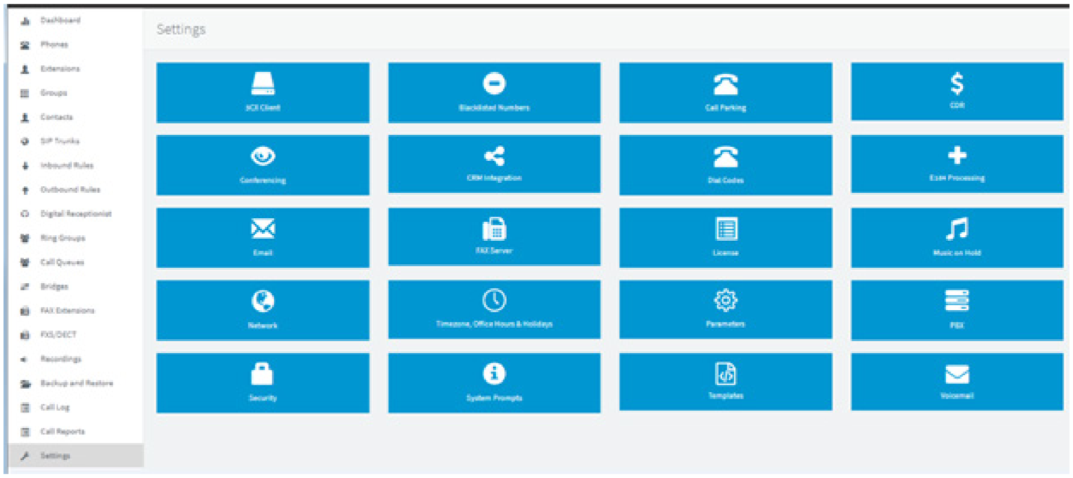

- As the images show, the “Call Options” section can be left as the default as Fusion Connect does not support video calling.
- The “Advanced” section can also be left as the default, except for the “IP to use in Contact and SDP” option. The “Use this IP address” option must be selected and the 3CX’s Phone System External IP address will need to be entered.
- The “Codec Priority” section allows the priority for the supported codecs (G711 and G729) to be configured.

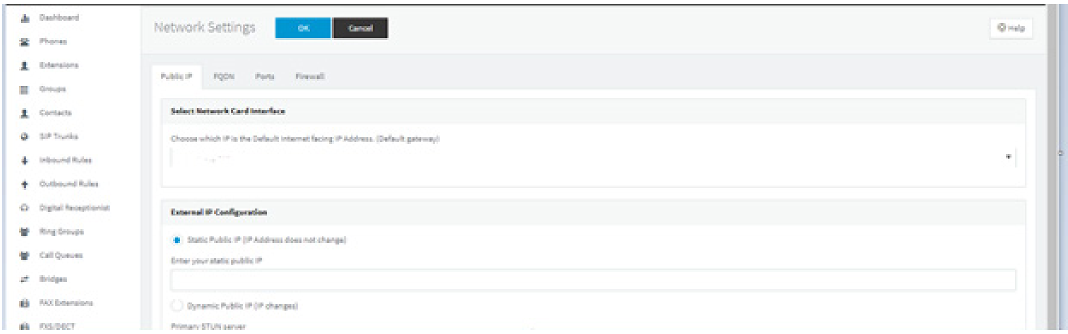
## Image 14, 15, and 16: Outbound Parameters

| | |
|---|---|
| Remote Party ID - Calling Party : Display Name | "OutboundCallerId" Outbound caller Id taken from Exte |
| Remote Party ID - Calling Party : User Part | "OutboundCallerId" Outbound caller Id taken from Exte |
| Remote Party ID - Calling Party : Host Part | "GWHostPort" gateway/provider host/port |
| P-Asserted Identity : Display Name | Leave default value |
| P-Asserted Identity : User Part | Leave default value |
| P-Asserted Identity : Host Part | Leave default value |
| P-Preferred Identity : Display Name | Leave default value |
| P-Preferred Identity : User Part | Leave default value |
| P-Preferred Identity : Host Part | Leave default value |
| P-Called-Party-ID : Display Name | Leave default value |
| P-Called-Party-ID : User Part | Leave default value |
| P-Called-Party-ID : Host Part | Leave default value |

• To change the outbound caller ID for individual extensions, please leave the options as the default (shown in the images above) and configure within the individual extensions.

**Images 17 and 18: Network and Stun Servers**




• By default, the Fusion Connect IP Based Template stun server is disabled. Navigate to the network section of the 3CX as shown above to ensure that the "Static Public IP" section is selected with the static IP of the 3CX PBX.