

3CX Technical Application
(For Fusion Static Configuration)
09/20/2017



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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

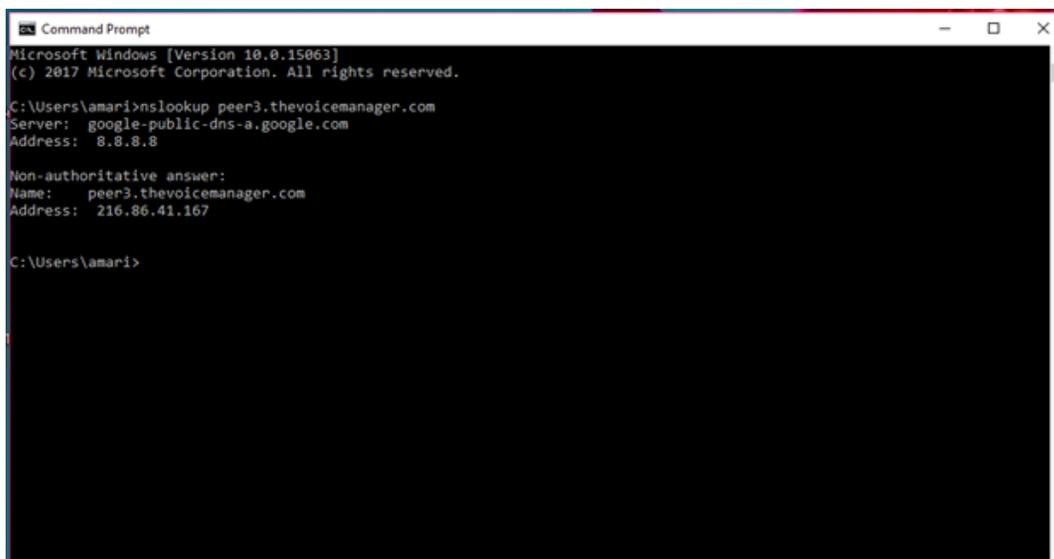
Service Records

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



```
Command Prompt
Microsoft Windows [Version 10.0.15063]
(c) 2017 Microsoft Corporation. All rights reserved.

C:\Users\amari>nslookup peer3.thevoicemanager.com
Server: google-public-dns-a.google.com
Address: 8.8.8.8

Non-authoritative answer:
Name: peer3.thevoicemanager.com
Address: 216.86.41.167

C:\Users\amari>
```

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:

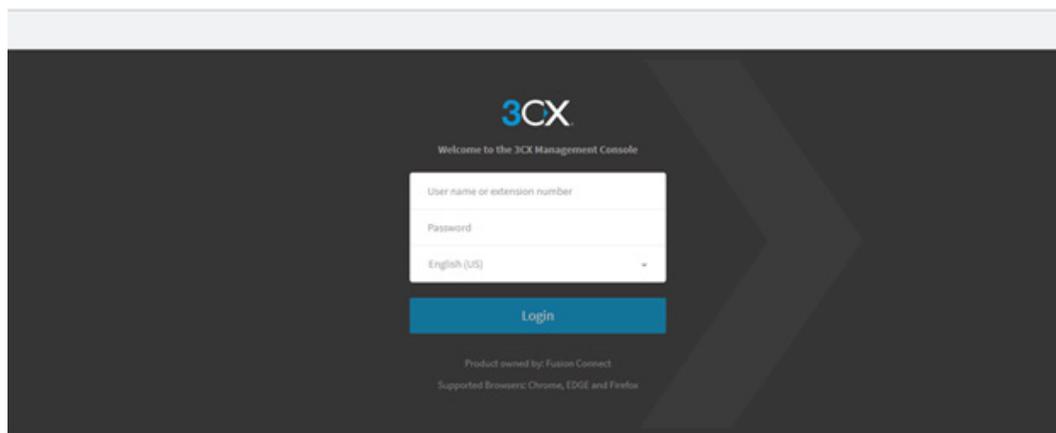
- G711u (or labeled as PCMU) - Which consumes approximately 87.2 Kbps of bandwidth per call over SIP
- G729 - Which consumes approximately 31.2 Kbps of bandwidth per call over SIP

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.

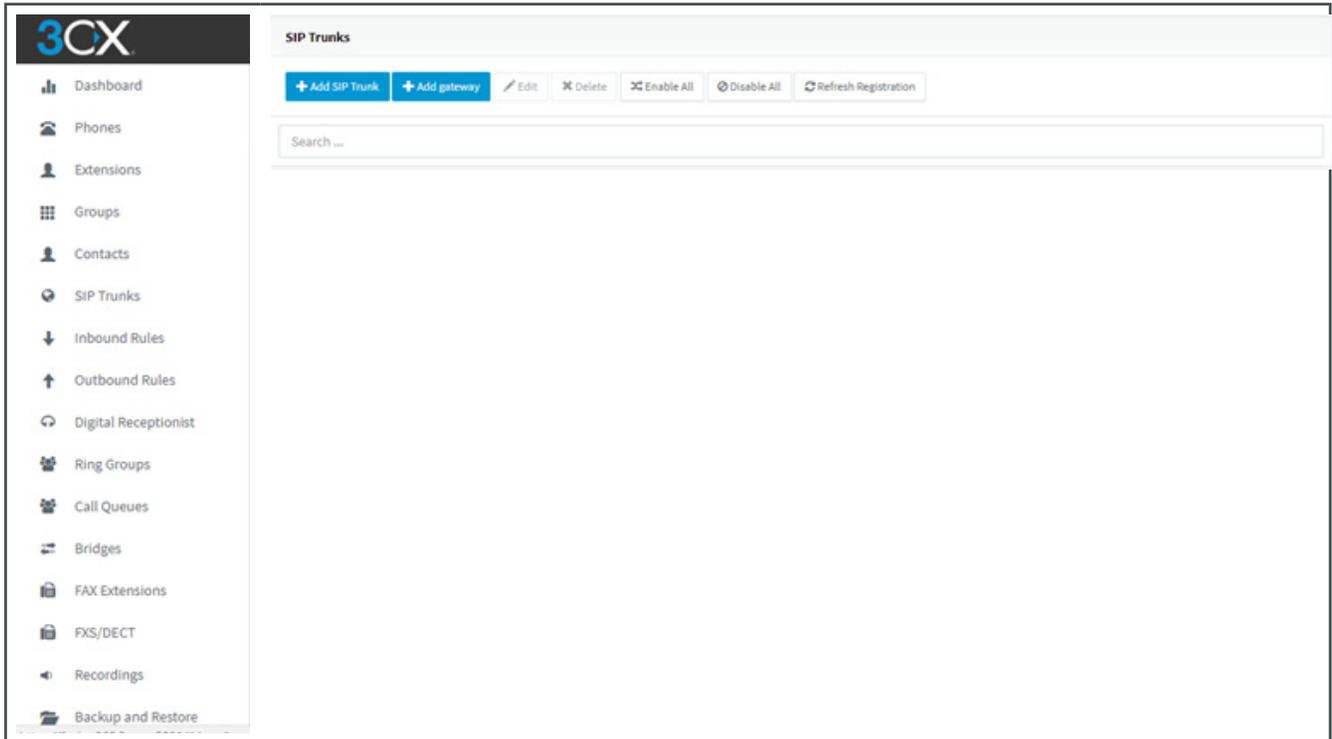
Configuring the 3CX Phone System

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 domain 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 Sessions (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 unless 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

The screenshot shows a web application interface for configuring Caller ID. At the top, there are tabs for 'General', 'DIDs', 'Caller ID', 'Options', 'Inbound Parameters', and 'Outbound Parameters'. The 'Caller ID' tab is active. Below the tabs, there is a heading 'Default caller ID' and a sub-heading 'Configure Outbound Caller ID' with a text input field. Below this, there are two sections: 'Inbound' and 'Outbound'. Each section has a table with columns for 'Source Patterns' and 'Replace Pattern'. The 'Inbound' section has a toolbar with buttons for '+ Add', 'X Delete', '↑ Move Up', and '↓ Move Down'. The 'Outbound' section also has a similar toolbar. The tables are currently empty.

- 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

The screenshot displays the configuration interface for Fusion Connect, specifically the 'Options' tab. The interface is divided into three main sections:

- Call options:** This section contains three checkboxes: 'Allow inbound calls' (checked), 'Allow outbound calls' (checked), and 'Disallow video calls' (unchecked).
- Advanced:** This section contains several checkboxes: 'PBX Delivers Audio' (checked), 'Supports Re-Invite' (unchecked), 'Support Replaces' (unchecked), 'Put Public IP in SIP Via Header' (unchecked), and 'SRTP' (unchecked). Below these is a 'Re-Register Timeout' field with the value '120'. At the bottom, there is a dropdown menu labeled 'Select which IP to use in 'Contact' (SIP) and 'Connection' (SDP) fields' with the option 'Use Default Settings' selected.
- Codec Priority:** This section features a '+ Add codecs' button and two buttons for 'Move Up' and 'Move Down'. Below these are two entries in a list: 'G.711 U-law' and 'G.729', each with a small 'x' icon to its right for removal.

- 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

General DIDs Caller ID Options Inbound Parameters **Outbound Parameters**

Outbound Parameters

Assign SIP header fields to 3CX Call Variables. Requires advanced SIP knowledge. Misconfiguration will cause your PBX to malfunction

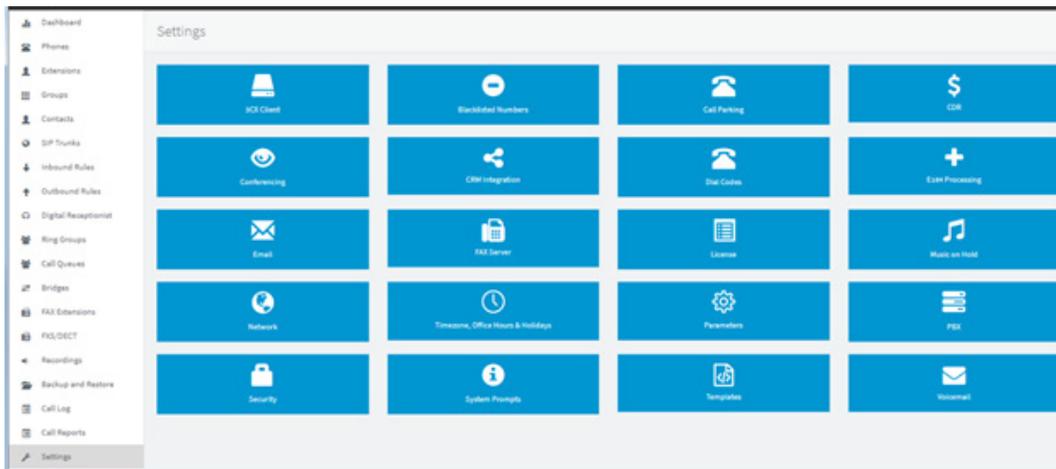
SIP Field	Variable	Custom Value
Request Line URI : User Part	"CalledNum" number that has been dialed (default: To-	
Request Line URI : Host Part	"GWHostPort" gateway/provider host/port	
Contact : User Part	"AuthID" authentication	
Contact : Host Part	"ContactUri" usually, content of Contact field	
To : Display Name	"CalledName" name that has been dialed (default: To->	
To : User Part	"CalledNum" number that has been dialed (default: To-	
To : Host Part	"GWHostPort" gateway/provider host/port	
From : Display Name	"OutboundCallerId" Outbound caller Id taken from Exte	
From : User Part	"AuthID" authentication	
From : Host Part	"GWHostPort" gateway/provider host/port	
User Agent : Text String	Leave default value	

FROM : Host Part	"GWHostPort" gateway/provider host/port
User Agent : Text String	Leave default value
Remote Party ID - Called Party : Display Name	Leave default value
Remote Party ID - Called Party : User Part	Leave default value
Remote Party ID - Called Party : Host Part	Leave default value
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

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.