

3CX Phone System Setup
09/20/2017



Contents:

Purpose and Scope.....3

General Setup Information.....3

Configuring the 3CX Phone System.....4

References & Resources.....8

Purpose and Scope

This setup guide is intended to show basic configuration of the 3CX Phone System (Version 15) and enable deployment of your FusionSIP Registration based SIP trunk.

Documentation was written with a technical audience in mind for re-sellers and installers of 3CX Phone systems. Fusion is limited to support of 3CX Phone Systems to the extent as detailed in this guide.

For more advanced configurations and features, please consult the 3CX documentation.

Website and support links referenced as external resources.

<https://www.3cx.com/>

<https://www.3cx.com/docs/>

<https://www.3cx.com/support/>

General Setup Information

You should always test your configuration prior to enabling in your production environment.

Please note it is the customers' responsibility to properly configure their PBX/Network to prevent security breaches. Fusion engineers are available to offer configuration advice, but under no circumstance shall their assistance constitute the assumption of responsibility by Fusion. In the event of an unauthorized access into a customer's PBX, expenses associated with traffic are irrefutable.

Before you begin, please ensure you have your Welcome Letter which contains your FusionSIP trunk configuration, CCS, authentication credentials (Username and Password), and IP addresses/port ranges required for your PBX. These will be needed to allow signaling and media from Fusion through your NAT or firewall.

Figure 1.1 references the specific welcome letter for the test trunk used to create this document. These entries will serve as the IP address and domain needed to configure your trunk.

*Your actual IP address and port ranges may differ. Consult your welcome letter for the most accurate information.

Traffic Type	IP Addresses	Domain Name	Protocol	Port Range
SIP Registration	216.86.41.69	sip2.thevoicemanager.com	UDP and TCP	5060/5061
SIP Static	216.84.41.68	peer2.thevoicemanager.com	UDP and TCP	5060
RTP - Registration	216.86.41.69	N/A	UDP and TCP	10000 to 65535
RTP - Static	216.86.41.68	N/A	UDP and TCP	10000 to 65535

Figure 1.1

Configuring the 3CX Phone System

Documentation created using FusionSIP Registration trunk with a fresh installation of 3CX (Version 15) utilizing FusionSip (Register) template. These values may be edited manually however we will be utilizing the template in this composition.

Note: Some screens are omitted if changes are not needed.

Step 1

After logging in as an administrator, configure your trunk by clicking on the “SIP Trunks” button found on the left toolbar. Click on the blue “+ Add SIP Trunk” button at the top of the new panel.

Select “FUSION CONNECT (Register)” as your provider. Enter your BTN (which is also your username) found in your Welcome Letter as the Main Trunk Number and click OK.

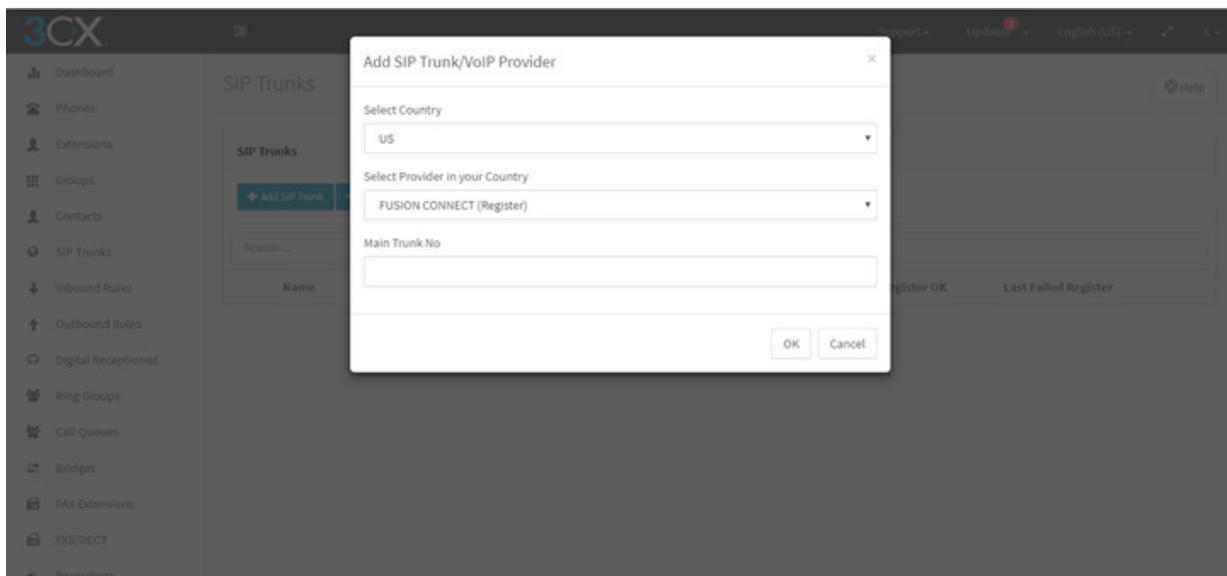


Figure 1.2

Step 2

At this point, many of the configuration settings have been filled in for you but you will need to configure the specifics based on your account information found in your Welcome Letter. Configure the trunk details and authentication as detailed below.

The screenshot shows the 'FUSION CONNECT (Register)' configuration window with the 'Trunk Details' tab selected. The window has 'OK' and 'Cancel' buttons at the top left and a 'Help' icon at the top right. Below the title bar are tabs for 'General', 'DIDs', 'Caller ID', 'Options', 'Inbound Parameters', and 'Outbound Parameters'. The 'Trunk Details' section contains the following fields:

- Enter name for Trunk:** A text box containing 'FUSION CONNECT (Register)'.
- Registrar/Server/Gateway Hostname or IP:** A text box containing 'sip2.thevoicemanager.com' and a port field containing '5060'.
- Outbound Proxy:** A text box (empty) and a port field containing '5060'.
- Number of SIM Calls:** A text box containing '5'.

Registrar/Server/Gateway Hostname or IP – Domain provided in Welcome Letter

For this trunk, sip2.thevoicemanager.com

Number of Sim Calls – trunk CCS

The screenshot shows the 'Authentication' configuration window. It contains the following fields and options:

- Type of Authentication:** A dropdown menu with 'Outbound - Outbound only' selected.
- Authentication ID (aka SIP User ID):** A text box containing '2164167082'.
- Authentication Password:** A text box with masked characters (dots) and a visibility toggle icon.
- 3 Way Authentication:** An unchecked checkbox.

Type of Authentication - Outbound – Outbound Only

Authentication ID – BTN

Authentication Password- password provided by Fusion

Step 3

Select the options Tab and enter the following:

The screenshot shows the 'Options' tab in the Fusion web interface. The 'Call options' section has three checkboxes: 'Allow inbound calls' (checked), 'Allow outbound calls' (checked), and 'Disallow video calls' (unchecked). The 'Advanced' section includes 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 '300'. A dropdown menu for 'Select which IP to use in 'Contact' (SIP) and 'Connection' (SDP) fields' is set to 'Use this IP Address', with the IP address '66.94.82.115' entered below it. The 'Codec Priority' section has a '+ Add codec' button and 'Move Up'/'Move Down' buttons. It lists two codecs: 'G.711 U-law' and 'G.729', each with a close button.

Re-Register Timeout – 300

Select which IP to use in 'contact' and 'connection' fields- Use this IP enter your external IP address (specific to your network)

Fusion offers support for G.711Ulaw and G.729 Annex (G.729)

Step 4

Select the Outbound Parameters Tab and enter the following to control caller ID at the extension level.

* Note: you will need to ensure the caller id is configured under the extensions.

FUSION CONNECT (Register) OK Cancel Help

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-number)	
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=display na	
To : User Part	"CalledNum" number that has been dialed (default: To-number)	
To : Host Part	"GWHostPort" gateway/provider host/port	
From : Display Name	"AuthID" authentication	
From : User Part	"AuthID" authentication	
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 Extension setti
Remote Party ID - Calling Party : User Part	"OutboundCallerId" Outbound caller id taken from Extension setti
Remote Party ID - Calling Party : Host Part	"GWHostPort" gateway/provider host/port
P-Asserted Identity : Display Name	"OutboundCallerId" Outbound caller id taken from Extension setti
P-Asserted Identity : User Part	"OutboundCallerId" Outbound caller id taken from Extension setti
P-Asserted Identity : Host Part	"GWHostPort" gateway/provider host/port
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

From: Display Name – “AuthID” authentication

From: User Part – “AuthID” authentication

P-Asserted Identity: Display Name – “Outboundcallerid” Outbound caller id taken from extensions setting

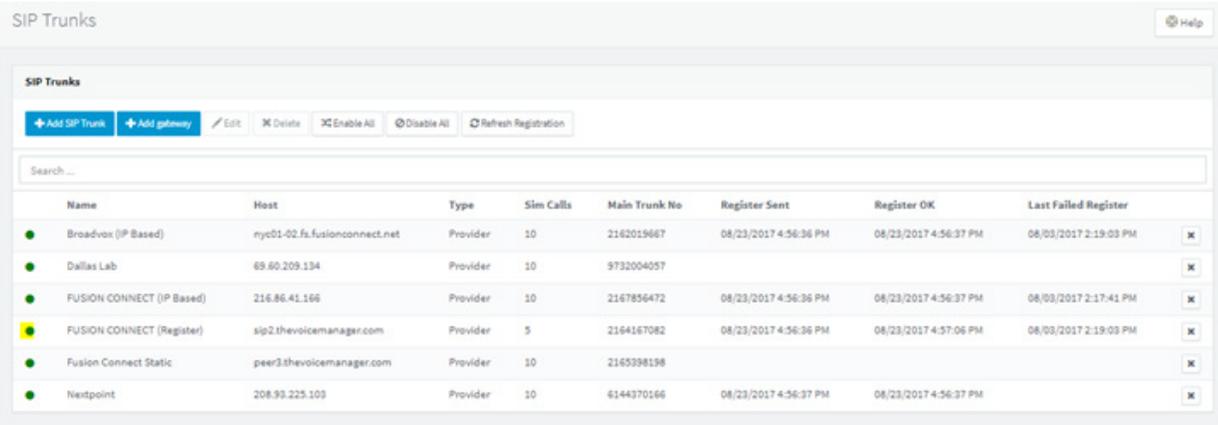
P-Asserted Identity: User Part – “Outboundcallerid” Outbound caller id taken from extensions setting

P-Asserted Identity: Host Part – “GWHostPort” gateway/providerhost/port

Step 5

Once complete, click “OK” at the top. Your FusionSIP trunk should now be registered. Confirm by viewing your SIP trunks. A green indicator light indicates success.

If it is not registered, please confirm the information has been entered correctly and contact technical support for further troubleshooting.



The screenshot shows the 'SIP Trunks' management page. At the top, there are buttons for '+ Add SIP Trunk', '+ Add gateway', 'Edit', 'Delete', 'Enable All', 'Disable All', and 'Refresh Registration'. Below these is a search bar. The main content is a table with the following columns: Name, Host, Type, Sim Calls, Main Trunk No, Register Sent, Register OK, Last Failed Register, and a delete icon. The table lists six trunks, all of which are 'Provider' type and have a green status indicator.

Name	Host	Type	Sim Calls	Main Trunk No	Register Sent	Register OK	Last Failed Register	
Broadvox (IP Based)	nyc01-02.fs.fusionconnect.net	Provider	10	2162019667	08/23/2017 4:56:36 PM	08/23/2017 4:56:37 PM	08/03/2017 2:19:03 PM	✕
Dallas Lab	69.60.209.134	Provider	10	9732004057				✕
FUSION CONNECT (IP Based)	216.86.41.166	Provider	10	2167856472	08/23/2017 4:56:36 PM	08/23/2017 4:56:37 PM	08/03/2017 2:17:41 PM	✕
FUSION CONNECT (Register)	sip2.thevoicemanager.com	Provider	5	2164167082	08/23/2017 4:56:36 PM	08/23/2017 4:57:06 PM	08/03/2017 2:19:03 PM	✕
Fusion Connect Static	peer3.thevoicemanager.com	Provider	10	2165398138				✕
Nextpoint	208.93.225.103	Provider	10	6144370166	08/23/2017 4:56:37 PM	08/23/2017 4:56:37 PM		✕

References & Resources

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