

Guide to setting up 3CX 15.5

Adding a SIP Trunk

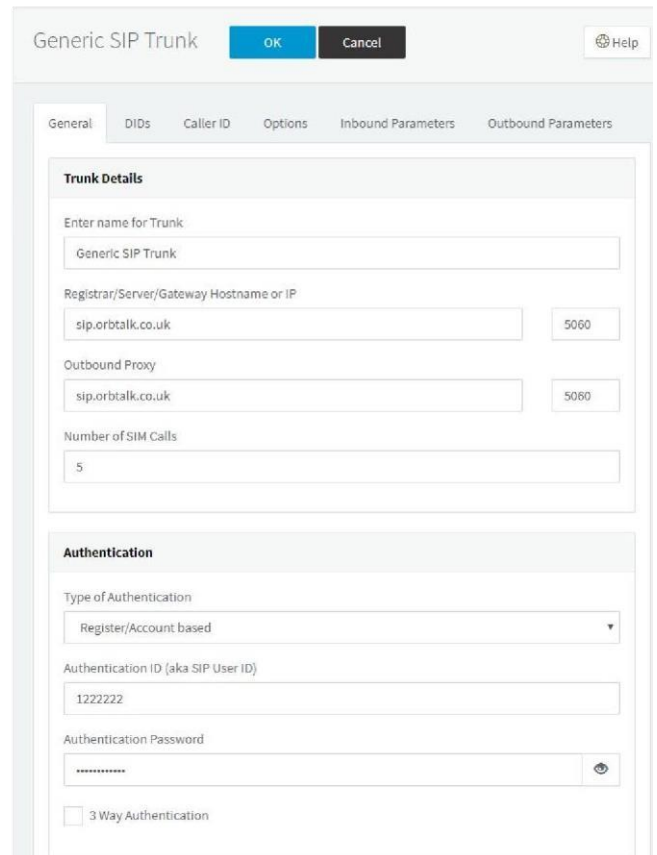
Go to the section called SIP Trunks -> then click "Add SIP Trunk" -> Select country "Generic" -> Select Provider in your country "Generic SIP Trunk" -> Enter "Main Trunk No."(you will obtain this from your "Manage Services" section in portal.orbtalk.co.uk -> it will be the under SIP Trunks -> Username(example 1222222) -> then click the play symbol to retrieve the password(reset password) and domain.



Trunk Details: Now use these details within the 3CX system for "Main Trunk N o."(example 122222) -> Then Enter a suitable name (Orbtalk) within "Enter name for Trunk" -> Registrar/Server/Gateway Hostname or IP:sip.orbtalk.co.uk -> Outbound Proxy:sip.orbtalk.co.uk -> Number of SIM Calls: equal to the amount of channels you have with us.

Next Section you are going to need to use the SIP Trunk details you collected from the Orbtalk portal, please follow the instructions:

Authentication -> Type of Authentication : Register/Account Based -> Authentication ID (aka SIP User ID): this is going to be the "Main Trunk No. (example 122222)" -> Authentication Password: "SIP Trunk password"



Now to go through the other options, please see below.

DIDs: Add the DID as international format like 442035888000.

Caller ID: Default Caller ID -> Configure Outbound Caller ID: "this will be the network caller ID that it goes to if you don't have one setup on the extension."

Options:

Call Options -> Allow Inbound Calls(yes), Allow Outbound Calls(yes), Disallow Video Calls(yes).

Advanced: PBX Delivers Audio(no), Supports Re-Invite(no), Support Replaces(no), Put public IP in SIP Via Head(no), Force invites to be send to IP of Register(no), SRTP(no), Re-register timeout (600) and Select which IP to use in 'contact'(SIP) and 'connection'(SDP)fields(Use default settings).

Codec Priority: G.711 U-law, G.711 A-law and GSM-FR.

Codec Priority

+ Add codecs Move Up Move Down

G.711 U-law	x
G.711 A-law	x
GSM-FR	x

Generic SIP Trunk OK Cancel

General DIDs Caller ID Options Inbound Parameters Outbound Parameters

Reformat Incoming or Outgoing Caller Identification numbers by configuring matching patterns. For more information click here

Default caller ID

Configure Outbound Caller ID

442035888000

Generic SIP Trunk OK Cancel

General DIDs Caller ID Options Inbound Parameters Outbound Parameters

Call options

- Allow inbound calls
- Allow outbound calls
- Disallow video calls

Advanced

- PBX Delivers Audio
- Supports Re-Invite
- Support Replaces
- Put Public IP in SIP VIA Header
- Force Invites to be send to IP of Registrar
- SRTP

Re-Register Timeout

600

Select which IP to use in 'Contact' (SIP) and 'Connection'(SDP) fields

Use Default Settings

Inbound Parameters:

Caller Number/Name Field Mapping:

CalledNum number that has been dialed (default: To->user): **Request Line URI: User Part**

CallerName caller's name (default: From->display name): - **To: User Part**

CallerNum caller's number (default: From->user): - **From: User Part**

Caller Number/Name Field Mapping:

Review the SIP header of the INVITE and specify where the following values should be present within the INVITE:

"CalledNum" number that has been dialed (default: To->user)

Request Line URI : User Part

"CallerName" caller's name (default: From->display name)

From : Display Name

"CallerNum" caller's number (default: From->user)

From : Display Name

Call Source Identification

Configure this option only when the SIP Trunk is IP based (peering), or does not support automatic inbound call detection. If you have multiple trunks from the same vendor or iss works best for this SIP Trunk

Request Line URI : User Part

"EnforcedOriginatorCallerID" To be used when you want to send Anonymous via PAI

Use both "Call Source Identification" rules and "Caller Number/Name -> CalledNum" field mappings (Note: Disables catch all routing capability)

Outbound Parameters:

Please ensure this is exactly as shown in **red** below on your 3CX System. Picture Below.

SIP Field Variable Custom Value

Request Line URI : User Part: **"CalledNum"** number that has been dialed (default:to->user)

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: **"CalledNum"** number that has been dialed (default:to->user)

To : User Part: **"CalledNum"** number that has been dialed (default:to->user)

To : Host Part: **"GWHostPort"** gateway/provider host/port

From : Display Name: **"OutboundCallerId"** Outbound caller Id taken from Extension settings in management console

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:

"OriginatorCallerID" Original caller number will be sent

Remote Party ID - Calling Party : User Part:

"OriginatorCallerID" Original caller number will be sent

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

SIP Field	Variable
Request Line URI : User Part	"CalledNum" number that has been d
Request Line URI : Host Part	"GWHostPort" gateway/provider host
Contact : User Part	"AuthID" authentication
Contact : Host Part	"ContactUri" usually, content of Contz
To : Display Name	"CalledNum" number that has been d
To : User Part	"CalledNum" number that has been d
To : Host Part	"GWHostPort" gateway/provider host
From : Display Name	"OutboundCallerId" Outbound caller
From : User Part	"AuthID" authentication
From : Host Part	"GWHostPort" gateway/provider host
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	"OriginatorCallerID" Original Caller nu
Remote Party ID - Calling Party : User Part	"OriginatorCallerID" Original Caller nu
Remote Party ID - Calling Party : Host Part	"GWHostPort" gateway/provider host
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

Orbtalk's ports and IP's required for the firewall(Optional)

Signalling (UDP port 5060)

185.158.58.7

185.158.57.7

Media IP's (UDP ports 10000 up to 65535)

185.158.58.5

185.158.58.6

185.158.57.5

185.158.57.6

3.1.166.30