

A. Configure OneStream:

1. Add SIP extension group to allow 3CX PBX to register to OneStream with
 Username: 3cx
 Password: 3cxpassword

Interface Type

Name

SIP Extension

Options

SIP Port (default 5060)
 NAT Traversal
 Call Limit (0=unlimited)

Extensions

	Username	Password
1	<input type="text" value="3cx"/>	<input type="text" value="••••••••••"/>
	<input type="button" value="New Extension"/>	

[Show Advanced Options](#)

2. Add Routes – 2 routes are required to allow Inbound and Outbound calls between the PBX and GSM:

Route	From Group	Dest. Addr.	Orig. Addr.	To Group	Mod. Dest.	Mod. Orig.	ACT	CB	Split/Fail
GSM > 3CX PBX	GSM	s	-	3CX PBX	-	-	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
3CX PBX > GSM	3CX PBX	?	-	GSM	-	-	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>



























B. Configure 3CX PBX

1. Click on "VOIP Providers", "Add Provider"
2. Enter a name for the provider (e.g. OneStream), select "Generic SIP Trunk" and click Next

Add VOIP Provider Wizard

Name of Provider ?

Choose a Provider:

<input type="radio"/>		Actio.pl	PL
<input type="radio"/>		Broadvox Fusion (IP Based)	US
<input type="radio"/>		Broadvox Fusion (Register)	US
<input type="radio"/>		CallCentric	US
<input type="radio"/>		Cbeeyond	Worldwide
<input type="radio"/>		CellIP	SE
<input type="radio"/>		EasyCall	GR
<input type="radio"/>		Engin	AU
<input type="radio"/>		G7Eleven	IE
<input checked="" type="radio"/>		Generic SIP Trunk	
<input type="radio"/>		Generic VoIP Provider	
<input type="radio"/>		InPhonex	Worldwide
<input type="radio"/>		nettel	DK
<input type="radio"/>		Nexvortex	US
<input type="radio"/>		Orbtalk	UK
<input type="radio"/>		Prionet	NL
<input type="radio"/>		SipCall	DE, CH
<input type="radio"/>		SIPNET	RU
<input type="radio"/>		Skype for SIP Beta	Worldwide
<input type="radio"/>		Spitfire	UK
<input type="radio"/>		VOIP Voice	IT
<input type="radio"/>		Voip-Unlimited	UK
<input type="radio"/>		Voz Telecom	ES
<input type="radio"/>		Weepee	BE
<input type="radio"/>		Wide VOIP	FR, LU
<input type="radio"/>		XeloQ	Worldwide

More 3rd party tested providers can be found here: <http://wiki.3cx.com/voip-provider/3rd-party-supported>

3. Enter the IP Address of the OneStream in the "SIP server hostname or IP" field and click Next

VOIP Provider Details:

Enter the hostname and port for your VOIP Provider's SIP Server

SIP server hostname or IP	<input type="text" value="192.168.102.91"/> ?
SIP Server port	<input type="text" value="5060"/> ?
Outbound proxy hostname or IP	<input type="text"/> ?
Outbound proxy port (default is 5060)	<input type="text" value="5060"/> ?

4. Fill in the account details as follows and click Next:

External Number = any number (not important but must be entered)




Authentication ID = Username in OneStream SIP Extension group (3cx)

Authentication Password = Password in OneStream SIP Extension group (3cxpassword)


Maximum simultaneous calls = 2

Account Details

Enter the Authentication ID, Password and number of your account

External Number	<input type="text" value="123"/>	
Authentication ID	<input type="text" value="3cx"/>	
Authentication Password	<input type="password" value="*****"/>	







Simultaneous Calls

Maximum simultaneous calls	<input type="text" value="2"/>	
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5. Choose the destination for incoming calls from GSM and click Next

Office Hours

Configure where calls should be routed during office hours.

<input type="radio"/> End Call		
<input checked="" type="radio"/> Connect to Extension	<input type="text" value="100 xtn 100"/>	
<input type="radio"/> Connect to Queue / Ring Group	<input type="text"/>	
<input type="radio"/> Connect to Digital Receptionist	<input type="text"/>	
<input type="radio"/> Voicemail box for Extension	<input type="text" value="100 xtn 100"/>	
<input type="radio"/> Forward to Outside Number	<input type="text"/>	
<input type="radio"/> Send fax to email of extension	<input type="text" value="email of extension 888"/>	

Same as Out of Office hours

6. Create a rule for calls to mobile phones by entering the following details and click Finish:

- Calls to numbers starting with = prefix for gsm calls
- Calls from extensions = range of extensions (e.g. 100-105)
- Route 1 > select OneStream trunk and set Strip Digits = 0

General

Rule Name ?

Apply this rule to these calls

Define to which outbound calls the rule must apply

Calls to numbers starting with (Prefix) ?

Calls from extension(s) ?

Calls to Numbers with a length of ?

Make outbound calls on

Configure up to 3 routes for calls. The second and third route will be used as backup. For each route, digits can be stripped or added.

Route		Strip Digits	Prepend
1	<input type="text" value="OneStream"/>	<input type="text" value="0"/>	<input type="text"/>
2	<input type="text"/>	<input type="text" value="1"/>	<input type="text"/>
3	<input type="text"/>	<input type="text" value="1"/>	<input type="text"/>

7. Edit the newly created VOIP provider and click on the "Advanced" tab.
Set the following options and click OK:

- Require registration for = In and Outgoing calls
- Which IP to use = Internal

General | **Advanced** | Outbound Parameters | Inbound Parameters | Source ID | DID

Provider Capabilities

Configure options related to the SIP capabilities of your provider

Supports Re-Invite ?

Supports 'Replace' ?

PBX Delivers Audio ?

Switch on Secure RTP (SRTP) ?

Registration Settings

Configure options related to the SIP capabilities of your provider

Time between registration attempts (in seconds) ?

Require registration for: ?

Which IP to use in 'Contact' field for registration:

External(STUN resolved) ?

Internal ?

Specified IP ?

Codec priorities

Specify which codecs to use and according to which priority.

Available Codecs

?

?

Assigned Codecs

?

?

?

8. Edit the VOIP Provider and click on the "Source ID" tab

Under "SIP Field" select "From : Host Part"

Under "Variable" select "GWHostPort"

Click "Add/Update" and then OK to confirm

General | Advanced | Outbound Parameters | Inbound Parameters | **Source ID** | DID

Call Source Identification
The source of incoming calls must be identified. Configure how 3CX Phone System should identify calls from this provider.

Matching Strategy: Match All Fields

SIP Field: From : Host Part | Variable: "GWHostPort" gateway/provider host/port

Buttons: Add/Update, Delete, Clear Selection

SIP Field	Variable	Custom Value
From : Host Part	"GWHostPort" gateway/provider host/port	

Source identification by DID
If Call Source identification is based on dialled number and DIDs are in use, you need to specify these DIDs here. Specify a Mask, or select individual DIDs

SIP Field containing DID numbers: Request Line URI : User Part

Source Identification by DID

Buttons: Add Mask, Add DID, Delete

Configuration is now complete – confirm that registration is successful by viewing the OneStream Home/Status Page:

SIP Extensions

Username	IP Address	Status
3cx	192.168.102.66	Registered
