

Unicorn60x0 IP ANALOG GATEWAY – 3CX CONFIGURATION

BASIC CONFIGURATION OF THE Unicorn60x0 WITH 3CX

Due to the various deployment possibilities of the Unicorn60x0 and 3CX, this configuration represents a basic Unicorn – 3CX configuration as using SIP Accounts. Individual configurations can be customized and may slightly differ from either of these basic configurations.

METHODS OF CONFIGURATION

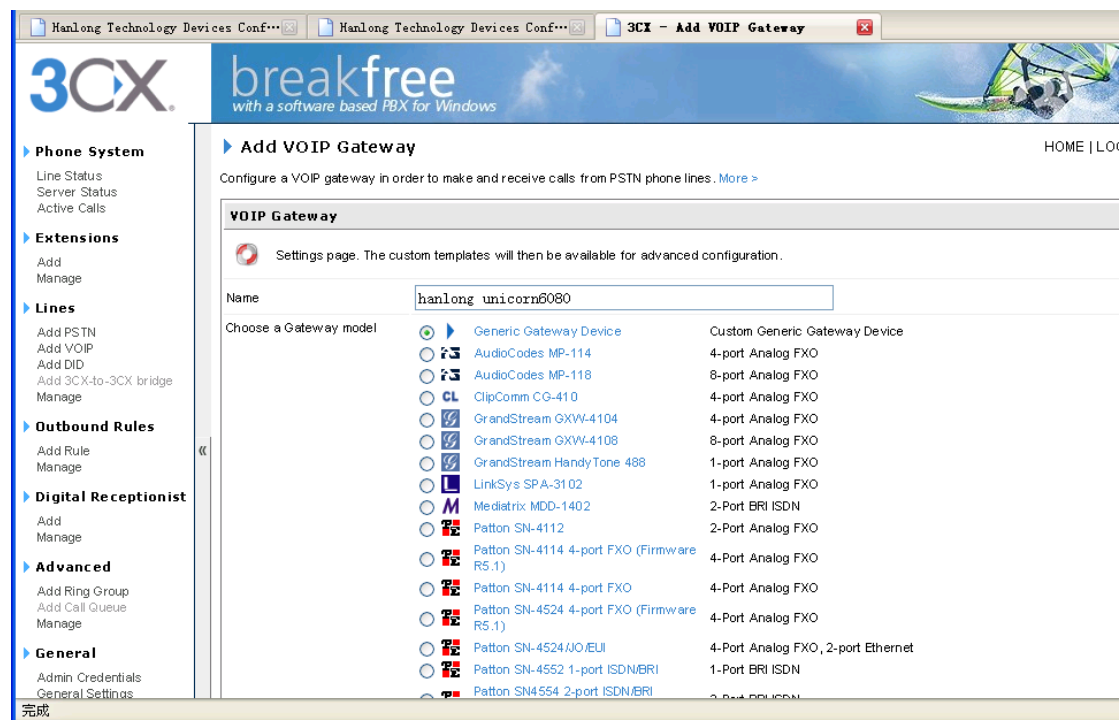
Configure the Unicorn60x0 with SIP accounts in 3CX; this will enable you to configure the Unicorn60x0 behind a NAT/firewall (used for one-stage and two-stage dialing).

Please reboot your system prior to making test calls.

SIP ACCOUNTS CONFIGURATION

Configure the SIP Accounts for the Unicorn60x0

- a) Lines -> Add PSTN -> Choose a Gateway model select “Generic Gateway Device” -> Name, input such as “hanlong unicorn6080”



Click “Next”:

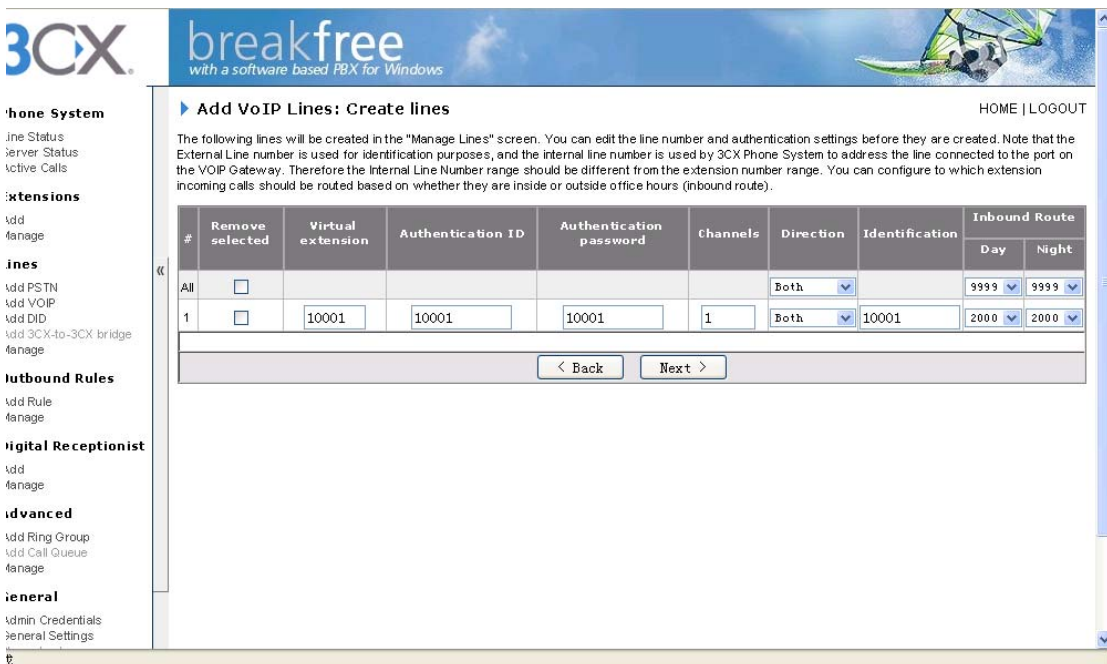


“Gateway Hostname or IP” input [Hostname or IP of unicorn6080](#)

“Number of ports” input “1”

“Type” select to “Analog”

Click “Next”



“Inbound Route” Select extensions which receive call from PSTN

Click "Next"

The screenshot shows the 3CX Breakfree web interface. The top navigation bar includes the 3CX logo and the 'breakfree' brand name with the tagline 'with a software based PBX for Windows'. A sidebar on the left contains a menu with categories: Phone System, Extensions, Lines, Outbound Rules, Digital Receptionist, Advanced, and General. The main content area is titled 'Create Outbound Rule' and includes a sub-header 'Outbound Call Rules allow you to define which PSTN line(s) or VOIP provider outbound calls should be placed on.' The configuration form is divided into sections: 'General' (Rule Name: 'Rule for hanlong unicorn6080'), 'Apply this rule to these calls' (Calls to Numbers starting with (Prefix): '0'), and 'Make Outbound Calls on' (three routes for 'hanlong unicorn6080' with 'Strip Digits' set to '1'). At the bottom of the form are 'Finish' and 'Skip' buttons. A 'HOME | L' link is visible in the top right corner. The bottom left corner of the page has the Chinese characters '完成' (Complete).

Click "Finish"

The screenshot shows the 3CX Breakfree web interface after the gateway creation. The top navigation bar is identical to the previous screenshot. The sidebar menu is also the same. The main content area is titled 'Gateway Created' and contains the following information: 'Lines 10001 to 10001 have been created for hanlong unicorn6080', 'You will need to enter these settings into the VOIP Gateway:', 'Proxy server IP or FQDN: 192.168.0.169:5060', and 'Line 10001' details: 'Internal number: 10001', 'Authentication ID: 10001', and 'Authentication Password: 10001'. A link for a detailed description is provided. The bottom left corner of the page has the Chinese characters '完成' (Complete).

Lines -> Manage -> Click "10001" -> Line(s) Configuration -> Other Options -> "Maximum simultaneous calls" change to "8".(When use Unicorn6080) or "4"(When use Unicorn6040)

Phone System
Line Status
Server Status
Active Calls

Extensions
Add
Manage

Lines
Add PSTN
Add VOIP
Add DID
Add 3CX-to-3CX bridge
Manage

Outbound Rules
Add Rule
Manage

Digital Receptionist
Add
Manage

Advanced
Add Ring Group
Add Call Queue
Manage

General
Admin Credentials
General Settings
Phone book
Get 3CX VOIP Client
Activate Licence
System prompts
Version Updates

Line(s) Configuration
Enter the details of your voip provider account and configure line and routing options.

PSTN Line No 10001 on hanlong unicorn6080

External Number

Authentication ID

Authentication Password

Route inbound calls in office hours to >>>>

Route inbound calls out of office hours to >>>>

Route FAX Calls >>>

Other Options <<<

Specify other options for this line. [More >](#)

Internal Number (Use different range as extension range)

Outbound Caller ID

Allow outbound calls on this line

Answer incoming calls on this line

Maximum simultaneous calls

click "OK"

Line Management HOME | LOGOUT

Edit lines by clicking on the corresponding lines. [More >](#)

Add PSTN
 Add VOIP
 Add DID
 Add 3CX-to-3CX bridge
 Edit Selected
 Delete Selected
 Manage Gateways & Providers
 Office hours

<input type="checkbox"/>	Identification	Virtual extension	Gateway / Provider Name	Gateway / Provider / DID	Direction	Channels	Fax	Office Hours	
								Open	Closed
<input type="checkbox"/>	10000	10000	unicorn6040_1	Gateway	both	4	No Fax	2000	2000
<input type="checkbox"/>	10001	10001	hanlong unicorn6080	Gateway	both	8	No Fax	2000	2000

b) On the Unicorn60x0, enter a SIP server IP address or FQDN under the Profile 1 web configuration page.

Under FXO Lines web configuration page set the following:

PROFILE OPTIONS -> Connect to Asterisk -> select "Yes".

Unicorn60x0 will auto-select an idle Line to make outbound call to PSTN.

PROFILE 1 OPTIONS	
Account Active:	<input type="radio"/> No <input checked="" type="radio"/> Yes
SIP Server:	<input type="text" value="192.168.0.169"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)
NAT Traversal	<input type="radio"/> No <input checked="" type="radio"/> No, but send keep-alive <input type="radio"/> STUN <input type="radio"/> UPNP
Connect to Asterisk	<input type="radio"/> No <input checked="" type="radio"/> Yes

PROFILE OPTIONS -> Caller ID Transport Type -> select "Relay via SIP From"

Caller ID Transport Type	<input type="text" value="Relay via SIP From"/>
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"Unconditional Call Forward to VOIP" to forward incoming PSTN calls to. Using 3CX configuration, example, this would be 2000.

Unconditional Call Forward to VOIP	User ID	Sip Server	Sip Destination Port
	<input type="text" value="2000"/>	@ <input type="text" value="192.168.0.169"/>	<input type="text" value="5060"/>

c) Under the FXO Port web configuration page, enter the SIP User IDs, Authentication IDs, and Authentication passwords as well as their corresponding profiles.

Channels	SIP User ID	Authentication ID	Authentication Password	Profile ID
1	10001	10001	***	Profile1
2				Profile1
3				Profile1
4				Profile1
5				Profile1
6				Profile1
7				Profile1
8				Profile1

So, all call from PSTN will be forward to extension which user id is 2000. When extensions dial "0xxxx.", the call will be routed to Unicorn60x0; Unicorn60x0 will select an idle line call to PSTN.

OTHER GENERAL SETTINGS

A. PSTN Settings

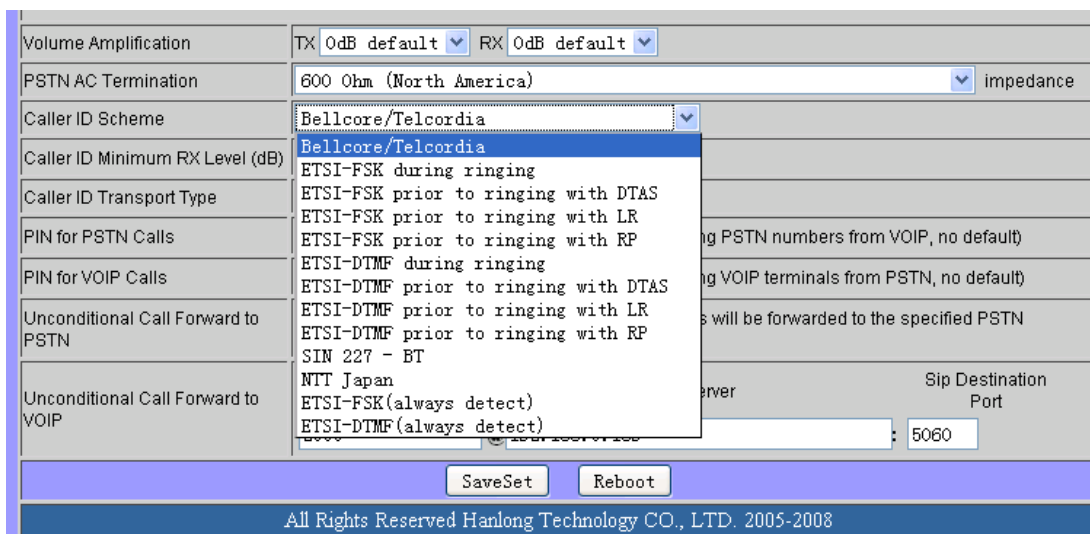
The following fields need to be set according to the PSTN Service provider (or PSTN PBX):

1. Call Progress Tones located under the *SUPER OPTIONS* page (Dial Tone,

Ring-back Tone, Busy Tone, Reorder Tone, and Confirmation Tone). These should be set according to the PSTN Service Provider or PSTN PBX that you are using with the gateway. The format and syntax are shown on the web GUI. By default, they are set according to North American PSTN settings.

2. Caller ID Scheme located under *FXO Lines Web Configuration* page. There are five different options to choose from: Bellcore, ETSI_RING, ETSI_TAS, DTMF and NTT.

For Caller ID detect, It is important to set Caller ID Scheme of FXO.



B. DTMF Methods (to be used ONLY for PSTN to VOIP calls, not for VOIP to PSTN).

The gateway allows several combinations of DTMF sending methods from PSTN to VoIP.

DTMF Method Flexible DTMF transmission method, User interface of In-audio, RFC2833, and/or SIP Info

DTMF Settings are in Profile pages.

- **DTMF in-audio:** Send DTMF as in-band (in-audio).
- **DTMF via RTP (RFC2833):** Send DTMF via RTP (According to RFC 2833).
- **DTMF via SIP INFO:** Send DTMF via SIP INFO message.

Please ensure the DTMF methods you set match those enabled on 3CX. RFC2833 is the recommended method;

C. NAT Settings

1. If you place the gateway within a private network behind a firewall, we recommend using a STUN Server. In the STUN Server field under the Advanced Settings web configuration page, enter a STUN server IP or FQDN. You can find free public

STUN servers on the internet. Set NAT Traversal (STUN) under the Profile web configuration pages to Yes.

2. Depending on your network settings, you may set the Use Random Port setting under the Advanced Settings web configuration page to Yes or No. If using a public IP address, set this setting to No.

D. Preferred Vocoder (Codec)

Under the *Profile Web Configuration* pages, you can choose from five different codecs:

1. PCMU (or G711u)
2. PCMA (or G711a)
3. G729A/B
4. G723.1

Please ensure that the codecs you set match those enabled on 3CX. Codecs may be enabled system-wide in 3CX or for specific users.

MAKE A TEST CALL

After configuring the Unicorn60x0, reload the chan_sip.so module and the extensions in 3CX, or simply restart the service. The commands vary depending on the version of 3CX. Update the changes made to the Unicorn60x0 and reboot the device. You may now place inbound and outbound test calls to test your configuration.