

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.

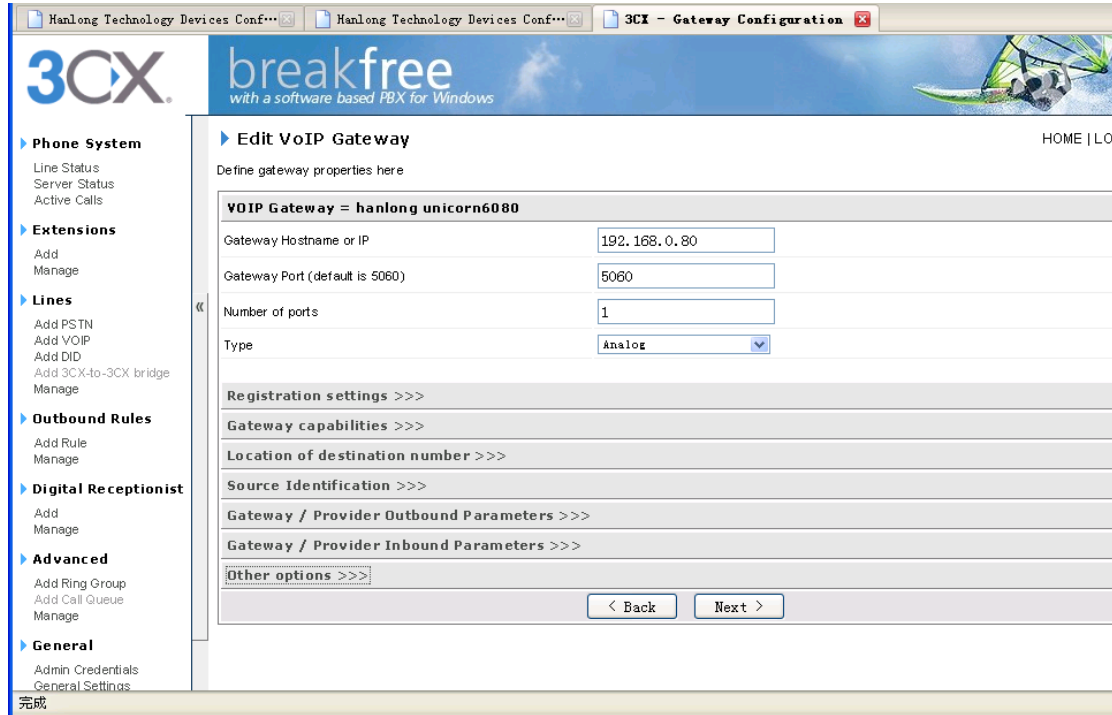
SIP ACCOUNTS CONFIGURATION

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”:

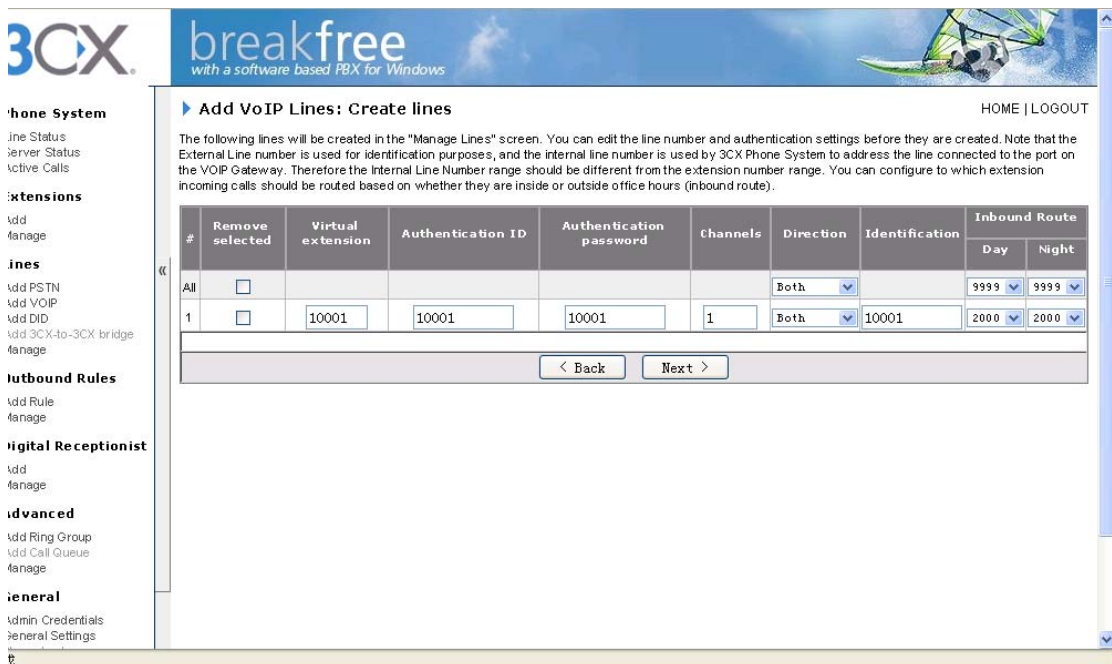


“Gateway Hostname or IP” input [Hostname or IP of unicorn6080](#)

“Number of ports” input “1”

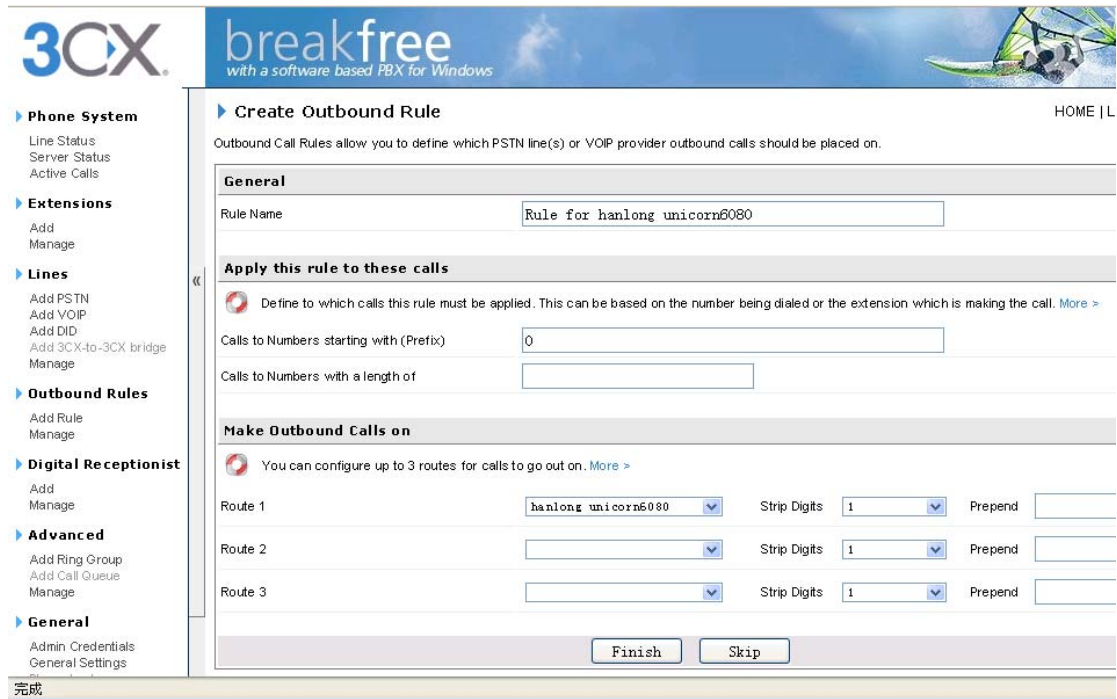
“Type” select to “Analog”

Click “Next”



“Inbound Route” Select extensions which receive call from PSTN

Click "Next"



The screenshot shows the 'Create Outbound Rule' configuration page in the 3CX Breakfree interface. The left sidebar contains navigation menus for Phone System, Extensions, Lines, Outbound Rules, Digital Receptionist, Advanced, and General. The main content area is titled 'Create Outbound Rule' and includes a 'HOME | L' link. Below the title, it states: 'Outbound Call Rules allow you to define which PSTN line(s) or VOIP provider outbound calls should be placed on.' The configuration is divided into three sections: 1. 'General': A text field for 'Rule Name' containing 'Rule for hanlong unicorn6080'. 2. 'Apply this rule to these calls': A sub-section with a red circle icon and text: 'Define to which calls this rule must be applied. This can be based on the number being dialed or the extension which is making the call. More >'. It includes two input fields: 'Calls to Numbers starting with (Prefix)' with '0' and 'Calls to Numbers with a length of' which is empty. 3. 'Make Outbound Calls on': A sub-section with a red circle icon and text: 'You can configure up to 3 routes for calls to go out on. More >'. It contains three rows for 'Route 1', 'Route 2', and 'Route 3'. Each row has a dropdown menu (Route 1 is set to 'hanlong unicorn6080'), a 'Strip Digits' dropdown (all set to '1'), and a 'Prepend' text field. At the bottom of the main area are 'Finish' and 'Skip' buttons. A '完成' (Complete) button is located at the bottom left of the page.

Click "Finish"



The screenshot shows the 'Gateway Created' configuration page in the 3CX Breakfree interface. The left sidebar is identical to the previous screenshot. The main content area is titled 'Gateway Created' and includes a 'HOME | L' link. Below the title, it states: 'Lines 10001 to 10001 have been created for hanlong unicorn6080'. It then says: 'You will need to enter these settings into the VOIP Gateway:'. The settings listed are: 'Proxy server IP or FQDN: 192.168.0.169:5060', 'Line 10001', 'Internal number: 10001', 'Authentication ID: 10001', and 'Authentication Password: 10001'. At the bottom, there is a link: 'For a detailed description how to do this for popular gateways'. A '完成' (Complete) button is located at the bottom left of the page.

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/Update

▶ 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**


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


HOME | LOGOUT


 Add PSTN


 Add VOIP


 Add DID

 Add 3CX-to-3CX bridge

 Edit Selected

 Delete Selected

 Manage Gateways & Providers

 Office hours

| ☐ | 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 PORTS web configuration page set the following:

PROFILE OPTIONS ->Ports Using The Profile Share With One Common Account

- If set to "Yes", Unicorn60x0 will use the first account among the FXO PORTS of using the same profile.
- If set to "No", you need configure one port one account in FXO PORTS page.

PROFILE OPTIONS -> Auto Select Idle Port (For Outgoing Call).

- If set to “Yes”, Unicorn60x0 will auto-select an idle Line to make outbound call to PSTN.
- If set to “No”, you need configure one port one account in FXO PORTS page, then it is the business of 3CX that it decide which idle FXO Port for outbound call.

| PROFILE 3 OPTIONS | |
|---|---|
| Account Active | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| SIP Server | <input type="text"/> (e.g., sip.company.com, or IP address) |
| Outbound Proxy | <input type="text"/> (e.g., proxy.provider.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 |
| Ports Using The Profile Share With One Common Account | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Auto Select Idle Port (For Outgoing Call) | <input checked="" type="radio"/> No <input 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"/> |
|--------------------------|---|

“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="@ 192.168.0.189"/> | <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.

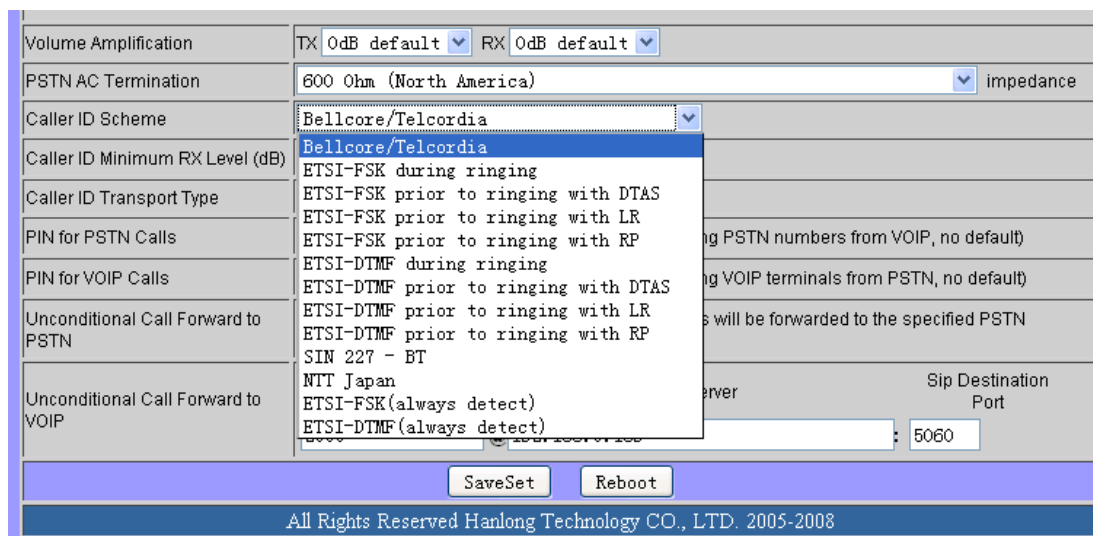
OTHER GENERAL SETTINGS

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.