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

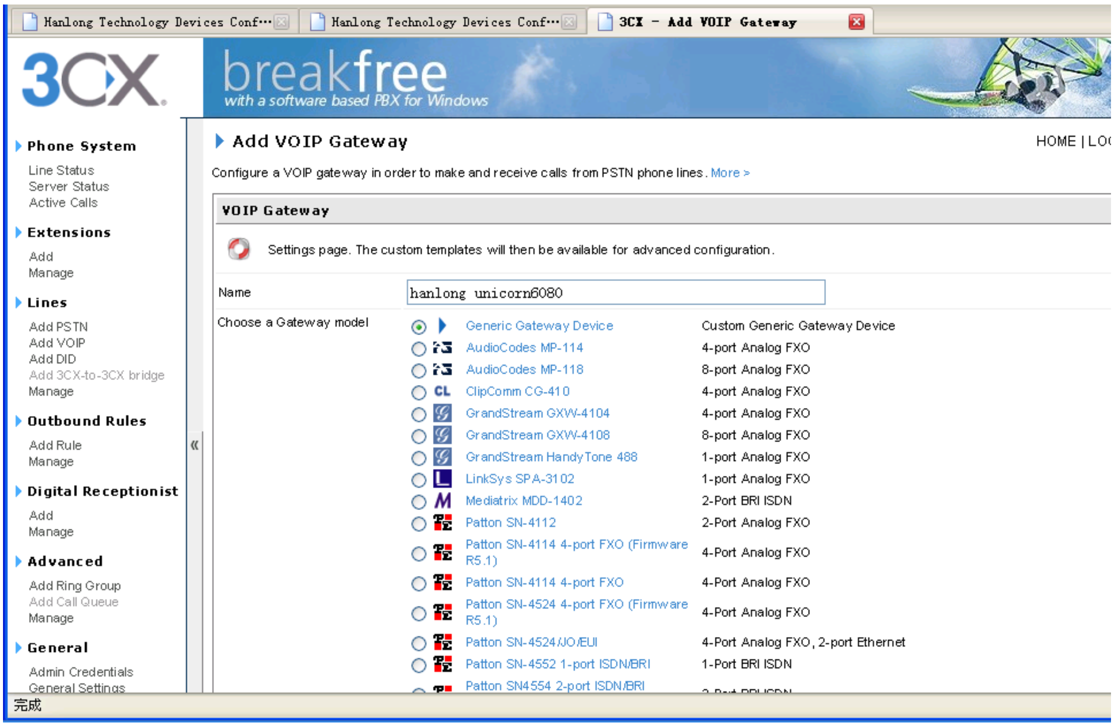
Configure the Unicorn60x0 with SIP accounts in 3CX; this will enable you to configure the Unicorn60x0 behind a NAT/firewall (used for one-stage and two-stage dialing).

Please reboot your system prior to making test calls.

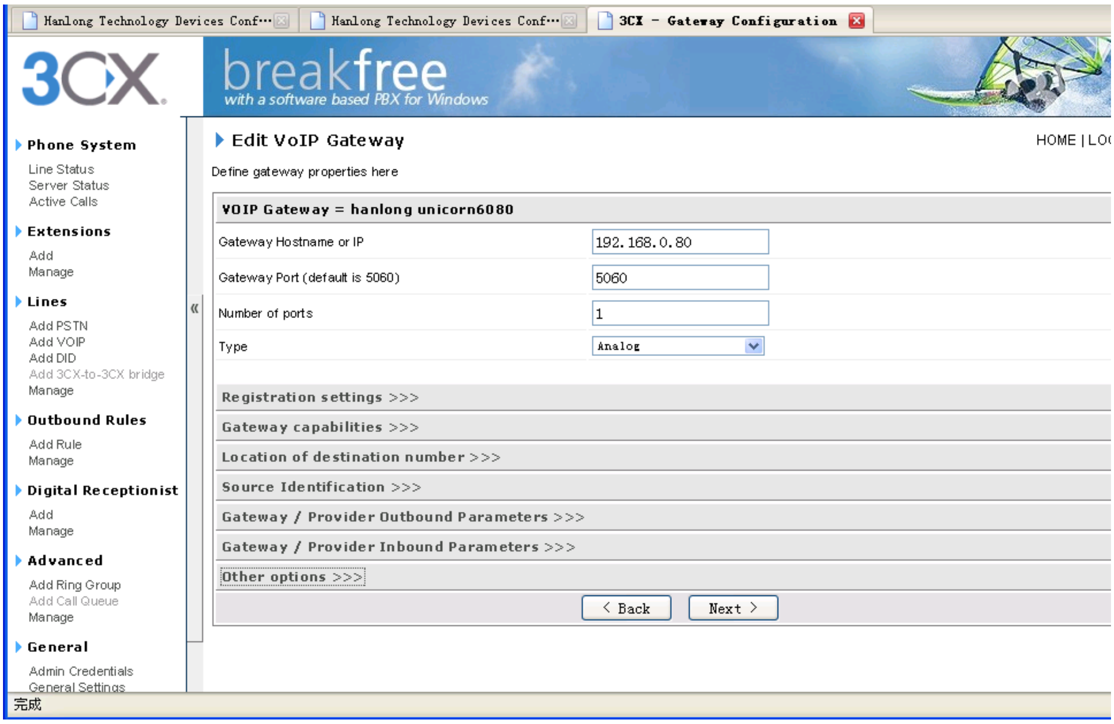
SIP ACCOUNTS CONFIGURATION

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"

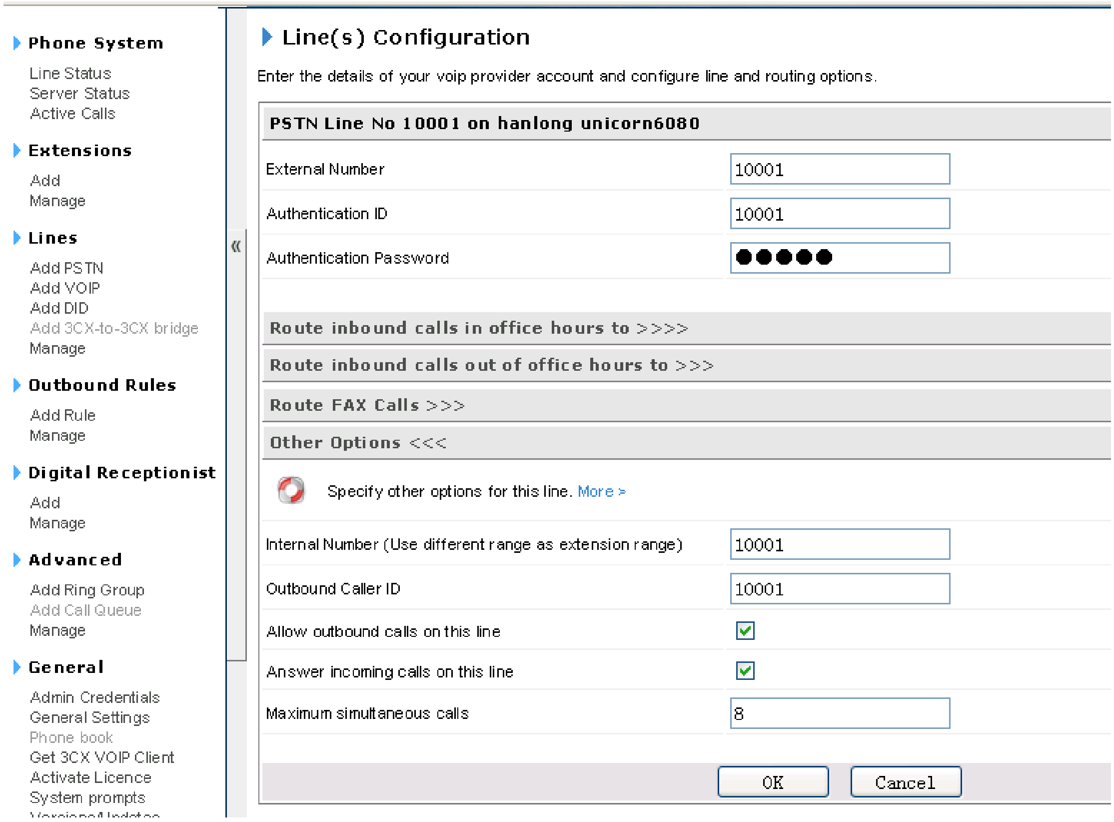
Click "Next"

"Inbound Route" Select extensions which receive call from PSTN

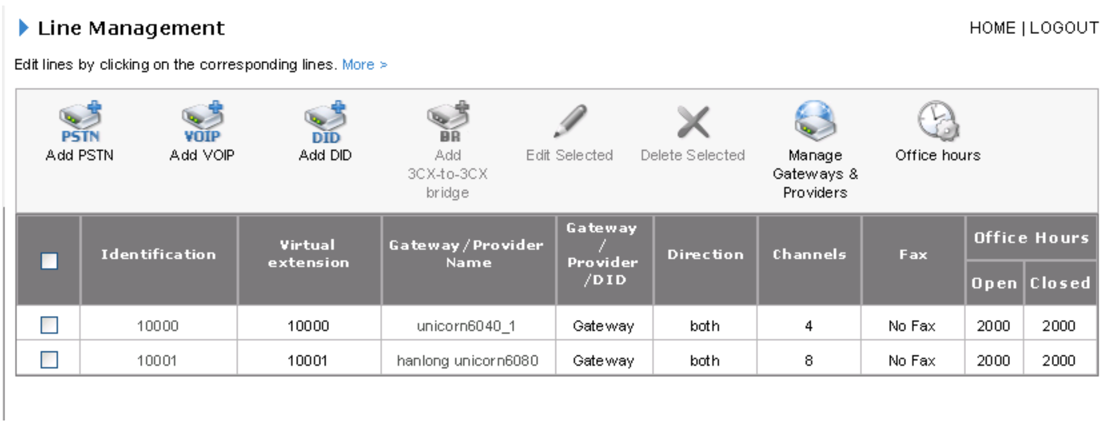
Click "Next"

Click "Finish"

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



click “OK”



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

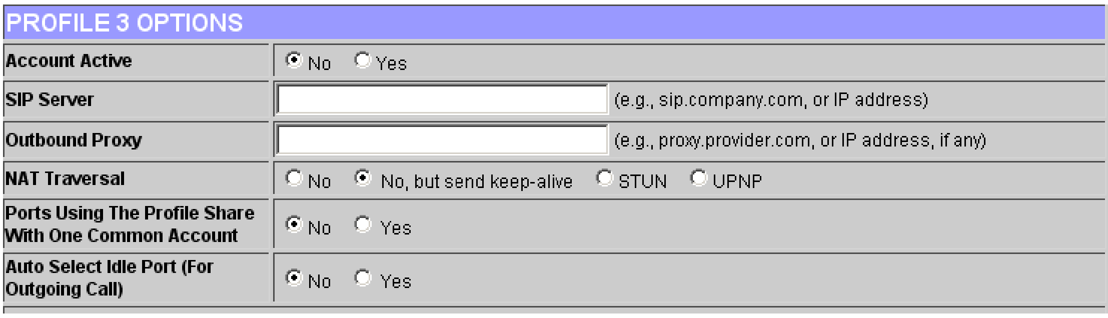
Under FXO PORTS web configuration page set the following:

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

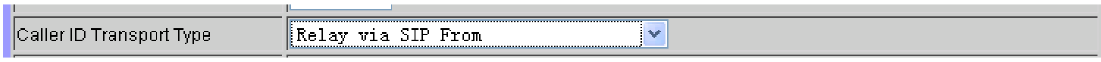
- **If set to “Yes”, Unicorn60x0 will use the first account among the FXO PORTS of using the same profile.**
- **If set to “No”, you need configure one port one account in FXO PORTS page.**

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

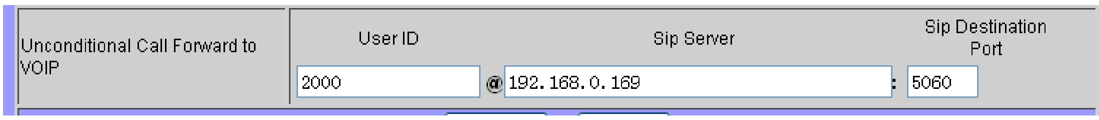
- **If set to "Yes", Unicorn60x0 will auto-select an idle Line to make outbound call to PSTN.**
- **If set to "No", you need configure one port one account in FXO PORTS page, then it is the business of 3CX that it decide which idle FXO Port for outbound call.**



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



"Unconditional Call Forward to VOIP" to forward incoming PSTN calls to. Using 3CX configuration, example, this would be 2000.



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

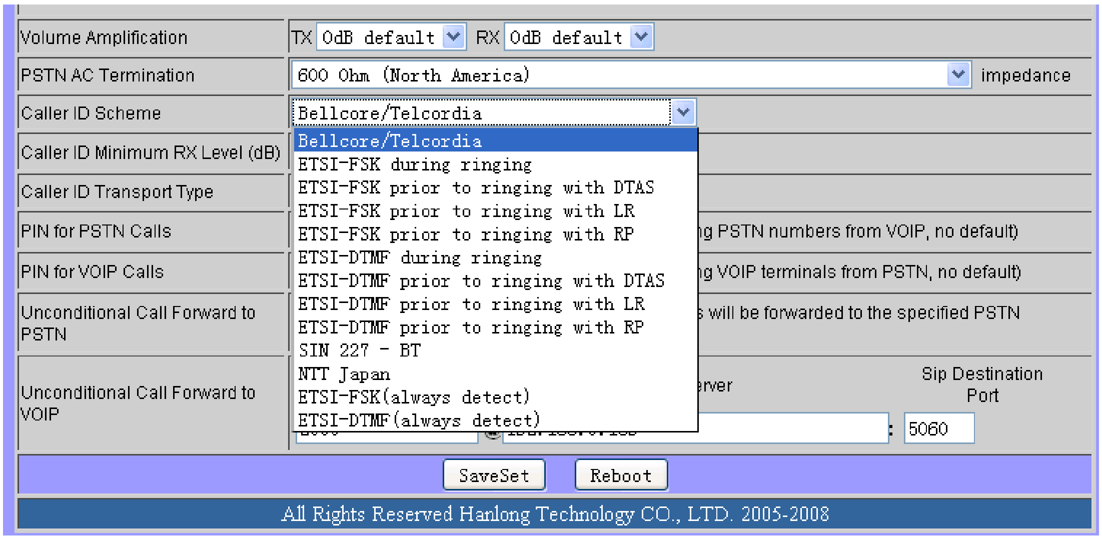
A. PSTN Settings

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.