Unicorn60x0 IP ANALOG GATEWAY – ASTERISK™ CONFIGURATION

BASIC CONFIGURATION OF THE Unicorn60x0 WITH ASTERISK™

Due to the various deployment possibilities of the Unicorn60x0 and Asterisk™, this configuration represents a basic Unicorn – Asterisk™ 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 Asterisk™; 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) In the sip.conf file, add SIP accounts for the Unicorn60x0's channels. You may add up to four or eight depending on the number of channels you have:

[100]

type=friend

secret=password

host=dynamic

context=internal

insecure=port

callerid=${CALLERID(num)}   ;for show PSTN number

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

c) Under PROFILE OPTIONS 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).

> ➢ **Don't need to set this option.**

PROFILE 3 OPTIONS	
Account Active	◉ No ○ Yes
SIP Server	[ ] (e.g., sip.company.com, or IP address)
Outbound Proxy	[ ] (e.g., proxy.provider.com, or IP address, if any)
NAT Traversal	○ No ◉ No, but send keep-alive ○ STUN ○ UPNP
Ports Using The Profile Share With One Common Account	◉ No ○ Yes
Auto Select Idle Port (For Outgoing Call)	◉ No ○ Yes

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

Caller ID Transport Type	Relay via SIP From ▾

d) 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	100	100	***	Profile1
2				Profile1
3				Profile1
4				Profile1
5				Profile1
6				Profile1
7				Profile1
8				Profile1

One-stage Dialing:

1. In the extensions.conf file, specify what numbers dialed from the VoIP side will be sent through the Unicorn60x0. To specify ten-digit numbers and eleven-digit numbers conforming to NANPA specifications as well as 911 emergency services to be sent through the Unicorn60x0, use the following syntax in the same context as the VoIP caller or in an included context:

    a. exten => _NXXNXXXXXX,1,Dial(SIP/${EXTEN}@100)

b. exten => _1NXXNXXXXXX,1,Dial(SIP/${EXTEN}@100)

c. exten => 911,1,Dial(SIP/${EXTEN}@100)

2. Numbers such as (626) 956-0260, 1(617)566-9300, and 911 will be dialed through the Unicorn60x0.

4. Calls coming into the Unicorn60x0 from the PSTN side will be directly auto-dialed to a configurable VoIP extension when using one-stage dialing. Set the auto dial extension in the extensions.conf file, within the same context that the Unicorn60x0 is in or in an included context. <u>For example</u>: forward incoming PSTN calls to extension 200, which may be a SIP device or a greeting (generally stored in /var/lib/Asterisk™/sounds):

a. exten => 200,1,Dial(SIP/200)

Or

b. exten => 200,1,Background(greeting_filename)

5. Under FXO Lines web configuration page set the following:

"Unconditional Call Forward to VOIP" to forward incoming PSTN calls to. Using Asterisk™ configuration, example, this would be 200.

Unconditional Call Forward to VOIP	User ID		Sip Server		Sip Destination Port
	200	@	192.168.0.250	:	5060

Two-stage Dialing (used only when configured with SIP accounts)

To use two-stage dialing, dial the SIP accounts configured on the channels of the Unicorn60x0. For example, if one of the SIP accounts is SIP/100, we add this line in the extensions.conf file, within the same context as the VoIP caller or in an included context:

exten => 100,1,Dial(SIP/100)

With this rule, VoIP users may call extensions 100 and the call will be connected with the Unicorn60x0 channel, where they will receive the PSTN dial tone to make PSTN calls.

On the Unicorn60x0, under FXO Lines web configuration page set the following:

"Unconditional Call Forward to VOIP"does not need to be set as it is not used. Incoming PSTN calls will simply receive VoIP dial tone, where the PSTN caller can then dial a VoIP number.

3. Most settings on the Unicorn60x0 web configuration pages are set to default values. Be sure to configure the Unicorn60x0 settings according to the local environment.

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.

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 Asterisk™. RFC2833 is the recommended method; which can be set under the [general] area in the sip.conf file with this line:

dtmfmode=rfc2833

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 Asterisk™. Codecs may be enabled system-wide in Asterisk or for specific users. For example: to enable PCMU system-wide, place the following lines under the [general] settings in the sip.conf file:

allow=ulaw

MAKE A TEST CALL

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