

Use IP PBX System

How to config IP PBX on Ecsow Dialer / Voice Broadcast

There use Asterisk to demonstrate how to config with IP PBX.

Step 1, Creating SIP Accounts from your Asterisk.

There actually have two options for creating a SIP account on Asterisk, the configuration file would be **sip.conf** for chan_sip, or **pjsip.conf** for chan_pjsip/res_pjsip. (Note: chan_pjsip is only available in Asterisk 12 or later.) These files reside in the Asterisk configuration directory, which is typically **/etc/asterisk**.

For example, we add the sip account karen-demo on sip.conf.

```
[karen-demo]
type = friend
username = karen-demo
secret = 1234
context = tutorial
host = dynamic
```

Step 2, Config the SIP account on Ecsow Dialer.

- **SIP ID/ User Name** - This should be set to **karen-demo** on Ecsow. This username corresponds directly to the section name in square brackets in sip.conf.
- **Domain** - The location of the server which the phone should register to. This should be set to the IP address of your Asterisk system.
- **Authentication User** - On Asterisk-based systems, this will be the same as the SIP user name above.
- **Outbound Proxy** - This is the server with which your phone communicates to make outside calls. This should be set to the IP address of your Asterisk system.

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The image shows a software window titled 'Options' with a close button in the top right corner. The window contains several tabs: 'General', 'SIP Account', 'Do-Not-Call', 'Voice Activity Detection', 'Call Distribution', 'Call Transfer', and 'Preset'. The 'SIP Account' tab is selected. Below the tabs, there is a section titled 'Account information from your SIP VoIP service provider' containing five input fields: 'Display Name' (Asterisk), 'SIP ID/User Name' (karen-demo), 'Domain' (172.16.26.4), 'Authorization User' (karen-demo), and 'Password' (masked with dots). Below this is the 'Outbound Proxy' section with the label 'Send outbound calls via:' and two radio buttons: 'Domain' (unselected) and 'Proxy' (selected). A text box next to the 'Proxy' radio button contains '172.16.26.4'. The 'Concurrent Calls' section has a label 'Number of concurrent calls' and a text box containing '1'. Below this is a note: '* SIP service must support multiple channels if you want to run concurrent calls'. At the bottom of the main content area is a button labeled 'Advanced Settings'. At the bottom right of the window are 'OK' and 'Cancel' buttons.

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