

	SIP	IAX2	STUN	DIAL PEER	MCAST
BASIC					
NETWORK					
▶ VOIP					
PHONE					
FUNCTION KEY					
MAINTENANCE					
SECURITY					
LOGOUT					

SIP Line: SIP 1

Basic Settings >>

Status	Registered	Domain Realm	<input type="text"/>
Server Address	<input type="text" value="117.176.159.157"/>	Proxy Server Address	<input type="text"/>
Server Port	<input type="text" value="5060"/>	Proxy Server Port	<input type="text"/>
Authentication User	<input type="text" value="800"/>	Proxy User	<input type="text"/>
Authentication Password	<input type="password" value="*****"/>	Proxy Password	<input type="password"/>
SIP User	<input type="text" value="800"/>	Backup Proxy Server Address	<input type="text"/>
Display Name	<input type="text" value="800"/>	Backup Proxy Server Port	<input type="text" value="5060"/>
Enable Registration	<input checked="" type="checkbox"/>	Server Name	<input type="text"/>

If you use DDNS, pls use ddns instead of IP at here

Almost the same as you register a local extension, but in the server address blank you should use public IP or the DDNS domain name instead of private IP address.

If it runs on TCP or TLS, please click "Advanced SIP Settings" and choose TCP or TLS in the "Transport Protocol" dropdown list.

Transport Protocol	<input type="text" value="UDP"/>
Use VPN	<input type="checkbox"/>
Enable DND	<input type="checkbox"/>

Notice:

SIP extension over TCP/TLS protocol also needs to enable TCP/TLS for the extension. Please select the transmission protocol on the extension configure page before registering from the phone.

And enable the NAT & Remote extension option

VoIP Settings

NAT: <input checked="" type="checkbox"/>	Transfer Protocol: <input type="text" value="UDP+TCP"/>	SRTP: <input type="checkbox"/>
Qualify: <input checked="" type="checkbox"/>	Remote Extension: <input checked="" type="checkbox"/>	
DTMF Mode: <input type="text" value="RFC2833"/>	Permit IP: <input type="text"/>	

Video Options