



# Sipphone VoIP Service Provider

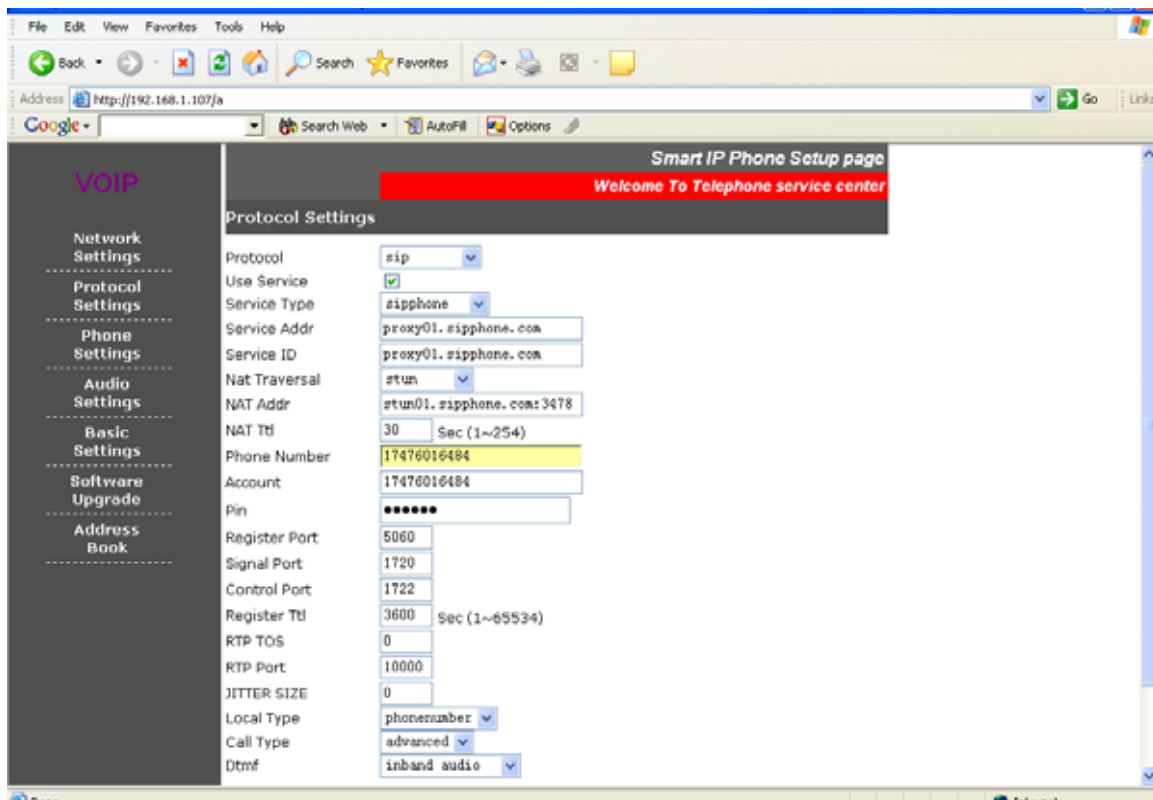
network settings			
iptype	dhcp	ppp id	ppp pin
local ip	192.168.1.71	subnet mask	255.255.255.0
dns	202.96.46.151	dns2	202.96.128.68
router ip	192.168.1.254	mac	00-09-45-64-02-ac
protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	common	service addr	proxy01.sipphone.com
service id	proxy01.sipphone.com	nat traversal	disable
nat addr		nat ttl	240
phone number	17473861939	account	17473861939
pin	1234	register port	5060
signal port	1720	control port	1722
dtmf payload	101	rtp port	1722
local type	account	call type	advanced
rtp tos	0		
phone settings			
use dialplan	disable	dial number	
ddcode	86	ddprefix	00
ddprefix	0	innerline	disable
innerline prefix	0	dual mode	disable
dualmode prefix	0	ring type	dtmf0
use digitmap	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>
fed nonanswer	<input type="checkbox"/>	fed always	<input type="checkbox"/>
fed busy	<input type="checkbox"/>	answer	30
audio settings			
vad	<input checked="" type="checkbox"/>	agc	<input type="checkbox"/>
codecl	g7231	codec2	g729
codec3	g711u	codec5	gsm
g 723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
speaker out (0-31)	30		
other settings			
password	1234	super password	12345678
debug	output all	ntp ip	210.59.157.10
use daylight	<input type="checkbox"/>	upgrade addr	

or

network settings			
iptype	dhcp	ppp id	ppp pin
local ip	192.168.1.109	subnet mask	255.255.255.0
dns	202.96.128.68	dns2	202.96.134.188
router ip	192.168.1.1	mac	00-09-45-62-18-95
protocol settings			
protocol	sip	use service	<input checked="" type="checkbox"/>
register ttl	60	service type	sipphone
service addr	proxy01.sipphone.com	service id	proxy01.sipphone.com
nat traversal	stun	nat addr	stun01.sipphone.com:3478
nat ttl	30	phone number	17476016484
account	17476016484	pin	972708
register port	5060	signal port	1720
control port	1722	jitter size	0
rtp port	1722	local type	phonenumber
call type	advanced	dtmf	inband audio
phone settings			
use dialplan	disable	dial number	
ddcode	86	ddprefix	00
ddprefix	0	innerline	disable
local prefix	0	nonlocal prefix	0
answer	0	ring type	dtmf0
use digitmap	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
forward number	82378801	fed poweroff	<input type="checkbox"/>
fed nonanswer	<input type="checkbox"/>	fed always	<input type="checkbox"/>
fed busy	<input type="checkbox"/>		
audio settings			
audio type	auto	audio frames	2
g 723.1 high rate	<input checked="" type="checkbox"/>	vad	<input checked="" type="checkbox"/>
agc	<input type="checkbox"/>	handset in	5
handset out	19	handset out	19
speaker out	18		
other settings			
password	1234	super password	19800211
debug	output all	ntp ip	210.59.157.10
use daylight	<input type="checkbox"/>	upgrade addr	
timezone	(GMT+08:00) Beijing, Hong Kong, Urumqi		

Save/Reboot

or



After successful setting, the phone can be used to call other phones at the same Proxy Server as follows:

Pick up the handset or press Speaker key, and then input phone number, such as 17473861936 or 17474745000, then press # or Call key to end dialing.

## FreeWorldDialup VoIP Service Provider

The screenshot displays a comprehensive configuration page for a VoIP service. It is organized into several sections:

- network settings:** Includes fields for local ip (192.168.1.108), subnet mask (255.255.255.0), router ip (192.168.1.1), dns (202.96.128.68), dnst (202.96.134.188), and wan (00-09-45-62-18-95).
- protocol settings:** Features protocol (sip), use service (checked), register ttl (60), service type (common), service addr (fwdNAT.pulver.com:5082), service id (fwd.pulver.com), nat traversal (disable), nat addr, nat ttl (30), phone number (631443), account (631443), pin (\*\*\*\*\*), register port (5060), signal port (1720), control port (1722), jitter size (0), rtp tos (0), rtp port (1722), local type (account), call type (advanced), and dtmf (inband audio).
- phone settings:** Includes use dialplan (disable), dial number, idcode (86), idprefix (00), idprefix (0), inline (disable), local prefix (0), nonlocal prefix (0), answer (0), ring type (dtmf), use digitmap (unchecked), forward number (82378801), fed power-off (unchecked), fed nonuser (unchecked), fed always (unchecked), and fed busy (unchecked).
- audio settings:** Includes audio type (auto), audio frames (2), g.723.1 high rate (checked), vad (checked), ago (unchecked), sec (checked), handset in (5), handset out (19), and speaker out (18).
- other settings:** Includes password (1234), super password (19800211), debug (output all), sip ip (210.59.197.10), use daylight (unchecked), upgrade addr, and timezone (GMT+08:00)Beijing, Hong Kong, Urumqi).

A "Save/Reboot" button is located at the bottom of the configuration area.

or

The screenshot shows a "Smart IP Phone Setup page" with a red banner that reads "Welcome To Telephone service center". A sidebar on the left lists various settings categories: Network Settings, Protocol Settings, Phone Settings, Audio Settings, Basic Settings, Software Upgrade, and Address Book. The "Protocol Settings" section is currently active and displays the following configuration:

- Protocol: sip
- Use Service:
- Service Type: fwd
- Service Addr: fwd.pulver.com
- Service ID: fwd.pulver.com
- Nat Traversal: utun
- NAT Addr: stun.fwdnet.net:3478
- NAT Ttl: 30 Sec (1-254)
- Phone Number: 631445
- Account: 631445
- Pin: \*\*\*\*\*
- Register Port: 5060
- Signal Port: 5068
- Control Port: 5068
- Register Ttl: 3600 Sec (1-65534)
- RTP TOS: 0
- RTP Port: 10000
- JITTER SIZE: 0
- Local Type: phonenumber
- Call Type: advanced
- Dtmf: inband audio

At the bottom of the settings area, there are "OK" and "cancel" buttons.

After successful setting, the phone can be used to call other phones at the same Proxy Server as follows:

Pick up the handset or press Speaker key, and then input phone number, such as 10001, 10000, then press # or Call key to end dialing.

## Inphonex VoIP Service Provider

network settings			
ip type	dhcp	ppp id	
local ip	192.168.1.71	subnet mask	255.255.255.0
router ip	192.168.1.254	ppp pin	
dns	202.106.46.151	dns2	202.96.128.68
mac	00-09-45-64-02-ac		
protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	common	service addr	inphonex.com
service id	inphonex.com		
nat traversal	disable	nat addr	
nat ttl	240		
phone number	6266746	account	6266746
pin	1234		
register port	5060	signal port	1720
control port	1722		
dtmf	inband audio	dtmf payload	101
rtp port	1722		
local type	account	call type	advanced
rtp tss	0		
phone settings			
use dialplan	disable	dial number	
dddcode	86	dddprefix	00
dddprefix	0		
inner line	disable	innerline prefix	0
dual mode	disable	dualmode prefix	0
ring type	dtmf0	use digitmap	<input type="checkbox"/>
call waiting	<input type="checkbox"/>		
forward number	82378009	fed poweroff	<input type="checkbox"/>
fed noanswer	<input type="checkbox"/>		
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>
answer	30		
audio settings			
vad	<input checked="" type="checkbox"/>	agc	<input type="checkbox"/>
acc	<input checked="" type="checkbox"/>		
codec1	g7231	codec2	g729
codec3	g711a	codec5	gsm
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
speaker out (0-31)	30		
other settings			
password	1234	super password	12345678
debug	output all		
snmp ip	210.59.157.10	use daylight	<input type="checkbox"/>
upgrade addr			

After successful setting, the phone can be used to call other phones at the same Proxy Server as follows:

Pick up the handset or press Speaker key, and then input phone number, such as 537775, 600, then press # or Call key to end dialing.

# Stanaphone VoIP Service Provider

network settings			
ip type	dhcp	ppp id	
local ip	192.168.1.71	subnet mask	255.255.255.0
router ip		router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68
mac		mac	00-09-45-64-02-ac

protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	stanaphone	service addr	sip.stanaphone.com
service id		service id	sip.stanaphone.com
nat traversal	disable	nat addr	
nat ttl		nat ttl	240
phone number	9145099111	account	9145099111
pin		pin	1234
register port	5060	signal port	1720
control port		control port	1722
dtmf	inband audio	dtmf payload	101
rtp port		rtp port	1722
local type	account	call type	advanced
rtp tss		rtp tss	0

phone settings			
use dialplan	disable	dial number	
dddcode	86	dddcode	10
iddcode		iddprefix	00
dddprefix		dddprefix	0
inner line	disable	innerline prefix	0
dual mode	disable	dualmode prefix	0
ring type	dtmf0	use digitmap	<input type="checkbox"/>
call waiting		call waiting	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>
fed noanswer		fed noanswer	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>
answer		answer	30

audio settings			
vad	<input checked="" type="checkbox"/>	agc	<input type="checkbox"/>
acc		acc	<input checked="" type="checkbox"/>
codec1	g7231	codec2	g729
codec3	g711a	codec5	gsm
codec3		codec3	g711u
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
speaker out (0-31)		speaker out (0-31)	30

other settings			
password	1234	super password	12345678
debug		debug	output all
snrp ip	210.59.157.10	use daylight	<input type="checkbox"/>
upgrade addr		upgrade addr	

After successful setting, the phone can be used to call other phones at the same Proxy Server as follows:

Pick up the handset or press Speaker key, and then input phone number, then press # or Call key to end dialing.

# Seawolf VoIP Service Provider

The screenshot shows a configuration window for a VoIP service provider. The interface is organized into several sections, each with a title and a grid of settings. The settings include various fields for IP addresses, ports, service IDs, and audio parameters. The 'network settings' section includes fields for ip type, local ip, subnet mask, dns, and mac. The 'protocol settings' section includes fields for use service, register ttl, service addr, nat traversal, phone number, register port, dtmf, local type, and call type. The 'phone settings' section includes fields for use dialplan, iddcode, inner line, dual mode, ring type, forward number, fed always, dial number, iddprefix, innerline prefix, dualmode prefix, use digitmap, fed poweroff, fed busy, dddcode, dddprefix, call waiting, fed noanswer, and answer. The 'audio settings' section includes fields for vad, codec1, codec2, codec3, codec4, codec5, g.723.1 high rate, audio frames, handset in, handset out, and speaker out. The 'other settings' section includes fields for password, super password, debug, sntp ip, use daylight, and upgrade addr.

network settings			
ip type	static	ppp id	
local ip	192.168.1.71	subnet mask	255.255.255.0
dns	202.106.46.151	dns2	202.96.128.68
ppp pin		router ip	192.168.1.254
		mac	00-09-45-64-02-ac

protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	common	service addr	prepsip.xrainbow.com
nat traversal	stun	nat addr	64.0.51.125
phone number	95069200002	account	95069200002
register port	5060	pin	1234
dtmf	inband audio	signal port	1720
local type	account	control port	1722
		rtsp port	1722
		rtsp tso	0
		call type	advanced

phone settings			
use dialplan	disable	dial number	
iddcode	86	iddprefix	00
inner line	disable	innerline prefix	0
dual mode	disable	dualmode prefix	0
ring type	dtmf0	use digitmap	<input type="checkbox"/>
forward number	82378008	call waiting	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed poweroff	<input type="checkbox"/>
		fed noanswer	<input type="checkbox"/>
		answer	30
		fed busy	<input type="checkbox"/>

audio settings			
vad	<input checked="" type="checkbox"/>	age	<input type="checkbox"/>
codec1	g7231	codec2	g729
codec4	g711a	codec5	gsm
codec3		codec6	
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
		speaker out (0-31)	30

other settings			
password	1234	super password	12345678
sntp ip	210.59.157.10	use daylight	<input type="checkbox"/>
		debug	output all
		upgrade addr	

After successful setting, the phone can be used to call PSTN phone.

Pick up the handset or press Speaker key, and then input phone number, such as 01082378008, then press # or Call key to end dialing.

# SinoSip VoIP Service Provider

network settings			
ip type	dhcp	ppp id	
local ip	192.168.1.71	subnet mask	255.255.255.0
router ip	192.168.1.254	ppp pin	
dns	202.106.46.151	dns2	202.96.128.68
mac	00-09-45-64-02-ac		
protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	common	service addr	www.sinosip.net
service id	www.sinosip.net		
nat traversal	disable	nat addr	
nat ttl	240		
phone number	10082378888	account	10082378888
pin	1234		
register port	5060	signal port	1720
control port	1722		
dtmf	inband audio	dtmf payload	101
rtp port	1722		
local type	account	call type	advanced
rtp tss	0		
phone settings			
use dialplan	disable	dial number	
iddcode	86	iddprefix	00
ddcode	0	ddprefix	0
inner line	disable	innerline prefix	0
dual mode	disable	dualmode prefix	0
ring type	dtmf0	use digitmap	<input type="checkbox"/>
call waiting	<input type="checkbox"/>		
forward number	82378009	fed poweroff	<input type="checkbox"/>
fed noanswer	<input type="checkbox"/>		
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>
answer	30		
audio settings			
vad	<input checked="" type="checkbox"/>	age	<input type="checkbox"/>
acc	<input checked="" type="checkbox"/>		
codec1	g7231	codec2	g729
codec3	g711a	codec5	gsm
codec4		codec6	
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
speaker out (0-31)	30		
other settings			
password	1234	super password	12345678
debug	output all		
snmp ip	210.59.157.10	use daylight	<input type="checkbox"/>
upgrade addr			

After successful setting, the phone can be used to call other phones at the same Proxy Server as follows:  
Pick up the handset or press Speaker key, and then input phone number, then press # or Call key to end dialing.

## DeltaThree VoIP Service Provider

network settings			
ip type	dhcp	ppp id	
local ip	192.168.1.71	subnet mask	255.255.255.0
router ip		router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68
mac		mac	00-09-45-64-02-ac

protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	common	service addr	natrelay.deltathree.com
service id		service id	natrelay.deltathree.com
nat traversal	disable	nat addr	
nat ttl		nat ttl	240
phone number	21648888	account	21648888
pin		pin	12345
register port	5060	signal port	1720
control port		control port	1722
dtmf	inband audio	dtmf payload	101
rtp port		rtp port	1722
local type	account	call type	advanced
rtp tss		rtp tss	0

phone settings			
use dialplan	disable	dial number	
dddcode	86	dddprefix	00
dddcode		dddprefix	0
inner line	disable	innerline prefix	0
dual mode	disable	dualmode prefix	0
ring type	dtmf0	use digitmap	<input type="checkbox"/>
call waiting		call waiting	<input type="checkbox"/>
forward number	82378008	fed poweroff	<input type="checkbox"/>
fed noanswer		fed noanswer	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>
answer		answer	30

audio settings			
vad	<input checked="" type="checkbox"/>	age	<input type="checkbox"/>
acc		acc	<input checked="" type="checkbox"/>
codec1	g7231	codec2	g729
codec3	g711a	codec5	gsm
codec3		codec3	g711u
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
speaker out (0-31)		speaker out (0-31)	30

other settings			
password	1234	super password	12345678
debug		debug	output all
snmp ip	210.59.157.10	use daylight	<input type="checkbox"/>
upgrade addr		upgrade addr	

After successful setting, the phone can be used to call other phones at the same Proxy Server as follows:

Pick up the handset or press Speaker key, and then input phone number, such as 10001 then press # or Call key to end dialing.

## Typical Configuration for Service Providers Using H.323 Protocol

## G-Card

The screenshot shows a configuration window for a G-Card. The interface is organized into several sections:

- network settings:** Includes fields for ip type (dhcp), local ip (192.168.1.71), subnet mask (255.255.255.0), router ip (192.168.1.254), dns (202.106.46.151), dns2 (202.96.128.68), mac (00-09-45-64-02-ac), ppp id (test1), and ppp pin (test1).
- protocol settings:** Includes use service (checked), register ttl (124), jitter size (0), service type (common), service addr (211.72.238.147), nat traversal (citron), nat addr, nat ttl (240), phone number (XXXXXXXX), account, pin, register port (1720), signal port (1720), control port (1722), dtmf (control string), dtmf payload (101), rtp port (1722), local type (phonenumber), call type (advanced), and rtp tss (0).
- phone settings:** Includes use dialplan (disable), dial number, iddcodes (86), iddprefix (00), ddprefix (0), inner line (disable), innerline prefix (0), dual mode (disable), dualmode prefix (0), ring type (dtmf0), use digitmap, call waiting, forward number (82378009), fwd poweroff, fwd noanswer, fwd always, and answer (30).
- audio settings:** Includes vad, agc, acc (checked), codec1 (g7231), codec2 (g729), codec3 (g711u), codec4 (g711a), codec5 (gsm), g.723.1 high rate (checked), audio frames (2), handset in (0-15) (13), handset out (0-31) (20), and speaker out (0-31) (30).
- other settings:** Includes password (1234), super password (12345678), debug (output all), sntp ip (210.59.157.10), use daylight, and upgrade addr.

After successful setting, the phone can be used to call other phones at the same GK as follows:

Pick up the handset or press **Speaker** key, and then input phone number, such as 7313002,7313006, 1001, 2345, then press # or **Call** key to end dialing.

# iTalk

The screenshot displays the iTalk configuration interface, organized into several sections:

- network settings:** Includes fields for ip type (dhcp), local ip (192.168.1.71), dns (202.106.46.151), ppp id (test1), subnet mask (255.255.255.0), dns2 (202.96.128.68), ppp pin (test1), router ip (192.168.1.254), and mac (00-09-45-64-02-ac).
- protocol settings:** Includes use service (checked), register ttl (124), jitter size (0), service type (etalk), service addr (211.96.27.15:2008), service id, nat traversal (disable), nat addr, nat ttl (240), phone number (51800040), account (0119100801), pin (395679962271), register port (1720), signal port (1720), control port (1722), dtmf (control string), dtmf payload (101), rtp port (1722), local type (account), call type (advanced), and rtp tss (0).
- phone settings:** Includes use dialplan (disable), dial number, iddcodes (86), iddprefix (00), iddprefix (0), inner line (disable), innerline prefix (0), dual mode (disable), dualmode prefix (0), ring type (dtmf0), use digitmap (unchecked), call waiting (unchecked), forward number (82378009), fwd poweroff (unchecked), fwd noanswer (unchecked), fed always (unchecked), fed busy (unchecked), and answer (30).
- audio settings:** Includes vad (unchecked), agc (unchecked), acc (checked), codec1 (g7231), codec2 (g729), codec3 (g711u), codec4 (g711a), codec5 (gsm), g.723.1 high rate (checked), audio frames (2), handset in (0-15) (13), handset out (0-31) (20), and speaker out (0-31) (30).
- other settings:** Includes password (1234), super password (12345678), debug (output all), sntp ip (210.59.157.10), use daylight (unchecked), and upgrade addr.

After successful setting, those phones with digitmap can be dialed like ordinary phones as follows:

- 1) Place domestic calls: pick up the handset or press **Speaker** key, and then input "city code+phone number", such as a number from Shanghai 02158547153; after a few seconds, the number will be called out.
- 2) Place international calls: pick up the handset or press **Speaker** key, and then input "country code+city code+phone number", such as a number from U.S.A 0014089821818, or a number from Taiwan 0088627997222; after a few seconds, the number will be called out.

AS for those phones without digitmap, after above operation, please then press **#** or **Call** key to end the dialing

## IPN

The screenshot displays a network configuration interface with the following sections and values:

network settings			
ip type	dhcp	ppp id	test1
local ip	192.168.1.71	subnet mask	255.255.255.0
dns	202.106.46.151	dns2	202.96.128.68
ppp pin	test1	router ip	192.168.1.254
mac	00-09-45-64-02-ac		

protocol settings			
use service	<input checked="" type="checkbox"/>	register ttl	124
service type	common	service addr	XXX.XXX.XXX.XXX:19900
nat traversal	citron	nat addr	
phone number	51800040	account	0119100801
register port	1720	signal port	1720
dtmf	control string	dtmf payload	101
local type	cat account	call type	advanced
jitter size			0
service id			
nat ttl			240
pin			72783816
control port			1722
rtp port			1722
rtp tso			0

phone settings			
use dialplan	disable	dial number	
idcode	86	idprefix	00
inner line	disable	innerline prefix	0
dual mode	disable	dualmode prefix	0
ring type	dtmf0	use digitmap	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>
dddcode			10
dddprefix			0
call waiting			<input type="checkbox"/>
fed noanswer			<input type="checkbox"/>
answer			30

audio settings			
vad	<input type="checkbox"/>	age	<input type="checkbox"/>
codec1	g7231	codec2	g729
codec4	g711a	codec5	gsm
codec3			g711u
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2
handset in (0-15)	13	handset out (0-31)	20
speaker out (0-31)			30

other settings			
password	1234	super password	12345678
snrp ip	210.59.157.10	use daylight	<input type="checkbox"/>
debug			output all
upgrade addr			

After successful setting, the phone can be used to call other phones using IPN as follows:

Pick up the handset or press **Speaker** key, and then input phone number, such as **19905010005**, then press **#** or **Call** key to end dialing.

Those phones with ditmap can be dialed like ordinary phones as follows:

1) Place domestic calls: pick up the handset or press **Speaker** key, and then input "city code+phone number", such as a number from Shanghai 02158547153; after a few seconds, the number will be called out.

2) Place international calls: pick up the handset or press **Speaker** key, and then input "country code+city code+phone number", such as a number from U.S.A 0014089821818, or a number from Taiwan 0088627997222; after a few seconds, the number will be called out.

AS for those phones without digitmap, after above operation, please then press **#** or **Call** key to end the dialing

## eTalk

The screenshot displays the eTalk configuration window, organized into several sections:

- network settings:** Includes fields for ip type (dhcp), local ip (192.168.1.71), dns (202.106.46.151), subnet mask (255.255.255.0), dns2 (202.96.128.68), ppp id (test1), ppp pin (test1), router ip (192.168.1.254), and mac (00-09-45-64-02-ac).
- protocol settings:** Includes use service (checked), register ttl (124), jitter size (0), service type (etalk), service addr (202.83.204.213), service id (EINSGK), nat traversal (disable), nat addr, nat ttl (240), phone number (51800040), account (0117EINS), pin (409466119596), register port (1720), signal port (1720), control port (1722), dtmf (control string), dtmf payload (101), rtp port (1722), local type (account), call type (advanced), and rtp tso (0).
- phone settings:** Includes use dialplan (disable), dial number, iddcodes (86), iddprefix (00), iddprefix (0), inner line (disable), innerline prefix (0), dual mode (disable), dualmode prefix (0), ring type (dtmf0), use digitmap (unchecked), call waiting (unchecked), forward number (82378009), fwd poweroff (unchecked), fwd noanswer (unchecked), fed always (unchecked), fed busy (unchecked), and answer (30).
- audio settings:** Includes vad (unchecked), age (unchecked), acc (checked), codec1 (g7231), codec2 (g729), codec3 (g711u), codec4 (g711a), codec5 (gsm), g.723.1 high rate (checked), audio frames (2), handset in (0-15) (13), handset out (0-31) (20), and speaker out (0-31) (30).
- other settings:** Includes password (1234), super password (12345678), debug (output all), sntp ip (210.59.157.10), use daylight (unchecked), and upgrade addr.

After successful setting, those phones with digitmap can be dialed like ordinary phones as follows:

- 1) Place domestic calls: pick up the handset or press **Speaker** key, and then input "city code+phone number", such as a number from Shanghai 02158547153; after a few seconds, the number will be called out.
- 2) Place international calls: pick up the handset or press **Speaker** key, and then input "country code+city code+phone number", such as a number from U.S.A 0014089821818, or a number from Taiwan 0088627997222; after a few seconds, the number will be called out.

AS for those phones without digitmap, after above operation, please then press **#** or **Call** key to end the dialing

network settings					
ip type	dhcp	ppp id		ppp pin	
local ip	192.168.1.71	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-09-45-64-02-ac
protocol settings					
use service	<input checked="" type="checkbox"/>	register ttl	124	jitter size	0
service type	auvtech	service addr	xxxxxxxx	service id	
nat traversal	disable	nat addr		nat ttl	240
phone number	88801002	account	.apgw.palm.palm2	pin	
register port	1720	signal port	1720	control port	1722
dtmf	inband audio	dtmf payload	101	rtp port	1722
local type	account	call type	advanced	rtp ton	0
phone settings					
use dialplan	disable	dial number		ddcode	10
idcode	86	iddprefix	00	iddprefix	0
inner line	disable	innerline prefix	0		
dual mode	disable	dualmode prefix	0		
ring type	dtmf0	use digitmap	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>	fed noanswer	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>	answer	30
audio settings					
vad	<input checked="" type="checkbox"/>	age	<input type="checkbox"/>	acc	<input checked="" type="checkbox"/>
codec1	g7231	codec2	g729	codec3	g711u
codec4	g711a	codec5	gsm		
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2		
handset in (0-15)	13	handset out (0-31)	20	speaker out (0-31)	30
other settings					
password	1234	super password	12345678	debug	output all
snmp ip	210.59.157.10	use daylight	<input type="checkbox"/>	upgrade addr	

After successful setting, those phones with digitmap can be dialed as follows:

1) Call other device under the same system: pick up the handset or press **Speaker** key, and then input number of the other party, such as 88801001; after a few seconds, the number will be called out.

2) Place outline calls:

a) Call PSTN phones: pick up the handset or press **Speaker** key, and then input "country code+city code+phone number", such as 14089821818; after a few seconds, the number will be called out.

b) Call numbers in other area: pick up the handset or press **Speaker** key, and then input "country code+city code+phone number" ; such as a number from U.S.A 0014089821818 or a number from Taiwan 0088627997222; if the number is in China, please input "city code+phone number", such as a number in Shanghai: 02158547153; after a few seconds, the number will be called out.

AS for those phones without digitmap, after above operation, please then press **#** or **Call** key to end the dialing

# Asiasoft

network settings					
ip type	dhcp	ppp id		ppp pin	
local ip	192.168.1.71	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-09-45-64-02-ac
protocol settings					
use service	<input checked="" type="checkbox"/>	register ttl	124	jitter size	0
service type	asiasoft	service addr	220.202.10.180:1080	service id	
nat traversal	disable	nat addr		nat ttl	240
phone number	100010	account	100010	pin	8155
register port	1720	signal port	1720	control port	1722
dtmf	control string	dtmf payload	101	rtsp port	1722
local type	account	call type	advanced	rtsp tso	0
phone settings					
use dialplan	disable	dial number		ddcode	10
iddcode	86	iddprefix	00	ddprefix	0
isur line	disable	isurline prefix	0		
dual mode	disable	dualmode prefix	0		
ring type	dtmf0	use digitmap	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>	fed noanswer	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>	answer	30
audio settings					
vad	<input type="checkbox"/>	agc	<input type="checkbox"/>	aec	<input checked="" type="checkbox"/>
codec1	g7231	codec2	g729	codec3	g711u
codec4	g711a	codec5	gsm		
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2		
handset in (0-15)	13	handset out (0-31)	20	speaker out (0-31)	30
other settings					
password	1234	super password	12345678	debug	output all
snmp ip	210.59.157.10	use daylight	<input type="checkbox"/>	upgrade addr	

After logging on the phone can be used to call PSTN phones

## Ewoophone

The screenshot displays the Ewoophone configuration interface, which is organized into several sections:

- network settings:** Includes fields for ip type (dhcp), local ip (192.168.1.71), subnet mask (255.255.255.0), router ip (192.168.1.254), dns (202.106.46.151), dns2 (202.96.128.68), mac (00-09-45-64-02-ac), ppp id, and ppp pin.
- protocol settings:** Includes use service (checked), register ttl (124), jitter size (0), service type (common), service addr (202.96.137.24:2008), service id, nat traversal (disable), nat addr, nat ttl (240), phone number (51800040), account (51800040), pin (47448155), register port (1720), signal port (1720), control port (1722), dtmf (control string), dtmf payload (101), rtp port (1722), local type (auto), call type (advanced), and rtp tss (0).
- phone settings:** Includes use dialplan (disable), dial number, dddcode (10), iddcode (86), iddprefix (00), dddprefix (0), inner line (disable), innerline prefix (0), dual mode (disable), dualmode prefix (0), ring type (dtmf0), use digitmap, call waiting, forward number (82378009), fwd poweroff, fwd noanswer, fwd always, fwd busy, and answer (30).
- audio settings:** Includes vad, acp, acc (checked), codec1 (g7231), codec2 (g729), codec3 (g711u), codec4 (g711a), codec5 (gsm), g.723.1 high rate (checked), audio frames (2), handset in (0-15) (13), handset out (0-31) (20), and speaker out (0-31) (30).
- other settings:** Includes password (1234), super password (12345678), debug (output all), sntp ip (210.59.157.10), use daylight, and upgrade addr.

After successful setting, the phone can be used to call other phones using Ewoophone cards as follows:

Pick up the handset or press **Speaker** key, and then input phone number, such as **51825840**, then press # or **Call** key to end dialing

## Typical Configuration for Service Providers Using **MGCP Protocol**

network settings					
ip type	dhcp	ppp id		ppp pin	
local ip	192.168.1.71	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-09-45-64-02-ac
protocol settings					
use service	<input checked="" type="checkbox"/>	register ttl	128	jitter size	0
service type	common	service addr	211.56.89.4	service id	
nat traversal	disable	nat addr		nat ttl	240
phone number	21048888	account	ngcptest1	pin	ngcptest1
register port	2427	signal port	1720	control port	1722
dtmf	inband audio	dtmf payload	101	rtp port	1722
local type	account	call type	advanced	rtp tos	0
phone settings					
use dialplan	disable	dial number		ddcode	10
idcode	86	idprefix	00	ddprefix	0
inner line	disable	innerline prefix	0		
dual mode	disable	dualmode prefix	0		
ring type	dtmf0	use digitmap	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>	fed noanswer	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>	answer	30
audio settings					
vad	<input checked="" type="checkbox"/>	age	<input type="checkbox"/>	sec	<input checked="" type="checkbox"/>
codec1	g7231	codec2	g729	codec3	g711u
codec4	g711a	codec5	gsm		
e.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2		
handset in (0-15)	13	handset out (0-31)	20	speaker out (0-31)	30
other settings					
password	1234	super password	12345678	debug	output all
snip ip	210.59.157.10	use daylight	<input type="checkbox"/>	upgrade addr	

After successful setting, the phone can be used to call other phones at the same Call Agent as follows:

Pick up the handset or press Speaker key, and then input phone number, then press # or **Call** key to end dialing

## Typical Configuration for Service Providers Using **Other Protocol**

## Net2Phone

The screenshot displays the Net2Phone configuration interface, organized into several sections:

- network settings:** Includes fields for ip type (dhcp), local ip (192.168.1.71), subnet mask (255.255.255.0), router ip (192.168.1.254), dns (202.106.46.151), dns2 (202.96.128.68), mac (00-09-45-64-02-ac), ppp id, and ppp pin.
- protocol settings:** Includes use service (checked), register ttl (124), jitter size (0), service type (common), service addr (216.53.5.52), service id, nat traversal (disable), nat addr, nat ttl (240), phone number (21648888), account (4084506324), pin (3315), register port (5060), signal port (1720), control port (1722), dtmf (control string), dtmf payload (101), rtp port (1722), local type (account), call type (advanced), and rtp tss (0).
- phone settings:** Includes use dialplan (disable), dial number, iddcodes (86), iddprefix (00), iddprefix (0), inner line (disable), innerline prefix (0), dual mode (disable), dualmode prefix (0), ring type (dtmf0), use digitmap, call waiting, forward number (82378009), fwd poweroff, fwd noanswer, fwd always, fwd busy, and answer (30).
- audio settings:** Includes vad (checked), age, acc (checked), codec1 (g7231), codec2 (g729), codec3 (g711u), codec4 (g711a), codec5 (gsm), g.723.1 high rate (checked), audio frames (2), handset in (0-15) (13), handset out (0-31) (20), and speaker out (0-31) (30).
- other settings:** Includes password (1234), super password (12345678), debug (output all), sntp ip (210.59.157.10), use daylight, and upgrade addr.

After successful setting, those phones with ditmap can be dialed like ordinary phones as follows:

- 1) Place domestic calls: pick up the handset or press Speaker key, and then input "city code+phone number", such as a number from Shanghai 02158547153; after a few seconds, the number will be called out.
- 2) Place international calls: pick up the handset or press Speaker key; and then input number: dial U.S.A number, please input "1+city code+phone number", such as 14089821818; dial number in other countries, please input "country code+city code+phone number", such as a number from Taiwan 0088627997222; after a few seconds, the number will be called out.
- 3) Call other IP phone using Net2phone card: pick up the handset or press Speaker key; and then input "\*72+phone number", such as \*724081586324; after a few seconds, the number will be called out.

AS for those phones without digitmap, after above operation, please then press # or Call key to end the dialing

# IAX2

network settings					
ip type	dhcp	ppp id		ppp pin	
local ip	192.168.1.71	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-09-45-64-02-ac
protocol settings					
use service	<input checked="" type="checkbox"/>	register ttl	124	jitter size	0
service type	common	service addr	iax2.fwdnet.net	service id	
nat traversal	disable	nat addr		nat ttl	240
phone number	53775	account	53775	pin	1234
register port	4569	signal port	1720	control port	1722
dtmf	inband audio	dtmf payload	101	rtp port	30000
local type	account	call type	advanced	rtp tss	0
phone settings					
use dialplan	disable	dial number		dddcode	10
iddcode	86	iddprefix	00	dddprefix	0
inner line	disable	innerline prefix	0		
dual mode	disable	dualmode prefix	0		
ring type	dtmf0	use digitmap	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
forward number	82378009	fed poweroff	<input type="checkbox"/>	fed noanswer	<input type="checkbox"/>
fed always	<input type="checkbox"/>	fed busy	<input type="checkbox"/>	answer	30
audio settings					
vad	<input checked="" type="checkbox"/>	age	<input type="checkbox"/>	acc	<input checked="" type="checkbox"/>
codec1	g7231	codec2	g729	codec3	g711u
codec4	g711a	codec5	gsm		
g.723.1 high rate	<input checked="" type="checkbox"/>	audio frames	2		
handset in (0-15)	13	handset out (0-31)	20	speaker out (0-31)	30
other settings					
password	1234	super password	12345678	debug	output all
snmp ip	210.59.157.10	use daylight	<input type="checkbox"/>	upgrade addr	

After You can use service from [iax2.fwdnet.net](http://iax2.fwdnet.net) to test it. You can call 613 for echo test. For a incoming call test, go to [www.freeworlddialup.com](http://www.freeworlddialup.com), choose menu->advanced->iax, use the "call me" link, input your account and pin, the system will call you in a minute. Some parameters explanation:always select use service