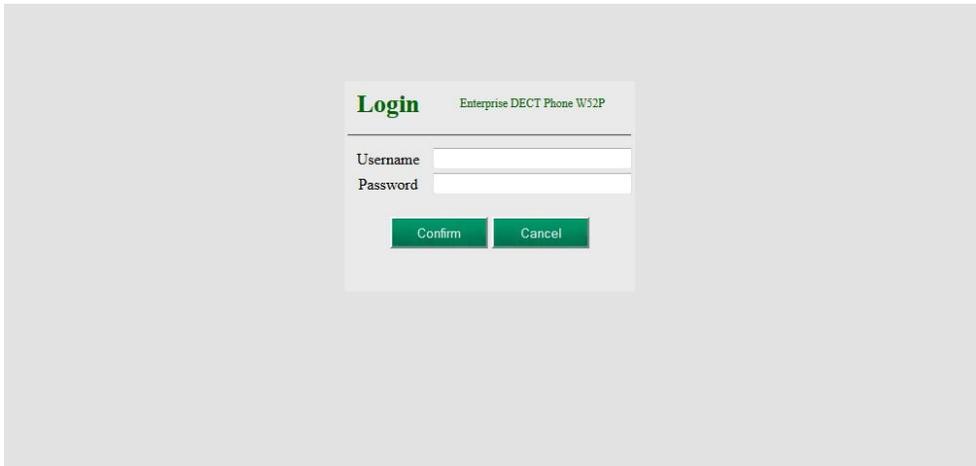How to Configure Yealink W52P:
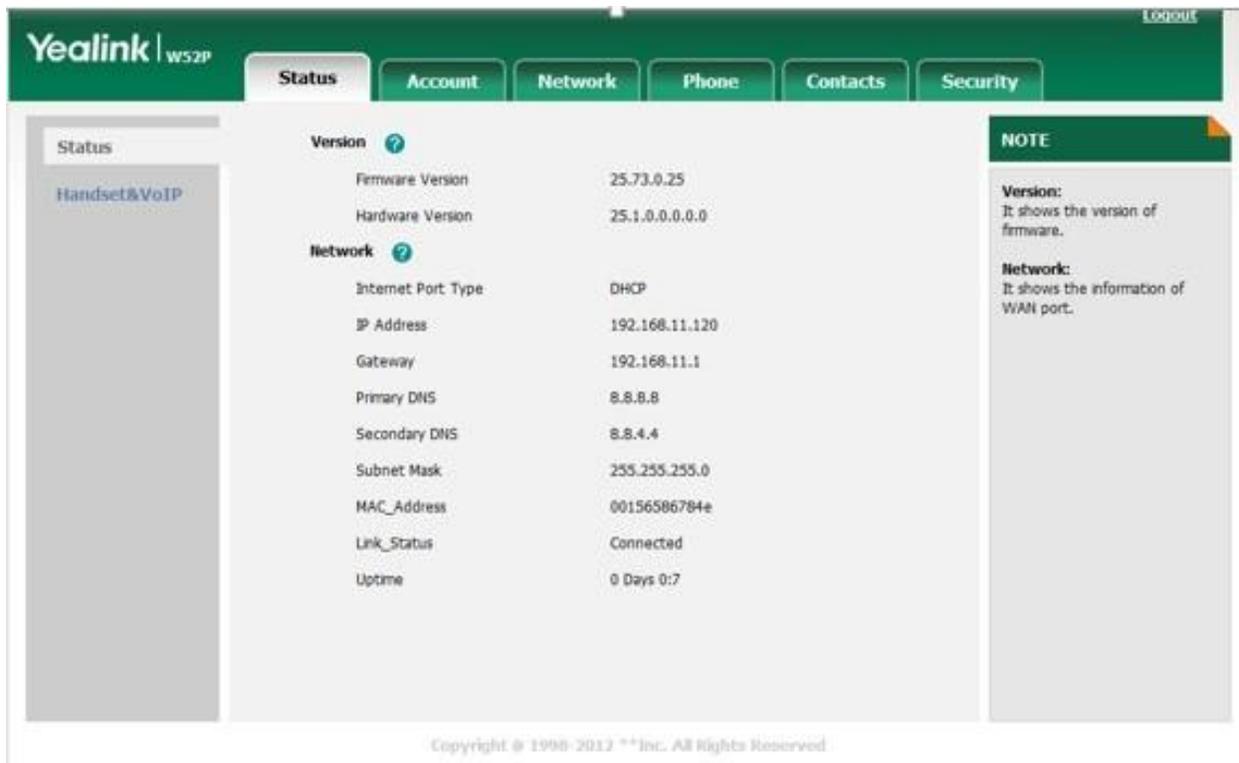
NOTE: A single Anyphone line has a maximum concurrent call capacity of 4 calls. Hardware capacity should also be taken into consideration. This means that Yealink W52P can handle up to have 4 concurrent calls.

To begin configuring your Yealink for use on Intermedia's network, you will need to pull up the graphical user interface. For this, do the following:

1) Press the center button on the phone
2) Select Status
3) Select Base
4) Enter the IP address listed in the account into a web browser
5) Log in using the credentials admin/admin

You will then be presented with a screen that looks like this:

Yealink | W52P   Logout

Status   Account   Network   Phone   Contacts   Security

Status

Handset&VoIP

Version
Firmware Version   25.73.0.25
Hardware Version   25.1.0.0.0.0.0
Network
Internet Port Type   DHCP
IP Address   192.168.11.120
Gateway   192.168.11.1
Primary DNS   8.8.8.8
Secondary DNS   8.8.4.4
Subnet Mask   255.255.255.0
MAC_Address   00156586784e
Link_Status   Connected
Uptime   0 Days 0:7

NOTE

Version:
It shows the version of firmware.

Network:
It shows the information of WAN port.

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The first thing that you want to do is update the firmware of the base unit to 25.73.0.40. You can find that
here: http://www.yealink.com/SupportDownloadfiles_detail.aspx?ProductsID=534&CateID=308&flag=1
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Once you have the file downloaded, simply upload it to the proper page, pictured below:

You will now need to navigate to the Account Page. You will need to enter in the SIP configuration information for your AnyPhone device on this page. These credentials can be found in your Hostpilot Account. The User Name and Server address will be whatever is found in your SIP Configuration for this device.

Auto attendants	SIP Configuration	SIP User Name ⑦	900123456
Lines	Outbound caller ID	SIP Authorization ID	900123456
Devices	Failover routing	SIP Password	**********
Hold music	911 address	SIP Domain	UC70.telecomsvc.com:6060
Groups	Call history	Outbound Proxy	UC70.telecomsvc.com:6060
	Groups	Type ⑦	Generic ▼
Order history		Preferred Codec ⑦	⦿ G.729 ◯ G.711
Additional settings		Secondary Codec ⑦	◯ G.729 ⦿ G.711 ◯ None

The Label and Display Name fields will be your Ext information. It will look like this when completed:

Yealink | ws2P

| Status | Account | Network | Phone | Contacts | Security |

Basic
Advanced
Codec
Number Assignment
Handset Name

Account	Account1 ▼ ❓
Register Status	Registered
Line Active	Enabled ▼ ❓
Label	323 ❓
Display Name	Ext 323 ❓
Register Name	900123456 ❓
User Name	900123456 ❓
Password	•••••••• ❓
Enable Outbound Proxy Server	Enabled ▼ ❓
Outbound Proxy Server	uc70.telecomsvc.com Port 6060 ❓
Transport	UDP ▼ ❓
Backup Outbound Proxy_Server	Port 5060 ❓
NAT Traversal	Disabled ▼ ❓
STUN Server	Port 3478 ❓
Proxy Require	❓
Anonymous Call	Disabled ▼ ❓
Anonymous Call Rejection	Disabled ▼ ❓
SIP Server 1 ❓	
Server Host	uc70.telecomsvc.com Port 6060 ❓
Server Expires	3600 ❓
Server Retry Counts	3 ❓
SIP Server 2 ❓	
Server Host	Port 5060 ❓
Server Expires	3600 ❓
Server Retry Counts	3 ❓

NOTE

Display Name
SIP service subscriber's name which will be used for Caller ID display.

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

NAT Traversal
Defines the STUN server will be active or not.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the value should be: com.nortelnetworks.firewall

Confirm   Cancel

After you have it looking like this, press the confirm button. The device will then attempt to register.

Now your device is registered! However, there is a little more configuration that needs to be completed:

Navigate to Accounts > Advanced and make these changes:

- Keep Alive Type = Default

- UDP Keep Alive Interval (seconds) = 30

- DTMF Type = RFC2833 or SIP INFO

- DTMF Payload = 101

- 100 Reliable Retransmission = Enabled

- SIP Registration Retry (seconds) = 30

- Local SIP Port: This needs to be set to 6xxx (Where xxx is equal to your Ext #) in this example we are configuring Ext 323 so we are setting the Local SIP Port to 6323. This allows each phone to use its own port and prevent call failures related to multiple requests for the same port.

- Caller ID Header: Set this to RPID-FROM

- Subscribe MWI To Voice Mail: Enabled

Yealink | W52P

| Status | Account | Network | Phone | Contacts | Security |

Basic

Advanced

Codec

Number Assignment

Handset Name

Account	Account1 ▼	❓
Keep Alive Type	Default ▼	❓
UDP Keep Alive Interval(seconds)	30	❓
Local SIP Port	6323	❓
RPort	Disabled ▼	❓
SIP Session Timer T1(seconds)	0.5	❓
SIP Session Timer T2(seconds)	4	❓
SIP Session Timer T4(seconds)	5	❓
DTMF Type	RFC2833 ▼	❓
How to INFO DTMF	DTMF-Relay ▼	❓
DTMF Payload(96~127)	101	❓
100 Reliable Retransmission	Enabled ▼	❓
Subscribe Register	Disabled ▼	❓
Subscribe for MWI	Disabled ▼	❓
MWI Subscription Period(0~84600)(seconds)	3600	❓
Caller ID Header	RPID-FROM ▼	❓
Use Session Timer	Disabled ▼	❓
Session Timer(seconds)		❓
Refresher	Uac ▼	❓
Use user=phone	Disabled ▼	❓
Voice Encryption(SRTP)	Disabled ▼	❓
Ptime(ms)	20 ▼	❓
Shared Line	Disabled ▼	❓
SIP Send MAC	Disabled ▼	❓
SIP Send Line	Enabled ▼	❓
SIP Registration Retry Timer(0~1800) (seconds)	30	❓
Subscribe MWI To Voice Mail	Enabled ▼	❓
Conference Type	Local Conference ▼	❓
Conference URI		❓

[Confirm]   [Cancel]

NOTE

Advanced
The Advanced parameters for administrator.

Please also remember to set the codec to that which is defined with you SIP credentials in the HostPilot portal. See the below Image, currently we only support G.729 and G.711u codecs. You will need to remove any other codecs listed in the "Enabled Codecs".

Next, go to Network > Advanced, and make these changes:

- Voice QoS = 46

- SIP QoS = 46

- Maximum RTP Port = 50000

- Minimum RTP Port = 30000

Almost there! Next, go to Phone > Preferences, and make the following change

- Set DHCP Time = Disabled
- Set the correct Time Zone
- Enable or Disable Daylight Saving Time based on your locations DST cycle (If you do not configure this correctly inbound calls will likely fail)

Now to configure your voicemail number, which is *the phone's direct number*, not numbers that forward to the phone. Go to the Phone menu, and make this change in the box at the bottom of the screen. It will look like this:

*** Note: You will need to put the same Voicemail Number in as the Busy and No answer Target. If you do not do this or forget to enable Busy and/or No Answer then calls will fail to go to Voicemail.

Finally, we need to associate the handsets and lines. Go to Account > Number Assignment. Currently Intermedia only support 1 Handset per line, and any other configuration may not be support by Intermedia Voice Support.

Now you are good to go! Please feel free to review the following articles for Firewall Rules and Port Requirements for the service to work correctly.

- https://kb.intermedia.net/Article/3119
- https://kb.intermedia.net/Article/3042