

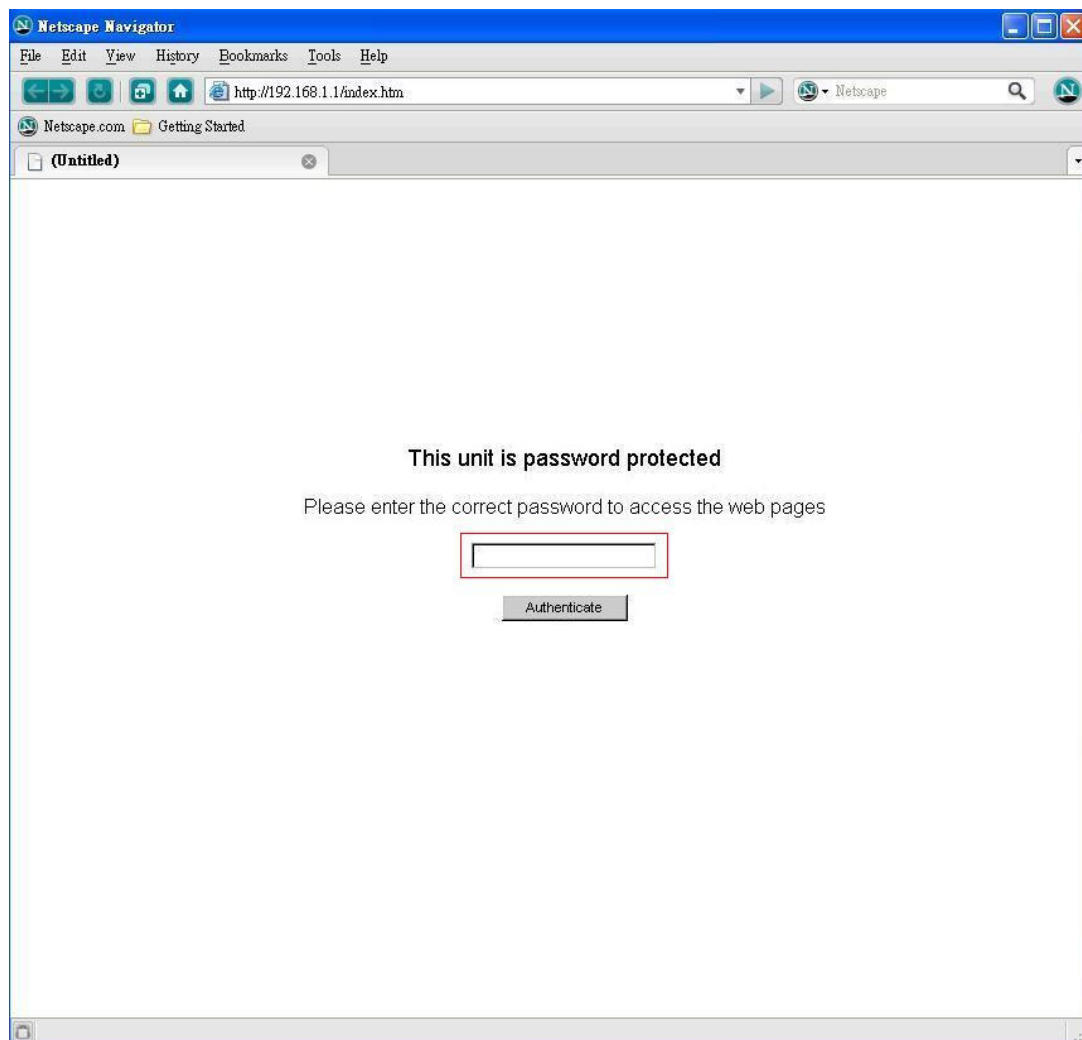
# VoIP-440S Setup example

1. Power on the VoIP-440S
2. Use the computer to connect the config port to setting VoIP-440S



The computer setting DHCP mode then will get the DHCP IP from VoIP-440S.

3. The default password is **root**



#### 4. Setting the SIP account info

The screenshot shows the OvisLink web interface. The top navigation bar includes links for SIP, SIP Extensions, OOB Signalling, ToS/DiffServ, and VLAN. The left sidebar contains links for Home, WAN, SIP, CODECS, System, Download, Logout, and Reset. The main content area is titled 'SIP Server Settings' and includes a status bar showing the current server, domain, and voice port. Below this, there are input fields for various SIP parameters, some of which are highlighted with red boxes. The 'Gateway Settings' section includes a dial plan field and three checkboxes for dial function settings. A table below lists four lines with their respective phone numbers, caller IDs, ports, AEC settings, user names, and passwords. The first line's user name and password are highlighted with red boxes. At the bottom, a note states that leaving a setting blank will force the unit to use information obtained via DHCP and/or DNS, and a 'Save SIP Settings' button is highlighted with a red box.

**OvisLink**  
The Total Networking Solutions

Home  
WAN  
**SIP**  
CODECS  
System  
Download  
Logout  
Reset

**SIP** SIP Extensions OOB Signalling ToS/DiffServ VLAN

**SIP Server Settings** (Current Server: sip.voipbuster.com : 5060 ; Domain: sip.voipbuster.com; VoicePort: 5060)

\* Server Address: sip.voipbuster.c (IP or FQDN)  
\* Port: 5060  
Domain Name: sip.voipbuster.com  
Voice Port: 5060  
☒ Send Registration Request with Expire Time 300  
Outbound Proxy IP: (IP or FQDN)  
Outbound Proxy Port:  
Stun Server IP: stun.voipbuster.. (IP or FQDN)  
Stun Server Port: 3478

**Gateway Settings**

Dial Plan:

☐ # use as a quick dial function  
☐ To enable # to be recognized as dial number  
☐ To enable \* to be recognized as dial number

	Phone Number	CallerID Name	Port	AEC On	User Name	Password	Dial-in Pla
Line1:	201	201	5060	OFF	markairlive	.....	
Line2:	202		5061	ON			
Line3:			5062	ON			
Line4:			5063	ON			

\* Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS

Save SIP Settings

5. Setting the codec info

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## CODECS

### Audio/CODEC Configuration

Selected	Silence Suppression
<b>G711U</b>	OFF
<b>G711A</b>	OFF
<input type="checkbox"/> G723	OFF
<input type="checkbox"/> G726	OFF
<input checked="" type="checkbox"/> <b>G729</b>	ON

**Packetization** 10ms

**Jitter Buffer**

☒ Adaptive Jitter Buffer: 100ms (maximum playout delay in milliseconds)  
☐ Fixed Jitter Buffer: 40ms (fixed playout delay in milliseconds)  
☐ Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection

Save CODEC Configuration

6. After save and reboot, then start to make VoIP calls.