Ref #'s RSZ03112210-1&2 94 SAYFADIR / PAGES

RSZ03112210-3

RSZ0311210-16

BITTEL IP TELEFONLAR - PHONES

THE ENGLISH USERs MANUAL
IS IN THE FOLLOWING PAGES

INGILIZCE KULLANIM KLAVUZU,
DEVAM EDEN SAYFALARDADIR

Modeller;
HA9888(68)TSD-IP
HA9888(67)TSD-IP
HA9888(66)TSD-IP

Asagida ki tanim ve tarifler,
cercek urunden az da olsa farkilik gosterebilir
Lutfen kullanimda gercek urunu esas aliniz.
Telefon ozellikleri, not verilmeden degisebilir.

Once lutfen Kullanim Klavuzu'nu okuyun
Cihazi dogru kullanmak ve kullanima
hazir duruma getirmek icin lutfen bu klavuzu okuyun.

Otel / Moteller icin iyi seyler yapiyoruz ..

Genel Bakis

SIP kelimesi, "Session Initiation Protokol" terminin bas harflerinden olusmaktadir. SIP telefon, "IP protokol tabanli iletisim agi"nda ses'in iletisim isini yapar. Aglara ornekler, LAN (Local Area Network), MAN

(Metropolitan Area Network) ve Internet. Bittel SIP IP telefon ailesinin, ozel otel istekleri, ustun ses kalitesi, kolay kullanim, ekonomik fiziksel tasarim, gelismis servisleri ile iletisim alanina onemli katkilari olmustur.

İslev ve ozellikler

2.1. Donanim

- CPU: BCM1190/DRAM:8MX16 bit / 4MB Flash hafiza
- Ethernet Port—2X 10/100M baglantilar
- AC/DC adaptor –Giris AC110-230V, Cikis DC 5V1A or POE
- Guc degisimi---1.5W (pasif modda)/ 1.8W (aktif)
- Ortam

Calisabildigi ortam: -0 to 50° C (32° to 122° F)
Depolama ortami: -20° to 65° C (-4° to 149° F)
Nem: 10 to 65%

2.2. Yazilim

- Otomatik IP adresi ve diger parametreleri otomatik. almak icin DHCP destegi
- Web'ten parametre ayarlari
- HTTP uzerinden firmware'in otomatik termin ve konfigurasyonu
- HTTP/FTP uzerinden otomatik guncelleme;
- Coklu Audio Codec destegi: G.711A/u, G.723.1; G729a/b; G.722 aritmetik ses kodu
- VAD (Voice Activity Detection) ,CNG (Comfort Noise Generation), Dynamic Jitter Buffer
- G. 168 96ms Echo iptali
- ITU-T stveard sinyalleme tonu ve DTM cikarim ve algilama
- DTMF Iletimi: dahili (ic band) audio; RFC2833; SIP INFO
- Firewall ve gonderim'de NAT destegi
- Arama yonlendirme / bekleme
- 500 girislik Telefon Kitabi / Kisileri Hizli Arama destegi (0≤N≤10)
- Ses ayari; ahize, hoparlor, muzik
- Hot Line

2.3. Standart ve Protokol

- SIP (RFC2833; RFC3261; RFC2543; RFC2327; RFC3264; RFC3265; RFC3550)
- IEEE 802. 3 /802. 3 u 10 Base T / 100Base TX
- IEEE 802. 1P/Q Tag VLAN
- QoS: Destek Cos Layer3 Qos (Diff-Serv) ve Tos Layer 2 Qos (802. 1P/Q)

- TCP/IP: Transmission Kontrol Protokolu / Internet Protokolu
- CMP: Internet Kontrol Mesaj Protokolu / Syslog: The BSD syslog Protokolu
- RTP: Real-time Transport Protokolu / RTCP: Real-time Transport Kontrol Protokolu
- DHCP: Dinamik Host Konfigurasyon Protokolu
- VAD/CNG: Saving Broadbve
- DNS: Domain Isim Sunucu / TFTP: Trivial Dosya Transfer Protokolu
- HTTP: Hyper Text Transfer Protokolu /SNTP: Basit Network Time Protokolu

Kurulus

3.1.Parca Listesi

Kurulum
Kurulumdan once parcalari kontrol edin. Eksik varsa bayiyi arayin.
1) SIP Ana Kutu
2) Ahize
3) Ahize kordonu
4) Universal Guc Adaptoru (sadece adaptor guc destegi icin)
5) Turkce ve Ingilizcde bir kullanim kitapcigi
7) Ethernet Kablosu
8) Vida torbasi (BT-2008(67-S/T-A icin)
9) POE kablo (BT-2008(67/68 icin)

1.2 Guc ve ag baglantisi

3.2.1 66IP/67IP/68IP/69IP

Ahize kordonunu takin; 3.3'e bakin
Ag baglantisi
Adaptor gucu : POE kablonun RJ45 ucunu HUB switch'in LAN port'una takin
ve diger ucunu da, telefonun WAN port'una takin.

POE'den guc : Ethernet kablonun ucunu telefonun WAN port'una takin ve diger ucunu da, HUB switch'in LAN port'una takin
Normal net mod DHCP 'dir, Bakin 5.3.
Telefonda Guc :

Adaptor gucu：Adaptorden cikan ucu POE kablonun adaptor giris prizine takin.
POE Modu ：Ethernet kablonun ucunu telefonun WAN port'una takin
ve diger ucunu da Pbx'in POE protuna takin

3.2.2 IP telefonun baglantisi

Ahize kordonunu takin
Ag baglantisi: Ethernet kablonun ucunu telefonun WAN port'una takin ve ve diger ucunu da,
HUB switch'in LAN port'una takin, Bakin 5.3.
Telefonda Guc;
Adaptor gucu：Adaptorden cikan ucu POE kablonun adaptor giris prizine takin.
POE Modu ：Ethernet kablonun ucunu telefonun WAN port'una takin
ve diger ucunu da Pbx'in POE portuna takin

3.4 Ses Kontrolu ve anten baglantilari (model 68)
Arkadan cikan kablo ucunu, ana govdenin “ **AUDIO CONNECT** ” girisine dikkatlice takin

3.5 Miknatisli alanlarindan uzak durun, miknatisli tornavida kullanmayin

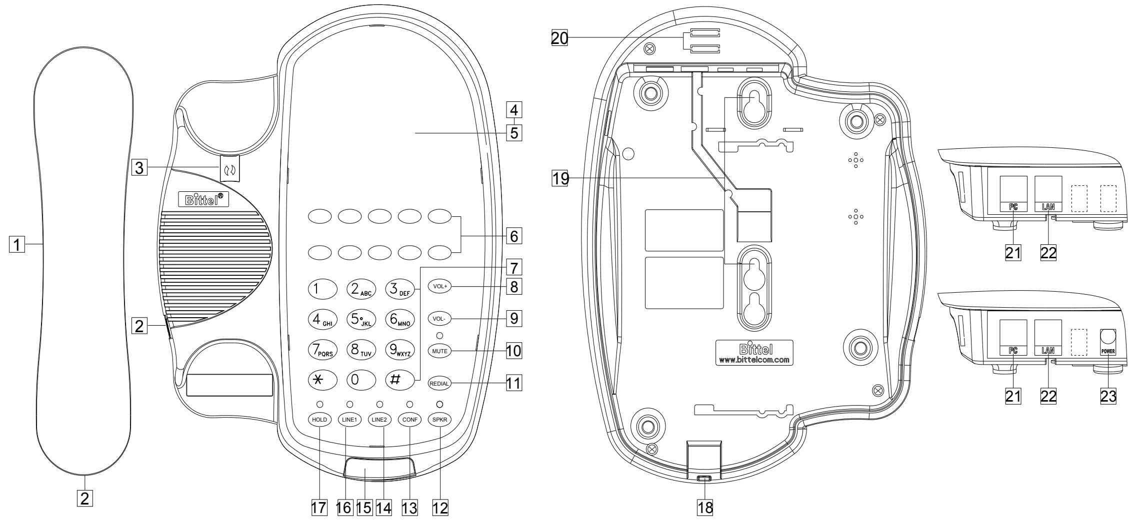
3.6 Kurulumu kuru ve temiz mekanda yapin

3.7 Telefonun calisacagi bolgede, “kisa devre, akim soku, yag, su akmasi olmamasi”na dikkat edin

Yer ve Kontroller:

4.1. Gorunus

(**IP-S/T**) SIP On Yuz (Bakin Fig 4.1.1)



4.1.1 58 IP

1. Ahize **2. Ahize kordonu giris** **3.Duvar montaj klibi**

4. On karton **5. Plastic overlay** **6. Hafiza tusu** **7. Digital tusu**

8. VOL+ **9. VOL-** **10. Kes tusu** **11. Tekrar Cevir tusu**

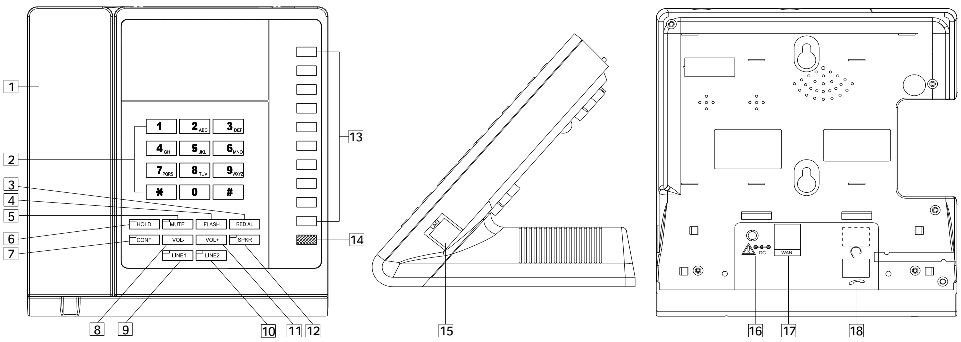
12. SPKR tusu **13. CONF tusu ((IP-T) model)** **14. HAT 2 tusu ((IP-T) model)**

15. Message tusu **16.HAT 1 tusu ((IP-T) model)** **17. Tut tusu** **18. MIC**

19. Duvar kablo gidis oyugu **20. Placard oyugu** **21. PC port**

22. LAN port **23. Guc**

(67IP-S/T) SIP On yuz (Bakin Fig 4.1.2)



4.1.2 (67IP-T-)/ (67IP-S)

1. Ahize 2. Sayisal tuslar 3.Tekrar Cevir tusu 4. Flash tusu

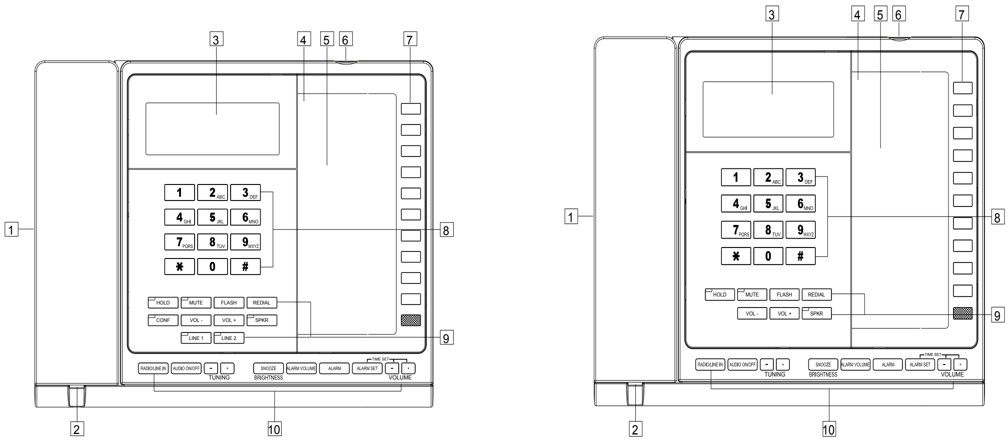
5. Mute tusu 6. Tut tusu 7. Konferans tusu ((67IP-T)*model*) 8. VOL-

9.LINE 1 tusu (67IP-T) model) 10. LINE 2 tusu ((67IP-T) model)

11. VOL+ 12. SPKR tusu 13. Hafiza tusu 14.MWL tusu 15.. LAN port

16. Power port 17. WAN port 18. Ahize kordon girisi

(68IP-S/T) SIP Telefon On yuz (Bakin Fig 4.1.3)



4.1 .3 68IP-T /68IP-S

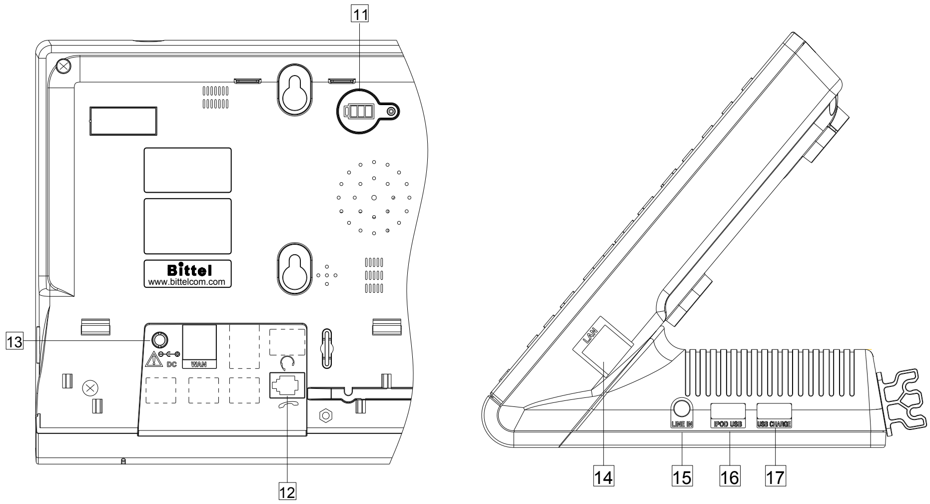
1. Ahize 2. Digital tusu 3. LCD 4. On karton 5 On Karton 2

6. Kagit Tutucu 7. Hafiza tusu 8. Digital tusu

9. TeleTelefon function tusu --- Tut tusu / Mute tusu /Flash tusu / Tekrar Cevir tusu / Konferans tusu ((68IP-T)*model*) / VOL- / VOL+ / SPKR tusu / LINE 1 tusu ((68IP-T)*model*) / LINE 2 tusu, (68IP-T) 'de

10. Ses tusu

(68IP-S/T) SIP On yuz (Bakin Fig 4.1.4)

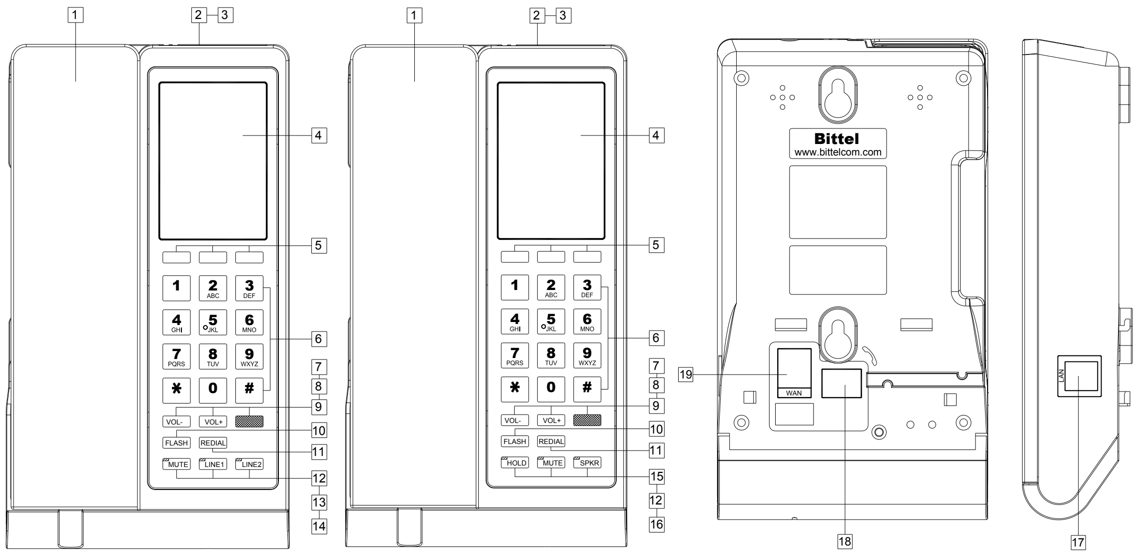


4.1 .4 68IP-T/ 68IP-S

11. Pil yuvasi 12. Ahize kordon girisi 14. LAN port 15. LINE IN port

16. IPod USB port 17. USB SARJ portu

(66/69IP-S/T) SIP On yuz (Bakin Fig 4.1.5)



1. Ahize 2. On karton 3. Karton tutucu 4. Arcylic yuz 5. Hafiza tusu 6. Sayisal tuslar

7. VOL- tusu 8. VOL+ tusu 9. Mesaj uyari isigi 10. Flash tusu 11. Tekrar Cevir tusu

12. Ses kesme tusu 13. LINE 1 tusu (sadece 2 hatli model), 14. LINE 2 tusu (sadece 2 hatli model)

15. Tut tusu (sadece 1 hatli model) 16. SPKR tusu

17. LAN Portu 18. Ahize giris 19. WAN Port

4.2. Fonksiyon tuslari

Tus	Ne ise yarar ?
HIZLI CEVIRME TUSLARI	Ahize elde, tek dokunusla karton uzerinde ki otel servisileri, otomatik cevrilir.
MUTE	Gorusme aninda, tusa basildiginda ses kesilir, tekrar basildiginda acilir.
VOL+	Her basista, ahizede veya hoparlorde ki ses yukselir
VOL-	Her basista, ahizede veya hoparlorde ki ses alcalir
SPEAKER	Bu tus, “ahizesiz gorusme” ozelligini acar veya kapatir
TUT	Gorusmeyi askiya alir, tekrar basildiginda askidan cozer
FLASH	Gorusme altinda catalti yapar (ahize klipsinin tiklanmasi)
MESSAGE	Bu isik, ayni zamanda tus'tur ve sesli posta servisini cevirir.
REDIAL	Son numara tekrar cevrilir.
CONF	2 hat devrede iken, bu hatta ki 2 kisiyi, 3 lu gorusmeye alirsiniz
LINE 1	Hat 1 aktive olur.
LINE 2	Hat 2 aktive olur.

4.3 Ses portlari fonksiyonu

● LINE IN portu—— iPod, MP3 ve diger portlar icin
● iPod USB port—— iPod, MP3 ve digerrleri akim sarji icin
● USB Charge port —— Diger mobil ektrik komponentler akim sarji icin

4.4 LED Fonksiyonlari

SPKR LED	Acik: Ahizesiz gorusme devrede
MESSAGE LED	Acik: Dinlememis mesajlar var Kapali: Yok
HOLD LED	Acik: Gorusme Tut'ta
MUTE LED	Acik: Dis sesler gitmiyor
HAT 1 LED	Acik: Hat 1 acik
HAT 2 LED	Acik: Hat 2 acik
CONF LED	Acik: 3 'lu gorusme

Web 'te Konfigurasyon

Sifre tanimi

2 tip web kullanim alani vardir; misafir girisi, yonetici girisi. Yonetim girisinde herseye hakimsiniz. Misafir girisi herseyi gorur ancak Sunucu, **adres ve port, SIP(1-2)** kalemlerini degistirebilir.

Girisler (birbirinde ayri olmalidir):

- Kullanici Modu
  - ◆ User name：misafir
  - ◆ Pass word：misafir

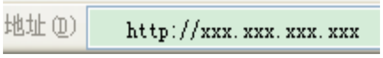
- Yonetici Modu：
  - ◆ User name：yonetici
  - ◆ Pass word：yonetici

Not: Gizli tuslarin ayarlari icin, onceden tanimlanmis sifre :123

5.2. Web browser icin konfigurasyon

Telefon ve PC network'e baglandiginda, PC'de ki logo'yu tiklatin ve Explorer'i acin ;

(yerel bir IP almak icin, telefonda “MUTE” tusuna basin).



Web logon'nunda, telefon konfigurasyonu, standart port 80 degilse,

http://xxx.xxx.xxx.xxx : xxxx/, girin, aksi takdirde kullanici Sunucu'yu bulamaz.

Asagida ki ekrandan sisteme giris yapin.

Logon arayuzu

Not:

1. Her degisiklikten sonra kayit icin ,【APPLY】'a basin
2. Kopruleme (bridging) yok ise, set'lerin port adresleri soyledir:
   IP : 192.168.10.1
   66IP, 67IP, 68IP ve 69IP : 192.168.10.1 Bakin 5.3.2.2 .
3. 1 hatli telefonun set'i **SIP1 veya SIP2** 'dir. oncelik **SIP1** 'dir.
4. Bu arayuzun WAN port'u tanimida, IP icin ilgili port, LAN portu,
   66IP/67IP/68IP/ 69IP icin ilgili port, WAN portudur.
   Bu arayuzun LAN port'u tanimida, IP icin ilgili port, PC portu,
   66IP/67IP/68IP/ 69IP icin ilgili port, LAN portudur

Not: Fabrika cikis network cikis ayarlari, DHCP'dir. Kullanici destek ortami DHCP olan network'lere bagladiginda, set, otomatik olarak network'e baglanir. DHCP ortami yoksa, PPPoE veya Static IP modu ayarlanmalidir.

Hafiza tuslari :

1）Basin, 【1】tusuna min. 10s, telefon, Static IP mode'a donecektir.
2）Basin, 【2】tusuna min. 10s, telefon, DHCP mode'a donecektir.
3）Basin, 【3】tusuna min. 10s, telefon, PPPoE mode'a donecektir.

Not: Network'u sectikten sonra, IP adres, Ag Maskesi, Baz Router Gateway ve DNS'in ayarlari, WAN alt seviye menusunde olabilir. ***Bakin*** **5.3.2。**

5.1 Web fonksiyonel tanimlamalar

5.1.1 Ana Ayarlar

5.1.1.1 Durum goruntusu

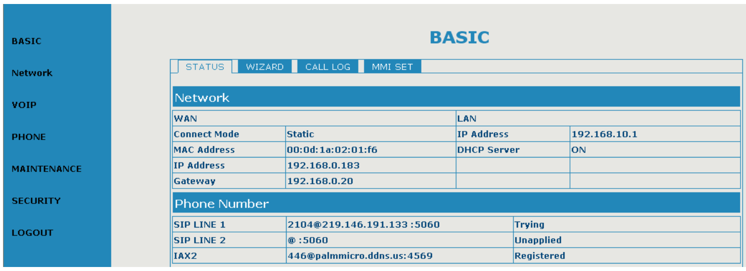


Fig. 5.3.1.1 Durum goruntusu

Network	WNA ve LAN'in o an ki durum goruntusu: Dahil olan, WAN IP（Static, DHCP, PPPoE） ve IP adres, MAC adres, Pre-ayari Gateway IP adres, LAN, IP adres, DHCP Sunucu ON/OFF modunda..+
Telefon Numarasi	SIP LINE 1－2 ve IAX2 'nin o an ki durumunu gosterir. En alt satirda, telefonun versiyon ve kullanim tarihi gosterilir.

5.1.1.2 Konfigurasyon Rehberi

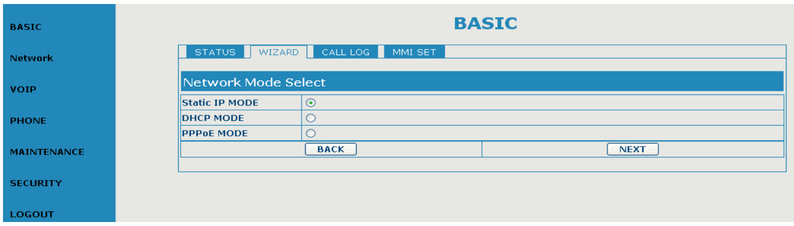


Fig. 5.3.1.2-1 Network Modu

Static IP MODE	SIP tedariki statik IP adresi verdiginde bunu secin. Secimden sonra, kullanici sunlari doldurmalidir; static IP adresi / subnet mask'i / gateway/main domain name, v.s. . Sayet bilmiyor ise, Sunucu tedarik yetkilisi veya network yonetim sorumlusuna basvurabilir.
DHCP MODE	Bunu sectiginizde, DHCP Sunucudan bilgi ototmatik alinir, girise gerek yoktur.
PPPoE MODE	Bu secildiginde, kullanici ADSL hesabi ve sifte girmelidir. Bakin in **5.2.**
Not: Network ortamina gore, gerekli uygun modu secin. Statik IP mod secildiginde, (Hat 1 default) kalemini ayarlamak ve ayar kalemlerini gormek icin, 【NEXT】'e basin. Once ki menuye donmek icin 【BACK】'a basin.	

1） Static IP Ayarlari

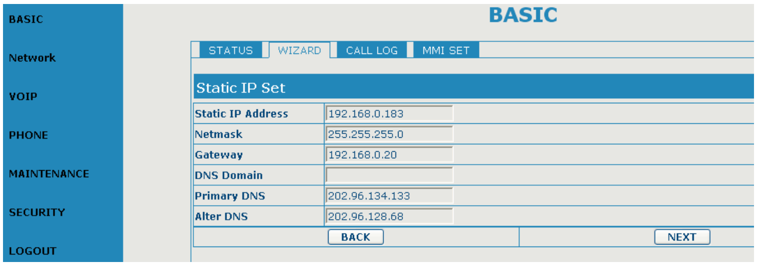


Fig. 5.3.1.2-2 Static IP Set'i

Static IP adresi	Dagitik IP adres'i girin
Net mask	Dagitik subnet mask girin
Gateway	Dagitik on-ayar gateway adresi girin
DNS Domain	“DNS domain” isim eklenti tesisi. Kullanici bir Domain isim girer ve DNS cozumlenmezse, analiz icin; telefon set'i domain isim adresi'nden sonra, bir domain isim ekler.
Birincil DNS	Ana DNS Sunucu adres'i girin
Sonra ki DNS	Yedek DNS Sunucu adres'i girin

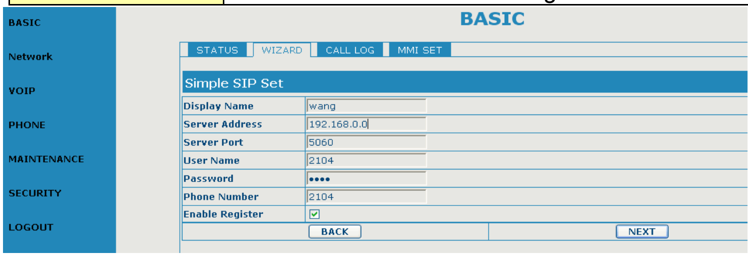


Fig. 5.3.1.2-3 SIP Hesap Ayarlari

Goruntu Ismi	Set edildikten sonra, aranan, arayani gorebilir. Ingilizce harfler mevcuttur.
Sunucu adres	Sunucu adresi, SIP kayit tanimlamalari, domain name gibi adresleri destekler.
Sunucu Port'u	Sunucu sinyal portu, SIP kayit tanimlamalari,
Kullanici Ismi	Hesabin SIP kaydi.
Sifre	Hesap Numarasi ve sifrenin SIP kaydi.
Telefon Numara	Numaranin SIP Sunucu'ya kaydi
Kayit Etkinlestirme	Izinli ve yasakli kayit ayarlari.

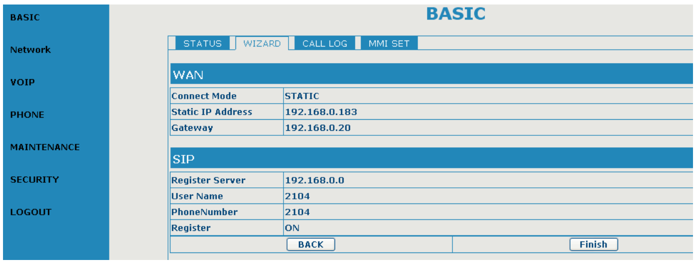


Fig 5.3.1.2-4 Statik Ayar Bilgisi

5.3.1.2-4 Statik Konfigurasyonun ayarinin el ile nasil yapilacagini gosterir,【Finish】'e basin.

2）DHCP Ayari

DHCP modu secin, SIP parametreleri ve onlari gormek icin【NEXT】'e basin.

Once ki sayfaya donmek icin【BACK】'e basin

3） PPPoE Konfigurasyonu

PPPoE modu secin, On line aygitin SIP parametreleri ve Hesap No. , sifre almak icin icin【NEXT】'e basin. Once ki sayfaya donmek icin【**BACK**】'e basin. Statik IP mod'ta ki gibi.

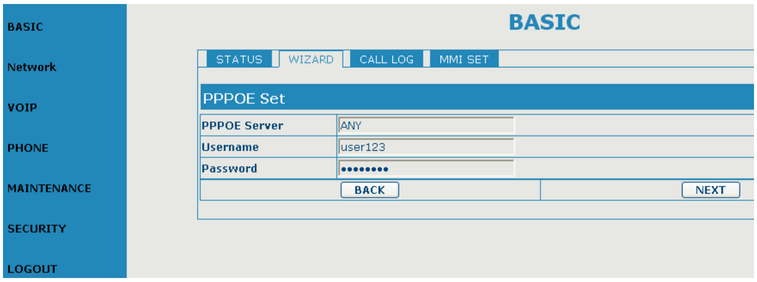


Fig . 5.3.1.2-5 PPPoE Sunucu

PPPoE Sunucu	Sunucu ismi, sayet PPPoE tedarikcisinin ozel bir istegi yok ise, genel pencere ile acilir.
Username	ADSL hesap numarasi girisi.
Password	ADSL sifre girisi.

Not： Set islemi bittikten sonra,【APPLY】tusuna basin, sonra telefon gerekli ayarlari kayit ve reboot edecektir, tekrardan sonra arama icin kayit olmus numaralar kullanilacaktir.

5.1.1.3 Arama Bilgisi

Bu sayfada aranan bilgisi (numaralar) gorunur.

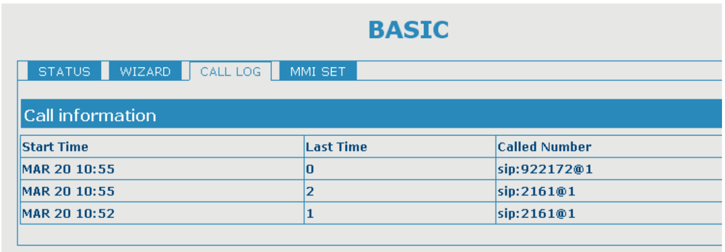


Fig.5.3.1.3 Aranmis Bilgisi

Baslangic Zamani	Baslangic Zamani'ni kayit eder
Son Zaman	Arama zamani, ikinci bir unitedir
Arama Numarasi	Kayitli hesap numarasi, arama protokolu ve kullanilmis hat.

5.1.1.4 MMI Ayari

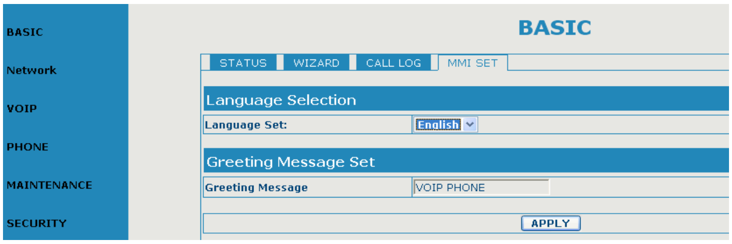


Fig. 5.3.1.4 MMI Ayari

Dil Secimi	"Gorunum dili" secmek icin, standart : Ingilizce
Selam Bilgisi	Telefonun LCD ekraninda ki "merhaba" mesajidir. 16 Latin alfabesi ve 5 Cin alfabesinden olusur. Sandart mesaj: VOIP PHONE.

5.1.2 Network Ayari

5.1.1.5 WAN

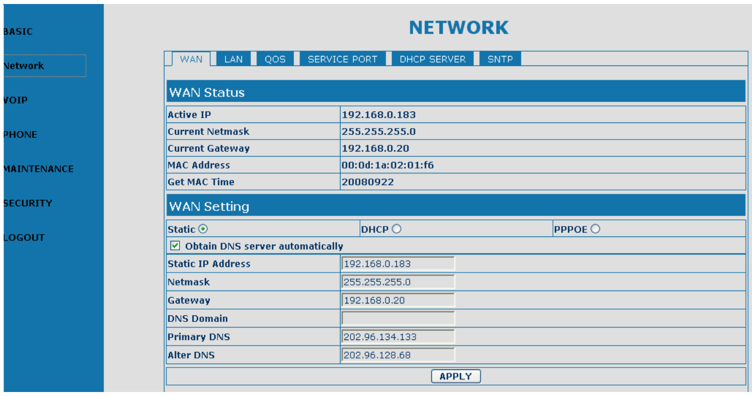


Fig 5.3.2.1 WAN Ayari

WAN Durumu	
Aktif IP	Aktif telefon IP ;
O anki Network	Aktif arka-web maskesi ;
O anki Gateway	Aktif on-ayar gateway IP'si ;
MAC adres	MAC adres ;
MAC Zamani almak	MAC adresi alim tarihi
WAN Ayari	
1) Wan Ayari : Gercek ortam icin uygun network mod'unu secin. Bakin 6.3.1.2. **2**) Bundan sonra, asada ki web sayfasi 5.3.1.2. 'nin detay ayarlarini gosterecektir **3**) IP degistigi zaman Web sayfasi cevap vermez, bu arada telefona baglanmak icin yeni bir IP adresi girilmesi gereklidir. **4**) Network ID'si; dagitilan DHCP Sunucu ile sistemin LAN'i tarafindan kullanilan ayni network ID'sinde ise, telefon, WAN'i belirlemek icin DHCP IP'yi kullanacaktir.	

Telefon, start up'ta IP almak icin DHCP client'i kullandigi anda, ayni telefon LAN network ID'sini modifiye edecektir. Telefon, WAN'i belirlemek icin IP kabul'unu su durumda red eder;
Sayet IP calisma durumu ve network ID'si alma isi'nde DHCP client'i kullanmada, LAN ayni ise.

5.2.2.2 LAN Ayari

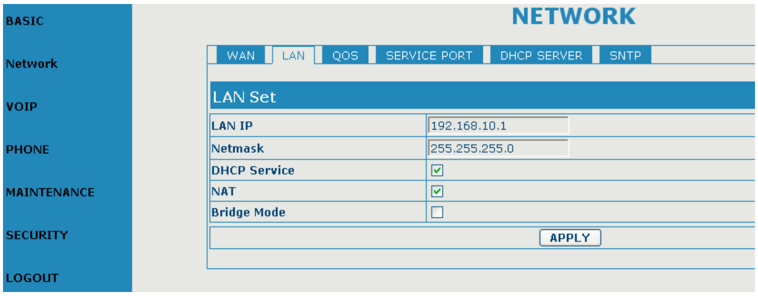


Fig. 5.3.2.2 LAN Ayari

LAN IP	LAN static IP ayari icin
Netmask	LAN Subnet mask ayari icin
DHCP Servisi	Aktif LAN end DHCP Sunucu. Telefon DHCP kiralama form'unu modifiye edecek, IP 'ye gore ayarlari kayit ecektir. Kayitlar suna gore olacaktir; LAN IP 'nin, subnet mask ve IP tanimindan sonra. Kullanici, DHCP servis ayarlarinin gecerli olmasi icin PC'yi yeniden baslatmalidir.
NAT	Aktif NAT。
Kopruleme Modu	Kullanmak; bu islem telefonu, PC port icin IP adres ayari set'ini tutmak icin degildir. LAN ve PC port'lari ayni network'e baglanir. [SUBMIT] 'e basildiginda telefon restart eder.

Not :   IP'de; telefonun LAN portuna ilintili olarak WAN portunun konuslanmasi.

5.1.1.6 QoS

Sonucta sistem, DiffServ ayari bazinda, 802.1Q/P Protokolu'nu desteklemektedir.
Burada, degisik VALAN ID'si kullanmak icin, sistem, VLAN VoiceLAN ve
DataLAN ayar fonsiyonlarina sahiptir.
Degisik VLAN ID eklemek icin, “Data VLAN sistem ayarlari” sunlari yapabilir;
sinyallesme, dil akimi, diger sinyal akimlarinin sabitlenmesi..

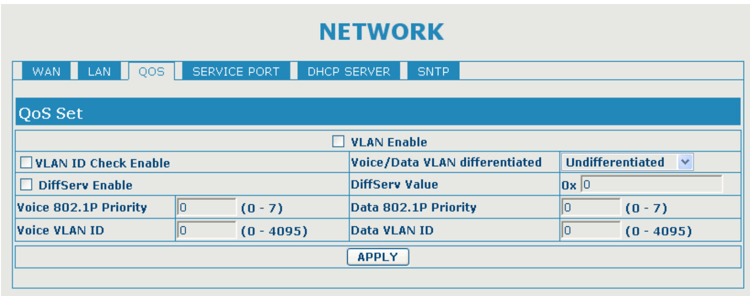


Fig. 5.3.2.3 QoS Ayari

VLAN Etkin	Aktif VLAN fonksiyonu
VLAN ID Etkin Kontrol	VLAN ID 'ye mutlaka uymali. Data paketi kendi VLAN IF-D'den farkli ise veya VLAN ID kaybolmus ve proses etmiyor ise. VLAN aktif degilse, bu cesit data proses edilir.
Voice/Data VLAN degismis	Ayar sunun icindir; ses/Data VLAN'da fark yaratmak veya yaratmamak, etiket farkliligi veya data etiketlenmemis durumu.
DiffServ Etkin	Yeni bulunan veya DiffServ. servis ayarlari.
DiffServ Deger	DiffServ parametre ayalari , normal seviye 0x00.
Ses 802.1P Onceligi	Ses gucunun ayari / sinyal data paketi 802.1p onceligindedir.
Data 802.1P Onceligi	Data 802.1p ayari / ses yok / sinyalleme 802.1p 'de.
Ses VLAN ID	VLAN ID'nin ses/sinyallesme data ayari.
Data VLAN ID	VLAN ID 'nin etiketini kullanmak icin, VLAN ID ve sessiz/sinyalesme data paket ayarlari.

Not:
1) VLAN Baslama; Ses/Data VLAN, “**degismemis**” olarak degisiklik gosterirse, butun paketler, ses VLAN ID 'sini etiket olarak kullanirlar.
2) VLAN Baslama; Ses/Data VLAN, “**etiket degismis**” olarak degisiklik gosterir ve Diff.Serv'i etkisiz yaparsa, sistem ses ve data'yi farkli gormeyecek ve butun paketler Ses VLAN ID 'sini etiket olarak kullanirlar.
3) VLAN Baslama; Ses/Data VLAN, “**etiket degismis**” olarak degisiklik gosterir ancak Diff.Serv'i etkin yaparsa, sistem ses ve data'yi farkli gorecek ve VLAN ID 'yi ard arda atiyacaktir..
4) VLAN Baslama; Ses/Data VLAN, “**data etiketsiz**” olarak degisiklik gosterirse, ses/sinyal paketi ,Ses VLAN ID'yi etiket olarak kullanacak, ancak data paketleri VLAN etiketlenmiyecektir..
5) Voice/Data VLAN'nin **degismemis veya degismis** olmasina bakilmaksizin VLAN etkinlesirse, butun paketler VLAN etiketi almayacaklardir.
DiffServ. etkinlestigi anda, butun paketler DiffServ degeri alacaktir.
6) Kullanici dikkat etmeli; standart olan VLAN ID Kontrol'un etkinlesmesi: etkinlestigi anda telefon VLAN ID ile uyusmayacaktir. Diger VLAN ID'ler bizimle uyusmadiginda paketler dikkate alinmayacaklardir.Buna karsin, telefon farkli VLAN ID 'leri olan komple paketleri kabul eder.
7) VLAN set edildiginde, kullanici statik mod ile IP alabilir. Aksi takdirde VLAN'da IP alamaz ve noktadan noktaya arama yapamaz.

5.1.1.7 Service Port

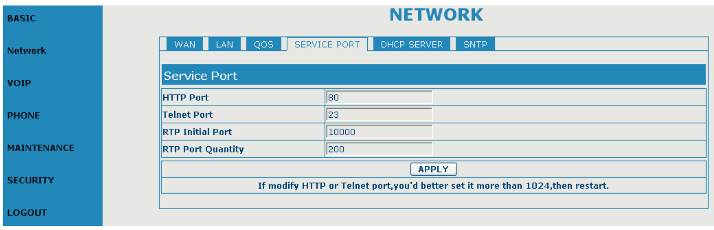


Fig. 5.3.2.4 Service Port

HTTP Port	Web portunu set etmek; fabrika cikisi 80 port'dur. Sistemi guclendirmek

	ve guvenli tutmak icin, standart olmayan 80 port'unu tutun. Degisiklikten sonra durumu kaydedin ve sistemi tekrar baslatin. Login islemini boyle yapin : http://xxx.xxx.xxx.xxx : xxxx.
Telnet Port	Telnet portunu set etmek; fabrika cikisi 80 port'dur.
RTP Birincil Port	Telefon RTP start portunu set etmek; bu port dinamik dagilimdadir.
TP Port Adet	Telefon RTP MAX Numarasi portunu set etmek; fabrika cikisi 200 pcs'dir.
Not: 1) İlgili sayfa degisikliginden sonra telefonu kayit ve restart edin, yoksa yaptiklarinizin telefon uzerinde bir etkisi olmayacaktir. 2) Kullanici, Telnet ve HTTP port numalarini modifiye ederse, port numarasini 1024 tutmak iyidir, cunku bu sistem reserve haldedir. Butun ayarlar sonunda telefon restart olmali. 3) HTTP port ayari “0” olarak alinirsa, HTTP servis engellenir.	

5.3.2.5 DHCP Sunucu

DHCP servis ayari icin kullanici bu sayfada, dinamik IP ayarlarini gorup ayarlayabilir ve DHCP tablosunu da kontrol edebilir.

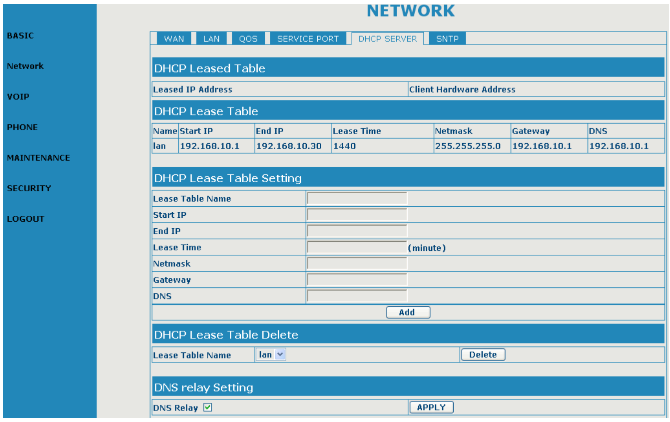


Fig . 5.3.2.5 DHCP Sunucu

DHCP Leased Tablo	DHCP tahsisli IP-MAC resmetme tablosu. Telefonu PC port'u baglandigi zaman, tablo baglanan yeri, IP ve MAC adreslerini gosterecektir.
Lease Tablo ismi	Eklenen lease tablo ismi.
Start IP	Eklenen lease tablo IP 'nin baslangici. DHCP urununu kullanmak icin PC'ye serbest bir IP atandigi zaman lease tablonun basindan aramaya basliyacaktir.
End IP	Eklenen lease tablo IP 'nin sonu. PC, ag noktasina baglanan IP numalarinin baslangictan bitise tesbiti. IP, PC noktasina baglandiginda DHCP, Start IP ve End IP arasinda olmalidir.
Lease Zaman	Lease tablo'ya eklenen lease IP zamani.
Netmask	Lease tablo'ya eklenen Net mask.
Gateway	Lease tablo'ya eklenen sabit gateway IP'si.
DNS	Lease tablo'ya eklenen sabit DNS Sunucu IP'si. Lease tablo'ya DHCP eklemek icin **ADDED** ve **SUBMITT**'a basin.

Not :

1) Lease tablo C tipi ag segmenti'nin IP adedinden az olmalidir. Tavsiye; sabit degerleri koruyun.

2） Etkin kilmak icin, sayet DHCP tablosu degisti ise, kayit ve restart edilmelidir.

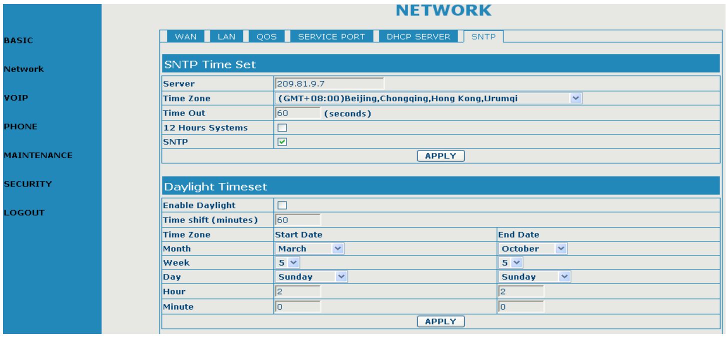
3) Lease tablo'nun ismini secin ve 【DEL】' e basin ve veri DHCP tablosundan bunu silecektir.

4) DHCP ayar gosterimi dakika'dir.

5) IP modelin, LAN acilimi telefonun PC port'u ile ilgilidir.

5.1.1.8 SNTP

Zamanlama, SNTP Sunucu'nun “otomatik zaman bolgesi ayarlari”ndan alinir, ancak el ile mauel olarak ta ayarlanabilir.



F ig. 5.3.2.6-1 SNTP ayari

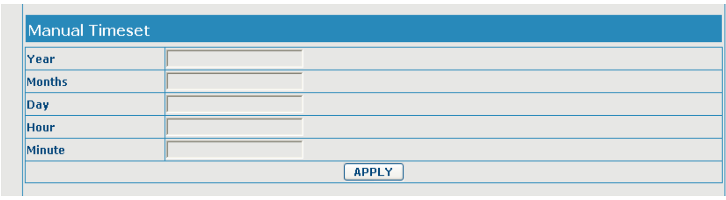


Fig. 5.3.2.6-2 SNTP ayari

Sunucu	SNTP Sunucu adres ayari
Zaman Bolgesi	Zaman bolgesini etkinlestirmek
Zaman bitis	Sunucu'nun senkron tipi icin fasila. Sabit/standart zaman 60s.
12Saat Sistemi	12 Saat Sistemi'ne gecmek Sabit/standart, 24Saat format'idir.
SNTP	Etkinlestirme; /kesme SNTP servisi ;
Etkinlestirme; Gunisigi	Etkinlestirme; zaman/saat tasarrufu
Zaman kaydirma(dak.)	Etkinlestirme; “gun kazanma sistemi”nin zaman kaydirmasi.
Ay	Etkinlestirme; zaman/saat tasarrufu 'nun AY start ve son'u
Hafta	Etkinlestirme; zaman/saat tasarrufu 'nun HAFTA start ve son'u
Gun	Etkinlestirme; zaman/saat tasarrufu 'nun GUN start ve son'u
Saat	Etkinlestirme; zaman/saat tasarrufu 'nun SAAT start ve son'u
Dakika	Etkinlestirme; zaman/saat tasarrufu 'nun DAKIKA start ve son'u

Not: El ile ayarlandiginda, her ayar calisma'da doldurulmali ve verilmelidir.

5.1.2 VOIP

5.1.2.1 SIP

SIP Sunucu ayar konfigurasyonu asagidadir.

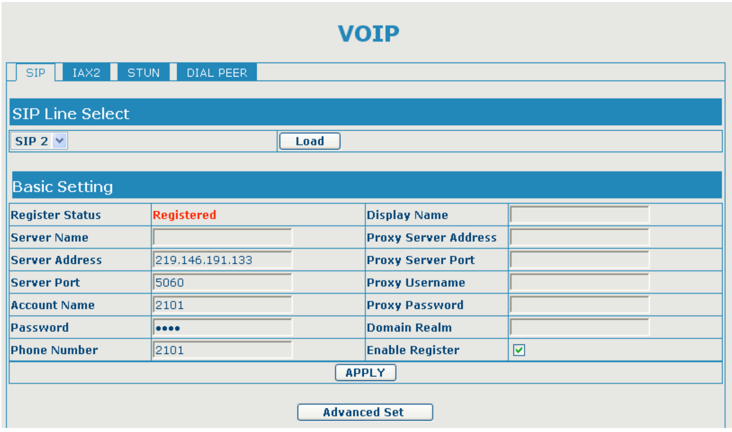


Fig. 5.3.3.1-1 SIP Genel ayar

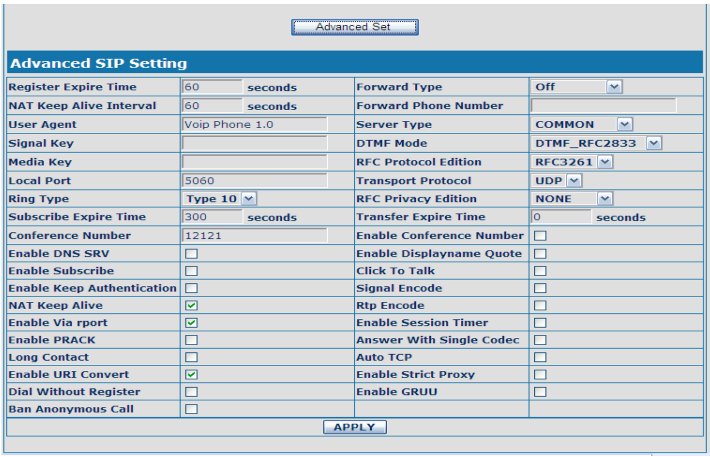


Fig. 5.3.3.1-2 SIP Gelismis SIP ayar

SIP Hat Secimi

Secim icin 2 SIP account'u vardir, SIP account hattini buna gore secin.
Secilen SIP account'una gelmek icin [Load] 'u tiklayin

Genel ayar	
Kayit Durumu	SIP Kayit Durum Gostergesi. Kayit basarili, [**Registered**] seklinde gorunucektir. Kayit basarisiz, [**Unregistered**] seklinde gorunucektir. Kayit yok, [**Unapplied**] seklinde gorunucektir.
Sunucu Ismi	Sunucu Isim tedariki
Sunucu Adresi	SIP adres yapilandirilmasi, domain isim tipinde Sunucu adres destegi.
Sunucu Port	SIP kaydi Sunucu command port'u yapilandirilmasi.
Account Ismi	SIP kaydi account yapilandirilmasi.
Sifre	SIP kaydi sifre yapilandirilmasi.
Telefon Numarasi	SIP kaydi ve telefon numarasi yapilandirilmasi. Numara yok ise, kayit olmaz..
Display Name	Yapilandirma; Latin alfabesi ile isim goruntusu
Proxy Sunucu adresi	Yapilandirma; Proxy Sunucu IP adresi Normalde, SIP Tedarikci's, Proxy Sunucu ve ilgili kaydini teklif eder. Proxy Sunucu'i konfigurasyon ve kaydi aynidir. Sayet IP adres konfigurasyonu onlardan ayri ise, Proxy Sunucu'i konfigurasyon ve kaydini da ozel yapiniz.
Proxy Sunucu Portu	Yapilandirma; Proxy Sunucu Port'u
Proxy Kullanci Ismi	Yapilandirma; Proxy Sunucu Kullanici ismi
Proxy Sifre	Yapilandirma; Proxy Sunucu Sifre
Domain Bolgesi	Yapilandirma; SIP Local Domain Bolgesi. Sayet, Sunucu yerel bir domain isaret etmiyor ise, yerel domain Sunucu ile ayni adres ve domain uzerine yapilanir. Sistem, Sunucu Adres kaydini domain bolgesi olarak otamatik girecektir ve kullanicilar bunu tekrar girmek durumunda kalmazlar.
Etkinlestirme; Kayit	Yapilandirma; Kaydi Etkinlestirme'yi cozme
Gelismis Set	
Zaman Bitis Kaydi	Yapilandirma; Zaman dolumu, standardi 60 saniyedir. Sayet “kayit zamani” **“buyuktur”** veya **kucuktur** “tesbit zamani”ndan ise, telefon tavsiye edilen zamana uyacaktir.
NAT Etkinlik Araligi	Yapilandirma; Etkinlik Araligi. Sayet SIP Sunucu'de Tesbit Fonksiyonu acik tutulmussa, telefon, **NAT 'i Canli Tutun interval**'i geldiginde anliyacaktir.
Kullanici Temsilcisi	Yapilandirma; Kullanici Temsilcisi
Sinyal tusu	Yapilandirma; Sinyal tusu
Medya tusu	Yapilandirma; Medya tusu
Dahili port	Yapilandirma; Sip port hatti

Hot Line Numarasi	Yapilandirma; Hot Line Numarasi OFF-HOOK 'ta (telefon acik) Hot line numarasi yok ise calismaz,
Ring Tipi	Yapilandirma; Her Ring Tipi hatti
Zaman bitis transferi	Atanan transfer sonlandirildiginda, gercek arama, zaman bitis'inden sonra sonlanacaktir. Standart 0 dir, yani telefon kapatildiginda telefon BYE kodu gonderir.
Abone Etkinlestirme	Basarili kayittan sonra abone bilgisi, bu diger kullanici durumu ve voice mail'ini de kayit edebilir.
Tastik Kaydi, Etkinlestirme	Yapilandirma; Tastik Kaydi, etkinlestirme veya etkinlestirmeme. Sunucu, kayit onayi aldigindan karsi teyit gonderir.
NAT'i canli tutma	Yapilandirma; otomatik tesbit Sunucusu. Lutfen zaman fasilasini, **NAT'i canli tutma**'dan daha azda tutun ve Sunucu ayari cok kisa ise bu islemi acin.
Rapor uzerinden Etkinlestirme	Yapilandirma; RFC3581 destegi var ya da yok. SIP Sunucu destegine ihtiyac duyan raporlama sistemi dahili network'leme icin kullanilir. Dahili ve harici network baglantisi'nda NAT irtibati icin kullanilir.
PRACK Etkinlestirme	Yapilandirma; SIP PRACK destegi etkinlestirme veya etkinlestirmeme. (geri arama zil tonu renkledirme icin) Tavsiye; standart ayar
Uzun Kontakt	Yapilandirma; Daha cok parametreli kontakt alani; SEM sistemi ile beraber calisir.
Etkinlestirme; URI Donusumu	%23 in transmisyonda ki **URI donusum #**'si
Kayit'siz Tuslama	Yapilandirma; Kayit olmadan Proxy Aramasi
Anonymous Cagri Yasagi	Yapilandirma; Anonymous Cagri Yasagi
Iletim Tipi	Forward Tipini secin (Standardi OFF) ● Kapali : Arama tipini kapali tut. ● Mesgul : Gelen arama, mesgul'de tanimli bir numaraya yonlenir. ● Cevap yok : Belirlenen arama tanimlinacak bir surede acmaz ise, gelen arama, mesgul'de tanimli bir numaraya yonlenir. ● Her zaman : Gelen arama, her zaman tanimli bir numaraya yonlenir . Telefon gelen arama, yonlenmeden once “arandigi”ni gosterir.
Telefon Numarasi Iletim	Yapilandirma; Telefon Numarasi ilet
Sunucu Tipi	Sunucu tipini sec veya ozel tip belirle.
DTMF Modu	DTMF mod'unu belirt, asagida ki gibi: ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO Degisik ISP internet'i degisik mod. verebilir.
RFC Protokol Baskisi	Yapilandirma; Protokol Yayini. Sayet telefon, CISCO5300 v.s. SIP1.0 Gateway'i ile uyumlu ise, lutfen Protocol Yayini RFC2543'i konfigure edin.

Transport Protokolu	Yapilandirma; Transport Protokolu :TCP/UDP
RFC Privacy Baskisi	Yapilandirma; RFC Privacy'yi destekle veya destekleme, RFC3323 ve RFC3325 yayini
Zaman Yanma Tesbiti	Yapilandirma; Zaman Yanma
Konferans numarasi	Yapilandirma; Konferans Arama numarasi
Etkinlestirme; Konferans numarasi	Etkinlestirme; Konferans Arama fonksiyonu
Etkinlestirme; DNS SRV	RFC2782 destegi
Bas ve Konus	Yapilandirma; Bas ve Konus（uygulama yazilimi gerekir）
Sinyal Encode'u	Yapilandirma; Signal Enkod'u
Rtp Encode'u	Yapilandirma; Rtp Enkod'u
Etkinlestirme; Zamanlama	Yapilandirma; Rfc4028'i destekle veya destekleme, ; SIP session'lari tazele
Tek Kodec ile Cevap	Arandiginda tek Kodec ile cevap
Otomatik TCP	Yapilandirma; Kodec 1.300 byte'tan buyuk ise, TCP prokolunu dondur.
Etkinlestirme; Strict Proxy	Ozel Sunucu ile uyum (Kod gonderirse, tekst icinde adres kullanma, orjinalini al)
Etkinlestirme; GRUU	Yapilandirma; GRUU
Etkinlestirme; Display name Quote	Yapilandirma; Sunucu ile uyumlu olmak icin, Quote goruntu ismi ile encode gonder

5.1.2.2 IAX2

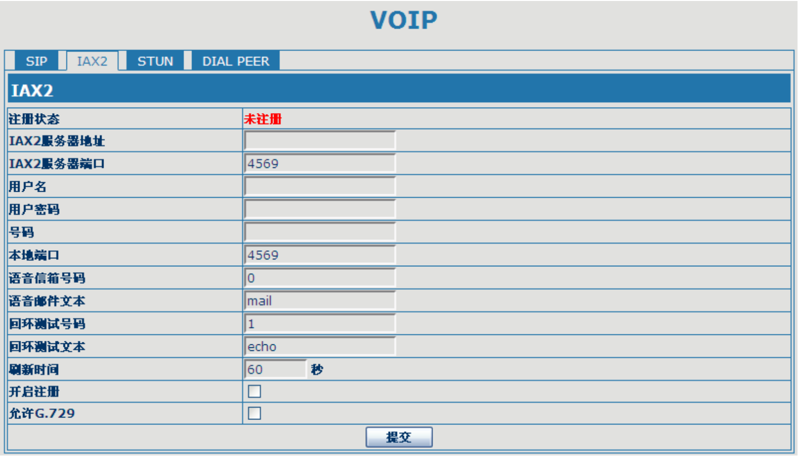


Fig. 5.3.3.2 IAX2 ayari

Kayit Durumu	IAX2 Register Durum Goruntusu. Basarli kayitta “**Register**” gosterecektir, yoksa “**Unregistered**” gorunecek.
IAX2 Sunucu Addr	Yapilandirma; IAX2 Sunucu adres, It can be domain isim tipi olablir.
IAX2 Sunucu Port	Yapilandirma; IAX2 Sunucu Port'u
Hesap Ismi	Yapilandirma; IAX2 Sunucu Hesap Ismi
Hesap Sifresi	Yapilandirma; IAX2 Sunucu Sifre
Telefon Numarasi	Yapilandirma; IAX2 Sunucu Telefon numarasi
Dahili Port	Yapilandirma; IAX2 Sunucu Local Port
Voice Mail Numarasi	Yapilandirma; Voice Mail Numarasi. Voice yazi tipinde ise, telefon giris yapamaz. IAX2 voice mail destegi oldugunda bu teksti voice mail numarasi yerine girebilir.
Voice Mail'in Echo Test'i	Yapilandirma; Voice Mail Text if IAX2support voice mail.
Numaranin Echo Test'i	Yapilandirma; Echo Test detegi vey / yok. Sayet sistem Echo'yu destekliyor ise ve Echo testi tekst halinde ise, bu Echo test numarasi ile degisir.
Text'in Echo Test'i	Yapilandirma; Echo Test Text
Zaman tazeleme	IAX2 tazeleme kaydi. Zaman saniyedir. Tavsiye, kullanici secimi; 60s – 3600s .
Etkinlestirme; Kayit	Yapilandirma; Etkinlestirme / Kaydi Etkinlestirmeme
Etkinlestirme; G.729	Yapilandirma; G.729 destegi var / yok. G.729 destegi icin telefon codec gonderir. Idefisk kullanilirsa (G.729 destegi icin degil) arama idefisk'i PC'yi bozar (break down).

5.1.2.3 Stun (Sasirtma)

STUN fonksiyonu asagida ki gibidir: VANT IP'si ve SIP sinyal port'undan cikmak icin, kullanici, STUN altinda, IP ve PORT CONTACT alaninin SIP kayit alanini SHIFT eder ve kayda girer.

Ilgili kullaniciyi bulmak icin gelen aramalar oldugunda yukarida ki kayittan emin olun.

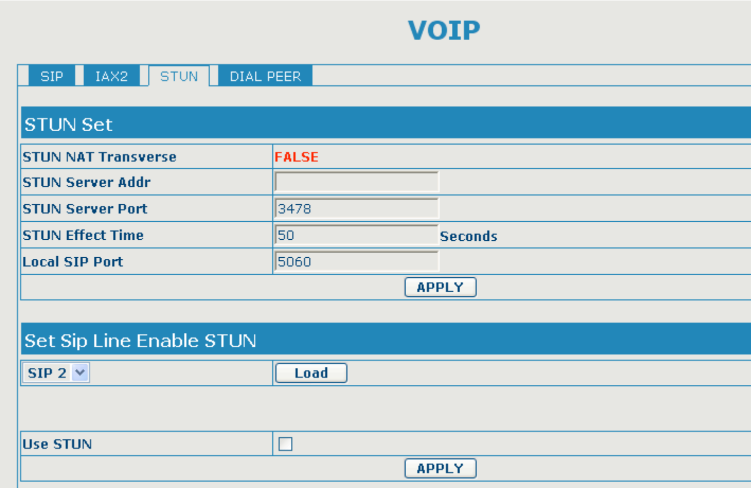


Fig. 5.3.3.3 STUN ayari

STUN Etkin Zaman	STUN Etkin Zamani etkinlestirme. Sayet NAT Sunucusu NAT mapping'i belli zaman sonunda idle durumda bulursa, mapping'i cozer ve sistem mapping islemini canli ve efektif tutmak icin STUN data paketi gondermeye ihtiyac duyar..
Yerel SIP Port'u	Yerel SIP Port'u zamani etkinlestirme; Standardi 5060 tir. (Yerel SIP Port ayarindan sonra, SIP desisen port icin iletisime girer).
Sip Hatti Etkinlestirme; Stun	
Stun'i kullanmak ve SIP hesap ayari icin hat secin, secilecek 2 hat vardir. **Load** tusuna basilarak secilir.	
Stun kullanimi	SIP STUN'i Etkinlestirme / Etkinlestirmeme

Not: SIP STUN, NAT'a SIP penetrasyon'unun gerceklesmesi icin kullanilir.

Sayet telefon, STUN Sunucu IP ve Port'unu (standardi 3478) konfigure eder ve SIP Stun'i etkinlestirirse, kullanici, "**NAT 'a pentrasyonun gerceklesmesi**" icin genel/adi SIP Sunucu'yu kullanabilir.

5.2.3.4 AYNISINI (PEER) cEVİRMEK

IP numara tablosunun islevselligi, kullaniciyi Internet uzerinden kolay arama yapmasi icindir.

Bu tablo ile kullanici, cok daha esnek cevirme kurallari secebilmenin tadini yasar.

Ornek; sayet kullanici IP adresini bilirse, istenen numarayi "aynalama arama" ile (peer to dial) arayabilir.

Canli ornek; dusunun ki IP adresi 192.168.0.181,

kullanici bir kural koyar, soyle ki; “123 cevrildiginde 123, 192.168.0.181 'in yerini alsin” .

Dial Peer Table						
Number	Destination	Port	Mode	Alias	Suffix	Del Length
123	192.168.0.181	5060	SIP	no alias	no suffix	0

Ornek, kullanici Pekin'e bir PSTN arama yapacak ise, arama kurali soyledir.Numara “1” ile baslayan butun aramalar bu arama kuralina baglidir. Ornek, kullanicinin aramak istedigi numara; 01062213123 ama 162213123 cevirdi; telefon girilmeyen rakkamlari dolduracaktir.

Dial Peer Table						
Number	Destination	Port	Mode	Alias	Suffix	Del Length
1T	0.0.0.0	5060	SIP	rep:??010	no suffix	1

Hafiza tasarrufu ve luzumsuz rakkam girisinden kurtulmak icin:

Number	Destination	Port	Mode	Alias	Suffix	Del Length
13xxxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

1. “ x “ cevrilen digit'le uyusur. Yukarida ki tanimli konfiurasyona gore, 13 ile baslayan 11 rakkamli numaradan sonra telefon cevrilen numaralara “0” ekleyecektir.

2. “[]” tanimi digit'e uyan siniri tanimlar. Bu bir sinir olabilir, virgullerle ayrilmis sinir'lar olablilir veya rakkamlar listeleri olabilir. 135 ten 139 'a kadar, 11 rakkamli numara cevrildikten sonra, telefon otomatik bir “0” ekleyecektir. Bu telefonu kullanirken, web'e girmeden, degisik kullanici isimlerinden cevirmeyi yaparsiniz.

Ozel ayarlar soyledir:

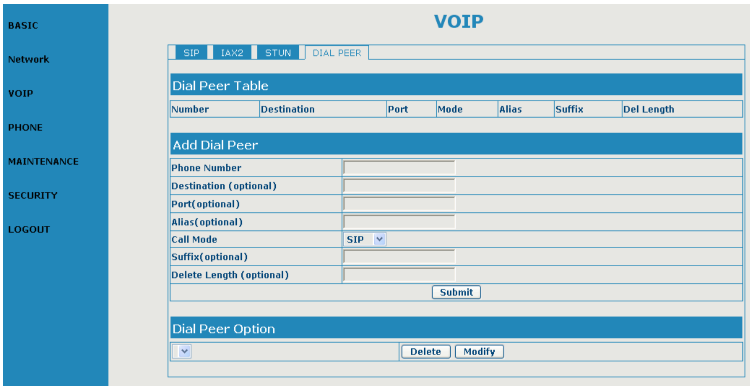
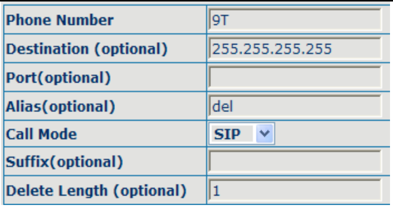
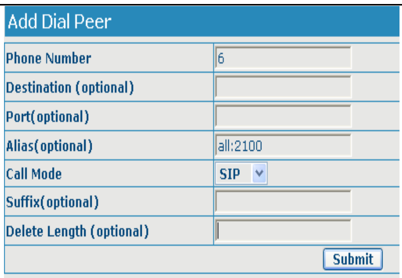
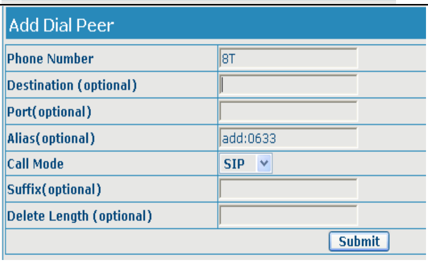
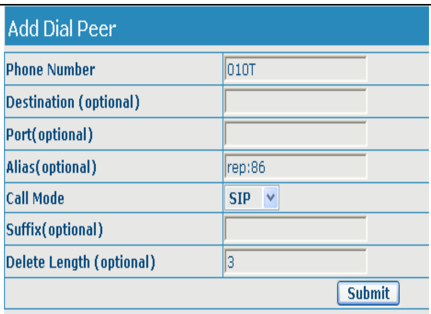
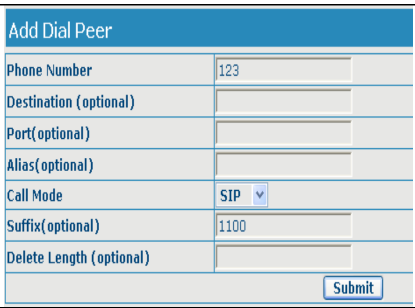


Fig. 5.3.3.4 DIAL PEER

<table>
<tr><td>Telefon numarasi</td><td>Aranan numaralari eklemek, 2 yolla olur.Birincisi tam uyum. Bunda, kullanici, tam uyumu saglmak icin, telefon numarasini cevirmesi gerekir. Digeri ise ornekleme metodudur (PSTN bolge kodunun on orneklemesi gibi); burada on bir numara girersiniz ve telefon arka numaralari belirler. Oncul numara maksimum 30 rakkamli, x formatta ve belirli numara kapsamlidir.</td></tr>
<tr><td>Yon</td><td>Yon adres yapilandirmasi. Kullanici peer to peer aramak isterse,
IP adresi veya ilgi (domain) ismi ve bu Somut IP, telefon DNS Sunucu tarafindan bolgelendirilir. Yapilandirma olmassa IP ayari 0.0.0.0. seklinde addedilecektir.
Bu yapiladirma opsiyoneldir.</td></tr>
<tr><td>Port</td><td>Sinyal portunun ozel ayari (opsiyoneldir), SIP'in standardi 5060'dir.</td></tr>
<tr><td>Muadil</td><td>Benzerleri (muttefikleri) belirlemek; bu islem opsiyoneldir ve ornekleme numaranin yerine gecer. Kullanici bunu kullanmazsa, ornekleme baz alinir.</td></tr>
<tr><td colspan="2">Not: Tanimlanan uzunluga gore ayar yapilan, 4 tip muadil (alias) vardir.
1) “xxx” in anlami; arayan numaranin onune xxx koyacaktir, bu aramayi kisaltir.
2) “xxx” in anlami; xxx, telefon numarasini degistirecektir ve bu hizli arama seklinde gorulecektir. Ornek; kullanci 1'i tanimli numara seklinde girerse, girilen bu numara gercek numara yerine addedilecek ve cevrilecektir.
3) del 'in anlami; tanimlanan uzunlukta ki numarayi telefon silecektir.
4) Rep: xxx anlami; telefon atanan numara ve ilgili uzunlugunu degistirecektir. Ornek, arayan, VOIP operatorleri tarafindan desteklenmis, PSTN(010-62281493) cevirirse, aranan gercek numara 010-62281493 'tur. Aranan numarayi, 9T ve rep:010 seklinde tanimlayabiliriz, sonra tanimlanan uzunluga 1 koyabiliriz. Bu sayede 9 ile baslayan tum aramlar 010+telefon numarasi seklinde algilanacaktir.</td></tr>
<tr><td>Arama Modu</td><td>Degisik sinyal protokolu secimi, Standardi SIP dir.</td></tr>
<tr><td>Eklenti (Suffix)</td><td>Suffix tanimi, Opsiyoneldir , olmazsa “eklenti yok” bilinir.</td></tr>
<tr><td>Silme Uzunlugu</td><td>Opsiyoneldir, ve bir numara ise tanimlanir.</td></tr>
<tr><td colspan="2">Coklu SIP hatlarinda cevirme ayari (Giris):
9T adresleme; sayet SIP1 kaydi ve dial-peer ayari yapilmis ise, butun aramalar, SIP1 server uzerinden gidecektir. Bu islem, numerik bilgi olarak, numaranin onunde “9” tuslandiginda olur.
8T adresleme, sayet ozel SIP2 kaydi ve dial-peer ayari yapilmis ise, butun aramalar, SIP2 server uzerinden gidecektir. Bu islem, numerik bilgi olarak, numaranin onunde “8” tuslandiginda olur.
<table><tr><th>号码</th><th>目的地</th><th>端口</th><th>类型</th><th>别名</th><th>后缀</th><th>删除长度</th></tr><tr><td>2T</td><td>0.0.0.0</td><td>4569</td><td>IAX2</td><td>del</td><td>no suffix</td><td>1</td></tr></table>2T; Bu kural su demek; kullanici IAX server'i konfigure ve kaydetti. IAX2 server'i kullanmak istiyor ise,aranan numaranin onune “2” koyacaktir.</td></tr>
</table>

Degisik muadil uygulamalara ornekler:

Web'e konum	Fonsiyon aciklamasi	Fonsiyon aciklamasi

<table>
<tr>
<td>
</td>
<td>Bu sayfa su demek;
9 ile baslayan herhangi numara SIP portal'dan gecer. Silme uzunlugu 1 'dir, yani butun secilen numaralarin 1. numaralari silenecek.</td>
<td>“92161” cevrilirse,
SIP Sunucu bunu
“2161” anlar.</td>
</tr>
<tr>
<td>
</td>
<td>Kullanici, 2'den sonra numara cevirirse, bu hizli arama seklinde algilanir. Bundan sonra kilerin hepsi, muadil numaradir.</td>
<td>“2” cevrilirse,
SIP Sunucu bunu
“2100” anlar.</td>
</tr>
<tr>
<td>
</td>
<td>Bu sayfada su olur;
Arama uzunlugunu dusurerek, bolge kodu veya prefix'ler otomatik eklenir. Burada muadiller eklemedir.</td>
<td>“6000” cevrilirse,
SIP Sunucu bunu
“06336000” anlar.</td>
</tr>
<tr>
<td>
</td>
<td>Spesifik arama, 86-6226 olursa, kullanici, PSTN(010-6226) cevirecektir. Sonra aranan numarayi, 010T seklinde belirler ve sonra rep:86 yapariz. Sonra, silme uzunluk tanimi 3 olur. Bundan dolayi 010 ile giden butun arama numalari 86+ ile degiserek gidecektir. Burada muadil rep. 'tir.</td>
<td>“010 6226” cevrilirse,
SIP Sunucu bunu
“866226” anlar.</td>
</tr>
<tr>
<td>
</td>
<td>Bu sayfa su demek;
Telefon numarasi 123'ten sonra otomatik olarak 1100 eklenecektir.</td>
<td>“123” cevrilirse,
SIP Sunucu bunu
“1231100” anlar.</td>
</tr>
</table>

5.1.3 Telefon ayari

5.2.4.1 DSP Yapilandirma

Bu sayfada, kullanici voice codec'i giris ve cikis ses ayarini belirleyebilir.

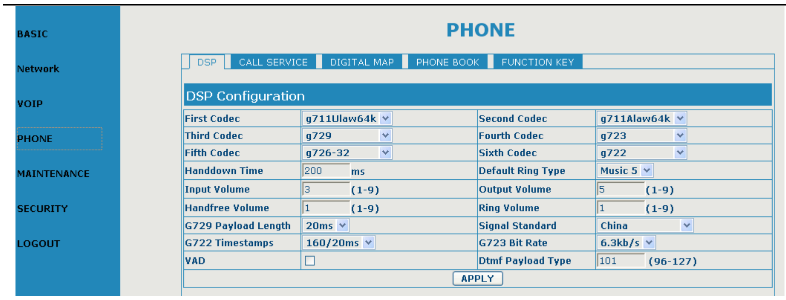


Fig. 6.3.4.1 DSP ayari

1. Codec	G.711A/u, G.722, G.723, G.729,G.726-32。 Bu 1. tercih; DSP codec:G.711A/u, G.722, G.723, G.729,G.726-32
2. Codec	Bu 2. tercih; DSP codec: G.711A/u, G.722, G.723, G.729,G.726-32。
3. Codec	Bu 3. tercih; DSP codec：G.711A/u, G.722, G.723, G.729,G.726-32。
4. Codec	Bu 4. tercih; DSP codec：G.711A/u, G.722, G.723, G.729,G.726-32。
5. Codec	Bu 5. tercih; DSP codec: G.711A/u, G.722, G.723, G.729,G.726-32。
6. Codec	Bu 6. tercih; DSP codec：G.711A/u, G.722, G.723, G.729,G.726-32。
Hand down Zamani	Standart zaman 200ms'tir. Sayet **hand down zamani**, belirlenmis zamandan asagida ise, telefon, “**hand down islemi**” ni kaale almayacaktir.
Giris Sesi	Ses giris derecesinin belirlenmesi
Cikis Sesi	Ses cikis derecesinin belirlenmesi
Eller serbest sesi	Eller serbest sesi derecesinin belirlenmesi
Zil Sesi	Zil Sesi derecesinin belirlenmesi
G729 Tasima uzunlugu	G729 Tasima uzunlugu derecesinin belirlenmesi
Standart Sinyal	Ulkelerde sinyal standarlari degisiktir
G722 Zamandamgasi	G722, 160/20ms ve 320/20 ms icin zaman damgasi secin, opsiyonaldir
G723 Bit Rate	G723 Bit rate, 5.3kb/s ve 6.3kb/s, opsiyonaldir
Sabit Zil Tipi	Sabit Zil Tipi ayari
VAD	Etkin olmak veya olmamak secimi. VAD'i etkin yaparsaniz, G729 tasima kapasitesi uzunlugu 20ms uzerine cikamaz.
DTMF Tasima Kap. Tipi	DTMF Tasima Kapasitesi Tipi Ayari

5.1.3.1 Call Service – Arama Servisi

Bu Web sayfasinda, kullanici, HotLine, Transfer Aramssi, Arama Bekletme, 3 Yollu Arama, Kara Liste, Beyaz Liste, Limit Listesi, v.s. gibi ogeleri konfigure eder.

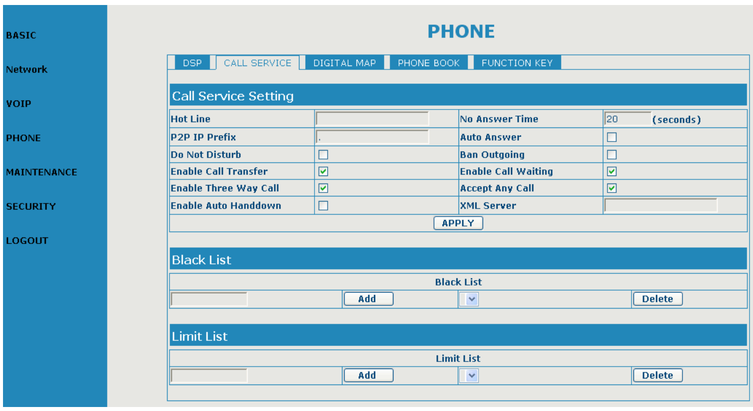


Fig. 5.3.4.2 Call Service – Arama Servisi

Hot Line	Hotline numarasi belirlemek. Hotline belirlendiginde, ahize kaldirildiginda, baska bir numara haricinde, direkt HotLine aranir.
Cevap Yok zamani	Cevap Yok zamani belirleme
P2P IP Prefix	Uctan uca arama'da On Ek'in belirlenmesi. Ornek; 192.168.1.119, cevireceksiniz ve P2P IP'nin On Ek'ini 192.168.1. yaptiniz, sadece #119 cevirin, 192.168.1.119. olacaktir.
Do Not Disturb / Rahasiz Etme	Bu secildiginde, hic bir arama kabul edilmez, arayanlar telefon mesgul sanir, fakat herhangi bir arama telefon normal sekilde calisir.
Etkinlestirme; Call Transfer Aramasi	Secildiginde, Call Transfer etkinlesir
Etkinlestirme; Arama Bekliyor	Secildiginde, Arama Bekliyor etkinlesir
Etkinlestirme; uc Yollu Arama	Secildiginde, uc Yollu Arama etkinlesir
Her Aramayi Kabul Et	Secildiginde, arayan numara telefona kayitli olmasa bile kabul edilir
Otom. Cevap	Arama geldiginde, telefon otomatik cevaplar
Arama Yasagi	Telefonda aramaya gecildiginde, devamli mesgul sesi verilir, arama yapilamaz
XML Sunucu	Sunucu adres tanimi, standardi, XML 'dir
Kara Lise	Black List -4119 . Kara Listeye tanimlanan numaralar, direkt telefon tarafindan reddedilir. Burada x herhangi bir numaradir. Ornek 4xx tanimladiniz, bu su demek, 4 ile baslayan ve 3 haneli herhangi bir numara aranmaz. NOKTA (.) su demek; herhangi bir kafadan atilan numaraya uyar. Ornek, 6. 6 ile ile basliyan her numaraya bu islemi yap ! Bu sekle uyan fakat kabul edilmek istenen bir numara da bu Kara Liste disina atilabilir. Bu numarayi Beyaz Liste'ye alirsa bu numara kabul edilir. Black List -4119 . 4119 haricinde butun numaralari yasakla
Limit Listesi	Bu numalara tanimlandiginda, ayni bu numalar aranamaz. Prefix bir numara verirseniz, ornek, 010; 010 ile baslayan butun numaralar aranmaz. Burada x herhangi bir numaradir. Ornek 4xx tanimladiniz, bu su demek, 4 ile baslayan ve 3 haneli herhangi bir numara aranamaz. NOKTA (.) su demek; herhangi bir kafadan atilan numaraya uyar. Ornek, 6. 6 ile ile basliyan her numaraya bu islemi yap !

Not: Kara Liste ve Limit Liste'ye 10 tanim yapilabilir.

5.1.3.2 Digital Map (Sayisal Harita)

Sistemin destekledigi sayisal mod'lar asagidadir.

1). “#” ile Son: İstedigin no.yu cevir, sonra # tusla.

2). Sabit uzunluk: telefon tanimli uzunlukta numaralari kabul eder.

3). Sure Bitti: Cevirdiniz ve bekleme suresi bitti, sistem numarayi yollar.

4). Kullanici tanimli: Kullanici numara veya prefix uzunlugunu belirleyebilir.

Kullanicilari, Pbx'te Harici Arama tuslamasinda, sekonder bir arama modunu canli tutmak icin, prefix no.sundan sonra, sistem, arama haritasinin arama kuralina gore tuslama sesini tazeliyecektir. Boylece kullanici tuslamaya devam edebilir. Arama bittiginde, telefon girilen prefix ve numarayi server'e gonderir.

Ornek, Sayisal Harita Kurali'na gore, burada bir “9,xxxxxxxx” kurali var. 9 tuslandiginda, telefon, ikinci bir cevir tonu gonderir, kullanici tuslamaya devam eder. Tuslama bittiginde, 9 ile baslayan numara aranir; gercekte 9 hanel ve 9 ile baslayan numara gitmistir.

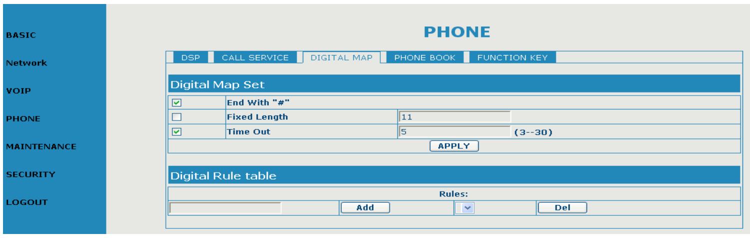


Fig. 5.3.4.3 Sayisal Harita Konfigurasyonu (Digital Map Configuration)

"#" ile biter	“#” cevirme ile biten, telefonu aktif veya pasif yapmak.
Sabit Uzunluk	Telefonun cevirecegi limitli hane tesbiti; Ornek; 11 handen fazla numara cevirdiniz, telefon ilk 11 numarayi cevirir.
Zaman asimi	Zaman bitim tanimlanmasi, Ornek; ceviren kisi tuslamayi 5 saniye yapmadi, telefon bu zaman oncesinde cevrilen numaralari cevirir.

Asagida, sayisal harita kuralini bulacaksiniz:

“[] “ , atanmis digitlerin sinirini belirler. Bu bir sinir, virgullerle ayrilmis sinir veya digitlerin listesi olabilir.

“x” herhangi bir cevrilen rakkama uyar.

“. “ hic numara olmayan veya keyfi bir numaraya uyar.

“Tn” n saniye icinde gonderilen numara sonunda ki, ilave zaman asimini belirler. N zorunlu ve 9 ve 0 saniye arasindadir. “Tn” cevirme duzeninin 2 karakterinden olmalidir. Sayet Tn, belirlenmezse, butun cevirme duzenlerinde, bu T0 olarak alinir.

“,” Alinan numara sonlanmasindan sonra gonderilen cevir tonu'dur.

Ornekler:

RULE
"[1-8]xxx"
"9xxxxxxx"
"911"
"99T4"
"9911x.T4"

[1-8]xxx: 1000 ve 89999 arasindaki butun numalar hemen cevrilmeli.

9xxxxxxx: 9 ile baslayan butun 8 haneli numalar hemen cevrilmeli.

911: 911 cevrildiginde, derhal gonder.

99T4: 4 saniye sonra 99 cevir.

9911x.T4: 9911 ile baslayan bir numara cevrildiginde,
4 saniye icinde en azindan 5 rakkam daha beklenir.

9,xxxx, 9 ile baslayan bir cevirme'de, 9 'dan sonra hemen tekrar cevir sesi gelir ve bitis tonunu bekler. Numaralar bittikten sonra, cevirme baslar, 2. cevirme olarak addedilir.

Not: “#” ile bitmede, sabit uzunluk, zaman asimi ve sayisal harita kullanilabilir.
Sistem, kurallara gore cevirme ve numara gondermeyi kesecektir.

5.1.3.3 TELEFON KITABI

Kullanici burada her isme, bir isim, telefon numarasi ve zil tipi belirleyebilir.

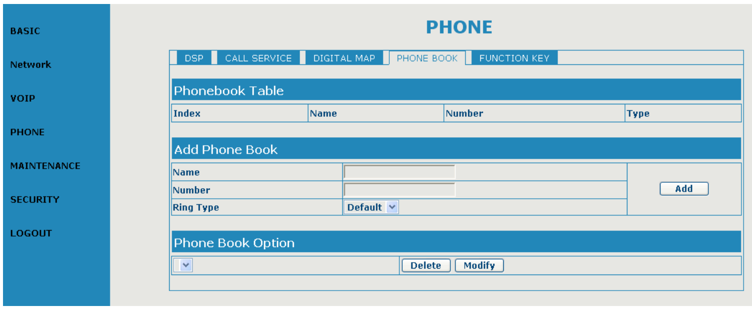


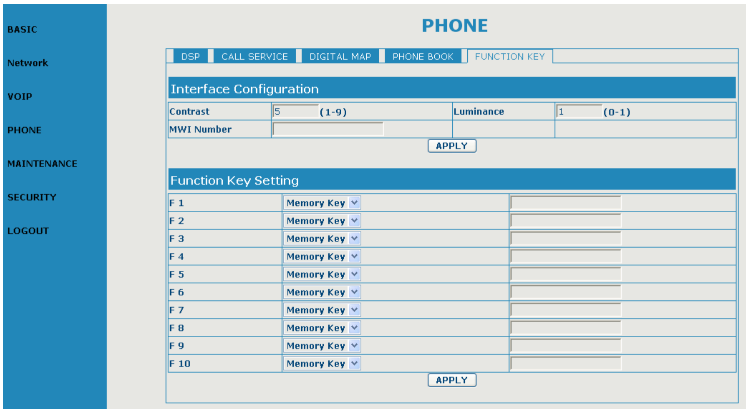
Fig. 5.3.4.4 Telefon Kitabi Ayari

Index	Name	Number	Type
1	ad	23	Default
1			

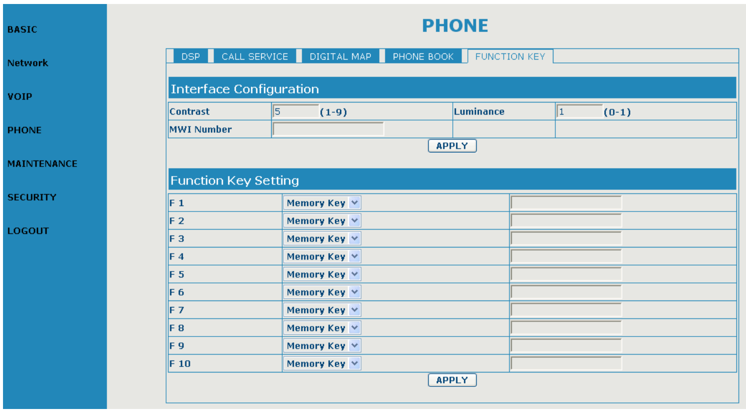
Telefon kitabinin detay ic bilgisi.

Isim	Ilgili telefonun ismi, bu cevrildiginde, lcd ekran numarayi gosterir.
Numara	Numara gir
Zil Tipi	Tipi belirle
“Modify” degistirme, “Delete” silme demektir.	
Not: Maksimum kapasite, 500 kayit 'tir	

5.3.4.5 FONKSIYON TUSU

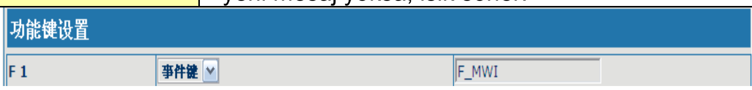


5.3.4.5 FUNCTION (İsLEM) tusu



Picture 5.3.4.5 Store tusu ayari

Kontrast	Konstrast seviye ayari,
Luminance – isik gucu	İsik Gucu seviye ayari
MWI Numarasi	Mesaj isigi sonup yanma seviyesi; mesaj geldiginde bu seviyede isik aktive olur, yeni mesaj yoksa, isik soner.

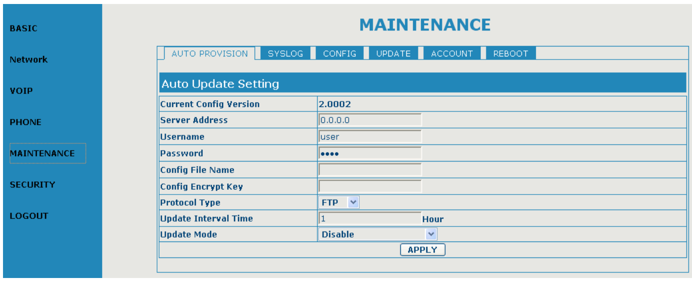


Kayit tusu：Her tusa belirli bir hareket kaydi atayabilirsiniz. Hat, “cevirme modu ayari” (SIP1,SIP2,Dialpeer,IAX2), mod suna benzer: SIP1:NAME1.

Harket Tusunun Fonksiyonu: degisik mod'lari monitor etmek.

5.3.5 Yonetim Ayari

5.3.5.1 Otomatik Konfigurasyon



5.3.5.1 Otomatik Konfigurasyon

Gecerli Konfig. Versiyonu	O an ki konfigurasyonu gosterir.
Sunucu adresi	Sunucu adres tanimi; Sunucu adresi, surada olabilir; IP form'da, yani 192.168.1.1，veya domain form'da，yani Ftp.domain.com. Ve sistem "Sunucu ayari, alt kutuphane fonksiyonunu da destekler, sistem, Sunucu belirlemesi yapabilir, yani sudur; sistem, Sunucu adresini, 192.168.1.1/ftp/config/ veya ftp.domain.com/ftp/config olarak belirleyebilir. Su demektir, ziyaret edilen Sunucu adresi, 192.168.1.1 veya ftp.domain.com dur, dosya /ftp/config/ e girilmistir. Alt directory'nin sonu OK veya "/" sizdir.
Kullanici ismi	Sunucu adres tanimi, FTP Sunucu ; TFTP Protokolu tanimina gerek yoktur; indirme 'de ftp protoklu kullaniliyor ise, standardi ftb kullanicidir.
Sifre	FTP Sunucunun kullanici sifresine gore, ilgili sifre tanimi.
Konfig.dosya ismi	Sistem konfigrasyon dosyasinin o an ki durumunun goruntulenmesi.
Encrypt Konfig. tusu	Konfigurasyon dosyasini karismis (encrypted) istiyorsaniz, karistirma sifresi girin.
Protokolu Tipi	3 tip secin, sunlar FTP,TFTP,HTTP.
Fasila zamani guncellemesi	Birim olarak saat
Guncelleme Modu	Guncelleme Mod'lari 1. Hukumsuz, "guncelleme yok" demektir. 2. Reboot'tan sonra guncelle: Reboot'tan sonra yukselt demektir. 3. Zaman araligi'ndan sonra guncelle: ne kadar sure ile yuksetilecek ?.

5.3.5.2 Systemlog (sistem kaydi) Systemlog (sistem kaydi), sistem program gidisi aninda, ilgili mesajlarin kaydi icin olusmus bir kullanici/Sunucu sistemi saglar. Programdan mesajlari alir, kaydeder ve onem ve tipine gore siralar. Ve ondan sonra kullanici tarifine gore dizer. Bu onem tanimi sisteme gozlem ve analiz gucu kazandirmistir. Syslog 8 seviyeye bolunmustur, bunar;

Level 0 - acil, duzeltme ikaz mesaji; sistem calismazsa

Level 1 - ciddi uyari, duzeltme ikaz mesaji; tehlikeli problem

Level 2 - kritik, ciddi hatlar, yani, sistem gerekli kaynagi bulamiyor ve yanlis dosya yukseltmesi

Level 3 - hata, sistemi negatif etkileyebilir

Level 4 - ikaz, sistem genel calismasinda tehlikeli degil, ancak ikaza deger, potansiyel tehlike..

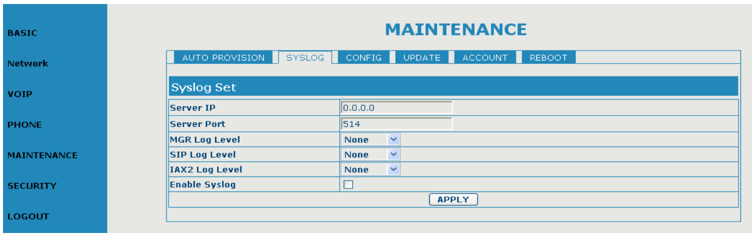
Level 5 - bilgi, sistem duzgun calisiyor, ancak su sartlara daha iyi calsima icin dikkat edin

Level 6 - duzeltme calismasi,

Level 7 - ayiklama, kullancilarin sistemin iyilestirmesi icin dikkat etmesi gerekir

Simdi ki durumda en alt seviye ayiklama harektleri syslog'a bilgi olarak gider ve telnet altinda dir**.**

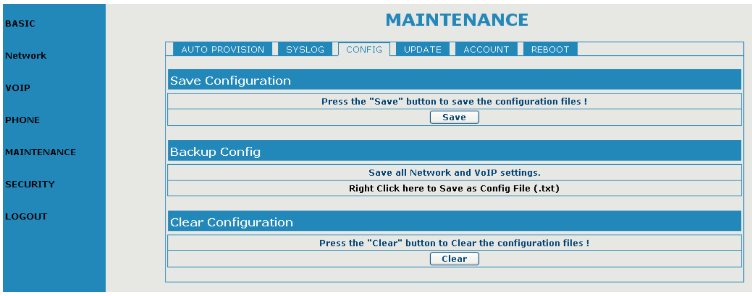
5.3.5.2



5.3.5.2 System log

Sunucu IP	tanimlama
Sunucu Port	tanimlama
MGR Log Level	tanimlama
SIP Log Level	tanimlama
IAX2 Log Level	tanimlama
Syslog etkinlestirme	Tanimlama

5.3.5.3 Konfigurasyon Tanimlamasi

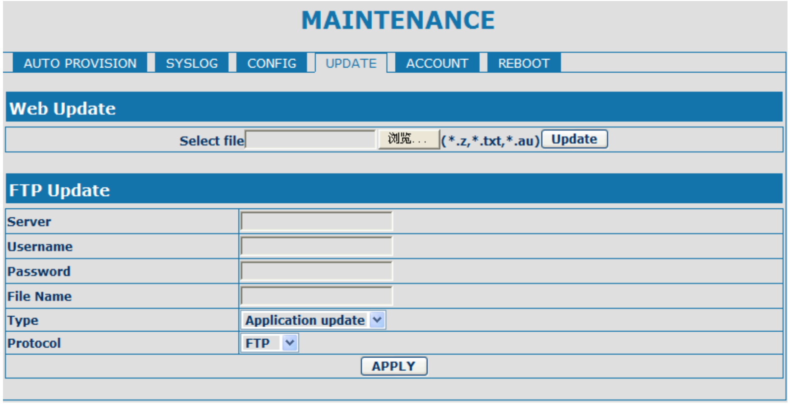


5.3.5.3 Konfigurasyon Dosyasi

Kayit	Dikkat: Telefonda yaptiginiz degisiklikler hemen gecerli olur. Kaydi unutursaniz, eski durumdan gider.
Yedekle	Fareyi sag tiklat ve "save target as" i sec, konfigurasyon dosyasini herseyi ile indiririsin.
Temizle	Sistemi varsayilan sekilde tani ve telefonu reboot et. Dikkat: Admin ile log olursan, Konfig. Temizle'de, hersey fabrika kaydina doner. Misafir olarak log olursan, “musteri numarasi” haricinde hersey temizilenir ve goreceli konfigurasyon o anda ki konfigurasyondur (SIP1-SIP2).

5.3.5.4 Konfigurasyon yukseltmesi

Bu sayfa ile, gecerli konfigurasyon dosyasi uzerinde telefon ayari.

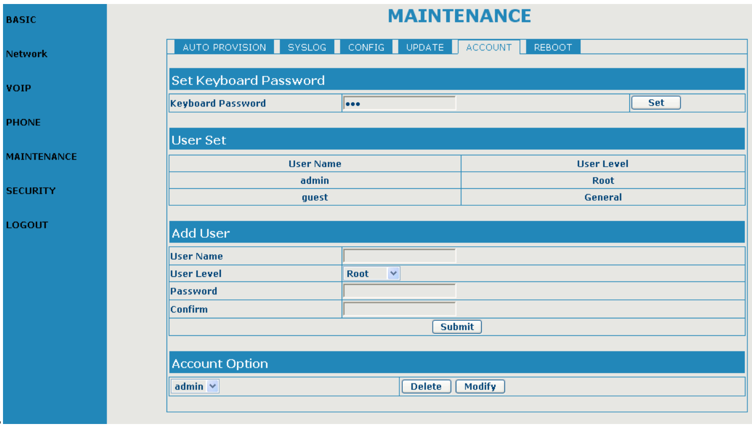


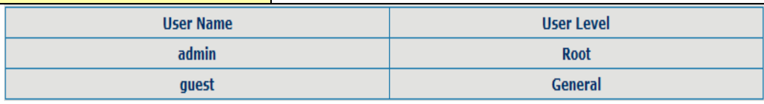
5.3.5.4 Yukseltme Yenilenmesi

Web Guncelleme	Web'e bakilarak, once ki konfigurasyonu bul (veya fabrika konfigurasyonu), o an ki telefona indir, bu, her birimin adim adim problemlerini onler. Ayni zamanda bu, dosya yukseltme ve mmiset indirmesi'dir de. “Update” I tikladiginizda, eketiftir.
Sunucu	Upload veya download olan FTP Sunucu adres ayari. Sunucu adresi, IP fromada, yani 192.168.1.1 olabilir veya domain formda, yani, ftp.domain.com olabilir. Bunun yaninda sistem, sistem alt kutuphane ayarlamasini da , yani, 192.168.1.1/ftp/config/ form, veya ftp.domain.com/ftp/config olan server adres ayarlamasi 'ni da destekler. Bu su demektir; ziyaret edilen Sunucu adresi, 192.168.1.1 or ftp.domain.com 'dur. Dosya su duzende olur; **/ftp/config/** Subdirectory's suffix, **“/”** suz o.k. dir .
Kullanici Ismi	FTP upload veya download kullanici isim tanimi. TFTP secilirse, kullanici ismi ve sifresini girmeye gerek yok
Sifre	FTP upload veya download kullanici isim tanimi.
Dosya ismi	“Yukseltilen dosya ismi” veya “konfigurasyon upload veya download” tanimi.
Dikkat: Export edilmis konfigurasyon dosyasi degistirilebilir. Bunun yaninda, bu module-importing islemini de destekler. Ornek; Sistem miprt etmek icin, konfigurasyon dosyasinda, sadece SIP modulu tutmaniz yeterlidir. Bundan dolayi diger modullerin konfigurasyonu, baska bir konfigurasyona import isleminde kaybolmazlar.	
Tip	Sistem tipi ayari: 1. Uygulama guncellemesi: sistem update dosyasinin download'u 2. Export dosya konfig. : FTP/TFTP Sunucuya yukle, isimle ve kaydet. 3. Export dosya konfig.: FTP/TFTP Sunucudan telefona indir, telefon reset'ten sonra durum o.k. Dir..
Protokol	FTP/TFTP server', sec

5.3.5.5 Hesap Konfig.

Web'te hesap ekleyip, cikarabilir veya hesaplarin otoritesini degistirebilirsiniz.

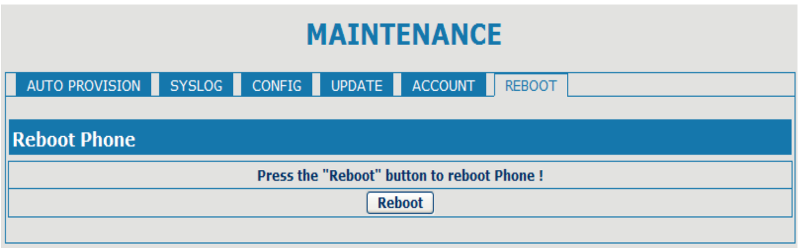


<table>
<tr><td>Klavye sifresi</td><td>“s” tuslu kalveyden telefondan telefona giris sifrelerini degistireblirsiniz. Rakkamdir.</td></tr>
<tr><td colspan="2">

User Name	User Level
admin	Root
guest	General

Bu tablo, o an ki kullanicinin “ciktigini” gosterir.</td></tr>
<tr><td>Kullanci İsmi</td><td>Kullanici isim tanimi.</td></tr>
<tr><td>Kullanci Seviyesi</td><td>Kullanici seviye tanimi.
Root kullanici her seyi yapar, genel kullanici sadece okur.</td></tr>
<tr><td>Sifre</td><td>Sifre giris.</td></tr>
<tr><td>Teyit</td><td>Sifrenin teyidi.</td></tr>
<tr><td colspan="2">Kullanici'yi sev ve degisiklik kabulu icin, Modify'a bas, hesabi silmek isitiyor isen Delete'e bas.
Yetki seviyesi genel olan, genel kullanici sadece ekleme yapar.</td></tr>
</table>

5.3.5.6 Reboot

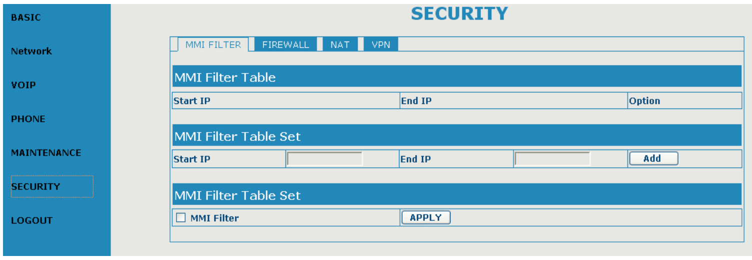


Telefonda yapilan degisiklikleri, gecerli yapmak icin telefonu reboot etmelisiniz, reboot'u tiklayin, telefon reboot gececektir.

Reboot 'tan once tum degisiklikleri, kaydedin.

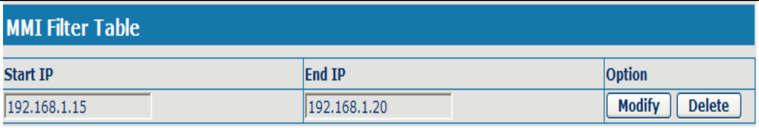
5.3.6 Guvenlik

5.3.6.1 MMI Filter

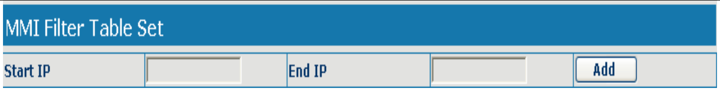


5.3.6.1 MMI Filter

Kullanici, sisteme ozel IP verebilir ki, bunlar onceden belirtilmis telefonun MMI' ina erisim ve telefon kontrolu icindir.



MMI Filter IP Table list:



Telefona erisen IP Adres Segment'lerinin eklenmesi veya silinmesi.

Start IP sutununda birincil IP Adresi verin, Bitis IP sutununda son IP Adresi verin,

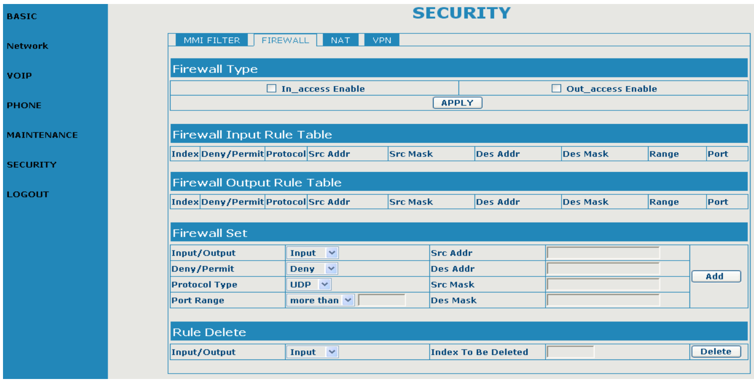
ve bunu IP segmentine eklemek icin Add'i tikleyin.

Secilen IP segmenti'ni silmek icin Delete'i tiklayin.

MMI Filter	MMI Filter'i etkinlestirmek veya ...memek icin secin; Etkin yapmak icin **Apply** 'a basin

Not : MMI filter siniri disinda, “IP ziyaret” set'i yapmayin, yoksa kullanici web'e giremez.

5.3.6.2 Firewall



5.3.6.2 Firewall Konfigurasyon

Bu arayuzde, kullanici, one bir firewall koyar ki,
Ozel network'lerden, yekisiz kullanicilar, engellesin (input kurali);
Internet erisiminden, yetkisiz ozel network'ler, engellesin (output kurali);

Firewall 2 tip kural destekler: Giris (input) kurali ve Cikis (ouput) kurali.
Herbiri ort. 10 kalem'i destekler.

Kullanici, web girisinden, input ve output kurallari ile firewall'u “etkin” veya “olu” kilar.
Sistem yetkisiz kisilerin erismini veya diger networklerin sisteme erismini engeller.
İki erisim listesini destekler; biri input paketlerinin filtelenmesi, digeri de output paketlerinin filtelenmesi ile ilgildir. Her tip liste, 10 kalemi ekleyebilir.

Referansiniz icin verilen ornek

☐ In_access Enable	☐ Out_access Enable

Input/Output	Input	Src Addr		Add
Deny/Permit	Deny	Des Addr		
Protocol Type	UDP	Src Mask		
Port Range	more than	Des Mask		

Ic erisim / In access Etkin;	In access kuralini calistirmak icin secin
Dis erisim / Out access Etkin;	Out access kuralini calistirmak icin secin
Input/Output	Input kurali veya Output kurali icin secin
Ret/Izin	Ret veya izin kurali icin secin
Protokol Tipi	Protokol tipi. TCP, UDP, ICMP, veya IP secin
Port Siniri	Siniri belirle
Src Adresi	Source (kaynak) adresi belirle. Tek bir IP adresi, network adresi, komple **0.0.0.0** veya ornegi, **192.168.1.0** olan ve ***.*.*.0** 'a benzer network adresi olabilir.
Des Adresi	Hedef adres (destination) tanimi. Tek bir IP adresi, network adresi, komple **0.0.0.0** veya ornegi, **192.168.1.0** olan ve ***.*.*.0** 'a benzer network adresi olabilir.

Output (Cikis) kurali eklemek istiyor isen, **Add** 'i tikla. Cikis terimini (output access) i etkinlestirmek icin **Apply** tusunu sec.
Boylece unite, **192.168.1.118** 'yi ping ettiginde, out access kuralina gore, sistem **192.168.1.118**'e data paketi gonderme istemini reddecektir. Network ID'si **192.168.1.0** olan diger birimleri ping'inde hersey normal olacaktir.

Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
1	Deny	ICMP	192.168.1.10	255.255.255.0	192.168.1.30	255.255.255.0	More than	0

Rule Delete

Input/Output	Input	Index To Be Deleted		Delete

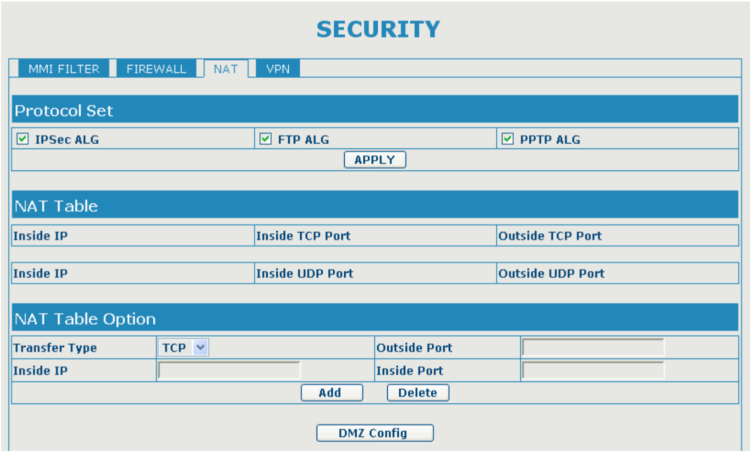
Secilen kurali silmek icin **Delete** 'I tikla

5.3.6.3 NAT Config - Konfigurasyonu

NAT, Net Adres Translation 'in bas harfleridir; IP adresi donusumu icin sorumlu bir protokoldur.

Daha acik bir degisle, IP donusumu ve acik kullanim icin olan ozel network portlari icin sorumlu bir birimdir.

Bu ayni zamanda genelde tanimladigimiz, IP adres map'lemesi icindir.



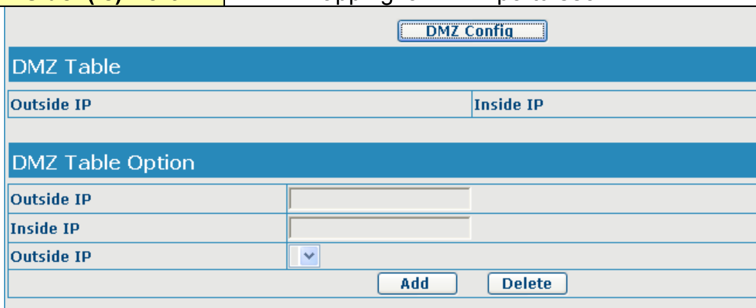
5.3.6.3 NAT Konfigurasyonu

IPSec ALG	Encryption (karistirma) teknigidir. IPSec ALG 'I etkin yapar veya birakirsiniz, standarti, etkin'dir.
FTP ALG	FTP, intranet IP paket gonderdigi zaman, intranet IP'yi extranet IP'ye donduren, “baglanti katman (connection layer)servisi'dir. FTP ALG 'yi secerek etkin yapar veya birakirsiniz, standarti, etkin'dir.
PPTP ALG	PPTP ALG 'yi secerek etkin yapar veya birakirsiniz, standarti, etkin'dir.

NAT TCP harita tablosu'nu gosterir.

NAT UDP harita tablosu'nu gosterir.

Transfer Tipi	Select the NAT mapping Protokolu style, TCP or UDP.
Outside Port	Set the LAN port of the NAT mapping
Inside (ic) IP	NAT mapping yapmak icin, LAN arayuz'e baglanmada, birimin IP adres set'i.
Inside (ic) Port	NAT mapping icin LAN portu sec.



Outside (dis) IP	Dis, Wan port IP adresi DMZ icin ;
Inside (ic) IP	Ic LAN port IP adresi of DMZ icin ;

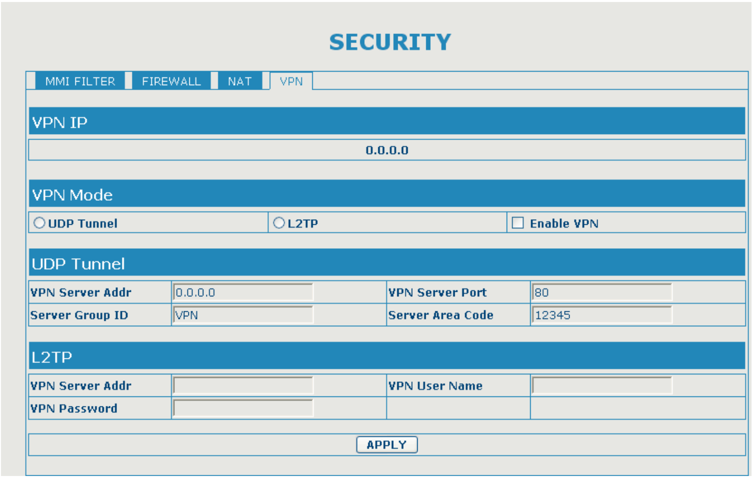
Secilen harita tablosunu silme icin **Delete** tusuna bas.

Not: 10M/100M uyumu, network kartidir ve sunun icin; 100M kopru modu'nun altinda, diger uniteler fiziksel hiz olcumleri ve ilgili testleri icin. Giden ses kalitesi icin gonderim performansinda NAT 'da bir takim feragatlar yaptik. Sistem idle mod 'da iken, gonderim full kapasite ile olur, yani 100M e yaklasan hizda, kalite garanti edilmez.

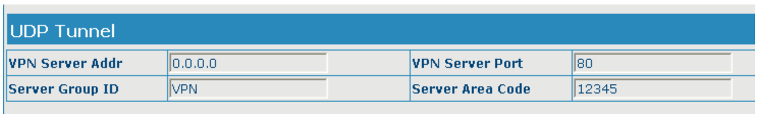
5.3.6.4 VPN

Bu web sayfasi, kullaniciya, genel network'ten guvenli baglanti modu erisim saglar.

Soyle denebilir, ozel bir erisim tuneli ile kullanici heryerden sisteme girebilir.

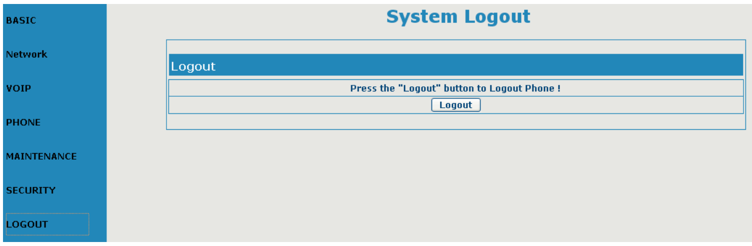


5.3.6.4 VPN Konfigurasyonu

VPN IP	O an ki VPN IP adres 'i gosterir.
	
Not: UDP Tunnel (VPN Tunnel) veya VPN L2TP secimi. Sadece o an ki durumdan birini secersiniz; kayit ve reboot edin.	
VPN Mode	Konfigurasyon, VPN mode 'u desekliyor mu ? .
VPN Konfigurasyon	
	
PN Sunucu Addr	Set VPN Sunucu adres.
VPN Sunucu Port	Set VPN Sunucu port Numarasi.
L2TP Konfigurasyon	
	
VPN Sunucu Addr	Set VPN L2TP Sunucu adresi, seti.

VPN Kullanici	Set VPN L2TP kullanici isim, seti.
VPN Sifre	İlgili VPN L2TP sifre, seti.

5.3.7 Logout - Cikis



Web'ten cikmak icin 【Logout】 u sec, diger giris 'te isim ve sifre ister.

5.4. TELEFON ETMEK

1. Arama / gorusme sesinin ayarlanmasi

Gorusme annda, ses siddetini ayarlamak icin, **VOL+/-** tusuna basin.
Ayarlandiktan ve ahize yerine konduktan sonra ses siddeti o seviyede kalir.

2. Ahize ve “ahizesiz gorusme (hoparlor ile)” modlari arasinda gecis

a. Ahize'den ve “ahizesiz gorusme” ye gecis
   Sadece **SPKR tusu**na basin ve ahizeyi yerine koyun
b. “Ahizesiz gorusme” den Ahize'li gorusmeye gecis
   Ahizeyi kaldirin

3. Aramayi cevaplama

a. Telefon caliginda, ahizeyi kaldirin veya **SPKR tusu**na basin.
b. Bir gorusme aninda, (telefon 2 hatli ise) ve 2. hattan da arama geliyor ise;
2. aramanin geldigi tusa bastiginizda, gelen arama ile gorusemeye gecersiniz.
ilk gorusme devre disi kalir.
Hangi hattin devre disi kalmamasini istiyor iseniz, **HOLD**'da basin, ve ardindan hangi hatti askiya almak istiyor isenin **o hattin tusuna** basin, o arama askida kalacaktir.

4. 3 yollu konferans gorusme

a. Gorustugunuz anda, **HOLD**'a basin, telefonun diger hattindan 2. gorusmeye gecebilirsiniz. Gectikten sonra **CONF tusu**na bastiginizda 3 kisi ayni anda karslikli gorusebilir.

b. **bu durumda** sadece hangi hat ile gorusmek istiyor iseniz, o hattin tusuna basin, sadece o hat ile gorusmeye baslarsiniz. Veyahut konferans aninda, **HOLD tusu**na basip, karsinizda ki 2 kisiyi aski'ya alabilir ve istediginiz **hat tusuna** basip, sadece o hat ile gorusebilirsiniz. Bundan sonra **CONF tusu**na tekrar bastiginizda, konfrans moduna tekrar donebilirsiniz.

5. Son arana numarayi tekrar cevirme

Son aradiginiz numarayi tekrar cevirmek icin sadece REDiAL tusuna basin.
Ancak gizlilik icin, bu ozellikte saklanan numralar 5 dakika sonra kaybolur, daha acik deyisle bu ozelik 5 dakika sonra yoktur.

6. HOLD (TUT) tusunu kullanmak

Ahize elde iken, **HOLD tusu**na basin ve ahizeyi govdeye birakin.
Gorusme yaptiginiz kisi, askida kalacaktir (**LED gosterge isigi** yanacaktir)
Tekrar ahizeyi kaldirdiginizda, karsinizda ki kisi aski'dan cikacaktir.

Paralel yapida telefon ikinci bir aramayi alir ise,
telefon otomatik **HOLD** moduna girebilir.

7. Mute (Kesme)

Bu tusa bastiginizda, sizden karsiya herhangi bir ses gitmez.
Tekrar bastiginiz ses gider.

8. Zil sesini ayarlamak

Govdinin arkasinda ki **Low** ve **Hi** (**Alcak** - **Yuksek**) seceneklerinden birini secerek zil ses siddetini ayarliyabilirsiniz.

9. Mesgul durumundan otomatik cikma

Bazi kereler, ahize govdedin uzerine tam olarak oturmayabilir.
Bu durumda telefon devamli mesgul konumdadir.
Tanimlanan belli bir sure sonunda telefon devamli **mesgul** durumunda calmaya zorlanirsa, **mesgul** konumda cikar ve **calmaya** baslar.

10. CALLER ID ozelligi

Telefon, ilk olarak o hatta calismaya basladiginda, gelen aramanin **DTMF** mi veya **FSK** (darbeli) mi

oldugunu otomatik olarak anlar ve bunu hafizaya atar. Bu anlama ozelligini degistirmek icin (baska bir hatta kullanilacak ise), telefon hattan cikarilmali ve akim tekrar verilmelidir.
Bu telefonlarda, “gizlilik” icin, gelen arama bilgisi belirli bir zaman sonra kaybolur.

11. Hizli arama tuslarini kullanmak

“Ahize elde” durumda, hizli arama tuslarindan birine bastiginizda,
o tus hangi dahili numara kaydi yapilmis ise, o numarayi arar.

Hafiza tusuna kayit :

Ahize yerinde modunda, **HOLD tusu**na basin+giris yapmak istediginiz **dahili no.**yu+**HOLD tusu**na+**hangi tusa atanacak** ise o tusa basin.

12. Mesaj isigi

Ahize govde'de modunda, **mesaj isigi tusu**na basin, telefon otomatik olarak,
programlanmis mesaj servisini arayacak ve o odanin mesajini, sistemin
ozelligine gore okuyacaktir.

Ahize govde'de modunda, **Mesaj isigi tusu**na basin, telefon otomatik olarak,
programlanmis mesaj servisini arayacak ve o odanin mesajini, sistemin
ozelligine gore okuyacaktir.

Mesaj isiginin hafiza tusuna kayit :

HOLD tusuna basin+giris yapmak istediginiz **dahili no.**yu+**HOLD** tusuna+**mesaj isigi tusu**na basin.

13. Tuslarin hizli bir sekilde hafiza programlanmasi

Bu islem icin ana merkezi ve el programlama cihazlari kallanilacaktir ve bu unitelerin ayri kullanim klavuzlari vardir.
Lutfen bunlari okuyun.

5.5. SES-AUDIO BOLUMU

1. Ses calmak :

Audio-ses calma moduna girmek icin, **AUDIO ON/OFF tusu**na basin,
daha sonra **RADiO/LINE IN tusu**na basarak, radyo ve direkt muzik calma
cihazi AciK moduna girebilirsiniz.

Press AUDiO ON/OFF key to enter or exist audio play mode, then press key t realized cyclical selection of LiNE iN and FM-RADiO mode.

● **iPod/MP3** cihazini calmak

iPod/MP3 'u ozel baglanti kablosu ile, **LINE IN** port'una baglayin ve daha sonra
radyo yeirne **LINE IN** modunu secerek,
ilgili unitenin ses vermesini aktif duruma getirebilirsiniz.

● **FM Radyo**

RADIO/LINE IN 'e basin ve **FM-RADIO modu**nu secin, daha sonra **TUNING+/-** tusuna basarak, arzu ettigimiz yayin kanalini bulun.

Tusa her basista, frekans 0.1 Mhz degisecektir. Bu tusa arzu ettiginiz yere gelene kadar basabilirsiniz. Kanali buldugunda arama gidisi duracaktir.

● Pause ve calma icin **AUDIO ON/OFF** a basin. **VOLUME +/-** basarak ses siddetini ayarlayabilirsiniz, (0 -30 seviye arasinda), fabrika standardi 15 'tir.

●**Urun Modunu Otomatik Tesbiti**

Ses modunda, telefon calar ise, bu islem otomatik olarak durakrsar.
Telefon konusmasi bittiginde telefon tekrar ses calma moduna girer.

2. Zamanin Ayarlanmasi

Zaman ancak, **AUDIO OFF** modunda(**LINE IN** veya **FM-RADIO** konumda degil) tanimlanabilir.

- **VOLUME- tusu**na ort. 1 saniye basin, LCD de saat digit flas edecektir, **TUNING+/- tusu**na basarak saati ayarlayabilirsiniz.
- Daha sonra **VOLUME- tusu**na tekrar basin, LCD de minute digit flas edecektir, **TUNING+/- tusu**na basarak dakikayi ayarlayabilirsiniz.
- **VOLUME- tusu**na basilarak İslem sonlanabİlİr.
- **VOLUME+ tusu**na basilarak, 12 saat veya 24 saat formatindan biri secilebilir.

3. Alarm'in kurulmasi

● Alarm Zamani

* ALARM SET tusuna basin , LCD de saat digit flas edecektir,
VOLUME+/- tusuna basarak saati ayarlayabilirsiniz.
* ALARM SET tusuna tekrar basin , LCD de minute digit flas edecektir,
VOLUME+/- tusuna basarak dakikayi ayarlayabilirsiniz.
ALARM SET tusuna basarak islemi bitirebilirsiniz.

2 Alarm'i cozmek

ALARM tusuna tekrar tekrar basarak, alarm'i kapatip acabilirsiniz.
Ekranin sag ust kosesinde, **alarm isareti** gozukecek veya gozukmeyecektir.

3 Alarm erteleme imkani

Alarm caldiginda **SNOOZE tusu**na basarak, alarm calmasini 10 dakika icin erleyebilirsiniz. 10 dakika sonra alarm tekrar calacaktir.

ALARM VOLUME tusu harincinde her hangi bir tusa basarak veya ahizeyi kaldirarak alarm'i iptal edebilirsiniz.

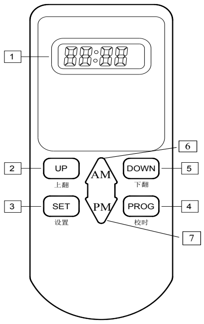
Alarm aninda muzik caliyor ise, sistem, **Alarm calma** moduna gecer.
AUDIO OFF modunda, **ALARM VOLUME tusu**na basarak, alarm volume gosterge seviyesini 5 er seviye artirabilir veya azaltabilirsiniz.
En yuksek 20, fabrika standardi 10'dur.

4. LCD isik siddetini ayarlamak
Alarm calma modu haricinde SNOOZE **tusu**na basarak LCD isik gucunu ayarlayabilirsiniz, 3 seviye vardir.

5. Sarj port'u
* Mini USB kablosu ile kucuk el cihazlarini sarj imkani icin,
  iki adet USB port'u vardir.
* iPod USB port'u ---iPod、iPhone Apple 'lar icin .
* USB CHARGE port'u——Gsm telefon ve kucuk el cihazlari icin.
* LiNE iN audio input port'u --- 3.5mm dual track audio hatti ile iPod/MP3 icin.

6. Homing system (ev bakim modu, house keeper icin)
ALARM tusuna devsamli 2 saniye basin, unite “0 modune” girecektir.
Housekeeping bakimi icindir.
onemli; 1 once ki misafirin butun alarm kayitlari sifirlanir !

7. Adjust time with remote control timing cloner ile saati ayarlama.
***** 1.5V 'luk piller takin. Polarite'ye dikkat edin (- +)



Timing cloner unitesi

1. LCD display time / gosterge
2. UP key——Saat yukari.
3.SET key——Saati ayarla.
4.PROG key ——Program zamani
5.DOWN key——Saat asagi .
6.AM tusu——12 saat format
7.PM tusu ——24 saat format

Zamani ayarlamak

●SET tusuna basin, saat digit'i yanip sonecektir, UP/DOWN ile ayarlayin ,

● SET tusuna tekrar basin, dakika digit'i yanip sonecektir,
UP/DOWN ile ayarlayin

●SET tusuna basarak i,slemi bitirin.

●AM tusuna basarak 12 saat format'i, PM tusuna basarak
24 saat format'ini secin.

Yukarida ki islem bittiginde, uniteyi telefonun 1 metre yanina yaklastirin ve PROG tusuna basin. Cihaz bilgiyi telefona yukleyecektir

Problem bulmak

1. cevir sesi yok

Hatti kontrol edin
Ahize ve telefon kablolarini kontrol edin

2. Zil veya devamli zil sesi yok

Telefon hattini kontrol edin

Ayni hatta ki telefon sayisina bakin ve Bittel temsilcisine bildirin

3. Gorusme aninda gurultu var

Butun hat ve kablolari ve baglanti yerlerini kontrol edin.

4. Caller iD yok

Caller iD ozelligi acik mi ?

Akim'i kontrol

5 . Ses yok

Akim var mi ?

10pins ribbon kabloyu ve baglantisini kotrol edin.

6. Radyo sesi net ve kaliteli gelmiyor

Anteni kontrol edin

Arzu edilen kanalin, bolgede kuvvetli yayini yok

FCC APPROVAL

FCC, Part 68 (ABD) kurallarina tamamen uymaktadir.
FCC 'ye uyumlu oldugu, model cikartmasinda kayit numarasi ile belirtilmekte ayrica (REN) (zil esit numarasi) belirtilmektedir. Bu bilgiler, ulkeninizin telefon sirketine verilmesi icin urun uzerinde belirtilmektedir.

CE APPROVAL

Complies with CE onay ve sartlarina tamamen uymakadir..

Fisler

RJ11C USOC standarti.

GARANTI

urun, alindigi tarihten 12 ay sure icinde tamamen uretim ve
imalat hatalarina karsi garanti kapsamindadir.

Asagida ki durumlar disinda Bİttel urunu onarma veya bir yenisi ile degistirmek sorumlulugundadir.

1) Normal kullanim disi,

2) Yetki verilmemis kisi veya kurumlarca onarim veya modifikasyon.

3) Bu urun, rutubetli, kirli, tozlu ortamlarda kullanilmaz.

4) Tasima veya depolama aninda islanma veya darbe

BU LiMiTED GARANTİ ALiCiYA oZEL HUKUSAL HAKLAR VERİR.

ALiCiNiN BUNUN YANiNDA BoLGELESEL BAKiM HAKLARi DA VARDiR.

Garanti, urunun Bittel veya onun kabul edilmis distributor'unden

alindiginda gecerlidir.

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No.1 Rizhao North Road,Rizhao,
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Uygunluk Taahhudu DOC_

Taahhut Numarasi: DOC110118 Tarih: 18 Ocak 2011

Biz, **Shandong Bittel Electronics Co., Ltd.**
No.1 Rizhao North Road, Rizhao Shandong, 276800, Cin

olarak taahhut ederiz ki, asagida sunulan modeller,
tamamen **EC Directive 99 / 5 / EC** standarlarina uygundur.

BT-2008(58IPTSD-10S)
BT-2008(66IPT)
BT-2008(66IPT-T)
BT-2008(67IPTSD-10S)
BT-2008(68IPTSD-10S)
BT-2008(69IPTSD- 3S)

ve bu urunler, ayni zamanda asagida belirtilen
istem ve harmonize standartlara tamamen uyumludur.
Urunler asagida ki standarlara gore gozlenmis, kontrol ve test edilmistir:

LVD	**EN 60950-1:2006**
EMC	**EN 55022:2006**
	EN 55024 : 1998 + A1:2001 + A2:2003
TBR	**ETSI ES 203 021 -2 V2.1.2 (2006-01)**
	ETSi ES 203 021 -3 v2.1.2 (2006-01)
	TBR 38 May 1998

Yetkili Imza: Isim: **Jiangwen Tian**
Mevki: Chief Engineer

Bittel

Shandong Bittel

Electronics Co.,Ltd

No.1,

Rizhao North Road, Rizhao,

Shandong,China

Declaration of Conformity

Declaration Number:DOC111101

Date: 2011.11.01

We, **Shandong Bittel Electronics Co., Ltd.**

No.1 Rizhao North Road, Rizhao Shandong, 276800, China

certify that the products described below are in conformity with **the EC Directive 99/5/EC**,

Model numbers,

HA9888(**12**)TSD

HA9888(**32**)TSD, HA9888(**32**)TS

HA9888(**38**)TSD, HA9888(**38**)TS, HA9888(**38**)TD

HA9888(**41**)T-18, HA9888(**41**)T-5, HA9888(**41**)T-0, HA9888(**41**)T-T

HA9888(**48**)TSD. HA9888(**48**)TS. HWD9888(**48**)TSD.

HA9888(**58**)TSD, HA9888(**58**)TS, HCD9888(**58**)TSD, HCD9888(**58**)TS, BT-2008(58IPTSD-10S)

HA9888(**66**)T, HA9888(**66**)T-T, BT-2008(66IPT), BT-2008(66IPT-T)

HA9888(**67**)TSD, HA9888(**67**)TS, BT-2008(67IPTSD-10S), HWD9888(**67**)TS

HA9888(**68**)TSD, HCD9888(**68**)TSD, BT-2008(68IPTSD-10S), HWD9888(**68**)TSD,

HA9888(**69**)TSD, BT-2008(69IPTSD-3S),

are in conformity with the following essential requirements and harmonized standards.

The products have been assessed by the application of the following standards;

LVD	**EN 60950-1:2006**
EMC	**EN 55022:2006**
	EN 55024 : 1998 + A1:2001 + A2:2003
TBR	**ETSI ES 203 021 -2 V2.1.2 (2006-01)**
	ETSI ES 203 021 -3 v2.1.2 (2006-01)
	TBR 38 May 1998

Authorized Signatory:

Jiang Wen Tian

Name: **Jiangwen Tian**

Title: **Chief Engineer**

Product Overview

SIP is the acronym for Session Initiation Protocol. SIP phone is the telephone that transmits voice over network based on IP protocol, for example LAN (Local Area Network), MAN (Metropolitan Area Network) and INTERNET.

This BITTEL series SIP IP phones provide an affordable IP phone system with ease of use, superior voice quality, economic physical design, advanced services, and features, especially own some hotel special features.

Features And Specification

2.1. Hardware Features

1. CPU: BCM1190/DRAM:8MX16 bit / 4MB Flash memory
2. Ethernet Port—2X 10/100M Connectors
3. AC/DC adaptor –Input AC110-230V, Output DC 5V1A or POE
4. Power Dissipation---1.5W (leave unused)/ 1.8W (activity)
5. Operating Requirements

Operation temperature: -0 to 50° C (32° to 122° F)

Storage temperature: -20° to 65° C (-4° to 149° F)

Humidity: 10 to 65%

b) Software Features

6. DHCP support for automatic allocation IP addresses and other parameters
7. Setting parameters through standard web browser (IE)
8. Automated provisioning of firmware and configuration via HTTP
9. Upgrade firmware via HTTP/FTP;
10. Multiple audio codec support: G.711A/u, G.723.1; G729a/b; G.722 voice code arithmetic
11. VAD (Voice Activity Detection) ,CNG (Comfort Noise Generation), Dynamic Jitter Buffer
12. G. 168 96ms echo cancellation
13. ITU-T standard signaling tone and DTM generation and detection
14. DTMF Transmission: Inband audio; RFC2833; SIP INFO
15. NAT support through-transmission technique and firewall

3. Call Forward / Call Waiting
4. Phone Book with 500 entries / N groups Speed Dialing Support (0≤N≤10)
5. Adjustable volume for both handset and speaker, ringer music selectable

6. Hot Line

c) Standard and Protocol

1. SIP (RFC2833; RFC3261; RFC2543; RFC2327; RFC3264; RFC3265; RFC3550)
2. IEEE 802. 3 /802. 3 u 10 Base T / 100Base TX
3. IEEE 802. 1P/Q Tag VLAN
4. QoS: Support for Cos Layer3 Qos (Diff-Serv) and Tos Layer 2 Qos (802. 1P/Q)
5. TCP/IP: Transmission Control Protocol / Internet Protocol
6. CMP: Internet Control Message Protocol / Syslog: The BSD syslog Protocol
7. RTP: Real-time Transport Protocol / RTCP: Real-time Transport Control Protocol
8. DHCP: Dynamic Host Configuration Protocol
9. VAD/CNG: Saving Broadband
10. DNS: Domain Name Server / TFTP: Trivial File Transfer Protocol
11. HTTP: Hyper Text Transfer Protocol /SNTP: Simple Network Time Protocol

Installation

3.1.Parts List

Check this following list before installation to make sure that you have received all items. If any item is not included in the package, please contact the distributor.

1) One SIP Main Case
2) One Handset
3) One handset cord
4) One Universal Power Adapter (only for adaptor power supply)
5) One product qualification and guarantee
6) One User Guide
7) One Ethernet Cable
8) One screw bag (for BT-2008(67-S/T-A)
9) One POE cable (for BT-2008(67/68)

5.2 Connecting the Power and Network

3.2.1 66IP/67IP/68IP/69IP

- Install the handset cord

Insert one end of the attached handset cord to jack on the handset, the other end to the jack on backside of the phone. Other operations please refer to 3.3

- Network connection

Power by adaptor ：Connect the RJ45 jack of POE cable line to the LAN port of HUB or switch , then connect the other RJ45 jack to the WAN port of the phone.

Power by POE：Connect the one end of the Ethernet cable line to WAN port of the phone, then connect the other end to the LAN port of HUB or switch.

Normal the net mode is DHCP. See 5.3.

- Power on the phone：

Adaptor mode：Insert the output jack of attached adaptor into the power jack of POE cable line.

POE Mode ：Insert the one end of the Ethernet cable line into the WAN port of phone , the other end into the POE port of the PBX.

- **58IP phone connection**

- Install the handset cord

Insert one end of the attached handset cord to jack on the handset, then insert the other end into the jack on the left side of the phone.

- Network connection

Insert the Ethernet cable into the LAN Jack. Plug the other end of the cable into HUB or switch. Normal the net mode is DHCP. See 5.3.

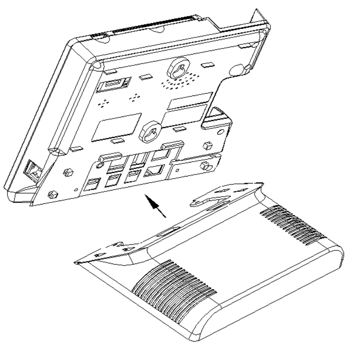
- Power on the phone：

Adaptor mode：Insert the output jack of attached adaptor into the power jack of POE cable line.

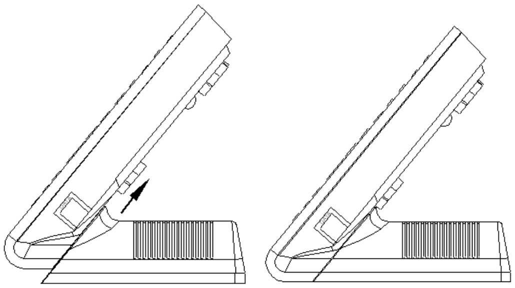
POE Mode ：Insert the one end of the Ethernet cable line into the WAN port of phone , the other end into the POE port of the PBX.

3.3 Connecting the Phone (67IP/68IP/69IP)

1. According to the direction of arrow, keep the back surface and base lock together.

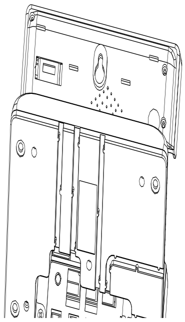


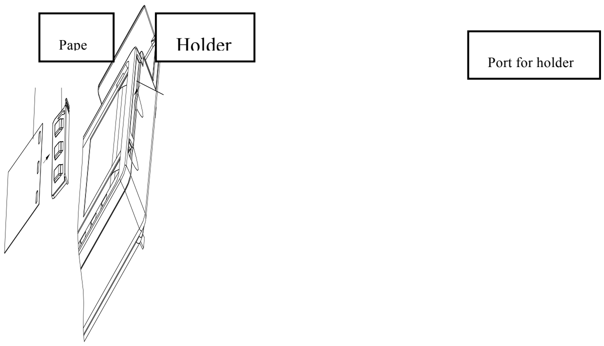
2. Push the base into the right position as the direction of arrow.



3. Fix with two screws..

4. Install paper faceplate and





3.4 Acoustics control and connection for signal wire (model 68 only)

Insert the signal ribbon cable which from the bottom side of upper case into the jack labeled “ AUDIO CONNECT ”, pay more attention to the plug and socket direction and make sure the connection is correct and firm.

3.5 As adopt the magnetic switch instead of the traditional hook-switch, install it far away from the strong magnetic field.

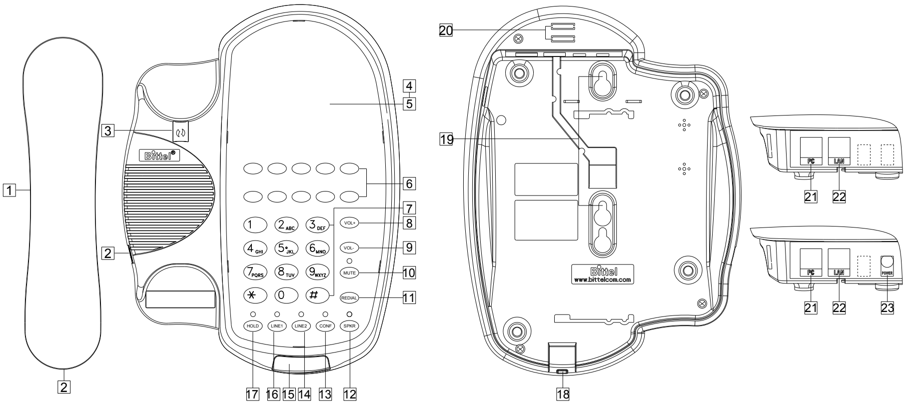
3.6 Install the phone in dry environment and avoid effected with damp.

3.7 The telephone should Fire-proof, shook-proof, clean the surface dust by soft cloth, prohibit to use chemical solvent .

LOCATION AND CONTROLS

4.1. Appearance

(**58IP**-S/T) SIP Phone Front Illustration (Refer to Fig 4.1.1)



4.1.1 58IP

1. Handset 2. Jack of handset cord 3.Wall mounting clip

4. Paper faceplate 5. Plastic overlay 6. Memory key 7. Digital key

8. VOL+ 9. VOL- 10. Mute key 11. Redial key

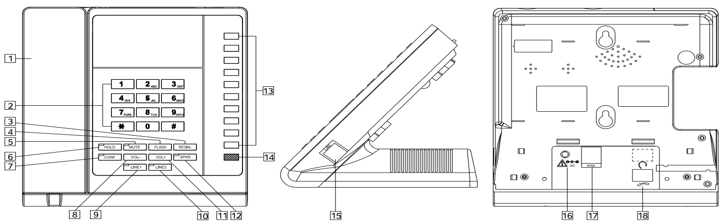
12. SPKR key 13. CONF key ((**58IP**-T)***model***) 14. LINE2 key ((**58IP**-T)***model***)

15. Message key 16.LINE1 key ((**58IP**-T)***model***) 17. Hold key 18. MIC

19. Wall mounting hole 20. Stand hole of placard 21. PC port

22. LAN port 23. Power port

(**67IP**-S/T) SIP Phone Front Illustration (Refer to Fig 4.1.2)



4.1.2 67IP-T-)/ (67IP-S)

1. Handset 2. Digital keys 3.Redial key 4. Flash key 5. Mute key

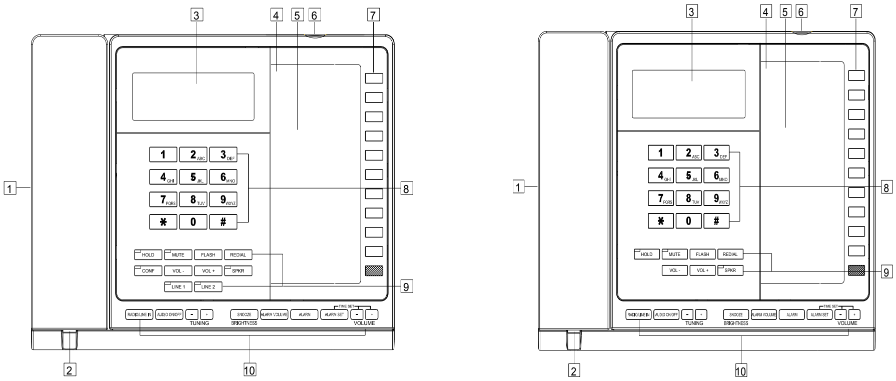
6. Hold key 7. Conference key ((**67IP**-T)***model***) 8. VOL-

9.LINE 1 key (**67IP**-T)***model***) 10. LINE 2 key ((**67IP**-T)***model***)

11. VOL+ 12. SPKR key 13. Memory key 14.MWL key 15.. LAN port

16. Power port   17. WAN port   18. Handset cord jack

(**68IP**-S/T) SIP Phone Front Illustration (Refer to Fig 4.1.3)



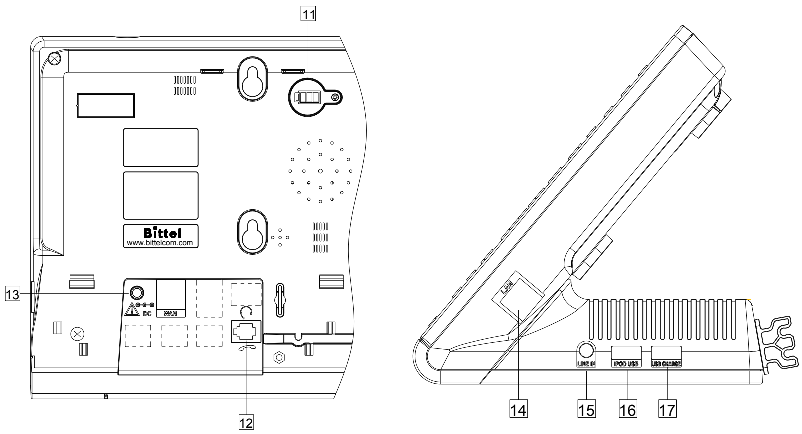
4.1 .3   68IP-T /68IP-S

1. Handset   2. Digital keys   3. LCD   4. Paper faceplate   5.Plastic faceplate

6. Paper Holder   7. Memory key   8. Digital key

9. Telephone function keys--- Hold key / Mute key /Flash key / Redial key / Conference key ((**68IP**-T)***model***) / VOL- / VOL+ / SPKR key   / LINE 1 key ((**68IP**-T)***model***) / LINE 2 key((**68IP**-T)***model***)   10. Acoustics keys

(**68IP**-S/T) SIP Phone Bottom and Side Illustration (Refer to Fig 4.1.4)

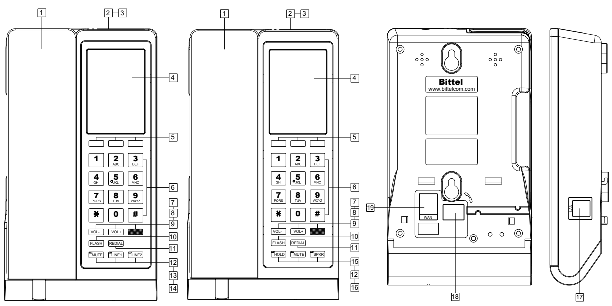


4.1 .4 68IP-T/ 68IP-S

11. Battery Box   12. Handset cord Jack   14.   LAN port   15. LINE IN port

16. IPod USB port   17. USB CHARGE port

(66/**69IP**-S/T) SIP Phone Bottom and Side Illustration (Refer to Fig 4.1.5)



1. Handset 2. Paper Faceplate 3. Paper Holder 4. Arcylic 5. Memory key 6. Digital kye 7. VOL- key 8. VOL+ key 9. Message Waiting Light

10. Flash key 11. Redial key 12. Mute key 13. LINE 1 key (two-line phone only), 14. LINE 2 key (only for two-line phone only) 15. HOLD Key (only for one line phone)

16. SPKR key (Only for one line phone) 17. LAN Port 18. Handset cord jack, 19. WAN Port

1. Telephone Function Keys

Keys	**Function**
ONE-TOUCH MEMORY KEYS	Speed-Dial key, each corresponds to a speed dial number, which will be called by a single press in off-hook mode.
MUTE	Mute local voice when in a call. Press again to restart the call. On-hook mode press this key show the IP .
VOL+	Increase the output volume of handset or speakerphone. Editing mode press it view up page.
VOL-	Decrease the output volume of handset or speakerphone. Editing mode press it view up page.
SPEAKER	Enter speakerphone mode. Press again to switch back to the handset mode.
HOLD	Temporarily hold the current call. Press again to un-hold.
FLASH	Off-hook or in SPKR mode, press this key once work as off-hook once.
MESSAGE	Press SKR, then press this key to play the pre-configured voice message according to notes..
REDIAL	Dial the last called number in on-hook mode.
CONF	Press it enter into three way conference when there are two lines working.
LINE1	Press it to active Line 1
LINE2	Press it to active Line 2

4.3 Acoustics ports function

●LINE IN port——Port for iPod ,MP3 and other audio signal connection.

●iPod USB port——Power charge port for iPod, iPhone and other devices.

●USB Charge port ——Power charge for other portable electrical equipments.

1. LED Functions

SPKR LED	On: the phone in the speaker call.
MESSAGE LED	On: There are unread voice messages. Off: No new voice message.
HOLD LED	On: The call is on hold.
MUTE LED	On: Local voice is muted when in a call;
LINE1 LED	On: Calling in line1
LINE2 LED	On: Calling in line2
CONF LED	On: Calling in three way

Configuration with Web Browser

1. Pre-setting pass word

There are two types for browser and order ： User mode and Manager mode. It will view and modify all items under Manager mode, but User mode only can view but can not modify SIP(1-2) items and server address and port.

According to the indication words to input the differ words , then enter into different mode:

4 User Mode

- User name：guest
- Pass word：guest

5 Manager Mode：

5.2 User name：admin

5.3 Pass word：admin

Noticee: For setting with hidden keys , the preinstall pass word :123

2. Configuration with web browser

When telephone and PC connected to network, double click logo on PC to open the IE；(press the “MUTE” to get the local IP).地址(D) http://xxx.xxx.xxx.xxx .If the telephone configuration port for Web logon is not 80 standard port, input http://xxx.xxx.xxx.xxx：xxxx/，otherwise user can not find the server.

After see following information and input the Username and Password then press 【Logon】enter into the interface.

Logon interface

Notice: 1. After modify each time , press【APPLY】will save this setting

2. If telephone not use bridging mode, 58IP's PC port address is 192.168.10.1; 66IP, 67IP, 68IP and 69IP's LAN's port address is 192.168.10.1. Details see 5.3.2.2 .

3. For setting 1-line IP phone, just need set SIP 1 or SIP 2. SIP 1 is priority.

4. When setting WAN port under this interface, the accordance port is LAN port for 58IP, the accordance port is WAN port for 66IP/67IP/68IP/ 69IP ; when setting LAN port under this interface, the accordance port is PC port for 58IP, the accordance port is LAN port for 66IP/67IP/68IP/ 69IP.

Noticee: The factory default network settings mode is DHCP, user just connect the device to such a networks environment which own DHCP service will link to the networks automatically. If user doesn't have DHCP environment then need to set the PPPoE or Static IP mode.

Memory Keys :

1) Press 【**1**】 key over 10s , telephone networks convert to Static IP mode automatically.

2) Press 【**2**】 key over 10s , telephone networks convert to DHCP mode automatically.

3) Press 【**3**】 key over 10s , telephone networks convert to PPPoE mode automatically.

Notice: ***After choose the networks mode ,the setting for IP Address, Net mask, Default gateway/Router and DNS may under the WAN low level menu . Details see* 5.3.2。**

3. Web functional descriptions

1. Basic setting

1. Status Display

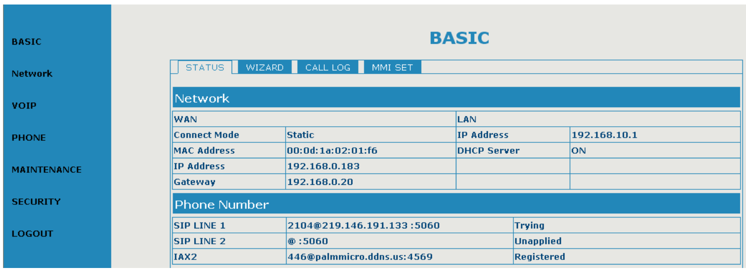


Fig. 5.3.1.1 Status display

Network	Display the current setting of WAN and \LAN: Including WAN IP（Static, DHCP, PPPoE ）and IP address，MAC address，Pre-setting Gateway IP address， LAN，IP address, DHCP server ON/OFF mode
Phone Number	Display the current number and status of SIP LINE 1—2 and IAX2. The bottom display is the telephone version and date used.

2. Configuration Guide

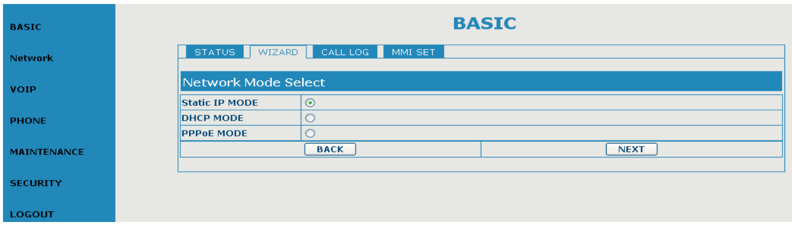


Fig. 5.3.1.2-1 Network Mode

Static IP MODE	Choose this item if the SIP supplier offer static IP address. After choose the item, user must fill following information: static IP address/ subnet mask/gateway/main domain name and others. If user don't know above information, it's better to ask assistant from server supplier or network management person .
DHCP MODE	Choose this mode, it can get such information automatically from DHCP server and no need to input any letters.
PPPoE MODE	Choose this mode, user must input ADSL account and pass words. See details in **5.2.**
Notice: Please choose the suitable mode based on the network environment . If choose static IP mode, press【NEXT】to set the items (LINE1 default) and view the setting items,. Press【BACK】to the previous web page.	

1）Static IP Setting

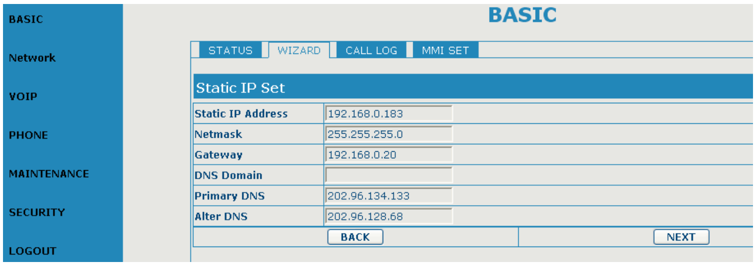


Fig. 5.3.1.2-2 Static IP Set

Static IP Address	Input the distributive IP address
Net mask	Input the distributive subnet mask.
Gateway	Input the distributive pre-setting gateway address
DNS Domain	Enactment DNS domain name suffix. If user input the domain name address and DNS can't analyze, the phone set will add the domain name after the domain name address then to analyze it.
Primary DNS	Input the main DNS server address.
Alter DNS	Input the spare DNS server address.

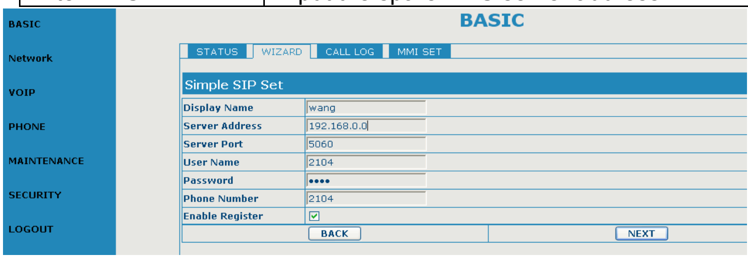


Fig. 5.3.1.2-3 SIP account setting

Display Name	Setting the display name, then the called party can see the calling party, English letters are allowable.
Server Address	Setting SIP register server address, support domain name kind address.
Server Port	Setting SIP register server signaling port .
User Name	Setting SIP register account.
Password	Setting SIP register the pass word of the account No.
Phone Number	Setting register number to SIP server.
Enable Register	Setting allowable / prohibited register

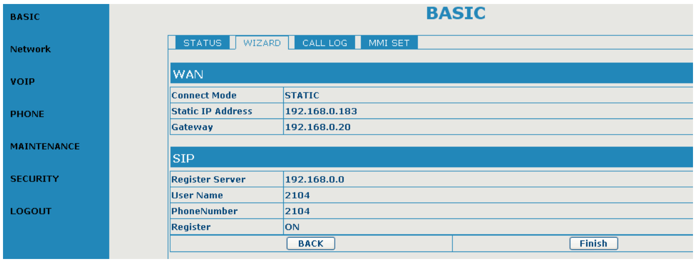


Fig 5.3.1.2-4 Static Setting Information

5.3.1.2-4 display the detail information of the static configuration manually, press【Finish】to finish the configure.

2）DHCP Setting

Choose DHCP mode, press【NEXT】to get the SIP parameters and view them , press【BACK】return back to the previous web page . Same as static IP mode。

3）PPPoE Configure

Choose PPPoE mode, press【NEXT】to get the account No., pass word and SIP parameters of the online device(LINE 1 default), press【BACK】return back to the previous web page . Same as static IP mode。

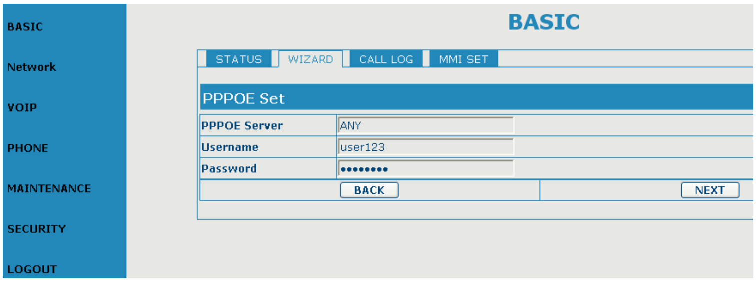


Fig . 5.3.1.2-5 PPPoE Server

PPPoE Server	Name of Server, if PPPoE supplier no special requirement, normally use the window default .
Username	Input the ADSL account number..
Password	Input the ADSL pass word.

Notice: After above setting finished, press【APPLY】key, then the telephone save the current setting and reboot ,after restart will use the registered account number to make callings.

3. Calling Information

Through this page to view all the called information (Numbers)

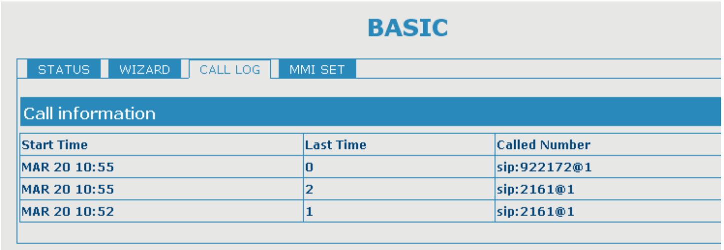


Fig.5.3.1.3 Called Information

Start Time	Recorder start time
Last Time	Calling time, a unit is second
Call Number	Recorded account No. and calling protocol and used line

4. MMI setting

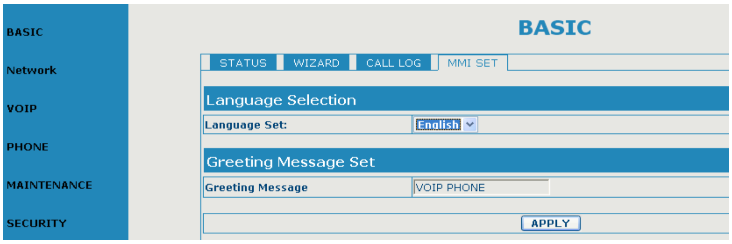


Fig. 5.3.1.4 MMI setting

Language Selection	Setting the display language .Factory default : English
Greeting Message	Greeting message is the words display on the LCD of the phone set. It support 16 digital English letters or Five Chinese letters . Factory Default: VOIP PHONE

2. Network Setting

1. WAN

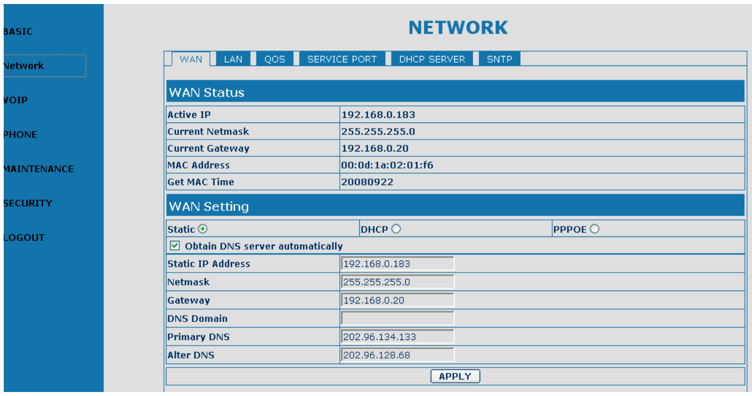


Fig 5.3.2.1 WAN Setting

WAN Status	
Active IP	Current telephone IP;
Current Network	Current sub-web mask;
Current Gateway	Current pre-setting gateway IP;
MAC Address	MAC Address;
Get MAC Time	Date of getting the MAC address.
WAN Setting	
1）Wan Setting：Choose the suitable network mode according to the real environment. Details see 6.3.1.2. **2**）After choose the network mode, web page below display the detail settings of 5.3.1.2. **3**）Web page no response if IP changed**,** at this moment only input new IP address can connect to the telephone. **4**）If networks ID which is distributed by DHCP server is same as network ID which is used by LAN of system, phone will use the DHCP IP to set WAN, and modify LAN's networks ID. when phone uses DHCP client to get IP in startup; if phone uses DHCP client to get IP in running status and network ID is also same as LAN's, phone will refuse to accept the IP to configure WAN.	

2. LAN Setting

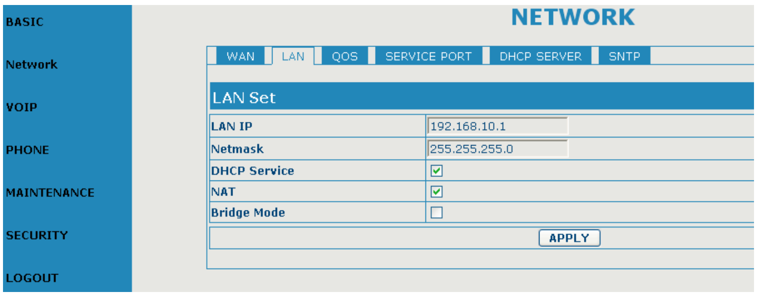


Fig. 5.3.2.2 LAN Setting

LAN IP	Setting LAN static IP
Netmask	Setting LAN Subnet mask
DHCP Service	Active LAN end DHCP server. Telephone will modify DHCP lease form and save the setting according to the IP and subnet mask after user modify the LAN IP. User need to restart the PC to take the DHCP service setting to effect.
NAT	Active NAT。
Bridge Mode	Use bridging mode（Transparent mode）, bridging mode keep the telephone not setting IP address for PC port. LAN port and PC port connect to same network . Telephone will restart after press [SUBMIT].

Notice：WAN port deployment of model 58IP corresponding LAN port of telephone.

3. QoS

The end system support 802.1Q/P protocol , support DiffServ Setting. Among them, VLAN own the function to setting Voice LAN and Data VLAN to use different VALAN ID. System setting Data VLAN own the ability to keep the signaling ,language flow and other digital flow to add other different VLAN ID, it is more flexible for VLAN system using .

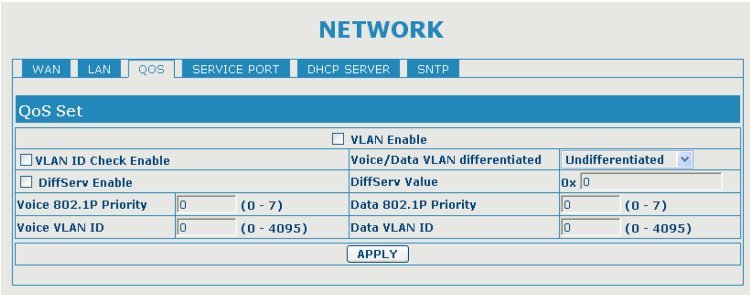


Fig. 5.3.2.3 QoS Setting

VLAN Enable	Active VLAN function
VLAN ID Check Enable	Matching the VLAN ID strictly. If data package is different to it's own VLAN ID or VLANID data package will lost and not to process. If VLAN not active, such kind data packages will be processed.
Voice/Data VLAN differentiated	Setting to make the differ for Voice/Data VLAN, undifferentiated、tag differentiated and data Untaged.
DiffServ Enable	Setting Innovation and Forbidden DiffServ.
DiffServ Value	Setting DiffServ parameters , normal level is 0x00.
Voice 802.1P Priority	Setting voiced sound / signaling data package 802.1p priority.
Data 802.1P Priority	Setting data 802.1p, unvoiced sound/ signaling to use 802.1p
Voice VLAN ID	Setting voice/ signaling data package of VLAN ID
Data VLAN ID	Setting data VLAN ID and data packages of unvoiced/ signaling to use tag of VLAN ID.
Notice: 1）Startup VLAN, if set Voice/Data VLAN differentiated as Undifferentiated, all packets will use the Voice VLAN ID as the tag. 2) Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and disable the DiffServ, then system will not distinguish the voice and data, all packets will use the Voice VLAN ID as the tag. 3) Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and enable the DiffServ, then system will distinguish the voice and data and add the VLAN ID each other. 4) Startup VLAN, if set Voice/Data VLAN differentiated as data untaged, then the packet of the signal/voice will use the Voice VLAN ID as the tag, but the data packets will not take the VLAN tag. 5) If Disable the VLAN, regardless to set the Voice/Data VLAN differentiated or not, all packets will not take the VLAN tag; If enable the DiffServ, all packets will only take the DiffServ value.	

6) User need not, enable the VLAN ID Check Enable that is default, If enable it, the phone will match the VLAN ID strictly. When others' VLAN ID dismatch with us, the packets will discard. Contrarily, the phone will accept the packets with the distinct VLAN ID.

7) User must gain the IP with the Static mode when you set VLAN, otherwise can't gain the IP in the VLAN and also can not dial with point to point.

4. Service Port

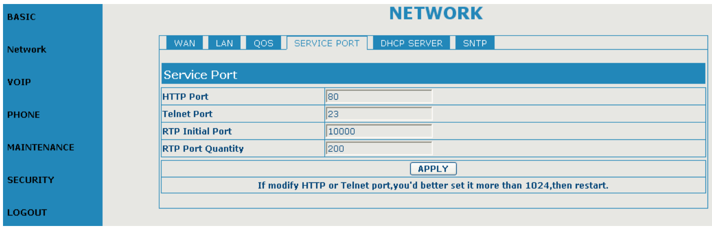


Fig. 5.3.2.4 Service Port

HTTP Port	Setting web browser port , factory default is 80 port. In order to strengthen system safety, suggest to set non-standard 80 port. Save setting after modification and restart telephone , please login with following style: http://xxx.xxx.xxx.xxx： xxxx.
Telnet Port	Setting telnet port, factory default is 23 port.
RTP Initial Port	Setting telephone RTP to start the port , this port is dynamic allocation
RTP Port Quantity	Setting the telephone RTP port MAX number, factory default 200 pcs

Notice:

1）Only stored and restarted the telephone after the current page modification , otherwise the phone can’t effect .

2) It’s better to keep the port number as 1024 if user modify the port number of Telnet, HTTP as this is the system reserve quantity. Must restart the telephone after above setting.

3）HTTP service will be prohibited if setting HTTP port as “0”,

5. DHCP Server

Through this page to set the DHCP service, user can defined the scope and other settings of dynamic IP ,r also can check DHCP lease table and others.

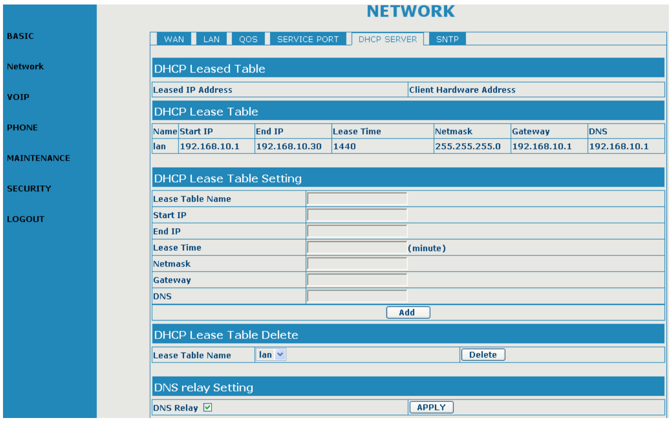


Fig . 5.3.2.5 DHCP Sever

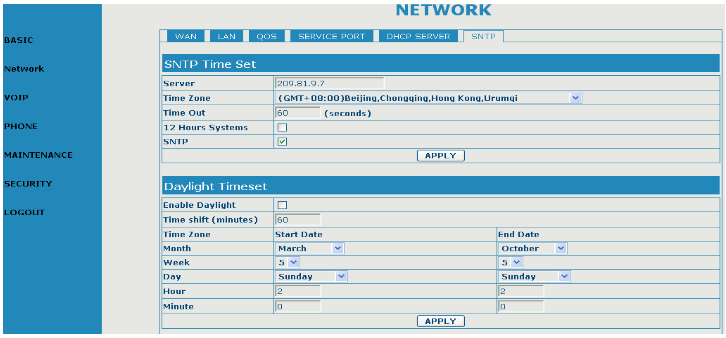
DHCP Leased Table	DHCP allocated IP-MAC mapping table. If the PC port of telephone connect with equipment then the table showed the IP AND MAC address of this equipment.
Lease Table Name	Added lease table name
Start IP	The start of the added lease table IP. When allocate free IP to PC port to use DHCP equipment it start to search from this lease table.
End IP	The end of added lease table IP. The IP number from start to end decided the IP numbers which connected to PC network equipment. The IP which connected to the PC equipment DHCP must among the Start IP and End IP.
Lease Time	The lease IP time which added lease table.
Netmask	The Net mask which added Lease Table.
Gateway	The default gateway IP which added lease table.
DNS	The default DNS server IP which added lease table. Press ADDED and SUBMITTO add DHCP lease table.

Notice:

1. The lease table must less than IP quantity of C style network segment. Suggest to keep the system default lease table and not to modify it.
2. If user modify the lease table of DHCP, it's need to save and restart to make the effect.
3. Choose the name of the lease table and press【DEL】and submit can delete it from DHCP lease table.
4. The display of the DHCP setting unit is minute.
5. LAN port deployment of model 58IP corresponding PC port of telephone.

6. SNTP

According to the real location to set the time zone and SNTP server to get the time and day saving time (DST)function automatically. Also can adjust the time manually.



F ig. 5.3.2.6-1 SNTP setting

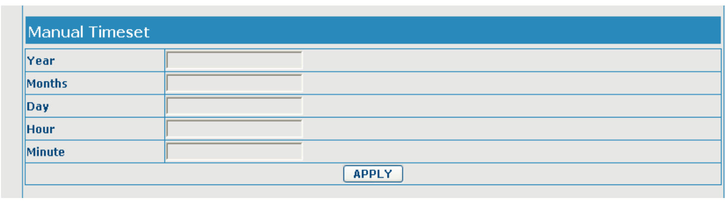


Fig. 5.3.2.6-2 SNTP setting

Server	Set SNTP server address
Time Zone	Enable the time zone
Time Out	The interval for inquire server to synchronous. Default time :60s
12 Hours Systems	Cut over to 12Hour format. Default is 24Hour format.
SNTP	Enable /Prohibit SNTP service ；
Enable Daylight	Enable day time saving
Time shift(minutes)	Enable the time shift of day time saving
Month	Enable the start and end month of day time saving
Week	Enable the start and end week of day time saving
Day	Enable the start and end day of day time saving
Hour	Enable the start and end hour of day time saving
Minute	Enable the start and end minute of day time saving

Notice：When setting manually, each setting need to fill and submit will work.

3. VOIP

1. SIP

Setting SIP server configuration as below

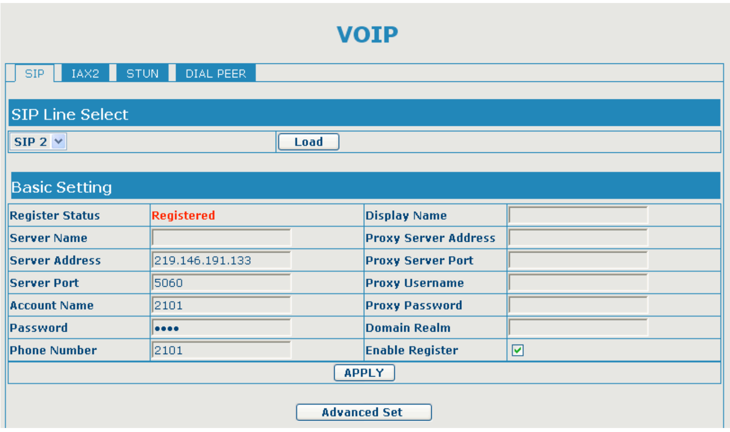


Fig. 5.3.3.1-1 SIP Basic Setting

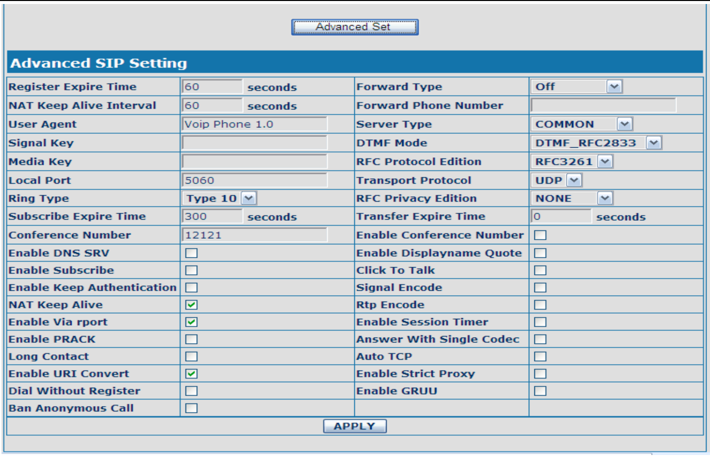


Fig. 5.3.3.1-2 SIP Advanced SIP Setting

SIP Line Selection	
There are two line SIP accounts for selecting, please select SIP account line accordingly. Click [Load] to switch to selected SIP account.	
Basic Setting	
Register Status	SIP Register Status display: Register successful will display [Registered] Register unsuccessful will display [Unregistered] None Register will display [Unapplied]
Server Name	Giving Server Name
Server Address	Configure SIP Address, support domain name type server address
Server Port	Configure SIP register server command port
Account Name	Configure SIP register account
Password	Configure SIP register account password
Phone Number	Configure SIP register account phone number. If no phone number , can not register
Display Name	Configure display name, English letter is allowed
Proxy Server Address	Configure Proxy Server IP Address（Usually, SIP provider offer service based on Proxy Server and register server, configuration of Proxy Server and register server is same. Once the configuration IP address of them is different, please configure Proxy Server and register server accordingly
Proxy Server Port	Configure Proxy Server port
Proxy Username	Configure Proxy Server Username
Proxy Password	Configure Proxy Server Password
Domain Realm	Configure SIP Local Domain Realm. If server did not designate local domain，local domain can configure same address and domain with server. System will input Register server address as domain realm automatically and users no need to input local domain

Enable Register	Configure Enable/Disable Register
Advanced Set	
Register Expire Time	Configure Register Expire Time, default is 60 second. If the Register Expire Time > or < setting time, The phone will change to the recommended time and re-register.
NAT Keep Alive Interval	Configure NAT Keep Alive Interval. If turn on the SIP server detection function, the phone will detect if the server answer by NAT Keep Alive Interval
User Agent	Configure User Agent
Signal Key	Configure Signal Key
Media Key	Configure Media Key
Local port	Configure each line sip port
Hot Line Number	Configure Hot Line Number in OFF-HOOK. It doesn’t work if no hot line number,
Ring Type	Configure Each line Ring Type
Transfer Expire Time	It will end the calling after expire time, when hanging up on Attended Transfer. The default expire time is “0”(It will send BYE code to end the calling after hanging up)
Enable Subscribe	Subscribe information after register successful, it can subscribe other user’s status and voice mail
Enable Keep Authentication	Configure support Authentication register or not. Then the server will confirm directly when receiving the request of Authentication register.
NAT Keep Alive	Configure auto-detection server. Please set time interval less than NAT Keep Alive Interval and turn on this function once the server setting is too short
Enable Via report	Configure support RFC3581 or not，report system is used for inner network, which need the support from SIP server. It was used for the NAT connection between inner network and outer network.
Enable PRACK	Configure support SIP PRACK or not（mainly used for coloring ringing back tone）Suggest user to use default setting
Long Contact	Configure Contact field with more parameters；It will work with SEM system together
Enable URI Convert	URI convert # to %23 in transmission
Dial Without Register	Configure Proxy Dial without register
Ban Anonymous Call	Configure Ban Anonymous Call
Forward Type	Select Forward Type(Default type is OFF） 6 Off：Off the forward calling type. 7 Busy： The incoming call will be forwarding to the appointed phone number 8 No answer：The incoming call will be forwarding to the appointed

	phone number once no answer within set time. 9 Always：The incoming call will be forwarding to appointed phone number directly. This telephone will indicate incoming call before forwarding
Forward Phone Number	Configure Forward Phone Number
Server Type	Select server type or special server type
DTMF Mode	Set DTMF mode as below 5. DTMF_RELAY 6. DTMF_RFC2833 7. DTMF_SIP_INFO Different ISP internet may provide different mode
RFC Protocol Edition	Configure Protocol Edition. Please Configure Protocol Edition RFC2543 when the phone will compatible with CISCO5300 etc SIP1.0 Gateway Default Protocol Edition is RFC3261
Transport Protocol	Configure Transport Protocol :TCP/UDP
RFC Privacy Edition	Configure support RFC Privacy or not, support RFC3323 and RFC3325 edition
Subscribe Expire Time	Configure Subscribe Expire Time
Conference number	Configure Conference Call number
Enable conference number	Enable conference call function
Enable DNS SRV	Support RFC2782
Click to Talk	Configure Click to Talk（Application software required）
Signal Encode	Configure Signal Encode
Rtp Encode	Configure Rtp Encode
Enable Session Timer	Configure support rfc4028 or not；refresh the SIP sessions
Answer With Single Codec	Answer With Single Codec when called
Auto TCP	Configure switch to TCP protocol automatically when the codec is more than 1300 byte.
Enable Strict Proxy	Compatible with special server(Do not use address in via text and use original of calling party when sending encode)
Enable GRUU	Configure support GRUU
Enable Display name Quote	Configure send encode with display name Quote to compatible with server

2. IAX2

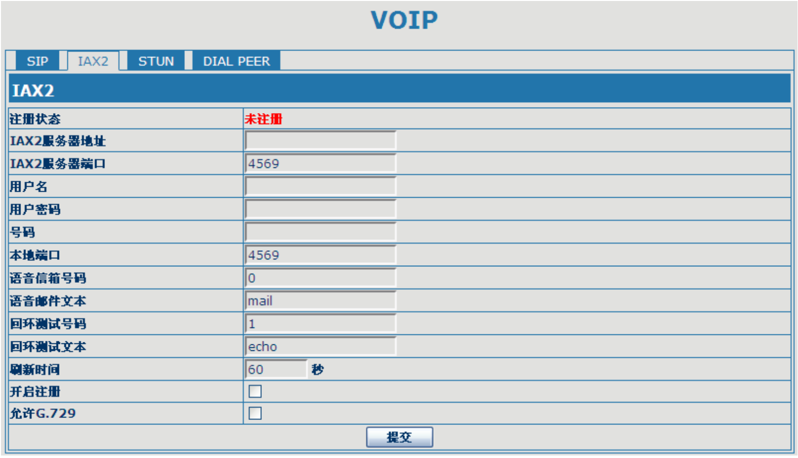


Fig. 5.3.3.2 IAX2 setting

Register Status	IAX2 Register Status display. It will display Registered if register successfully. Otherwise it will show Unregistered。
IAX2 Server Addr	Configure IAX2 Server Address，It can be domain name type
IAX2 Server Port	Configure IAX2 Server Port
Account Name	Configure IAX2 Server Account Name
Account Password	Configure IAX2 Server Password
Phone Number	Configure IAX2 Server Phone number
Local Port	Configure IAX2 Server Local Port
Voice Mail Number	Configure Voice Mail Number. If the voice mail is letter type, the phone can not input. It will input this letter instead of voicemail number if IAX2support voice mail.
Voice Mail Text	Configure Voice Mail Text if IAX2support voice mail.
Echo Test Number	Configure Support Echo Test or not. If the system support Echo and Echo Test is text type, then Echo Test Number will replace Echo Test Text.
Echo Test Text	Configure Echo Test Text
Refresh Time	IAX2 register Refresh Time，The time unit is second. Suggest user to select from 60s to 3600s
Enable Register	Configure Enable/Disable Register
Enable G.729	Configure Support G.729 or not. The phone sending codec support G.729. If use idefisk（Notice support G.729），then calling idefisk will make the PC break down.

3. Stun

The function of STUN as below: User through STUN to get the out network IP of VANT and out network port of SIP signal port , shift IP and PORT of CONTACT field of SIP registration bag , then to register . After above to keep sure if there are in-coming calls can find the accordance user.

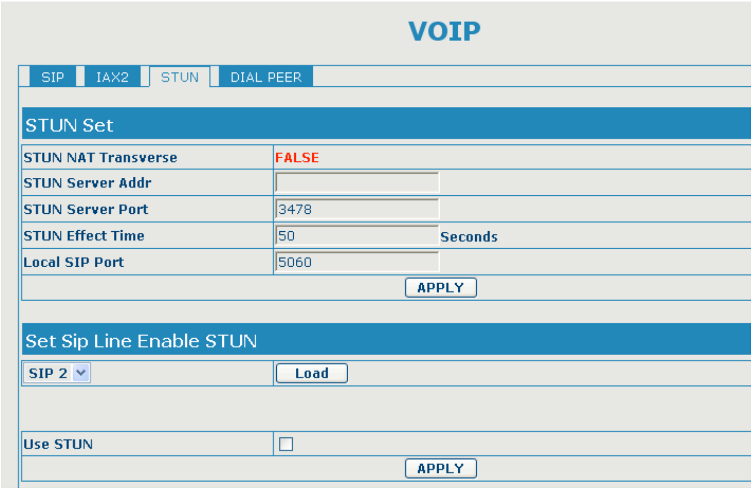


Fig. 5.3.3.3 STUN Setting

STUN Effect Time	Set STUN Effective Time. If NAT server finds that a NAT mapping is idle after time out, it will release the mapping and the system need to send a STUN data packet to keep the mapping effective and alive.
Local SIP Port	Set the Local SIP Port, the default is 5060(after setting the Local SIP Port, SIP will telecommunicate using the changed port)
Set Sip Line Enable Stun	
Choose line to set SIP account to use Stun, there are 2 lines to choose. User can switch by **【Load】** button.	
Use Stun	Enable/Disable SIP STUN

Noicte: SIP STUN is used to realize SIP penetration to NAT. If the phone configures STUN Server IP and Port (default is 3478), and enable SIP Stun, user can use the ordinary SIP Server to realize penetration to NAT.

4. DIAL PEER

The functionality of the IP numbers table is to enable user to make call through Internet. By this table user can enjoy more flexible dial rules. For example, if the user knows the number and IP address, and intends to use dial peer to dial that number directly. Supposing the IP address is 192.168.0.181, user can set a dial rule as the following content. Then user can dial 123 to replace 192.168.0.181 here.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
123	192.168.0.181	5060	SIP	no alias	no suffix	0

For example, if user want to dial a PSTN call to Beijing, just set a dial rule as follows. All numbers beginning with 1 can make a call according to this dial rule. For example, user want to dial 01062213123, but just dial 162213123 to realize the long distance call after make such kind setting.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
1T	0.0.0.0	5060	SIP	rep:??010	no suffix	1

To save the memory and avoid abundant input of users, add the follow functions:

Number	Destination	Port	Mode	Alias	Suffix	Del Length
13xxxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

1. " x " Match any single digit that is dialed.

If user makes the above configuration, after user dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.

2. "[]" Specifies a range that will match digit. It may be a range, a list of ranges separated by commas, or a list of digits.

If user makes the above configuration, after user dials 11 digit numbers started with from 135 to 139, the phone will send out 0 plus the dialed numbers automatically.

Using this phone you can realize dialing out via different accounts without switch in web interface. The following is the specific setting:

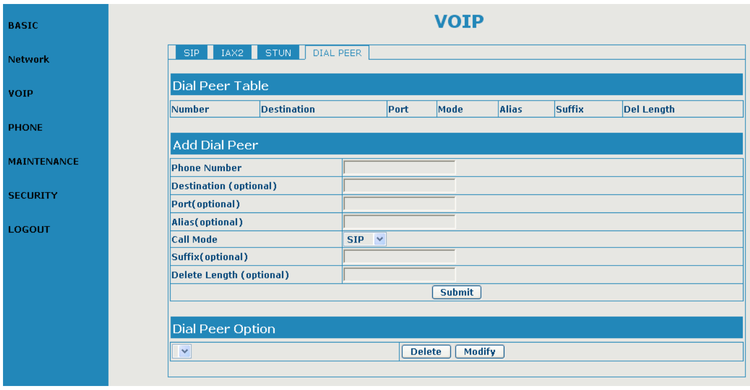


Fig. 5.3.3.4 DIAL PEER

<table>
<tr><td>Phone number</td><td>In order to add calling numbers, there are two types of matching conditions: one is full matching. In the full matching, user need dial the phone number to realize calling to what the phone number is completely mapped. The other is the prefix matching (equal to the prefix district code of PSTN). In the prefix matching, you need input the desired prefix number and T; then dial the prefix and a phone number to realize calling to what the prefix number is mapped. The prefix number supports at most 30 digit and x format and range of number scope.</td></tr>
<tr><td>Destination</td><td>Set Destination address. If user wants to set peer to peer call, please input destination IP address or domain name and the concrete IP address will be parsed by phone DNS server. If there is no configure, the setting IP will be considered as 0.0.0.0. This is optional configure item.</td></tr>
<tr><td>Port</td><td>Set the Signal port, this is optional configure item. the default is 5060 for SIP.</td></tr>
<tr><td>Alias</td><td>Set alias. This is optional configure item，which is a replace number for the number with prefix If user don't set Alias, it will show no alias.</td></tr>
<tr><td colspan="2">Notice: There are four types of aliases, which need to set corresponding to length appointed.
1) " xxx " it means that user need dial xxx in front of phone number, which will reduce dialing number length.
2) "xxx" means that xxx will replace some phone number, which will realize speed dial. For example, when user sets 1 as configure number, the real number will be transferred by setting all: number;
3) del: It means that phone will delete the number with length appointed.
4) Rep: xxx, it means that phone will replace the number with length and number appointed. For example, when user wants to dial PSTN(010-62281493) through ground service supplied by VOIP</td></tr>
</table>

operators, the real calling number is 010-62281493. We can set the calling number as 9T, and rep: 010, then set 1 in length appointed. Then all users who dial the phone numbers beginning with 9 will be replaced as 010+phone number. It is in keeping with the normal modes for users.	
Call Mode	Select different signal protocol, the default is SIP.
Suffix	Set suffix, this is optional configure item. It will show no suffix if user don't set it.
Delete Length	Set delete length and have the number the user input deleted. This is optional configure item.

Introduction of how to set up dial-peer to implement switch between multi- SIP lines

9T mapping: If you have registered a SIP1 server and set dial-peer，all calls will be sent via SIP1 server when you press the numeric key "9" in front of dialing destination phone numbers.

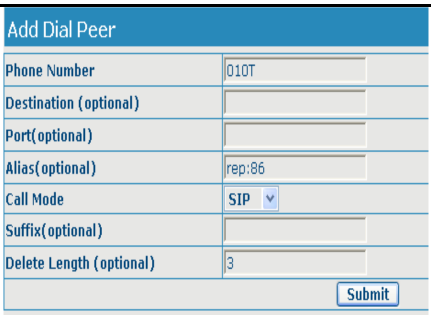
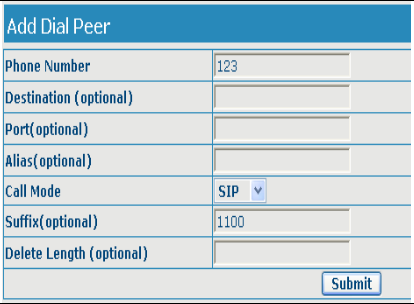
8T mapping: If you have registered a Private SIP2 server and set dial-peer，all calls will be sent via SIP2 server when you press the numeric key "8" in front of dialing destination phone numbers.

号码	目的地	端口	类型	别名	后缀	删除长度
2T	0.0.0.0	4569	IAX2	del	no suffix	1

2T The rule of 2T means user configures IAX server and registered. He need to dial the number with prefix 2 if he want to dial via IAX2 server

Examples of different alias application:

Set by web	**Function explanation**	**Example explanation**
Phone Number: 9T **Destination (optional)**: 255.255.255.255 **Port(optional)**: **Alias(optional)**: del **Call Mode**: SIP **Suffix(optional)**: **Delete Length (optional)**: 1	This page means any number beginning with 9 will send out through SIP portal. Delete Length is 1, meaning all number dialed will delete the first letter of number.	If you dial "92161", the SIP server will receive "2161"
Add Dial Peer Phone Number: 6 Destination (optional): Port(optional): Alias(optional): all:2100 Call Mode: SIP Suffix(optional): Delete Length (optional): Submit	This setting will realize speed dial function, after user dialing the numeric key "2", the number after **all** will be sent out, alias being **all**.	If user dial "2", the SIP server will receive 2100.
Add Dial Peer Phone Number: 8T Destination (optional): Port(optional): Alias(optional): add:0633 Call Mode: SIP Suffix(optional): Delete Length (optional): Submit	This page will realize the function of automatically adding district number or prefix, therefore reducing dialing length. Alias here is **add**.	If user dial "6000", the SIP server will receive "06336000"

	If user wants to dial PSTN(010-6226), while specified dialing should be 86-6226, then we can set dialed number as 010T, then rep:86, after that, set delete length as 3. Therefore, all numbers dialed beginning with 010 will be sent out with 010 replaced by 86+. Alias is **rep.**	If user dial "010 6226", the SIP server will receive "866226"
	This page means it will automatically add 1100 after phone number 123.	If user dial "123", the SIP server will receive "1231100"

4. Telephone Setting

1. DSP Configure

In this page, user can configure voice codec, input/output volume and so on.

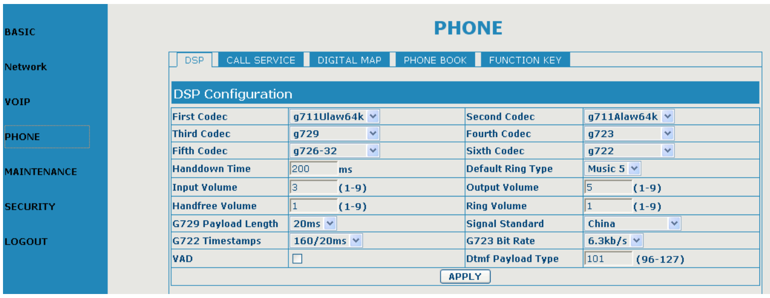


Fig. 6.3.4.1 DSP setting

First Codec	G.711A/u, G.722，G.723, G.729,G.726-32。 The fist preferential DSP codec:G.711A/u, G.722，G.723, G.729,G.726-32
Second Codec	The second preferential DSP codec:G.711A/u, G.722，G.723, G.729,G.726-32。
Third Codec	The third preferential DSP codec：G.711A/u, G.722，G.723, G.729,G.726-32。
Fourth Codec	The fourth preferential DSP codec：G.711A/u, G.722，G.723, G.729,G.726-32。
Fifth Codec	The fifth preferential DSP codec G.711A/u, G.722，G.723, G.729,G.726-32。
Sixth Codec	The sixth preferential DSP codec：G.711A/u, G.722，G.723, G.729,G.726-32。
Hand down Time	Specify the minimum reflection time of Hand down, the default is 200ms. If hand down time is less than set time, phone will ignore the hand down action.
Input Volume	Specify input volume grade.
Output Volume	Specify output volume grade.

Hand-free Volume	Specify hand-free volume grade.
Ring Volume	Specify ring volume grade.
G729 Payload Length	Specify G729 Payload Length.
Signal Standard	Signal standard: different countries have different signal standards.
G722 Timestamps	Choose Timestamps for G722, 160/20ms and 320/20 ms optional.
G723 Bit Rate	Set G723 Bit rate, 5.3kb/s and 6.3kb/s optional.
Default Ring Type	Set default ring type.
VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.
DTMF Payload Type	DTMF Payload Type: set DTMF Payload Type.

2. Call Service

In this web page, user can configure Hotline, Call Transfer, Call Waiting, three Way Call, Black List, white list, Limit List and so on.

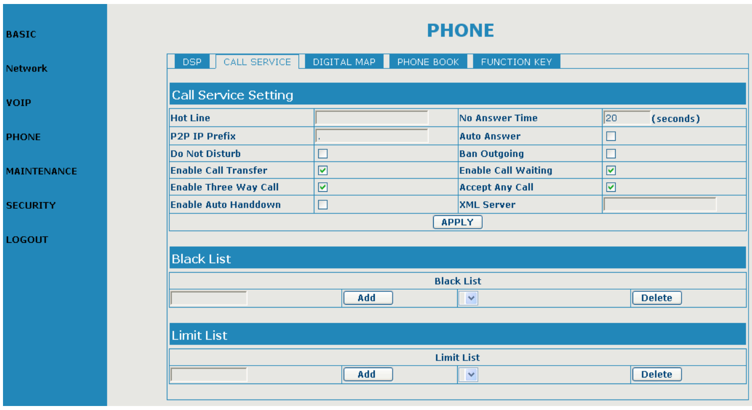
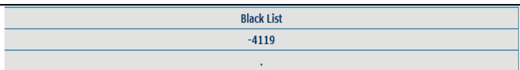


Fig. 5.3.4.2 Call Service

Hot Line	Specify Hotline number. If hotline is set, lift the handset then the hotline is dialed and any other number can't be dialed.
No Answer Time	Set No Answer Time.
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what you want to dial is 192.168.1.119, If you define P2P IP Prefix as 192.168.1., you dial only #119 to reach 192.168.1.119.
Do Notice Disturb	Select NO Disturb, the phone will reject any incoming call, the callers will be reminded that the phone is busy, but any outgoing call from the phone will work well.
Enable Call Transfer	Enable Call Transfer by selecting it.
Enable Call Waiting	Enable Call Waiting.
Enable Three Way Call	Enable Three Way Call.
Accept Any Call	If select it, the phone will accept the call even if the called number does not belong to the phone.
Auto Answer	Auto answer, the phone will auto answer when there is an incoming call.
Ban Outgoing	If user selects Ban Outgoing, then when pick up the phone, there will be busy tone suggesting phone is hung up.
XML Server	Set the server address and default file name of XML.
Black List	 Set Add/Delete Black list. If user does not want to answer some phone calls, add

	these phone numbers to the Black List, and these calls will be rejected. x and . are wildcard. x means matching any single digit. for example, 4xx means any number with prefix 4 whose length is 3 will be forbidden to dial out DOT (.) means matching any arbitrary number digit. for example, 6. means any number with prefix 6 will be forbidden to dial out. If user wants to allow a number or a series of number incoming, he may add the number(s) to the list as the white list rule. the configuration rule is -number, for example, Black List -4119 . means any incoming number is forbidden except for 4119. Notice: End with DOT (.) when set up the white list
Limit List	Set Limit List. Please input the prefix of those phone numbers which you forbid the phone to dial out. For example, if you want to forbid those phones of 010 as prefix to be dialed out, you need input 010 in the blank of limit list, and then you can not dial out any phone number whose prefix is 001;if the prefix if is 0, then all phone numbers with the prefix 0 can't be dialed. x and . are wildcard. x means matching any single digit. for example, 4xx means any number with prefix 4 which length is 3 will be forbidden to dial out. DOT means matching any arbitrary number digit. for example, 6. expresses any number with prefix 6 will be forbidden to dial out.

Notice: Black List and Limit List can record at most10 items respectively.

3. Digital Map

The dial modes this system supports are as follows:

1). End with “#”: dial your desired number, and then press #.

2). Fixed Length: the phone will intercept the number according to your specified length.

3). Time Out: After you stop dialing and waiting time is out, system will send the number collected.

4). User defined: user can define the number length of dialing and prefix of number.

In order to keep terminal users' secondary dialing mode when dialing the external line with PBX, when the prefix of phone number is dialed, system will restart dialing tone according to configuration rule of digital map, then the user continues to dial number, when the dial is over, the phone will send the number prefix and second dialing number to the server.

For example, there is a rule 9,xxxxxxxx in the digital map table. after dialing 9, phone will send the secondary dial tone, user may keep dialing. after finished, phone will call the number which starts with 9, actually the number sent out is 9-digit with 9.

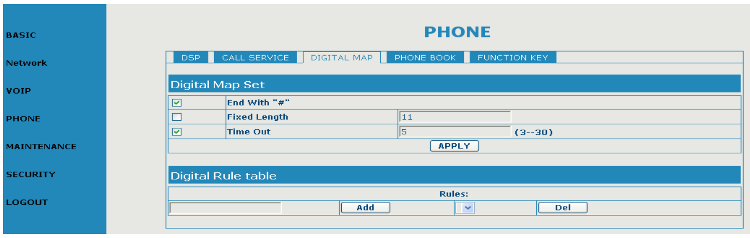


Fig. 5.3.4.3 Digital map configuration

Ending with "#"	Set Enable/Disable the phone ended with “#” dial.
Fixed Length	Specify the Fixed Length of phone ending with. For example, if the fixed length is 11, when the user dials after 11 digits, the phone will automatically dial 11 digits.
Time out	Set the time for timeout with second as unit. The default is 5 seconds. That means, if the user discontinues dialing for a period of 5 seconds after first dialing, then the phone will send the number dialed.

Below is user-defined digital map rule:

“[] “ Specifies the range of designated digits. It can be a range, or a range separated by commas, or a list of digits.

“x” Match any single digit that is dialed.

“. “Match any arbitrary number of digits including none.

“Tn” Indicates an additional time out period before digits are sent of n seconds in length. “n” is mandatory and ranges from 0 to 9 seconds. “Tn” must be the last 2 characters of a dial plan. If Tn is not specified it is assumed to be T0 by default on all dial plans.

“,” means dialing tone is sent after it stops receiving number.

Examples:

RULE
"[1-8]xxx"
"9xxxxxxx"
"911"
"99T4"
"9911x.T4"

[1-8]xxx: Means all four digit numbers between 1000 to 8999 to be dialed immediately.

9xxxxxxx: Cause 8 digit numbers started with 9 to be dialed immediately

911, Cause 911 to be dialed immediately after it is entered.

99T4 Cause 99 to be dialed after 4 seconds.

9911x.T4:Cause any number started with 9911 at least 5 digits to be dialed 4 seconds after dialing ceases.

9,xxxx，means a phone number beginning with 9, after dialing 9, will restart playing dial tone, then wait for ending tone; when finishing dialing four digit number, the phone will send out all 5 digit number. It is used for imitating second dialing.

Notice: End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously, System will stop dialing and send number according to your set rules.

4. PHONE BOOK

User can input the name, phone number and select ring type for each name here.

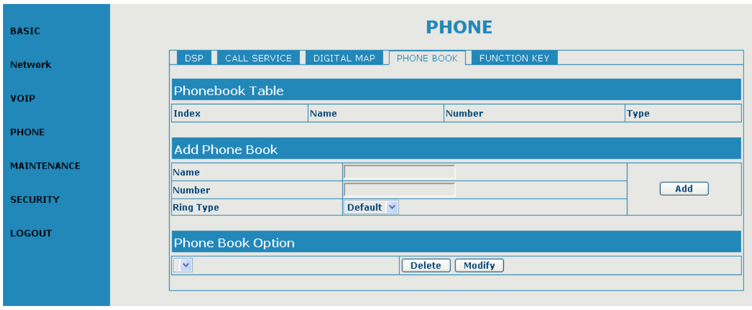
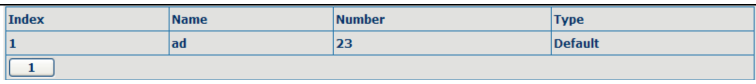
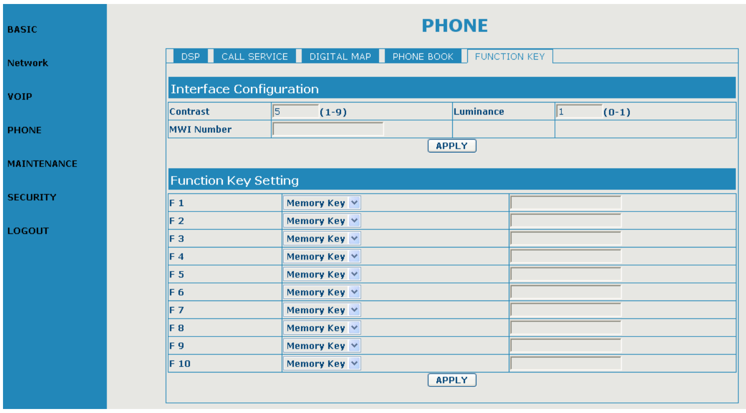


Fig. 5.3.4.4 Phone Book Setting

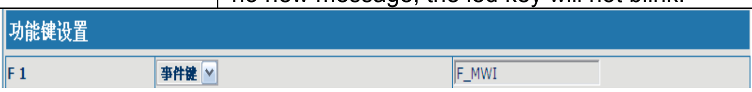
<table>
<tr><td colspan="2">

Show the detailed information of phonebook.</td></tr>
<tr><td>Name</td><td>It's the name of the phone. When the phone is dialed, LCD screen will show the number.</td></tr>
<tr><td>Number</td><td>Set phone number.</td></tr>
<tr><td>Ring Type</td><td>Set ring type.</td></tr>
<tr><td colspan="2">Click “Modify” to change the selected information and click the “Delete” to delete the selected record.</td></tr>
<tr><td colspan="2">Notice: the maximum capability of the phonebook is 500 items</td></tr>
</table>

5.3.4.5 FUNCTION KEY



Picture 5.3.4.5 Store Key Setting

<table>
<tr><td>Contrast</td><td>Set the contrast ratio；</td></tr>
<tr><td>Luminance</td><td>Set the level of luminance of the screen；</td></tr>
<tr><td>MWI Number</td><td>Set the number of MWL, when there is a message, the led key will blink, press MWI, it will automatically call the voice mail to receive the message; if there is no new message, the led key will not blink.</td></tr>
<tr><td colspan="2">

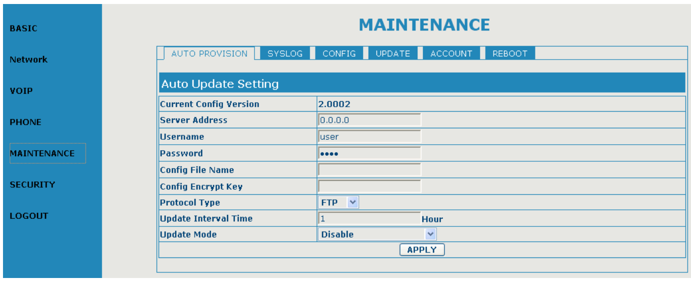
Store Key：You can store number for every store key，when necessary choose function key to connect to this key. Line "set the dialing mode"（SIP1,SIP2,Dialpeer,IAX2), mode similar</td></tr>
</table>

to：SIP1:NAME1

Function of Event Key：monitor different modes 。

5.3.5 Management Setting

5.3.5.1 Automatic Configuration



5.3.5.1 Automatic Configuration

Current Config Version	Display the version of the current configuration version.
Server Address	Set FTP server address. Server address can be in IP form, e.g. 192.168.1.1，or in domain form，e.g. Ftp.domain.com. And the system also supports the function of server setting subdirectory function, e.g. The system can set server address192.168.1.1/ftp/config/ or ftp.domain.com/ftp/config，meaning the server address visited is 192.168.1.1 or ftp.domain.com，file is put in /ftp/config/. The end of subdirectory is OK with or without “/”.
Username	Set the user name of FTP server；TFTP protocol no need to set；if download using ftp protocol, the default is ftp user.
Password	Configure the password correspondingly to user password of the FTP server;
Config File Name	Display the current version of the system configuration file.
Config Encrypt Key	If the configuration file which needs to be upgraded is encrypted, then enter the encrypted password here.
Protocol Type	Choose from three types of server protocols, including FTP,TFTP,HTTP.
Update Interval Time	Time for upgrade interval , using hour as unit.
Update Mode	Modes of Upgrade 4. Disable means no upgrade. 5. Update after reboot: means upgrade after reboot. 6. Update at time interval: means how long it needs to be upgraded.

5.3.5.2 Systemlog

Syslog provides a mature client/server system for recording the messages running in the system program. Syslog receives the messages from the program and classifies the messages in order of priority and type. Then it records the messages according to the rules set by the administrator. It's a healthy and coherent way of managing messages.

The current syslog can be divided into eight levels, which are:

Level 0—emergency, it's the debugging message when the system isn't working(like system breakdown or must reboot etc.), it's the highest level.

Level 1—alert, it's the debugging message when the system meets fatal problem.

Level 2—critical, serious mistakes, e.g: not enough resource for system, wrong upgrading file.

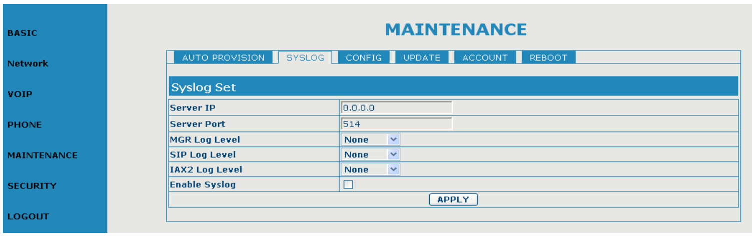
Level 3—error，may affect system in a negative way；

Level 4—warning，not affecting the operation of the system, but may have potential dangers worth noticing.

Level 5—not, the system operates normally under certain conditions but needs to pay attention to the operating environment and whether the parameters are right.

Level 6—info, debugging output information.

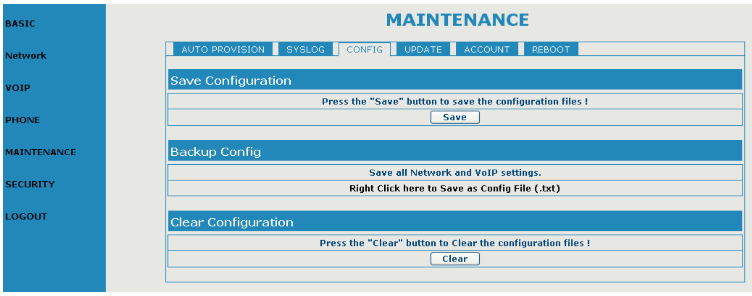
Level 7—debug, used especially for debugging message, mainly for outputing debugging message needed by research staff; It is the most basic debugging message and it has the most output messages. At present the lowest level of debugging message sent to syslog is info, debug is only present under telnet.



5.3.5.2 System log

Server IP	Set syslog server IP or domain.
Server Port	Set syslog server port.
MGR Log Level	Set MGR log level.
SIP Log Level	Set SIP log level.
IAX2 Log Level	Set IAX2 log level.
Enable Syslog	Set syslog on/off.

5.3.5.3 Configuration Setting

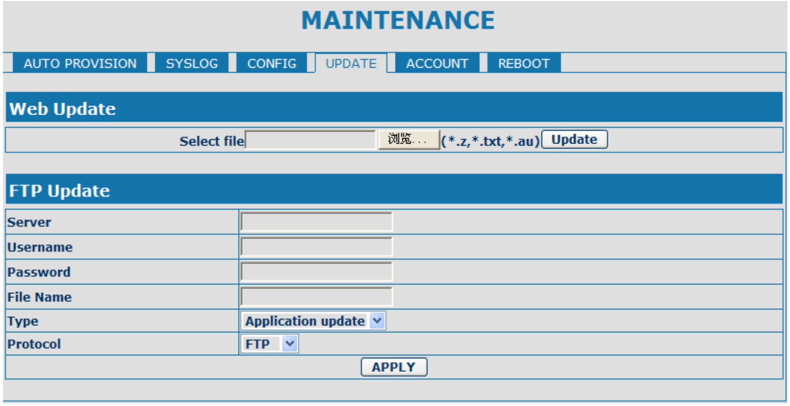


5.3.5.3 Configuration File

Save Config	Attention: The changes you made for phone will become effective immediately. If you fail to save your changes, phone will still use its set configuration
Backup Config	Right click mouse, choose "save target as", then you can download the configuration file with its suffix as .txt.
Clear Config	Set default for the system and reboot the phone. Attention: if log using admin, Clear Config will make all configuration back to factory configuration; if log using guest, Clear Config will clear away all other configurations beside account number and relative configuration of current version.(SIP1-SIP2).

5.3.5.4Upgrade Configuration

Set phone configuration based on existed configuration file through this page.

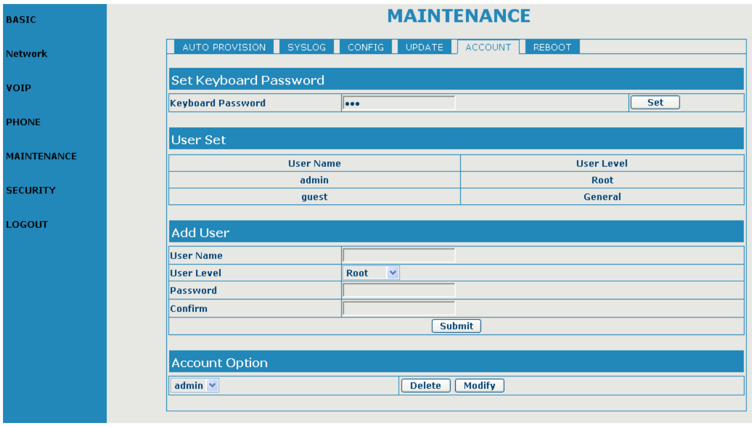


5.3.5.4 Renew Upgrade

Web Update	Through browsing, find previous saved configuration file(or configuration file provided by the manufacturer), then download it to present phone, thus eliminating the trouble of setting each item step by step. Also can download upgrade file, mmiset through this page. Effective when clicking Update.
Server	Set FTP server address uploaded or downloaded. Server address can be IP form, e.g:192.168.1.1, or domain form,e.g:ftp.domain.com. Besides, system supports server's subdirectory set function, e.g, server can set server address 192.168.1.1/ftp/config/ form, or ftp.domain.com/ftp/config, meaning the server address visited is 192.168.1.1 or ftp.domain.com. File is put under /ftp/config/. Subdirectory's suffix is ok without "/".
Username	Set the user name of FTP uploaded or downloaded. If choose TFTP, then no need to set user name and password.
Password	Set server password of FTP server uploaded or downloaded.;
File name	Set file name of system upgrade file or configuration file uploaded or downloaded.
Attention: The exported configuration file can be modified. Besides, it supports module-importing. For example, you only need to keep SIP module in configuration file then import it to system, therefore configuration of other modules will not lose after imported into other configuration.	
Type	Set System type: 1. Application update: download system update file 2. Config file export: Upload the config file to FTP/TFTP server, name and save it. 3. Config fie import: Download the config file to phone from FTP/TFTP server. The configuration will be effective after the phone is reset.
Protocol	Select FTP/TFTP server

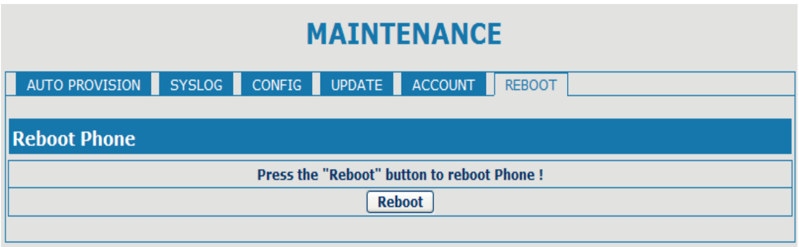
5.3.5.5 Account Config

You can add or delete user account, and change the authority of each user account in this web page



<table>
<tr><td>Keyboard Password</td><td>Set the password for entering the setting menu of the phone by the phone ‘s key board. The password is digit.</td></tr>
<tr><td colspan="2">
<table>
<tr><th>User Name</th><th>User Level</th></tr>
<tr><td>admin</td><td>Root</td></tr>
<tr><td>guest</td><td>General</td></tr>
</table>
This table shows the current user existed.</td></tr>
<tr><td>User Name</td><td>Set account user name.</td></tr>
<tr><td>User Level</td><td>Set user level, Root user has the right to modify configuration, General can only read.</td></tr>
<tr><td>Password</td><td>Set the password.</td></tr>
<tr><td>Confirm</td><td>Confirm the password.</td></tr>
<tr><td colspan="2">Select the account and click the Modify to modify the selected account, and click the Delete to delete the selected account.
General user only can add the user whose level is General.。</td></tr>
</table>

5.3.5.6 Reboot

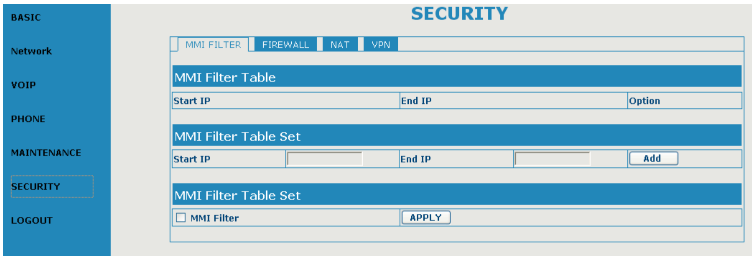


If user modified some configurations which need the phone’s reboot to be effective, you need to click the Reboot, then the phone will reboot immediately.

Notice: Before reboot, user need confirm that all configurations have been saved

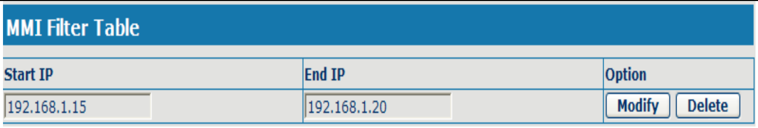
5.3.6 Security

5.3.6.1 MMI Filter



5.3.6.1 MMI Filter

User could make some device own IP, which is pre-specified, access to the MMI of the phone to config and manage the phone.



MMI Filter IP Table list:

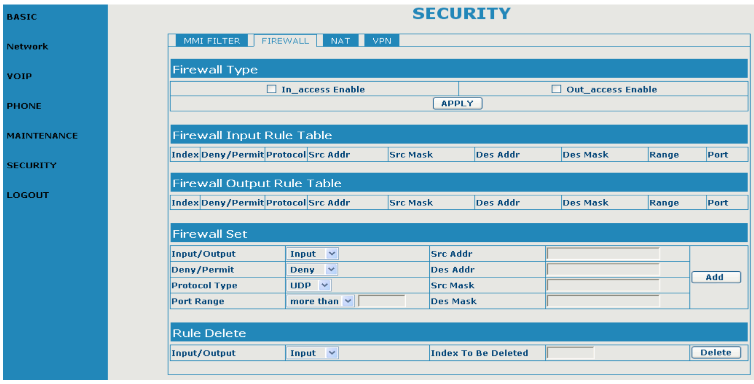
Add or delete the IP address segments that access to the phone.

Set initial IP address in the Start IP column, set end IP address in the End IP column, and click Add to add this IP segment. You can also click Delete to delete the selected IP segment.

MMI Filter	Select it to enable or disable MMI Filter. Click **Apply** to make it effective

Notice： Do not set the visiting IP outside the MMI filter range, otherwise, user can not log on the web.

5.3.6.2 Firewall



5.3.6.2 Firewall Configuration

In this web interface, user can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule).

Firewall supports two types of rules: input access rule and output access rule. Each type supports at most 10 items.

Through this web page, user could set up and enable/disable firewall with input/output rules.

System could prevent unauthorized access, or access other networks set in rules for security. It supports two access lists: one for filtering input packets, and the other for filtering output packets.

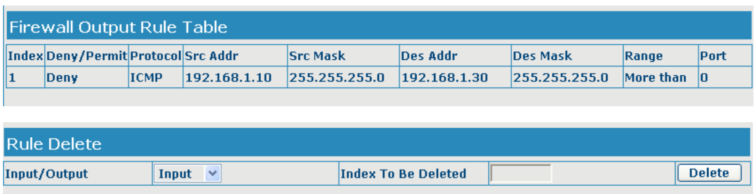
Each kind of list could be added 10 items.

We will give you an instance for your reference.

In_access enable	Select it to Enable in_ access rule
out_access enable	Select it to Enable out_ access rule
Input/Output	Specify current adding rule by selecting input rule or output rule.
Deny/Permit	Specify current adding rule by selecting Deny rule or Permit rule.
Protocol Type	Filter protocol type. You can select TCP, UDP, ICMP, or IP.
Port Range	Set the filter Port range
Src Addr	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0, for example：192.168.1.0.
Des Addr	Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*0, for example:192.168.1.0.

Click the **Add** button if you want to add a new output rule. Then enable out_access, and click the Apply button.

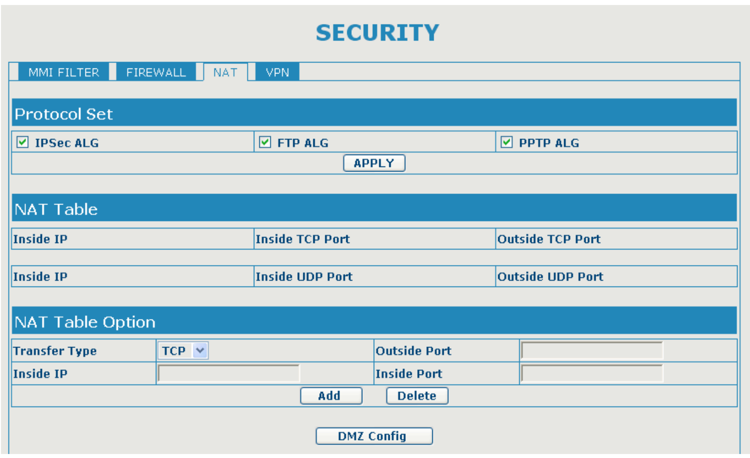
So when devices execute to ping 192.168.1.118, system will deny the request to send data package to 192.168.1.118 for the out_access rule. But if devices ping other devices which network ID is 192.168.1.0, it will be normal.



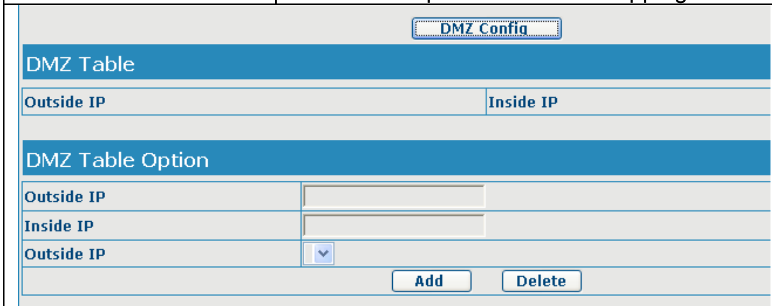
Click the **Delete** button to delete the selected rule.

5.3.6.3 NAT Config

NAT is abbreviated from Net Address Translation; it's a protocol responsible for IP address translation. In other word, it is responsible for transforming IP and port of private network to public, also is the IP address mapping which we usually say.



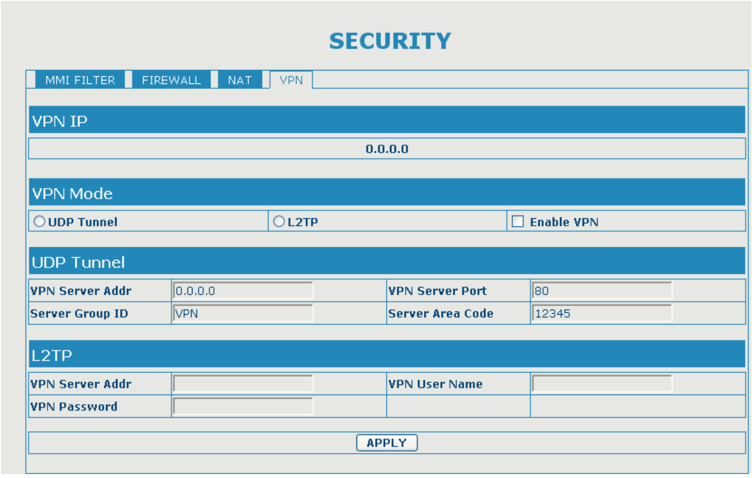
5.3.6.3 NAT Configuration

IPSec ALG	It is an encryption technology. Select it to enable/disable IPSec ALG, the default is enable.
FTP ALG	FTP is a service of connection layer which can transform intranet IP into extranet IP when intranet IP is sending out packet. Select it to enable/disable FTP ALG, the default is enable.
PPTP ALG	Select it enable/disable PPTP ALG, the default is enable.
 Shows the NAT TCP mapping table.	
 Shows the NAT UDP mapping table	
Transfer Type	Select the NAT mapping protocol style, TCP or UDP.
Outside Port	Set the LAN port of the NAT mapping
Inside IP	Set the IP address of device which is connected to LAN interface to do NAT mapping.
Inside Port	Set the LAN port of the NAT mapping
	
Outside IP	Set the outside Wan port IP address of DMZ.;
Inside IP	Set the inside LAN port IP address of DMZ;
Click the **Delete** button to delete the selected mapping table.;	

Notice: 10M/100M adaptive means the network card, and other equipment physical consultations speed, testing speed under bridge mode near to 100M, in order to ensure the quality of voice and communications real-time performance, we made some sacrifices of NAT under the transmission performance. Transmit with full capability only when system is idle, so can not guarantee that the transmission speed reach to 100M.

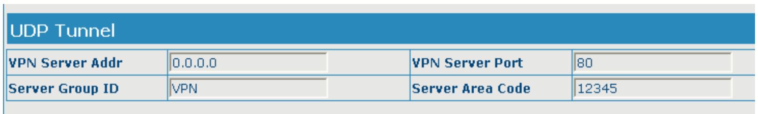
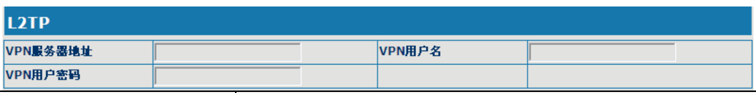
5.3.6.4 VPN

This web page provides user with a safe connect mode by which user can make remote access to enterprise inner network from public network. That is to say, User can set it to connect public networks in different areas into inner network via a special tunnel.

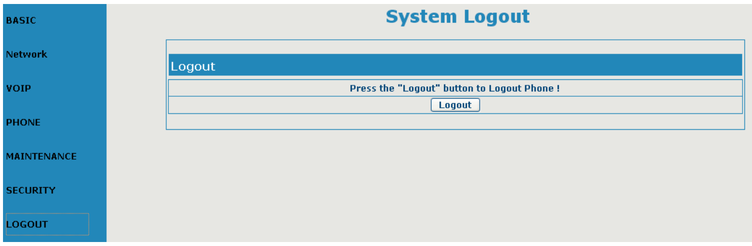


5.3.6.4 VPN Configuration

VPN IP	Shows the current VPN IP address.

	
Notice: Select UDP Tunnel (VPN Tunnel) or VPN L2TP. You can choose only one for current state. After you select it, you'd better save configuration and reboot your phone.	
VPN Mode	Whether the configuration supports VPN mode.
VPN Configuration	
	
PN Server Addr	Set VPN server address.
VPN Server Port	Set VPN server port number.
L2TP Configuration	
	
VPN Server Addr	Set VPN L2TP server address.
VPN User Name	Set VPN L2TP user name.
VPN Password	Set corresponding user name of VPN L2TP.

5.3.7 Logout



Press【Logout】exist the web, next visit need input user name and password accordingly.

Using the SIP Phone

4. Placing Phone Calls

1. Off-hook, On-hook and Speaker-phone Mode

With SIP phone on-hook, it can switch to the off-hook mode by picking up the handset. Press SPEAKER key and put down handset, the phone switches to the speaker-phone mode.

With SIP phone on-hook, press SPEAKER key to enter the speaker-phone mode. The SPEAKER LED on the phone is turned on when the phone is in speaker-phone mode. Pick up handset to switch to off-hook mode, LED turn off accordingly.

Notice: Off-hook active LINE1 preferentially and LINE1 enter into working mode in default mode.

2. Dial Directly

Pick up handset or press SPEAKER key, then input the deployed telephone number and press【 # 】key to dial out;

On-hook mode, press the desired number and press【 # 】 key to dial out .

3. Redial

Pick up handset or press SPEAKER key then press REDIAL key to dial the last call.

On-hook mode , press 【VOL+】 and 【VOL-】 to select the desired number , then pick up handset or press【SPKR】 to dial out the selected number.

Notice: The system will empty the calling recorders after reboot the telephone, at this moment press【**REDIAL**】in vain **.**

4. Conference

When one LINE in calling mode, press【**CONF**】key and input the phone number to keep the second

LINE in calling mode, press 【**CONF**】key again to enter into three way conference mode., press【**SPKR**】key will hang up both lines ; When one line in calling mode, HOLD the first line and make the other line in calling mode too , then press 【**CONF**】key into three way conference mode. In conference mode, press 【**HOLD**】 key to keep both line in hold mode, press 【**HOLD**】 key again to resume the hold mode, press【**SPKR**】 key to exist the calling mode.

Notice：One-Line phone doesn't own such features. For Two-line phone, if guest wants to use this function, user must active Call Waiting and Three Way Call function. For details please see 5.3.4.2.

5. Speed dial

When phone is in off-hook or speakerphone mode, press speed dial key (M1-M10) to call the number associated with each speed dial key in phone book as F1-F10.

5. Answering calls

1. Answering a call

Pick up the handset or press the SPEAKER key to answer a call. Put down the handset or press the SPEAKER key to hang up. If there is a new coming call to another line during current call, press the corresponding line to answer the call and hold the current call. After the new call finished press the corresponding line to hang up and release the held call automatically, then press the corresponding line to finish the call.

2. Call Hold

During a call, press HOLD key to put the other party on hold and send hold music to the other party. Hold LED will light up. Press HOLD key again, the current call will release call hold.
If a call was on hold, you can put down the handset or press the SPEARKER key to close the speaker. The current call will be on hold. When pick up the handset or press the SPEARKER key again, the current call will release call hold.

3. Call Waiting

When ***Call waiting*** option is enabled, if there is a new incoming call during the current call, it send out indicator sound， press the corresponding line key to place the current call on hold and answer the new incoming call. After the new call finished press the corresponding line to stop it and release the current held call .

Notice：-S series (Single line) model no this function.

4. Resume Answer

When the waited call is enabled, press the corresponding **【LINE1】** or **【LINE2】** key then stop the current calling and resume the original call.

Notice：-S series (Single line) model no this function.

6.3 Forward

For this part, please see detail information of 5.3.3.1

6.3.1 **Always**

Set the SIP-Advanced Set-Forward Type as "Always" by web setting , all the incoming calls will be forwarded to Forward-to Number..

6.3.2Forward when busy

Set the SIP-Advanced Set-Forward Type as "Busy " by web setting ,if a new call comes when the line is busy, the new call will be forwarded to ***Forward-to Number***.

6.3.3 Forward when no answer

Set the SIP-Advanced Set-Forward Type as "No Answer " by web setting, if a new call comes and is not answered in a period of time (defined by ***No Answer Time***), the new call will be forwarded to ***Forward-to Number***.

6.4 Auto-answer

Set the PHONE CALL SERVICE "type as "Auto Answer " by web setting, if a new call comes and is not answered in a period of time (defined by ***No Answer Time***), the new call will be answered automatically which means the speaker phone will be automatically turned on. Details see 5.3.4.2.

6.5 Mute

During a call, press MUTE key to mute local voice. The mute LED will light up, indicating that the other party cannot hear you.

Press MUTE key again to resume the conversation, the mute LED will turn off.

6.6 Checking Voice Message

A flashing message LED indicates there are new voice messages. In off-hook mode, please press MESSAGE key to call the pre-set voice message number to retrieve voice message. The message LED will turn off when there is no message.

7 ACOUSTIC PART (68 model only)

7.1 Play AUDIO

Press AUDIO ON/OFF key to enter or exist audio play mode, then press RADIO/LINE IN key to realized cyclical selection of INE IN and FM-RADIO mode.

7.1.1 Play iPod/MP3 input device

Connect the iPod/iPhone/MP3 to LINE IN port through the audio line, or

Connect the iPod/iPhone/MP3 with the fixed Audio Cable, or

Connect the iPod/iPhone/MP3 with the fixed iPod cable;

Then press the LINE IN key to select the LINE IN mode to realize the audio playing.

7.1.2 Play FM radio

Press RADIO/LINE IN key to select the FM-RADIO mode, then press TUNING+/-select the needed frequency .With each press the frequency changed 0.1MHz. press TUNING+/-for while to enter into searching frequency forward or backward , it will stop automatically after searching FM signal.

7.1.3 Press AUDIO ON/OFF to Pause or Play. Press VOLUME+/- to adjust the volume, total level 0-30, factory default is 15.

7.1.4 AUDIO OFF mode, press TUNING+ for a while , LCD display ON or OFF (ON or OFF the hotel mode), ON :volume 0—20, OFF: volume 0-30.

7.1.5 AUDIO OFF mode, press RADIO/LINE IN not release, LCD display Time, Month, Day , Year , DST Zone and easy to view the setting parameters.

7.1.6 Detect the device mode intelligently

If telephone ringing or off-hook under FM-RADIO or LINE IN mode, the telephone can PAUSE audio playing , after the phone call finished or ringing stopped it can resume to playing mode again.

7. 2. Setting Time

Under AUDIO OFF mode (not LINE IN or FM-RADIO working mode) to set the time

7.2.1 Press VOLUME- key 1S， hour digit flashing on the LCD , press TUNING+/- to adjust hour.

7.2.2 Then press VOLUME- key, minute digit flashing, press TUNING+/- to adjust minute.

7.2.3 Then press VOLUME- key or no operation to finish the setting

7.2.4 Press VOLUME+ key to adjust 12 hour format and 24 hour format.

7. 3. Set alarm

7.3.1 Set Alarm Time

* Press ALARM SET key , the hour digit display on the LCD , press VOLUME+/-， to adjust the hour.

* Then press the ALARM SET key, the minute display on the LCD, press VOLUME+/-to adjust the minute.

* Then press the ALARM SET key or no operation to finish the setting.

* Unlock the Alarm

Press ALARM key to start and close alarm. In start mode the alarm and its mark will be on the top right corner, when in close mode, the alarm and its mark disappear.

7.3.2 Alarm snooze feature

When alarm ringing, press any key and the alarm stop ringing for 10 minutes. After 10 minutes it will ring again. Except ALARM key press any other keys or telephone off-hook can stop ringing.

Alarm ringing during playing mode, it exist playing mode automatically.

7.3.3 In AUDIO OFF mode, press ALARM VOLUME key to adjust the alarm indicator volume, each press increase 5, the top level is 20, factory default is 10.

7.4 Adjust LCD brightness

Notice alarm ringing mode, press SNOOZE to adjust the LCD brightness, there are Mid. Hi .Close three levels for selecting.

7. 5. Charge port

There are two USB port , charge for some devices with mini USB line.

* iPod USB port ---For charging iPod、iPhone apple devices .

*USB Charge port——For charging mobile phone and other peripheral products .

* LINE IN audio input port --- With 3.5mm dual track audio lines connect iPod/MP3 to acoustics

7. 6. Homing system

Press ALARM for 2 seconds the device enter into homing system. It’s easy for housekeeping.

7.7 Change clock’s battery

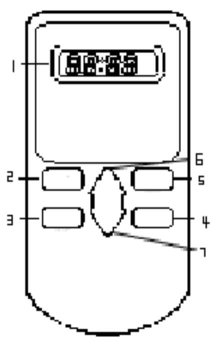
First please take off the battery from the battery cabinet on the back of the phone. Then please install new battery. The specification of the battery is 3V, 50Mah.

Notice: The anode of the battery should be toward outside. The battery should be changed every two years.

7.8 Adjust time with remote control timing cloner

7.8.1First install two 7# 1.5V batteries into the before using it. Pay attention to the polarity of the battery. Don’t mix the old and new batteries.

7.8.2 Remote Control Timing Cloner



1. LCD Display
2. DST SET key——Setting DST time.
3. TIME SET key—— Setting time and date.
4. TIME PROG key—— Adjusting time and date
5. DST PROG key —— Adjusting DST time.
6. UP key —— Upward key.
7. DOWN key——Downward key.

7.8.3 Set time with the Remote Control Timing Cloner

●Press TIME SET key, hour digits flashing, press UP/DOWN key to adjust,

●Press TIME SET key, press UP/DOWN key to adjust minute,

●Press TIME SET key, press UP/DOWN key to adjust month,

●Press TIME SET key, press UP/DOWN key to adjust day,

●Press TIME SET key, press UP/DOWN key to adjust year,

●Press TIME SET key, LCD display the time setting finished.

7.8.4 Set DST area code with the Remote Control Timing Cloner

●Press DST SET key, then press UP/DOWN key to set the area code:

00= DST OFF, 01=USA/Canada , 02=Europe/Russia/Swiss;

03= Mexico, 04=New Zealand

7.8.5 Adjust Time

●After above setting , press TIME PROG or DST PROG at the right upper direction within 10cm near the base LCD to adjust the TIME or DST Time.

7.7.6 Conversion of 12- Hour Format and 24- Hour Format

●When LCD display time, press TIME SET over 2 seconds to realize the conversion..

Notice: If the attached RCT Cloner is different than above image, please according to the attached object.

9. Maintenance and troubleshooting

1. No dialing tone of handset

Check the telephone network,

Check the connection of handset cord and straight line cord

2. No ringing or no continually ring

Please check the telephone line or the quantity of parallel phone set

3. Noise during talk

Please check the telephone line or please check the handset cord

4. Acoustics no sound

Please check if the external power supply is correctly connected.

Please check the 10pins ribbon cable and AUDIO CONNECTION

5. No signal or nersy noise after radio selected the channel

Please adjust the direction of the radio antenna.

Please check if there is local TV channel in the same frequency.

5. The phone can’t receive any call.

Please check if the it’s in “DOD” status.

If there is any problem, please don’t open the phone body, please contact us or our local representative. Thanks.

10. Restore factory settings

The phone will enter into the safety mode after press “#” twice when there is power. Then press “*#168”, the phone will restore factory settings.

11. WARRANTY

This product is warranted for a period of 12 months from the date of purchase against faulty materials or workmanship. If during this period a defect arises, we may repair or replace the product, at Bittel's discretion, provided that:

1) The product has not been used for any purpose other than normal use,

2) Unauthorized product repair or modifications have not been attempted.

3) The product has never been used in a harsh or corrosive environment.

4) No damage in transit

THIS LIMITED WARRANTY GIVES THE BUYER SPECIFIC LEGAL RIGHTS. THE BUYER MAY ALSO HAVE OTHER RIGHTS WHICH VARYFROM JURISDICTION TO JURISDICTION.

This warranty is only valid for merchandise purchased directly from Bittel or dealers or distributors Bittel Co. authorized.

12. FCC WARNING

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

CONTACTING BITTEL

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