

September Club Meeting

Date: Friday, September 26, 2008

Time: Socializing at 7 pm, Meeting at 7:30

Place: Covington School, 205 Covington Road, Los Altos

Topic: Homebrew Night

Summary: We invite everyone to bring a project to show and share. Each participant has 5 - 10 minutes to present a project and answer questions about it. The following prizes will be awarded (as decided by the audience): First place - \$40, Second place - \$30, Third place - \$20, and Fourth place - \$10. Projects must not have been presented at a previous Homebrew contest. Projects should be something you've constructed (kits are OK) and used. Projects should be HAM radio related.

The club offers Raffle prizes, refreshments (great coffee, great cookies) and technical advice at the meeting: Bring your questions for Dr. Know-It-All and get great answers. Be sure to attend for an enjoyable evening.

August Meeting Report

The August meeting speaker was Nimit Hongyim, K6XOX, who spoke about internet telephone service. He took up the project to tackle a technical challenge and to get phone service for free. In addition, there is no extra cost for long distance. One can sign up for service at specified web sites. An IP (internet protocol) phone, analog phone with ATA (analog telephone adapter), or wireless phone is required. You can also use your PC as a phone with the appropriate software and headphone. Nimit described and demonstrated how to use the Internet for telephone service. Internet telephony is made possible by a communication protocol called VOIP, which stands for voice over internet protocol. See more information in the article he authored "How to Setup Free Internet Phone Service for Home" later in this relay.



Nimit Hongyim, K6XOX
August Speaker



Bernard and Doyle
Raffle Winners

The MFJ-842 140-525 MHz Cross-needle SWR/Power Meter was won by Bernard Van Haecke, KI6TSF (the second raffle prize in as many months!). The Rigblaster no-mic w/usb was taken home by Doyle Kisler, KG6YUN. Bernard, KI6TSF, was also chosen for the Wish You Were Here prize of \$5 (that's four months in a row that the prize has been taken and the second month in a row for Bernard!).

President's Corner



Membership Meeting. Our next meeting is Friday, September 26th at 7pm (<http://www.fars.k6ya.org/meeting>).

The September meeting is "Homebrew Night." We invite all Hams to show off their most recent amateur radio projects to

our audience. Then the audience decides who gets the 1st (\$40), 2nd (\$30), 3rd (\$20), and 4th (\$10) place prizes. Haven't got a project. You still have a few days to put one together.

Board Members. Elections for Board Members of the FARS Board of Directors are scheduled for Nov. 21st (our November meeting). There will be an opportunity to nominate board members at the October 24st meeting. If you interested in joining the board, or would like to suggest someone for the board, please me or another board member know about your interest.

Am-Tech Day. The October Amateur Radio Technology Day is scheduled for Saturday, October 4th at the Stanford Linear Accelerator Center. This month we are operating the special event station N6T to celebrate the completion of four years of Am-Tech Day. Check the FARS web site (www.fars.k6ya.org/amtechday/) for the program schedule, directions, and the latest news.

Email Notices. Subscribe to the FARS Announcement list (www.fars.k6ya.org/mail/) to receive reminders of FARS activities and other news.

Flea Market. The last Electronics Flea Market of 2008 is scheduled for October 11th and is hosted by the Santa Clara County Amateur Radio Association. **Am-Tech Day.** The October Amateur Radio Technology Day is scheduled for Saturday, October 4th at the Stanford Linear Accelerator Center. Check the FARS web site (www.fars.k6ya.org/amtechday/) for the program schedule, directions, and the latest news. - de Mikel, KN6QI

Upcoming Events

- Sep 26 7:00 PM, [Club meeting](#), Covington School
FARS Annual Home Brew Night!!
- Oct 2 7:30 PM, Board Mtg at the Los Altos Town Crier
- Oct 4 8 AM to 9 PM, [Am-Tech Day](#), SLAC, 8AM-9PM
- Oct 11 [Electronics Flea Market](#), De Anza, 5 am to Noon
- Oct 17-19 Pacificon 2008, www.pacificon.org
- Oct 24 7:00 PM, [Club meeting](#), Covington School
- Nov 1 8 AM to 9 PM, [Am-Tech Day](#), SLAC, 8AM-9PM
- Nov 6 7:30 PM, Board Mtg at the Los Altos Town Crier
- Thursdays 8:00 PM, FARS net, 145.230(-), 100 Hz PL

See more events, [FARS Calendar](http://www.fars.k6ya.org/events/calendar) <<http://www.fars.k6ya.org/events/calendar>>

Pacificon 2008, October 17-19

Pacificon 2008, "The West Coast Ham Radio Convention," is presented by the Pacific Division of the American Radio Relay League and produced by the Mount Diablo Amateur Radio Club. There will be an Antenna Seminar on Friday, October 17th 8 am to 4 pm, which has limited registration, so sign up early. The regular activities are held Saturday and Sunday, October 18th and 19th. There is a breakfast Saturday morning and a Banquet Saturday evening. More information may be found at the Pacificon website, www.pacificon.org. Tickets may be ordered online using Paypal or sent in. Regular admission (which does not include the Antenna seminar, breakfast or banquet) may be purchased on site.

CLUB INFORMATION

President: Mikel Lechner, KN6QI
Vice President: Steve Stearns, K6OIK
Treasurer: David Cooper KE6PFF
Secretary: Rob Riley, KI6INR
Radio Officer: Phil Hawkins, KA6MZE
Training Officer: Kevin Weiler, K6XXX
Relay Editor: Mark Hardy, AF6DO

FARS Board: Dick Baldwin N6ATD, Robert Flemate KE6TFU, Pink Foster KG6ILA, Nimit Hongyim K6XOX, Kristen McIntyre K6WX, Barbara Neuhauser AE6RM.

Station Trustee: Stan Kuhl, K6MA
FARS Web Page: <http://www.fars.k6ya.org>
Download Relay: <http://www.fars.k6ya.org/relay>

Club members and non-members are encouraged to subscribe to the FARS Announcement list by browsing www.fars.k6ya.org/mail, clicking on Subscribe/Unsubscribe and following the instructions under "Subscribing to fars-announce".

You may submit announcements to the FARS Announcement at fars-announce@svpal.org. The list is moderated and messages will be posted as approved by the list moderator.

Contact the FARS board of directors at fars-board@svpal.org

Club meetings are held at 7 PM on the fourth Friday of each month except January (Winter Banquet); and sometimes there are changes for June (for field day) and Nov. & Dec (for holidays).

Annual club membership is \$20. Club badges are \$9. Visitors are always welcome! Directions in this newsletter. Talk-in: N6NFI (145.23-, 100 Hz) or W6ASH repeater (145.27-, 100 Hz).

FARS *Relay* is the official monthly newsletter of the Foothills Amateur Radio Society. Contributions to the newsletter from members, family, and guests are earnestly solicited!

Contributions are subject to editing and/or compression. All readable forms welcome.

Here is how to reach the editor:

Mark Hardy, AF6DO
Mail: P.O. Box 2248
Santa Clara, CA 95055
Voice: 408-243-0701 (Before 9 PM, preferred)
Email: mark.af6do@gmail.com, At FARS meetings.

Maybe Digital Radios Aren't the Problem

This article, by Glenn Bischoff, presents some problems that firefighters are having with digital radios and how the problems might be mitigated. See the following websites for more information and readers' feedback.

http://urgentcomm.com/commentary/newsletters/0822_digital_radio/

<http://urgentcomm.com/commentary/newsletters/digital-versus-analog-radio-0829/index.html>

<http://urgentcomm.com/commentary/newsletters/digital-radio-problems-feedback-0905/>

Am-Tech Day July 2008 Pictures



Field day 2008 160/80 meter experiences

My idea as field day captain for the past three years has been to bring something new to Field Day every year. Field Day 2007 saw the successful deployment of a 135 foot vertical antenna which was lifted by a 6 foot helium filled balloon, this vertical antenna was designed for 160 meter operation and worked marginally well on 80 meters. The vertical antenna used four 50' radials. With this antenna six contacts were made on 160 meters and about a dozen contacts on 80 meters. Why so few contacts? Noise, noise and more noise; due to the vertical nature of the antenna. So the issue this year was how to solve the noise problem. After talking to some top-bander (160 meter) friends of mine, it was decided to try something the society has not done before. This new idea was use separate transmit and receive antennas for the 160 meter station.

So the question became, what kind of receive antenna to use. My top-bander friend Jim Peterson K6EI, who is captain of the West Valley club field day operation over at Mora Hill just a stones throw from the FARS Maryknoll field day site, said to me, "why not put up a Beverage antenna, you have a large enough field for it." So based on this I designed a Beverage antenna that is 300 feet long and five feet off the ground; this antenna would just fit on the Field Day site. The Beverage antenna is a traveling wave antenna terminated at one end with a resistor equal to the characteristic impedance of the antenna. At the other end of the antenna is a 9:1 balun to match the antenna to the radio. Okay, but we are talking about two different antennas, one transmit and the other receive ported into the same radio. The answer to this is an automatic transmit and receive switch (T/R switch). Because of high loss incurred by the antenna being so close to the ground, a preamplifier is required to increase the signal to noise ratio (S/N) in order to pull the wanted signals out. What I needed was a pre-amp and a T/R switch in one box, and what I found was an Ameco PT-3 pre-amp that would suit these needs.

During the Friday before Field day I spent most of the day putting up the Beverage antenna so it could be tested that evening. The vertical antenna would not be deployed until Saturday night due to 160 and 80 meters not being active during the day because of D-layer absorption. So after deploying the Beverage antenna and setting up the GOTA station, I began testing the Beverage antenna. To test a Beverage antenna I listened to WWV and WWVH on 2.5 and 5.0 MHz. Because a Beverage antenna is directional and the antenna had it's major lobe facing Northeast while the null faced Southwest that meant that WWV located in Fort Collins, Colorado (Male voice) would come in strong while WWVH located in Hawaii (female voice) would come in weak if this antenna had directionality. The results indicated that the antenna had the expected directionality.

During the daytime the GOTA station worked 20 and 15 meters while at night the station worked 160 and 80 meters as non-GOTA, thus the 3A category. Next was to put up the 135 foot vertical antenna.

Just before the Pot luck Barbeque, I laid down the sixteen 50 foot radials in the paved parking lot and as the Sun started to set and the wind died down the helium balloon was filled to a

diameter of approximately 6 feet. The 18 gauge insulated antenna wire was attached to the balloon with a small fishing line lure swivel to counteract balloon wind rotation which attached to the antenna. Now with the antenna up it had to be matched. Matching was done with a matching network designed by Paul Zander AA6PZ and after much ado and fussing with the network the antenna was matched and ready to operate. Initially the first contacts were on 160 meters with the farthest being Arizona on voice. Because of the use of the T/R switch I could do an A/B comparison between the receive capability of the vertical antenna (TX and RX) and the Beverage antenna (RX) and the vertical antenna (TX) on the fly during the QSO by flop-flopping between the two configurations. While talking to the Arizona station I did that comparison. With the Beverage antenna, the station WS7R could be heard with great clarity with an S3 noise level, and when I switched to the vertical for receive WS7R could not be heard at all due to high S9+10 noise level. A/B tests were done also on a Utah station worked W7IVM on 80 meters with the same results. Stations were worked all night on both bands with good success. So now Proof of Concept has been done on this antenna system for 160 and 80 meters. What next? Since 160 meters is a nighttime winter band, there are plans in process to go back up to Maryknoll in January for the CQWW 160 meter CW contest and launch the balloon system again. This pretty much sums up the HF low band antenna for the FARS Field day 2008. I am already thinking of how to expand the capability of this antenna for both bands for Field Day 2009. If you are interested in working the 160 meter CQWW contest in January, please give me an email at sputnik@pacbell.net

- 73 es TNX de
Phil Hawkins, KA6MZE

Ham Radio's Rejuvenation

Electronic Design Analog/Power Editor, Don Tuite (NR7X), hypothesizes that "doing away with the code requirement last March has completed a rejuvenation of ham radio that was triggered by the World Trade Center attacks and Katrina." He is "looking for reader comments yea and nay." The article mentions FARS, other local clubs, the "monthly get together" (omits the name Am-Tech Day and FARS sponsorship) and the 9 am talk net (www.9amtalk.net). See more information at the following link.

<http://electronicdesign.com/Articles/Index.cfm?AD=1&ArticleID=19467&bypass=1>

Interesting Video

As a kid in the pre-TV era, some of my favorite things at the movies werethe B/W MGM shorts called, "Pete Smith Specialties". They were always whimsical and interesting. Pete Smith had an interesting delivery, later duplicated only by Harry Morgan (Dragnet, MASH). There is now a Pete Smith Specialty on Ham Radio on YouTube from 1939, before my movie-going. A little '30s melodramatic, but so what? It runs almost 10 min. <http://www.youtube.com/watch?v=vBGIdf0VjQ4>.

- de Les, W6TEE (via Peter, WA6VAQ)

How to Setup Free Internet Phone Service for Home

By Nimit K6XOX

Goal: Call at home or call my family overseas without a long distance fee like with my regular telephone or cell phone, without fixed line costs, without a PC running all day, using just a small box, able to connect to all [VoIP](#) (Info at -> <http://www.fcc.gov/voip/>) services, use Free service (<http://www.freeworlddialup.com>) and use an IP Phone (Internet protocol)

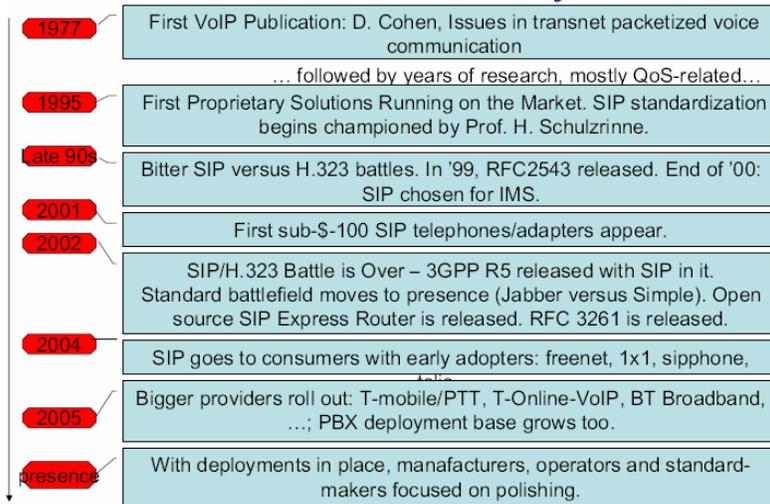
First of all, I just want to share how to benefit from a new technology solution that I found by accident. I tried to use the VOIP (Voice Over Internet Protocol) since I do not have a land line in my apartment and I use cable internet from Comcast. It might be useful for the communications in areas such as an emergency phone, with wireless internet such as field day site or any DX site or someone who always travel around the world and wants to get in touch with family back home but only carries along an ATA (Analog Telephone Adaptor) and plug into the hotel internet. For the Emergency purposes, you might be surprised that during an earthquake cell phone and landline services get overloaded - but with this alternative private line by Internet you can still get through. In other words, consider it like hot line to your family and friends in stead of RF radio. Don't you think this is great?

If you've never heard of VoIP, get ready to change the way you think about long-distance phone calls. VoIP, or **Voice over Internet Protocol**, is a method for taking [analog audio signals](#), like the kind you hear when you talk on the phone, and turning them into [digital data](#) that can be transmitted over the Internet. How is this useful? VoIP can turn a standard Internet connection into a way to place **free phone calls**. The practical upshot of this is that by using some of the free VoIP software that is available to make Internet phone calls, you're bypassing the phone company (and its charges) entirely.

There are many VOIP service providers out there, such as you seen on TV ads like Vonage or Comcast digital home phone service or even T-Mobile, that have jumped in to the new telephone system.

Let start with the History of the VOIP:

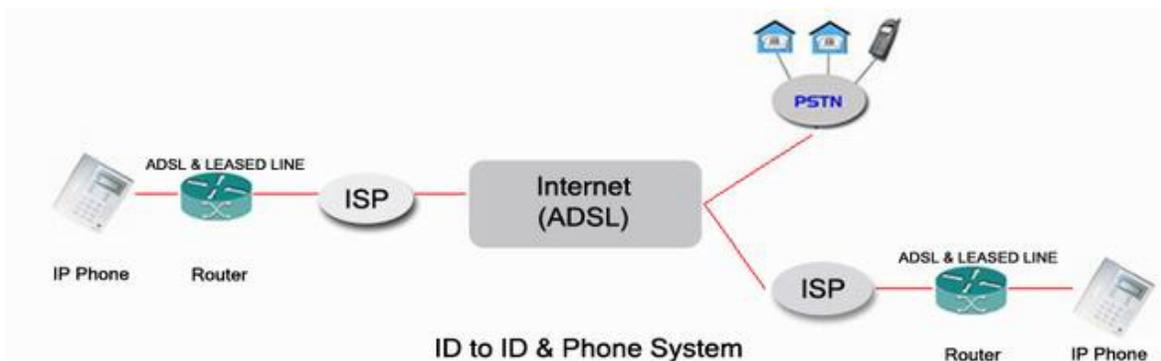
30 Years of VoIP History



Jiri Kuthan+Dorgham Sisalem, Tekelec, March 2007

I found this article from www.iptel.org. It is a good article in PDF format that is about 242 pages that clearly describe how to understand. I will not explain the technical detail because so many people might not want to hear and the technical issues and just want to know how to make it work, like a D.I.Y. weekend project, which is very useful for the XYL - hi hi .

Let see the simple diagram of how the system looks:



How does it Work?

The most common way VoIP works is that the end user establishes a hi-speed broadband connection, a router and a [VoIP gateway](#). Instead of a standard telephone line, the router sends the telephone calls over an Internet connection. The VoIP gateway, placed somewhere in direct proximity of the connected Internet converts the analog signals into digital format, which are further broken down into smaller chunks called 'packets', before sending it over the Internet, much like the way data is transmitted to and from the [computer](#). These packets are sent to their final destination and instructions for bringing back into an understandable form are embedded in them. It then goes through a VoIP gateway where the packets are reconverted into the original analog format utilizing a [PSTN](#) (Public Switched Telephone Network), thereby routing the call to the number the caller has dialed blending old school technology and hi tech delivery in a seamless and instantaneous way.

The system is setup at one of the free service providers called <http://www.freeworlddialup.com> (or FWD), that provided the SIP service and peering service. Those who used the SKYPE, Yahoo Msg or MSN messenger service already have an idea how it will work. This service has have both Soft phone (Software which run on PC same as SKYPE) and Hardware Phone (old Analog Phone with Adaptor or IP Phone with out leaving the PC on all day)

Benefit:

- Call Peer to Peer between FWD# to FWD#
- Out Bound Call Toll Free the Free World Dialup Toll Free Connection allows you to reach toll free numbers around the world. Here are the countries which are currently online:
 - * NL Dial *31(800)... to reach Netherlands Toll free
 - * UK Dial *44(800)... , *44(500)... or *44(808)... to reach United Kingdom Toll free
 - * US Dial *1(8xx) xxx xxxx to reach United States Toll free
- In Bound Call, one of free service provider www.ipkall.com IPKall is a PSTN (Public Service Telephone Network) to Existing VoIP service ONLY after you sign-in and they will give you Free Washington state phone number to your Internet phone such as area code 260,253,360 and 425 so that you can use your cell phone to call your family in other country but only pay for the domestic cost.
- Voice mail send call will go to voicemail and message will be sent to your email as .wav file <-I like this one !!

Get started:

1. First thing you need to get started is a *FWD* number.
 - Your FWD number is like your phone number, it allows anyone on the network to call you.
 - You can also call anyone else on the network using that person's FWD number.
2. Ok so now we need to get you registered.
 - Go to **my.FWD** and click **Sign up for FWD**. Go straight there » <http://account2.freeworlddialup.com>
 - You'll be asked to fill out some boring registration forms (a lot of this stuff makes up your personal profile). Just follow the onscreen instructions and you'll be fine. **No Need to be real but makesure Email is real
3. After registering you'll receive an email with some more information in it.
 - Remember to jot down your FWD number and password.
 - You can read the letter later, it'll still be there after you finish the quick start guide.

my.FWD

Home » my.Fwd » my.FWD

Sign Up

**EndUser Registration:
Personal Information**

FWD Number: _____

First Name: *

Last Name: *

Your Country: *

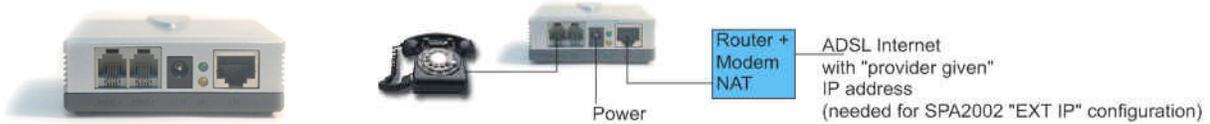
E-Mail: *

Next >> Reset

My Account
Received Calls
Missed Calls
Dialed Calls
My Profile
Extra Features
White Pages
Call Me
< Sign Up for Fwd >

Now, you have your FWD# then you need to setup the hard-Phone and Soft-Phone: 3 choices for now

1. **ATA** The simplest and most common way is through the use of a device called an ATA (analog telephone adaptor). The ATA allows you to connect a standard phone to your computer or your Internet connection for use with VoIP. The ATA is an analog-to-digital converter. It takes the analog signal from your traditional phone and converts it into digital data for transmission over the Internet. Providers like Vonage and AT&T CallVantage are bundling ATAs free with their service. You simply crack the ATA out of the box, plug the cable from your phone that would normally go in the wall socket into the ATA, and you're ready to make VoIP calls. Some ATAs may ship with additional software that is loaded onto the host computer to configure it; but in any case, it's a very straightforward setup. like this one Linksys SPA-2002 to connect from the Old Analog phone and connect to the LAN HUB or Router Cost around \$25 on EBAY sometime cheaper, I used this one and it was pretty easy to setup took me only 5 min to figure out and made it worked.



2. **IP Phone:** These specialized phones look just like normal phones with a handset, cradle and buttons. But instead of having the standard RJ-11 phone connectors, IP phones have an RJ-45 [Ethernet](#) connector. IP phones connect directly to your [router](#) and have all the hardware and software necessary right onboard to handle the IP call. [Wi-Fi phones](#) allow subscribing callers to make VoIP calls from any [Wi-Fi hot spot](#). Linksys SPA-921, It combine with the regular analog phone with digital in one set but it has LCD display to tell you caller ID and miss call and setup for Phonebook etc. I got this phone from EBAY around \$60 but it is worth for the money.

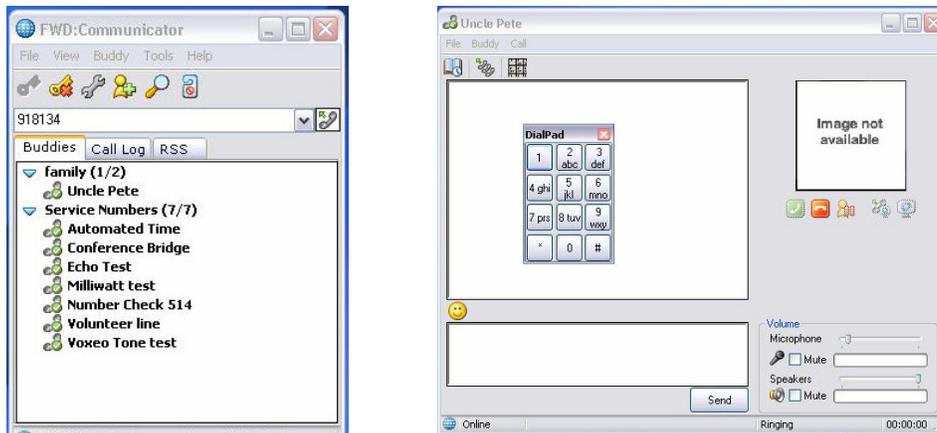


IP-Phone LINKSYS SPA-921

The **WIP300 Wireless-G IP Phone** operates in the 2.4GHz band, supports 802.11g and the latest VoIP SIP protocols. The large, full-color high resolution display features an intuitive user interface enabling users to easily and quickly configure the handset using Secure Easy Setup (SES). Cost around \$200

Mobile WiFi Phone for VoIP Service

3. **Computer-to-computer** -- This is certainly the easiest way to use VoIP. You don't even have to pay for long-distance calls. There are several companies offering free or very low-cost software that you can use for this type of VoIP. All you need is the software, a [microphone](#), [speakers](#), a [sound card](#) and an Internet connection, preferably a fast one like you would get through a [cable](#) or [DSL modem](#). Except for your normal monthly ISP fee, there is usually no charge for computer-to-computer calls, no matter the distance. In FWD that we call the FWD Communicator Free download from <http://www.freeworlddialup.com/products/?p=communicator>



I used one of my ATA boxes from Linksys in this case. Most other phones will utilize similar setup protocol.

After you have the Hardware then you need to get the WEB access to setup the IP phone or ATA.

The ATA that I used was setup after plugging in power and phone and LAN connector. Listen to headset and dial **** and you will hear voice of the system tell you the model and then you dial "110 and # to get the WEB IP to access the setup screen. After that you will see the INFO screen:

SIPURA technology, inc. **SPA-2002 Info screen shot** **Sipura Phone Adapter Configuration**

Info System User 1 User 2 [Admin Login](#) basic | [advanced](#)

System Information

DHCP:	Enabled	Current IP:	192.168.1.50
Host Name:	SipuraSPA	Domain:	hsd1.ca.comcast.net.
Current Netmask:	255.255.255.0	Current Gateway:	192.168.1.1
Primary DNS:	68.87.76.178		
Secondary DNS:	68.87.78.130 68.87.69.146		

Product Information

Product Name:	SPA-2002	Serial Number:	88012HA29960
Software Version:	3.1.5	Hardware Version:	2.0.1(5439)
MAC Address:	000E08FA8AF8	Client Certificate:	Installed

System Status

Current Time:	7/31/2008 17:37:14	Elapsed Time:	2 days and 22:49:01
Broadcast Pkts Sent:	0	Broadcast Bytes Sent:	0
Broadcast Pkts Recv:	8673	Broadcast Bytes Recv:	1130710
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	74789	RTP Bytes Sent:	11924588
RTP Packets Recv:	30321	RTP Bytes Recv:	4848864
SIP Messages Sent:	36763	SIP Bytes Sent:	12789735
SIP Messages Recv:	549	SIP Bytes Recv:	236524
External IP:	98.210.113.132		

Line 1 Status

Hook State:	On	Registration State:	Registered
Last Registration At:	7/31/2008 17:33:49	Next Registration In:	3365 s
Message Waiting:	No	Call Back Active:	No
Last Called Number:	*18008541366	Last Caller Number:	siwan
Mapped SIP Port:	5060		
Call 1 State:	Idle	Call 2 State:	Idle
Call 1 Tone:	None	Call 2 Tone:	None
Call 1 Encoder:		Call 2 Encoder:	

- FWD Standard Configuration Settings:
 - Username: *FWD Number*
 - Password: *FWD Password*
 - Domain: *fwd.pulver.com*
 - SIP Proxy: *fwd.pulver.com:5060*
 - STUN server: *stun.fwdnet.net:3478* (Phone must be STUN enabled)

Click to login as Admin and click advance setting, then setup for the line 1 as screen shot below:

SIPURA SPA-2002 Line 1 screen shot
technology, inc. **Sipura Phone Adapter Configuration**

Info System SIP Provisioning Regional **Line 1** Line 2 User 1 User 2 [User Login](#) [basic](#) | [advanced](#)

Line Enable: **yes**

Streaming Audio Server (SAS)
SAS Enable: no SAS DLG Refresh Intvl: 30
SAS Inbound RTP Sink:

NAT Settings
NAT Mapping Enable: **yes** NAT Keep Alive Enable: **yes**
NAT Keep Alive Msg: \$NOTIFY NAT Keep Alive Dest: \$PROXY

Network Settings
SIP TOS/DiffServ Value: 0x68 Network Jitter Level: high
RTP TOS/DiffServ Value: 0xb8 Jitter Buffer Adjustment: up and down

SIP Settings
SIP Port: 5060 SIP 100REL Enable: no
EXT SIP Port: Auth Resync-Reboot: yes
SIP Proxy-Require: SIP Remote-Party-ID: no
SIP Debug Option: none RTP Log Intvl: 0
Restrict Source IP: no Referrer Bye Delay: 4
Refer Target Bye Delay: 0 Referee Bye Delay: 0
Refer-To Target Contact: no Sticky 183: no

Call Feature Settings
Blind Attn-Xfer Enable: no MOH Server:
Xfer When Hangup Conf: yes Conference Bridge URL:
Conference Bridge Ports: 3

Proxy and Registration
Proxy: fwd.pulver.com Use Outbound Proxy: no
Outbound Proxy: Use OB Proxy In Dialog: no
Register: yes Make Call Without Reg: no
Register Expires: 3600 Ans Call Without Reg: no
Use DNS SRV: no DNS SRV Auto Prefix: no
Proxy Fallback Intvl: 3600 Proxy Redundancy Method: Normal

Subscriber Information
Display Name: **Name for Call ID** User ID: **FWD# xxxxxx**
Password: **FWD Password** Use Auth ID: no
Auth ID:
Mini Certificate:
SRTP Private Key:

Here it is very important to set the Dial Plan in order to call out with the correct template number as 1-8xx-xxx-xxxx or FWD# xxxxxx. All you need is to add the dial-plan string I provided to your ATA dial-plan context to call any US toll-free number without needing to prefix the dialed number with a *. Without the dial-plan string, you will need to prefix the dialed number with a * to place an outbound call to any US toll-free number using FWD. Copy the dial plan as below to replace with the default and add the some string to be able to call xxx and FWD# xxxxxx as below:-

(xxx|xxxx|xxxxx|xxxxxx|<:*>18[0678][0678]xxxxxxxS0|)

Hook Flash Tx Method: None FAX Disable ECAN: no
Release Unused Codec: yes

Dial Plan
Dial Plan: **(xxx|xxxx|xxxxx|xxxxxx|<:*>18[0678][0678]xxxxxxxS0|)**
Enable IP Dialing: **yes** Emergency Number:

FXS Port Polarity Configuration
Idle Polarity: Forward Caller Conn Polarity: Forward
Callee Conn Polarity: Forward

[Undo All Changes](#) [Submit All Changes](#)

Setup the SIP screen:

SIPURA
technology, inc.

Sipura Phone Adapter Configuration

Info | System | **SIP** | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2 | [User Login](#) | [basic](#) | [advanced](#)

SIP Parameters

Max Forward:	70	Max Redirection:	5
Max Auth:	2	SIP User Agent Name:	\$VERSION
SIP Server Name:	\$VERSION	SIP Reg User Agent Name:	
SIP Accept Language:		DTMF Relay MIME Type:	application/dtmf-relay
Hook Flash MIME Type:	application/hook-flash	Remove Last Reg:	no
Use Compact Header:	no	Escape Display Name:	no

SIP Timer Values (sec)

SIP T1:	.5	SIP T2:	4
SIP T4:	5	SIP Timer B:	32
SIP Timer F:	32	SIP Timer H:	32
SIP Timer D:	32	SIP Timer J:	32
INVITE Expires:	240	ReINVITE Expires:	30
Reg Min Expires:	1	Reg Max Expires:	7200
Reg Retry Intvl:	30	Reg Retry Long Intvl:	1200

Response Status Code Handling

SIT1 RSC:		SIT2 RSC:	
SIT3 RSC:		SIT4 RSC:	
Try Backup RSC:		Retry Reg RSC:	

RTP Parameters

RTP Port Min:	16384	RTP Port Max:	16482
RTP Packet Size:	0.030	Max RTP ICMP Err:	0
RTCP Tx Interval:	0	No UDP Checksum:	no
Stats In BYE:	no		

SDP Payload Types

NSE Dynamic Payload:	100	AVT Dynamic Payload:	101
INFOREQ Dynamic Payload:		G726r16 Dynamic Payload:	98
G726r24 Dynamic Payload:	97	G726r40 Dynamic Payload:	96
G729b Dynamic Payload:	99	NSE Codec Name:	NSE
AVT Codec Name:	telephone-event	G711u Codec Name:	PCMU
G711a Codec Name:	PCMA	G726r16 Codec Name:	G726-16
G726r24 Codec Name:	G726-24	G726r32 Codec Name:	G726-32
G726r40 Codec Name:	G726-40	G729a Codec Name:	G729a
G729b Codec Name:	G729ab	G723 Codec Name:	G723

NAT Support Parameters

Handle VIA received:	yes	Handle VIA rport:	yes
Insert VIA received:	yes	Insert VIA rport:	yes
Substitute VIA Addr:	no	Send Resp To Src Port:	no
STUN Enable:	yes	STUN Test Enable:	no
STUN Server:	stun.fwdnet.net:3478	EXT IP:	
NAT Keep-Alive Intvl:			45

Undo All Changes | Submit All Changes

That is all of the setup. Click Submit ALL Changes and the Box will go back to the Info screen and the Line is registered. If not please check the FWD# and Password. If everything worked as registered then you will hear the tone after you pick up the hand set and you can check the system by dialing 613 and # for the echo test. If you hear the echo voice then the system is good to go and go to www.freeworlddialup.com and click my.FWd and login to your account and click on Call me then you will hear the phone ring and complete the test. You can test by call 1-8xx toll free number to any company to test with this Dial plan above you do not need to dial * or *** just simply dial 1-8xx-xxx-xxxx.

After all the setup is finished then if you need to have the land line phone number same as regular PSTN, so far I only see Washington state provider providing the PSTN in bound call for free to FWD# , just go to www.IPKALL.com

Here how it works with IPKALL:

1. Register Your IP Phone with a VoIP service with www.freeworlddialup.com. Once you register your IP Phone with Voip Network, You are able to contact all of the other registered members if they are on-line when you are. There service provide as external SIP contact address like 612@fwd.pulver.net;
2. After you receive your SIP address, register that number at IPKALL.com and you will be assigned a local number in Washington State. Give out your Washington state number to friends, Family, Business associated and others around the world, using this number. They can call you for the cost of the domestic call and you will receive the call through your IP Phone device anywhere in the world.
3. If you don't answer or are not online, the call will go to voicemail and a message will be sent to your email as a .wav file



IPKall Signup

****ABSOLUTELY FREE**** Washington state phone number to your IP phone.

Important Announcement:

Please be aware that all accounts require a **valid email address** in order to receive your IPKall number. Accounts with invalid email addresses will be terminated. Accounts that are inactive for 30 days may be terminated for non-use.

To sign up for your own personal phone number, simply fill out the form below with your existing IP phone number.

[Log in to an existing account](#)

Choose your account type: SIP IAX

Choose Area Code for your IPKall Number:

SIP Phone number:

SIP Proxy: (ex. fwd.pulver.com)

Email Address:

Password:

of Seconds to ring before hanging up:



After you finish sign up for the IPKALL.com, you will received the email from IPKALL for the phone number with area code that you select such as my 1-360-717-xxxx. Then you can test by calling from your cell phone and you IP phone will start to ring. ALL DONE.

There are so many services and more flexibilities and more technical items than this, but for this article which need to be very simple to understand on how to setup there is no need to know how all the details work. More info that you can find and ask question to many people can be found at: - <http://forum.voxilla.com/ipkall-support-forum/> and <http://www.freeworlddialup.com/community/forum/>

If you have any problem with setup or questions, send me email at K6XOX at arrl dot net or call me at FWD# 918134

I just found out later Jeff Pulver who is founder of the <http://www.freeworlddialup.com> is HAM RADIO call sign WA2BOT in NY

Many of you might wonder how this happened to make a phone call for free and cheap call and free number,

Is it legal to use? Here is what I found:

February 20, 2004 Reaction: FCC Releases pulver.com/FWD Order

After carefully reviewing the text of the FCC's [Memorandum Opinion and Order](#) (Order) with regard to the pulver.com/FWD Petition, I realized that February 12, 2004 will be known as a landmark day in the history of communications. By declaring Free World Dialup's end-to-end IP communications as an unregulated interstate information service, the Federal Communications

Commission (FCC) has set the first critical brick of our broadband future, affirming its commitment to keep the Internet free from unnecessary regulation. Chairman Powell recently remarked that the Commission's decisions on Voice over Internet Protocol (VoIP) will be the most important in 100 years of regulatory history. This is not an understatement. The text of the Order provides much-needed regulatory clarity on existing law, clearing the way for investment and innovation in broadband applications and services, some we have yet to imagine.

I will watch with interest how the impact of this classification evolves in the VoIP rulemaking process, and reaffirm our commitment to working with policymakers on these issues. In the meantime, we commend the FCC for keeping the future of the Internet where it belongs - in the hands of consumers.

I look forward to also actively work with other regulatory authorities around the world as the IP Communications Industry continues to evolve.

Share this post: <http://192.246.69.231/jeff/personal/archives/000555.html>

Conclusion

VoIP is still in its infancy. While it holds great promise, it has some major technical hurdles to jump, such as emergency calling, and the need for an uninterruptible power source (i.e. PC battery backup). However, as VoIP is set to become more widely available, let's hope there will be reliable solutions in place for the existing problems in the coming years. Who knows? In another five years, we may have VoIP system sans a router and the VoIP service being more common than conventional phone networks we rely on so heavily today.

For the non-profit which main purpose with no commercial value but for education only.

Special Thank you, Credit and information from:

www.iptel.org , www.freeworlddialup.com, www.ipkall.com, <http://www.fcc.gov/voip/> , <http://www.howstuffwork.com>

FARS 2008 MEMBERSHIP RENEWAL FORM

Date: _____

PLEASE fill out the form for all new/renewal memberships.

Name(s) & Callsign(s) & Class (E-A-G-T-N-None): _____

Mailing Address: _____

Home phone: _____ Work phone: _____

Fax (H or W?) _____ Packet BBS Address: _____

E-mail: _____ ARRL Exp Date(s): _____

Preferred modes: (e.g. HF-SSB/VHF/QRP/Other): _____

I'm willing to Elmer new hams with: _____

Special topics of interest / suggestions for club meeting speakers:

Dues: \$20 per year, new members add \$9 for badge fee. **Please note:** Membership runs from January 1 to December 31.

Send your check payable to FARS, to:

David A. Cooper
PMB 41
270 Redwood Shores Parkway
Redwood City, CA 94065-1173



How to get to FARS Club meetings (Visitors always welcome)

Meetings are held at the Covington Elementary School (directions below) on the fourth Friday. Socializing at 7 PM with the regular meeting at 7:30 PM. There may be changes in the meeting dates for January, June, November, and December.

DIRECTIONS:

From Interstate 280. take the El Monte exit Northeast. Cross Foothill Expressway. (A) At the first traffic light turn right on Covington. (B) Immediately at the fork take the left street (Covington). Go about 1/10th of a mile. Turn left into the parking lot. The gym is the tall building to your right with red and white stripes.

From Foothill Expwy., take the El Monte exit and go Northeast; then follow directions as above at point (A).

From US101 or El Camino: take San Antonio Road west (to Foothill Expressway). Then follow directions above at point (A).

TALK-IN via the [N6NEI](#) (145.230-; 100Hz PL) repeater or the [W6ASH](#) 145.27- (100Hz PL) repeater.