

The KSU has a 25 pair cable that runs downstairs into the den for a Stromberg Carlson Call Director and also another 25 pair cable that runs upstairs into my room for a Western Electric 564 and an ITT 2564. The KSU is allocated four lines that are shared between the key phones.

A Comkey 416 system is also installed with two lines from the 1232. The Comkey is similar to a key system, but the KSU equivalent is built inside the master set. The other sets are smaller and are called slave sets, but all the telephones function exactly the same for the user.

The 1232 gets it hold music from a CD player that is connected to the PBX auxiliary input jack. The 1232 is connected to a paging system, and any user inside the PBX can make an announcement by dialing 320, or 321 or 322 for a specific paging zone. The 1232 also has an extremely sensitive audio monitor connected that can be accessed by dialing the correct trunk code. There is one telephone that is not connected to the PBX, and is instead connected directly to the ATT phone line. This telephone serves as a power failure phone, because the PBX does not operate in a power failure. Also connected to this telephone is a Caller ID unit to call people back with, because the 1232 precedes Caller ID and does not pass the date through. The 1232 has a remote access device connected to it.

Soon I will have a Mitel SX-200 ML and an Ericsson mechanical PBX set up. These PBXs will all be tied together. All four PBXs, the Asterisk, the Panasonic 1232, the Mitel, and the Ericsson will be remotely accessible.

The Asterisk machine operates on C*NET using 794 exchange. One great thing about Asterisk is that it does not require a powerful machine. It is running with a 2Ghz processor and 512MB of RAM. The Asterisk machine connects to the hub in my room. Connected to it are a Cisco 7960 and a Linksys 941 ip phones. I will soon be connecting an FXS and FXO ATA, which will allow me to tie together the Asterisk and 1232 PBXs. I have also learned how to connect a Nintendo DS using the SIP ap svsip. This C*NET switch currently hosts four people. The 794 prefix is run by PenguinTel, which is what I have named my C*NET service. PenguinTel would like to add more users, but we cannot add too many because of our limited bandwidth. The Asterisk machine communicates with its IP phones and remote users using the SIP protocol. Calls made between different exchanges are made using the IAX2 protocol, which is better suited for that purpose. The Asterisk machine is connected to two Sippgate lines, and one is used for an inbound portal from the PSTN (Public Switched Telephone Network) because Sippgate provides unlimited inbound calls and a DID number.

I started my interest in phones about a year ago, and since then have continued learning more and expanding my knowledge base. I'm not totally sure what I want to do in the future, but it will be something in the technical field.

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KyleBroMSW42.doc



Kyle Roberts
I am in the 10th grade in High
School, and play a Sax in the
Band, and I am only 50% geek!

PenguinTel.com

How I Have C*Net Set Up In My Home By Kyle Roberts



The network I have set up for my family's communication needs is a large network, but is not as complicated as it seems. One just needs to look at the network step by step to understand it.

First we'll look at the data side of it. Our internet service provider is Sonic.net, a great local DSL provider. It provides around 5mbps downstream and 768kbps upstream. I'd like to have more bandwidth, but unfortunately Petaluma and my neighborhood are limited in choices for internet. Eventually Uverse will become available, it has come to the neighborhood next to us, but that was a year ago and we still await it. Enough with that though. There is a Siemens DSL modem that is connected near the PBX, and connected to the modem is a Linksys wireless router.

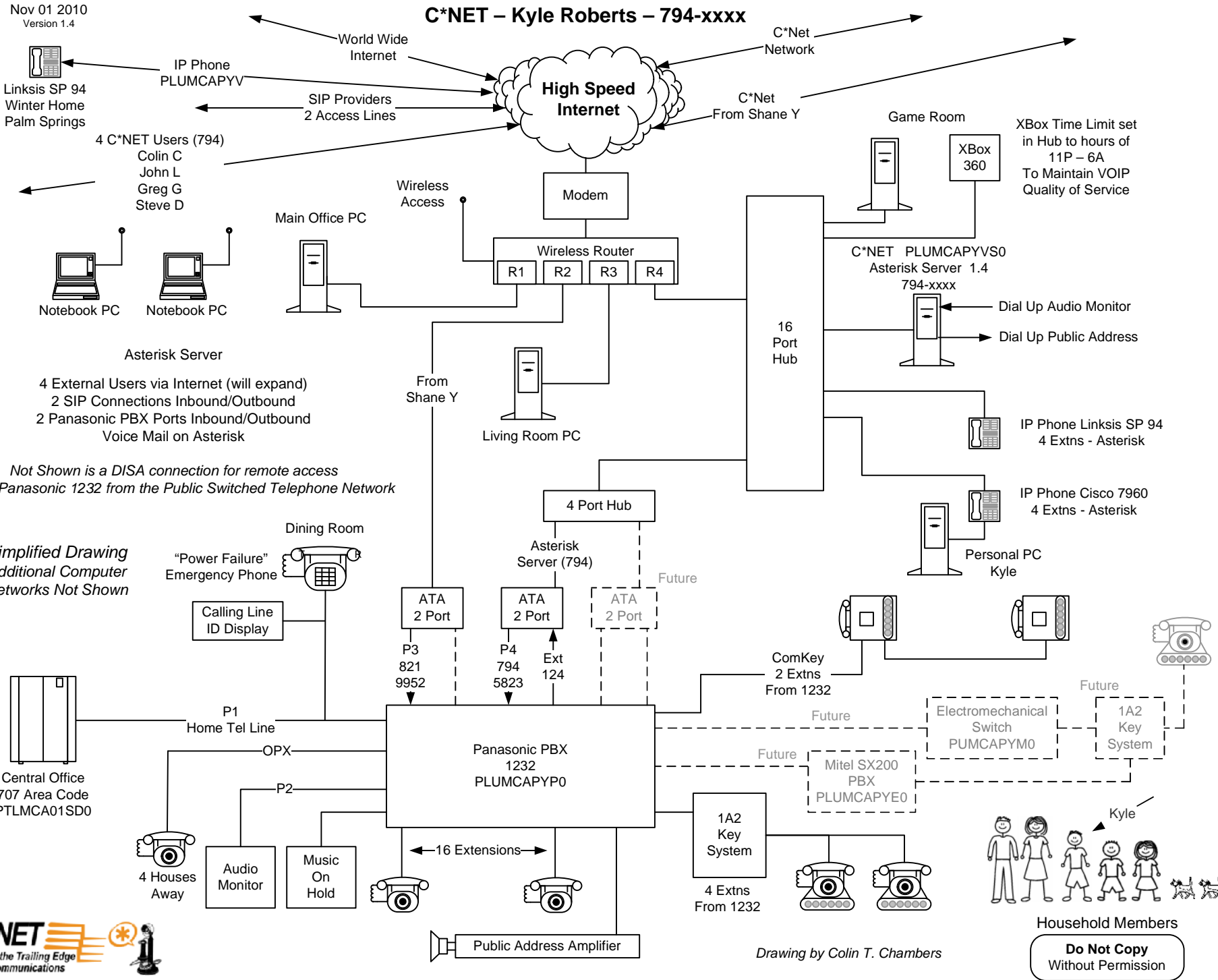
The Linksys router (WRT54GL) that is connected to is not your normal router. It looks like any other blue box Linksys product, but it runs Tomato, which is a Linux firmware. Tomato allows me to set options that would normally not be present on a \$50 router. I can set QOS (Quality of Service) settings, which is extremely important. QOS is essential for Asterisk and VoIP communications because "voice data" must always have first priority. If it does not, then the calls will be choppy and may even drop. Data over the internet is sent in packets, and at the user (client) end the data is reassembled and looks normal. It does not matter in what order the packets get there, all that matters is that they do get there.

VoIP (Voice over Internet Protocol) is different because the packets must be sent in the order they are created because you do not want to have your second word come before your first. Minimal latency (delay) is also important or the call will resemble a bad way. The router also does standard things like port forwarding for Asterisk. Certain devices that are more bandwidth intensive and may be disruptive to VoIP can be restricted to operate only on off peak call times.

This router is located downstairs for better WiFi coverage, and also because there are some computers downstairs. In my room upstairs I have a 16 port hub that is connected from the router with a 50 ft ethernet cable. I try to make all of my computers wired instead of wireless because it is faster. The only wireless devices that are connected are two laptops, a Wii, and a Nintendo DS that I have figured out how to use as a SIP enabled phone. The computers in my room connect to the hub, and the Asterisk switch also connects there because it is in my room. One computer is plugged into the back of the Cisco 7960 IP phone, because the phone has a two port hub built inside it. This is useful because if there is a computer that is far away from the router or hub, one can save cable by having to only run one long cable and then have a second, shorter cable plug into phone and computer.

That's the computer data side of things, so now we'll move to the analog telephony side. The main PBX that is connected is a Panasonic 1232. This PBX is unique because it supports all analog phones, including rotary phones. The 1232 is connected to an ATT land line, and also is connected to an ATA for my original C*NET connection that is hosted by Shane Young. The ATA connects directly to the router downstairs, because the PBX is also downstairs in the garage. Connected to the PBX is a Western Electric 551 (shoebox) KSU for the key phones.

C*NET – Kyle Roberts – 794-xxxx



Notes:

This is a simplified drawing and some details have been left out.

The 794 Prefix is at the home of Kyle Roberts. This location has a medium speed Internet connection. The Xbox limits the available bandwidth and QOS (Quality of Service) setting are necessary in the router/hub.

The entire family uses PC's at various times.

Printers are not shown, but are on the network.

NOT shown is the 795 Prefix, this is located on a separate Asterisk Server about 3 city blocks away. It is served by a Fiber Optic Network and has a higher available bandwidth.

Asterisk Servers can be accessed remotely.

It is possible to use FXO and FXS ports located on the Asterisk Server to provide services to the Future telephone systems.

An actual drawing of the complete detailed network would require the use of transparent overlays to show all the connections.



Drawing by Colin T. Chambers

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Comments:

This booklet and the drawings were made using Microsoft Visio and then pasted into Microsoft Word.

Most Network Engineers use Microsoft Visio to make Network Diagrams.

In Visio it is possible to link symbols to text files, pictures or Internet links.

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