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Enabling Voice Calls via a Wireless Broadband Router

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Abstract

Wireless networks are evolving rapidly, enabling services that until recently were almost exclusively bound to wired environments. With improvements to link quality, bandwidth and network management (Wireless Mesh Networks), the rapid growth and popularity of wireless networks is only going to continue.

With these improvements, opportunities to provide services such as Voice over Internet Protocol (VoIP) are becoming feasible. The LinkSys WRT54GL wireless broadband router is a cheap, yet powerful device that is able to run customised software on top of routing data wirelessly.

Wireless Mesh Networks, VoIP software and the WRT54GL will be combined in this Thesis to allow for a simple yet robust telephone network to exist in parallel with traditional data networking.

It is envisioned that the telephone network will be able to create and maintain a telephone network automatically. To enable this, additions to freely available VoIP software will be made and a decentralised telephone network will exist entirely on wireless routers.

A critical analysis will be carried out on all aspects of this Thesis and the feasibility and practicality of such a system will be investigated and presented.

Certification of Dissertation

I certify that the ideas, designs and experimental work, results, analyses and conclusions set out in this dissertation are entirely my own effort, except where otherwise indicated and acknowledged.
I further certify that the work is original and has not been previously submitted for assessment in any other course or institution, except where specifically stated.

Adam Jones

q1022786

Signature

Date

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Chapter 1

Introduction

This chapter highlights the way that the proposed system operates, and give an understanding of its' importance in real world applications. The aims of the project are expressed here as well as the work that was needed to be completed for the project to operate. A brief insight to existing solutions that are currently in use is also provided, which reinforces the need for this project.

1.1 Introduction

In recent times, two technologies have become the focus of much attention. Telephone systems and wireless networks. It is the purpose of this Thesis two combine these two elements in a way that allows for small to medium sizes communities[1] to create a simple yet robust telephone network that is accessible with cheap hardware and freely available software.

Telephone systems are shifting towards Internet Protocol (IP) based networks over traditional Public Switched Telephone Networks (PSTN). While it is widely accepted that problems exist when sending voice conversations over IP networks, much work has been completed to ensure that an acceptable quality of service is obtained.

Wireless networks are gaining much momentum as a viable option to wired environments. Early wireless networks were extremely limited by factors such as poor connection and bandwidth issues. As the technology matured, the quality of the wireless networks also increased. Modern wireless networks are still problematic but advances in this field are allowing the technology to become a popular networking tool.

While enabling voice transmissions to exist across wireless networks is not a new concept (mobile phones) the novel aspect to this Thesis is in the way that the network is created and maintained. This Thesis aims to create telephone network existing entirely on wireless routers with a management system that allows networks to automatically form and be destroyed in an ad hoc manner. To allow conversations to take place, specialised Voice over Internet Protocol (VoIP) software is loaded directly on the wireless router.

In the proposed system, each of the routers acts as a telephone exchange. Each of these exchanges is configured to share data to surrounding exchanges about phones that are connected to the local exchange. After client data has been shared between exchanges, calls are to be made over the wireless network. Based upon this, the hierarchy of the telephone network can be seen to be a peer to peer arrangement eliminating the need for a central governing body.

[1]In this context, communities is meant in the broader sense of the word. For example: an urban workplace or a remote township.

This Thesis enables a very simple telephone network to be created and maintained on wireless broadband routers. It illustrates that the proposed system is both feasible and practical using current technology in both hardware and software. Critical evaluation of the quality and capacity of the telephone system over wireless networks is carried out in this Thesis and will be further discussed.

1.2 Alternative Systems

Systems that are currently in use are based upon proven technologies that are well established but are all limiting in some way. These limiting factors relate to security, flexibility, management as well as installation and operating costs. Examples of currently used alternatives include traditional telephone systems, hand-held radio devices and Voice over Internet Protocol (VoIP) (both proprietary and freely available solutions).

Traditional wired telephone systems are extremely widespread and effective solutions that provide a satisfactory service. After many years and technological advances, the traditional telephone system is very robust method for enabling users to converse over great and small distances.

One of the problems of traditional telephone systems is way that they are installed and managed. Adding extra telephones and is not a simple task and involves specialized equipment and technical knowledge. This can be a costly exercise and may take some time for a technician to complete.

Another consideration is the physical layout of the area that is to have a telephone network. If the environment changes over time[2], or is hard to access, installations may not able to be completed.

The system being proposed in this Thesis is designed to operate over limited distances as a stand alone network. Compared to traditional telephone networks, the area covered by the network is very limited. It is envisioned that the concepts presented her will allow multiple wireless telephone networks to be linked with low latency, high bandwidth internet connections.

One of the greatest advantages of the proposed system is the ability for phones to be added to the network with little networking or telephone system knowledge. Management of the network is regulated automatically, removing the need for users to rely on a third party to make alterations to the telephone network. This Thesis also aims

[2]Open cut mines are a good example of the changes mentioned

to allow users in remote and/or difficult to access areas to have a reliable and robust telephone system.

Systems that are able to operate wirelessly, such as Citizen Band (CB) radio devices, also allow the creation of a communications systems that are able to operate in difficult areas. In addition to this, the systems are managed by the users and are free to use. The disadvantages that are associated with these systems include privacy, limited channels and range.

Typically CB radios have 40 channels that are available to carry audio, and each of these channels are open to anyone to listen to. This raises serious privacy issues that make the systems unsuitable for the broadcast of sensitive information. Another consideration is that 40 channels are quickly consumed by small numbers of users.

Once the number of users exceeds the available channels, careful channel management is needed and can become cumbersome. Range of the units is also a concern when using hand-held devices, without repeater stations to retransmit signals the system becomes limited.

The proposed system alleviates many of the problems discussed. Privacy is greatly increased with eavesdroppers made to attack the telephone system rather than passively listen in to broadcasts. Channel management is eliminated and replaced with telephone extension management[3] with more extensions added easily. Finally, the range of the network is controlled by the number of wireless routers that are in range of each other.

VoIP networks are a new and rapidly growing technology, many variations exist and all differ in some way. Many proprietary and freely available systems are available to be used. Proprietary systems are among some of the best communication devices[4] but require users to have access to the internet.

Many of the freely available versions of VoIP systems provide much the same functionality that is found in traditional systems and existing proprietary VoIP systems. This thesis uses a freely available telephone networking program, Asterisk, which is

[3]the number of unique extensions is almost endless

[4]The quality of Skype conversations often have a higher quality sound than traditional systems

one of the premier VoIP solutions available. This software is added to and modified to create the system being discussed here. A novel approach is being applied to telephone systems in this thesis and it is envisioned that the best elements of each the alternative systems presented, is utilised while the lacking areas of each is overcome.

1.3 Need for System

In many areas, whether they be remote sites or small areas in an urban environment, there exists a real lack of low cost, easily installed and maintained telephone systems. It is the aim of this project to provide a system that is suited to aforementioned situations. It can be seen in real world applications that implementing telephone systems are costly and difficult especially in challenging environments. Some examples include:

- Remote areas with little traditional network access.
- Offshore situations (Oil rigs, fleets).
- Temporary sites (Demountable offices).

In situations such as those listed above it can be imagined that a cheap and simple wireless telephone network could be an essential element.

1.4 Benefits of System

While the proposed system has obvious advantages over existing solutions including:

- Low cost of equipment.
- Ease of installation.
- Little maintenance of system.
- Ease of use (similar to traditional systems).
- Ability to form and maintain computer networks.

There are additional benefits to having such a system. These benefits are more to do with the router and its' abilities to transmit multiple types of data. As the network of nodes behave in a similar manner to wired Local Area Networks (LAN) many services are able to be distributed across the network. Some of the benefits include:

- Ability to distribute Internet access.
- Ability for the wireless nodes to perform multiple functions (data logging, controlling).
- Ability for the telephone to interface with traditional systems (call outside lines).

All of the above abilities are able to be allowed across the wireless connections. These functions are expected to be implemented when creating an actual system but are outside the scope of this Thesis.

1.5 Work Conducted

During the course of this project many obstacles had to be overcome in order to complete a working system. One of the key problems associated with the project was allowing each of the wireless nodes to be able to identify the telephone network servers around it. This was a major part of this project and this is covered in great detail in section 4.3.

Other, less intensive aspects that needed to be dealt with included:

- Determining the layout of the network and the devices in the network(Section 3.1).
- Installing the new firmware for the router(Section 4.1.1).
- Configuring the router to apply desired network configuration(Section 4.1.4).
- Installing/Configuring the telephone network software(Section 4.2).
- Testing the system suitably(Section 5.3).

At the conclusion of this process, a large discussion area is provided to determine the effectiveness of the system and this can be seen in chapter 6.

1.6 Chapter Summary

This chapter outlines the proposed system, its' need in the real world and gives a brief glimpse of how the system will operate. Throughout the following chapters, the elements discussed here will be explained in great detail and also their integration into the system will be shown.

Chapter 2

Literature Review

This chapter sets out to give an understanding of some of the key concepts and components that the system is built upon. Some of the underlying principles the pertain to the project are outlined below and are discussed in much detail throughout this chapter:

- Wireless Mesh Networks.
- Telephone Systems.
- Audio Codecs.
- Router Specific Firmware.

Each of the components listed above form the various sections of this chapter and are the building blocks that the project is formed upon.

2.1 Telephone Systems

Telephones have been changed the way that the world communicates. Until recently traditional telephone systems have enjoyed a monopoly of a very popular market. Since the introduction high bandwidth data networks, a newer method of sending voice between endpoints has been rapidly gaining momentum. These data networks are Internet Protocol based networks and as such this new method is aptly named Voice over Internet Protocol (VoIP).

Both of these technologies use differing methods to create voice links and enable telephone conversations. Traditional telephone systems employ circuit switched networks for connections to be made. Circuit switched networks can be imagine to have a direct line from source to destination. VoIP uses packet switched networks to transport the voice data. Packet switched networks have many, smaller paths to traverse to reach the destination. Over the next sections, these two systems will be explored.

Because both systems use digital signals to transmit the audio from source to destination. In order to reduce the amount of data that needs to be sent over the networks compression of the audio data is completed by applying compression algorithms. These are explored in Section 2.3.

2.2 Traditional Telephone Systems

Traditional telephone systems have, over time, embedded themselves into almost community. Initially starting out as totally analog systems, they have since evolved into a mixture of analog and digital systems[1]. The move towards digital transmissions has allowed for a higher volume of information to be transported through the same wired medium.

These telephone systems use circuit switched networks to enable a direct and exclusive line of communication between two points effectively linking the two end with a single

[1]Most long distance transmissions are converted to a digital signal shortly after leaving the source and are converted to an analog signal once they are close to the endpoint

wire. This allows for uninterrupted communication with constant latency.

2.2.1 Voice over Internet Protocol

As the name suggests, VoIP uses Internet Protocol as the medium to send voice over data networks. While traditional telephone networks use circuit switched network, VoIP uses packet switched network to send voice. Because of the design of packet switch networks, there are inherit problems associated with sending real time data over these networks.

The reason for this is the queueing and buffering of the data stream along the way to the destination. This has the following undesirable affects:

- The data packets are sent at regular intervals, but because of buffering (at intermediate routers) and queueing delays. The packets are not received at the regular intervals that there were sent in.[2]
- The delay can become quite large over large distances (buffering, queueing redirecting). Larger delays can make conversations unbearable for the end users[3].
- Lost packets can result in poor quality conversations. Typically this is seen in stressed networks or networks with unreliable connections.

These issues are currently preventing VoIP solutions from replacing traditional telephone networks. Much research is currently being done to reduce the effects of packet switched networks on real time data transmissions. It has been said that VoIP is preparing to revolutionise the way that telephone systems operate (Meggelen, Madsen & Smith 2007, p. 1-2) and that this conversion is rapidly arriving.

[2]Packets may even be received out of order.

[3]Typically over 300ms is unacceptable

2.2.2 Providers os Systems

Traditional telephone systems are generally maintained and owned by one or two companies. These companies own the infrastructure for the telephone network and as such have alot of control over the pricing access to the services.

VoIP implementations on the other hand do not generally control and any infrastructure but provide a means by which to enable telephone conversations to take place over data networks. Many VoIP providers exist with some proprietary solutions such as Skype^{TM} and other free distributions such as Asterisk.

2.3 Audio Codecs

Audio codecs (COder / DECoder) are used to encode analog signals that are within the human sonic range (typically 12Hz to 20 000 Hz) into a digitized approximation of the original signal. In a telephone system, analog signals are captured by a microphone on the handset then converted to a digital signal. Once this digital signal is received at the other end it is converted back to an analog signal and sent to a speaker so it can be heard. This conversion takes place so that the information is able to be sent on computer networks.

Typically, audio codecs are able to compress the digital signal in order to reduce the amount of data that needs to be transmitted between endpoints. The amount of compression that each codec is able to complete differs and as such the resulting quality of the reconstructed signal varies also. The compression of audio for transmission across a network is an important aspect to consider when designing a VoIP system. If the compression algorithm is poor, the bandwidth required by one telephone conversation may use most of the available bandwidth in the network. On the other hand, if the compression is too great, the quality of the telephone conversion may be degraded to a point where it is difficult to maintain a conversation.

There are many audio codecs available but only a handful that are supported by Asterisk. These codecs that are supported by Asterisk are very common audio codecs and offer a great choice for the selection of a suitable audio codec for the project. Most codecs that are integrated into equipment (mobile phones, IP telephones) are supported by Asterisk and various quality tests will be conducted on the audio quality.

The choice of audio codec is an important aspect of the project as one of the limiting factors of VoIP is the bandwidth available in the WMN. Although the G.711 is generally credited a higher quality reproduction of audio, an extensive range of tests were conducted on various age groups and it was found that the G.729A codec gave a better Mean Opinion Score (MOS) than the G.711 codec (4.02 versus 3.65 out of a possible 5) when voices were recorded and played back to the listeners (Light 2004).

For this reason, these two codecs are to be considered for the encoding for the telephone network. By using these two codecs it is hoped that a suitable selection can be made for anyone considering implementing the project being presented in this paper.

Table 2.1 shows the characteristics of the two codecs being considered for this project. It should be noted that the bit rate of the G.711 is eight(8) times that of the G.729A codec which further shows the importance of de-jitter, queueing delay and queueing loss when selecting an audio codec.

Codec	Creator	Codec Bandwidth (Kilo Bits / Second)	Frame Size (milliseconds)	Number of Frames per Second
G.711	ITU-T	64	20	50
G.729A	ITU-T	8	20	50

Table 2.1: Showing Codec Characteristics

Another important point to note is that of the ability of sending faxes using the codecs. While this is outside the scope of the project it may be implemented in a system based on this project and as such it should be noted that the G.711 codec can handle fax tones but G.729A will not. If a fax machine were to be connected to this system care must be taken to ensure that the G.711 codec is used.

2.3.1 G.711

The G.711 audio codec was released in 1972 with the primary focus on encoding speech for use in telephony applications. The actual process used to encode the analog voice signal to a digital signal for transmission across a network is called Pulse Coded Modulation (PCM). This technique uses bits to represent the input level of an analog signal and then preforms an Analog to Digital Conversion (ADC) on that signal to get a approximation of the original signal. This codec uses 8000 samples per second (8kHz) with eight(8) bits representing each sample hence using a total of 64kbps.

To aid in understanding this concept figure 2.1 show the way in which the analog signal (smooth line) is converted to a digital signal (jagged line).

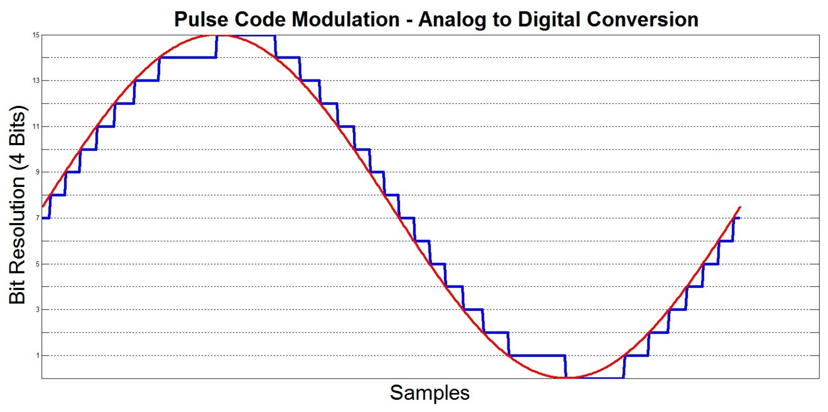


Figure 2.1: Pulse Coded Modulation

2.3.2 G.729A

The G.729 Audio codec uses a different encoding algorithm to the G.711 audio codec and that is Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP). Code Excited Linear Prediction (CELP) works by using linear predictions on excitations and was developed in 1985. This type of audio compression and the variations of it are one of the most widely applied speech coding algorithms in use.

A code book of typical input streams is used and when fed into the system can model many sounds produced by the human voice box. By using previous samples as a feedback to the system, pitch can be estimated to resemble the pitch of the speaker. Figure 2.2 shows how the system works when decoding of an audio signal taking place.

Essentially, to encode voice the encoder selects the best match to the sampled data from the codebook and transmits the reference to that code to the receiver and the receiver uses those references to reconstruct an approximation to original sample. This approach is more computationally expensive than the G.711 codec but results in a much lower transmission rate.

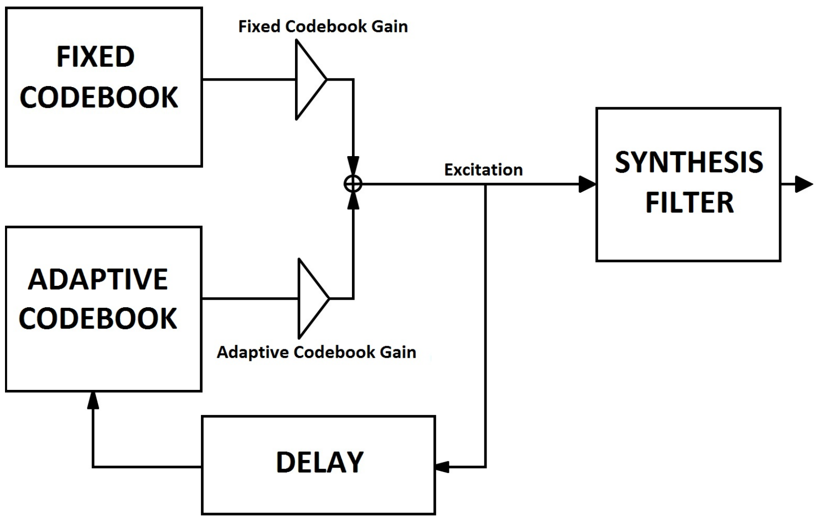


Figure 2.2: Code Excited Linear Prediction

2.4 Wireless Networks

Wireless communication as we know it today, had their beginnings when Guglielmo Marconi transmitted a radio frequency wireless transmission over a distance of 18 miles in 1895(Nicopolitidis, Obaidat, Papadimitriou & Pomportsis 2003, page 2). The first wireless data network was created in 1971, when the university of Hawaii shared data across islands. This network was named ALOHANET(Nicopolitidis et al. 2003, pages 7-8). Wireless data networks have evolved since their introduction and currently the IEEE 802.11 networks are the most popular standards for the wireless data networking.

2.4.1 Wireless Data Networks

Currently, the most common wireless standard is 802.11 introduced by the Institute of Electrical and Electronics Engineers (IEEE). Since defining the standard in 1997, the standard has been amended with with popular variants being 802.11a (1999) 802.11b (1999) and the 802.11g (2003)(Gast 2005, page 10). These standards have penetrated many areas and are still growing at a rapid rate. The main reason for the rapid expansion of wireless networks is primarily the cheap cost to users. Another reason for the success of the technology is the degree of freedom experienced. With most areas covered by wireless access, the ability to access network resources quickly and quickly has proved a popular system.

Most wireless networks consist of an access point and clients. These clients all communicate with the access point and the services are distributed out of the access point. These are labeled Wireless Local Area Networks[4] (WLAN) and are an extremely popular type of wireless network. Another popular wireless network is the wireless ad hoc network where temporary networks are formed when wireless devices come into range of one another. These sort of networks are generally formed with low power devices such as mobile phones and netbooks but can employ higher powered devices.

Recently, a new concept has been brought forward which is set to change the way that public networks operate. This network architecture is named Wireless Mesh Networks.

[4]Because of the similar architecture to wired Local Area Networks

2.5 Wireless Mesh Networks

“Wireless Mesh Networks (WMNs) are dynamically self-organized and self configured, with the nodes in the network automatically establishing an ad hoc network and maintaining the mesh connectivity” (Akyildiz & Wang 2005, p. S23).

Wireless Mesh Networks (WMN) are a developing technology that allows delivery of services to an area. The arrangement of a WMN consists of two categories of participants, which are mesh clients and mesh routers. The mesh routers form the backbone of the network and are usually considered to be immobile.

Mesh clients on the other hand are usually mobile radio devices that access resources that the mesh routers provide. Figure 2.3 shows the arrangement of a WMN with the dark shaded area representing the mesh routers and the light shaded area representing the mesh clients.

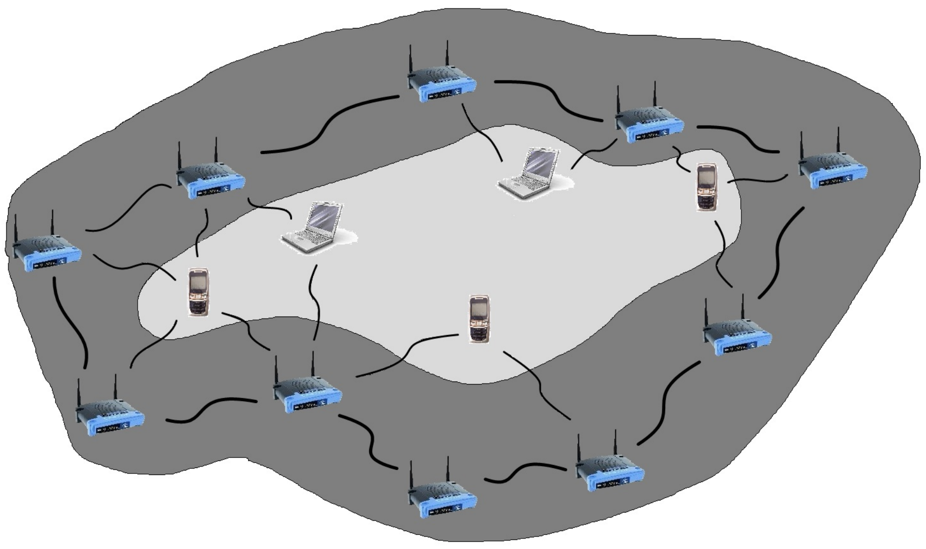


Figure 2.3: WMN Layout

This network can be seen in a number of applications throughout the world and some good examples of these are(Bruno, Conti & Gregori 2005, p. 124 - 125):

Intelligent Transportation Systems The Portsmouth Real-Time Travel Information System (PORTAL) is a system that relays real time information on public transport services. This information is able to be accessed by the public in order to better organise their travel around the city.

Public Safety The San Matteo Police Department in the San Francisco Bay Area uses a mesh network to co-ordinate the officers around the area in real time. This is in an effort to better organise assistance and co-operation between officers.

Public Internet Access Cerritos, California has an Internet Service Provider (ISP) that provides Internet access to paying customers using a WMN that is installed throughout the city.

WMN are described as communications networks that consist of radio nodes that form a network in an ad hoc manner. WMN have become the focus of many researchers in the past 10 - 15 years and much advancement has been made in this time. Although there have been vast improvements to the bandwidth and quality of signals, many areas such as packet routing exist where more research is necessary.

WMN are maturing quickly and with the price of hardware that is able to operate within this area, many projects and consumer products are being created that are taking advantage of this technology. One of the major aspects of this project is to correctly configure and maintain a WMN that is capable of more than just routing data between clients of the WMN but to provide a reliable and robust service that will run in parallel with data routing.

2.5.1 Mesh Routers

Mesh routers provide a backbone for the WMN and as such, enable the distribution for data routing and provision of resources. Common resources provided by WMN are Internet access, Intranet access, file transfers and phones systems. Mesh routers usually have more resources available to them as compared to mesh clients.

2.5.2 Mesh Clients

Mesh clients can be any radio device that accesses the WMN and uses the resources provided by the WMN. These devices can range from mobile phones, PDAs (Personal

Digital Assistant) and laptops or netbooks. In order to connect to the WMN these devices have to operate in the same radio frequency and be able to communicate with the mesh routers.

2.6 Wireless Mesh Network Routing Protocol

Routing protocols are responsible for the delivery of data across a network. In a wired environment (for example the Internet) this task is usually fairly straight forward as the routers are immobile and reachable. Once a route has been found in a wired network it is extremely unlikely to change. In environments where the topology changes from time to time (mobile wireless network) the routing protocol needs to be able to determine new paths to send the data along.

The design of wireless routing protocols has become a very active field of study for many researchers and this can be seen by the amount of proposals being published over the last 10 years. Thousands of designs have been designed, simulated and implemented in this field, each with a specific application catered for. Each of these proposed wireless routing protocols can be placed into three very broad categories, and they are: Reactive, Proactive and Hybrids (mixture of reactive and proactive).

The reason for the vast amount of routing protocols for wireless networks is due to the varying conditions of applications for wireless networks. Some of the aspects that are used to determine the most suited routing protocol can be seen below:

- The expected mobility of the wireless routers.
- The number of wireless routers that are part of the network.
- The type of data being sent across the network.
- The physical layout of the network.

Depending on the situation a certain type of routing protocol should be implemented. in the next three sections each of the categories will be looked at and an example of an available routing protocol will be discussed.

2.6.1 Reactive

Reactive routing protocols are named as such due to the way that they react to requests. When a request is made to send data to a certain destination, the routing protocol send requests to surrounding nodes on the network. This messages that are sent to surrounding nodes contain information pertaining to the requested destination, these broadcasts are passed on until the destination is reached. Once the destination receives the request the information is passed back along the path it was sent along back to the original sender. In most applications, multiple paths are received and then a suitable (usually shortest distance) path is selected and the data is sent along that path.

There are two main disadvantages to using a reactive routing protocol, and they can be seen below:

- The process of determining the routes to various nodes on the network can cause the network to become overwhelmed with path request packets. While one node would be unlikely to cause this, when multiple nodes are determining routes across the network it could be seen.
- There is a latency between attempting to send data and the data actually being sent. This time is due to the time required to determine a path to the destination.

Reactive routing protocols are particularly beneficial in wireless networks that are highly mobile in nature. One of the most common reactive routing protocols is Ad hoc On-demand Distance Vector routing protocol(AODV) (Perkins & Royer 1999).

2.6.2 Proactive

Proactive routing protocols determine the path to all nodes in the network (in a similar way to reactive routing protocols) but keep a table of all the paths to the nodes in a table. This table is stored locally and when a request is made to send data to a specific node, the node is looked up in the table and the path found there is used.

By using this method, data is quickly sent on the network. There are two major disadvantages to using this method and they are:

- The size of the tables (that are stored locally) grow with every node on the network. When many nodes are available in the network these tables can become very large and may cause problems associated with available memory.
- When a node is broken (powered down or moves out of range) this type of routing protocol may not recognise this fault before sending information along that path. This would lead to data loss or errors on the network.

Proactive routing protocols are seen to perform better in environments that have little to no mobility of nodes and that has nodes that are very reliable. One of the most common reactive routing protocols is Optimized Linked State Routing protocol(OLSR) (Clausen & Jacquet 2003).

2.6.3 Hybrid

Hybrid routing protocols attempt to use the best parts of both the reactive and proactive routing protocols whilst trying to avoid the disadvantages of both. One of the most common hybrid routing protocols is Hazy Sighted Link State routing protocol(HSLS). HSLS was developed by BBN technologies (*BBN Technologies website* 2009) in 2001 and revised in 2003 (Santivanez & Ramanathan 2003). HSLS keeps a table similar to the one used in proactive routing protocols but also uses the reactive local updates whenever a link is lost within two(2) hops of the node.

The disadvantages of this type of routing protocol are:

- Nodes receive updates from nodes that are further away less frequently and due to this may not have an up-to-date view of the network. These less frequent updates on distant nodes serve to reduce network bandwidth.
- Security mechanisms that are in-place may fail if key infrastructures are not in

reach of each other. However this is only likely in poorly designed (physical location) networks.

Hybrid routing protocols perform well in environments that are somewhat in-between those environments that reactive and proactive routing protocols are most suited to.

2.7 Security Issues

The implementation of security in data networks is an essential element that attempts to prevent attacks upon equipment and data within the network. Many security measures are available for networks and these include firewalls, anti-virus protection and controlling access to certain areas of networks.

Over the next sections, security in regards to wireless networks and Asterisk are evaluated.

2.7.1 Wireless Security

With the increasing amount of wireless networks, data security has become harder to defend against. Much research has been done on this topic and security measures such as WPA and WEP have been developed specifically aimed an wireless networks. The devices provide some protection against malicious attackers and should be enabled in the proposed system. This is especially true if sensitive material will be available on the network.

Wireless security is outside the scope of this Thesis and will not be implemented.

2.7.2 Asterisk Security

Security within Asterisk is treated very seriously, many features have been added to the VoIP system due to malicious attacks. One of the key problems is the ability for an attacker to use the connections available on a system tom make calls through the Asterisk server. Although the proposed system is not interfaced with an outside line, future implementations of this system may.

Eavesdropping on a conversation is a real concern for this Thesis. With enough knowledge and access to the wireless network, an attacker could force voice traffic to be routed through their connection (or recorded) and allow the two way conversation to be listened to.

Before employing this project in a real world situation, security would need to be implemented to guard against such attacks. As this Thesis is not being deployed for public use, this issue will not be further developed upon in the course of this Thesis.

2.8 Hardware

There were two main pieces of hardware used when creating and testing the system being proposed in this Thesis. The wireless routers that are used to perform the main functions of the system and the IP phones. Although the IP phone is not an essential part of the system (a software based phone could be used) it is included to give a full picture of the system.

Over the following two sections, both the wireless router and the IP phone will be discussed and the specifications and capabilities of each component given.

2.8.1 LinkSys WRT54GL Wireless Router

Figure 2.4: LinkSys WRT54GL

The WRT54G series of wireless router have been a popular line for Linksys mainly due to the versatility of the units. In particular, the WRT54GL wireless router has been popular and since its' introduction in 2005. Two versions exist of the WRT54GL and they are 1.0 and 1.1 (See table 2.2). The main difference between the two is that in version 1.0, the maximum size of the firmware image that could be uploaded using the preloaded firmware was 3MB. Later changed to 4MB when version 1.1 was released.

The WRT54GL supports the use of two wireless transmission standards and they are IEEE 802.11b and IEEE 802.11g. The characteristics for these two standards can be

Version	Processor Speed (in MHz)	RAM (in MB)	Flash memory (in MB)	Release Date
WRT54GL 1.0	200	16	4	2005
WRT54GL 1.1	200	16	4	2008

Table 2.2: Hardware specifications for the WRT54GL router

seen in table 2.3.

Protocol	Release Date	Frequency (GHz)	Typical throughput (Mbit/s)	Max net bitrate (Mbit/s)
802.11b	Sept 1999	2.4	4.3	11
802.11g	June 2003	2.4	19	54

Table 2.3: Specifications for 802.11b and 802.11g

On of the key selling points of this router is the fact that LinkSys made the firmware source publicly available. This has allowed community and commercial developers to revise the firmware image for specific applications. These extensions to the firmware allows the router to perform many extra functions on top of routing data wirelessly, some of the popular applications for the router are listed below:

Creation Internet Hotspots Many people have successfully created an Internet hotspot where wireless users are able to gain access to the internet through the wireless router. Some providers use an active portal where users are forced to register (or pay) before accessing the network (*Hot Spot PA website* 2009).

Controlling Robots There are some good examples that show the router controlling motion and web cameras (*ROBOSTUFF Website* 2009). The router controls the motors and servos as well as providing a way to access the robot wirelessly.

Adding Memory to the Router By modifying the hardware and adding an SD memory card reader to a spare (unused) communication port on the router an extra 2 gigabytes can be added to the memory of the router. Many application for this extra memory can be implemented such as data logging or creating a wireless file server (*Adam Kowalewski website* 2009).

More information regarding alternative firmware images is available in section 3.2. A complete listing of all of the technical information pertaining to the WRT54GL can be found on the LinkSys website (*WRT54GL Technical Information* 2009).

2.8.2 LinkSys SPA901 IP Phone

Figure 2.5: LinkSys SPA901

The SPA901 IP phone pictured in figure 2.5 is a robust unit that is able to communicate with the system being developed. The key features can bee seen in table 2.4. The phones' settings are able to be modified in two ways. Firstly the configuration is able to be modified by using a web interface. The alternate configuration is by using the Interactive Voice Response (IVR) menu where commands are entered by key presses on the keypad of the phone.

The phone has support for Session Initiation Protocol (SIP) which allows the phone to register with a service provider. The SPA901 also supports the audio codecs G.711 and G.729 which are being used in this Thesis. Also the ability for the phone to be set with either a static IP address or use a Dynamic Host Configuration Protocol (DHCP) client to lease an IP address is supported by the phone. In this paper the DHCP client will be running on all phones connected to the routers and therefore is a requirement that this phone is able to complete.

More in-depth technical information for the SPA901 IP Phone can be found on the

IP Phone Model	Supported Audio Codecs	Connection Protocol	Configuration
SPA901	G.711, G.726 G.729, G.723.1	Session Initiation Protocol (SIP)	web interface and key pad

Table 2.4: Specifications for the SPA901 IP Phone

Cisco website (*SPA901 Technical Information* 2009).

2.9 Chapter Summary

Throughout this chapter an attempt is made for the reader to gain an understanding of the various aspects of the project. It is these fundamental components that the project is built upon and as such these concepts are needed to be understood in order to gain an understanding into how the proposed system operates.

For the remainder of this document, these concepts will be integrated into the project and will combine to allow various aspects of the system. The underlying principles discussed in this chapter form the basis for which the rest of the project is built upon. The results found during the process of researching this project, and shown in this chapter, will be compared to actual results found in section 6.2.

Chapter 3

Design of Proposed System

This chapter covers the decisions and assumptions made about the proposed system in regard to the network, the devices in the network and how they combine to offer a robust telephone system. The key components of the system include:

- The wireless router.
- The IP phone.
- The firmware for the wireless router.
- The VoIP software (Asterisk).
- The various software portions that are responsible for the maintenance of the system.
- The purpose designed programs that allow the automated configuration of the telephone network.

Each Section of this chapter aims to give a brief insight into the operation of the various aspects of the system but also defines how that aspect integrates into the system as a whole.

3.1 Design Philosophy

One of the most fundamental aspects of this project is to first develop an idea of how the system will operate. This process required many considerations about the possible ramifications of the decisions made about seemingly straight forward aspects of the system. Of particular interest, the following aspects were first looked at before attempting any implementation:

- The expected mobility of the mesh routers.
- The portability of the IP phones and the ability to log onto a phone.
- Automatically connecting Asterisk servers.
- How the wired and wireless interfaces of the router would be handled.
- How the audio data is routed.

After deciding on how the system would operate, each of the key aspects of the system was looked at to determine the feasibility of the implementation. Over the following sections, these aspects will be discussed from a design perspective. Chapter 4 will discuss the implementation of these concepts.

3.1.1 Mobility and Reliability of Mesh Routers

The mobility of the mesh routers goes towards the overall design of the system and the selection of aspects such as the routing protocols used in the system. It is envisioned that the mesh routers will mostly remain immobile in the practical application of the system. The main reasoning for this is listed below:

- In order for a specific area of coverage for the WMN, the mesh routers (backbone) are needed to remain in a set place.
- Most phones will remain fixed in place (inside a residence or office) and these areas should remain relatively immobile.

- Most (if not all) routers will need to be plugged into a power source which limits the mobility of the units somewhat. The support for mobile devices to interact with the system.

Another important aspect to consider is that of reliability of the mesh routers. By reliability it is meant that there should be little to no time that the mesh router should be powered down. This is especially true in sparse networks where one node failure may cause a failure of the network. In figure 3.1 it can be seen that is the node in the middle in part (a) were to fail, that the entire network would become two separate WMN (b).

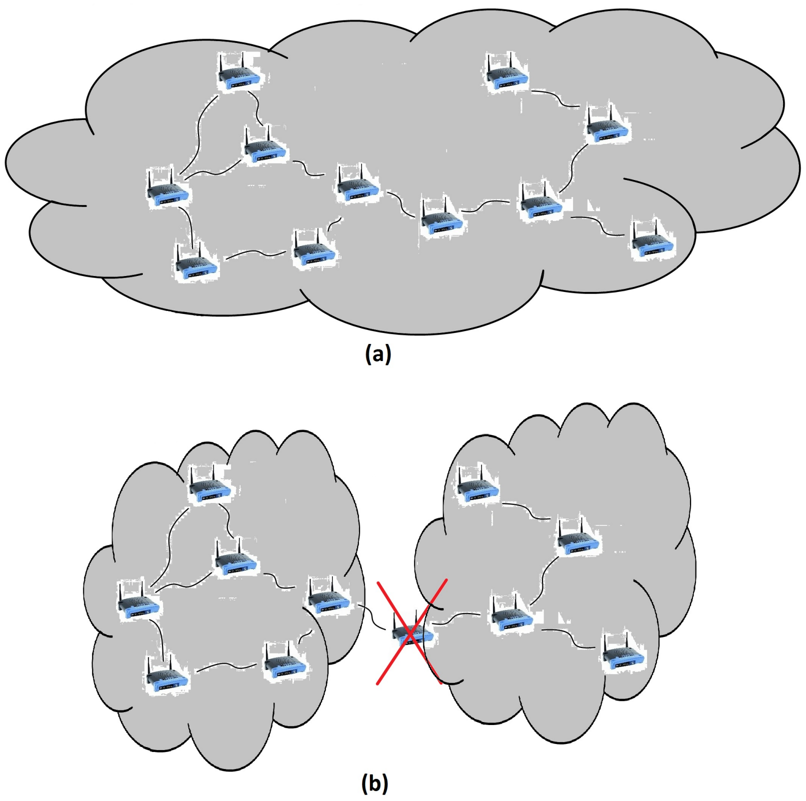


Figure 3.1: Showing a critical failure of WMN

In section 3.3.3 it was discussed that proactive routing protocols are particularly beneficial to relatively immobile wireless networks. From the above described design of the proposed system, it can be seen that this type of routing protocol would suit this application well. Section 4.1.3 discusses the actual selection of the routing protocol for this project.

3.1.2 Portability of IP Phones

The ability to move phones between mesh routers (Asterisk servers) is an attractive idea for two reasons. Firstly, from time to time users may switch areas or sites and would still like to have calls routed to their phone regardless of their physical location. Secondly, when creating a telephone network, the ability to plug in a phone and have it automatically configure itself with a mesh router would save a lot of time in the setup of the telephone network.

The ability for the user to move within the telephone network and still be able to have a phone within a close proximity can be completed in two ways. Firstly as described above, the phone can be physically taken with the user and plugged into a closer mesh router. Secondly, if the network has additional phones the user could log on to the phone (change the extension of the phone to be the users' extension). This ability to log on may be more commonly used as the physical removal and transport of the phone is cumbersome.

The ability for the phone to be moved to a different location and automatically re-join the telephone network has been implemented into the system by running a Dynamic Host Configuration Protocol (DHCP) server on the wired ports of the mesh router. In this way, when an IP phone is plugged into a router, it leases an IP address off the router and uses the default gateway (obtained as part of the DHCP server) to register with the Session Initiation Protocol (SIP) server that is running concurrently with Asterisk on the router (see section 4.1.4).

Once the SIP session has been configured and the Asterisk server is aware of the phone (see section 4.3) calls can be made and received by the phone.

Another major aspect to consider is how the IP phones will be allowed to move around the network. One potential solution to this is to have the IP phone always register with the same Asterisk server regardless of where it is on the network while another is to have it register with the Asterisk server that it is connected to.

In order to get a better understanding of this, consider Figure 3.2. In the figure it can seen that the IP phone is connected to Router 1 and that wireless connections exist between the routers numbers 1 through 4. It can also assumed that all the routers are running an Asterisk server. If the phone were always to register to Router 4, network traffic could be sent from Router 1 to Router 3 and then to the Asterisk server on Router 4 and a connection made. Additionally, if the IP phone were plugged into Router 2 the traffic could be directed from Router 2 through to Asterisk on Router 4.

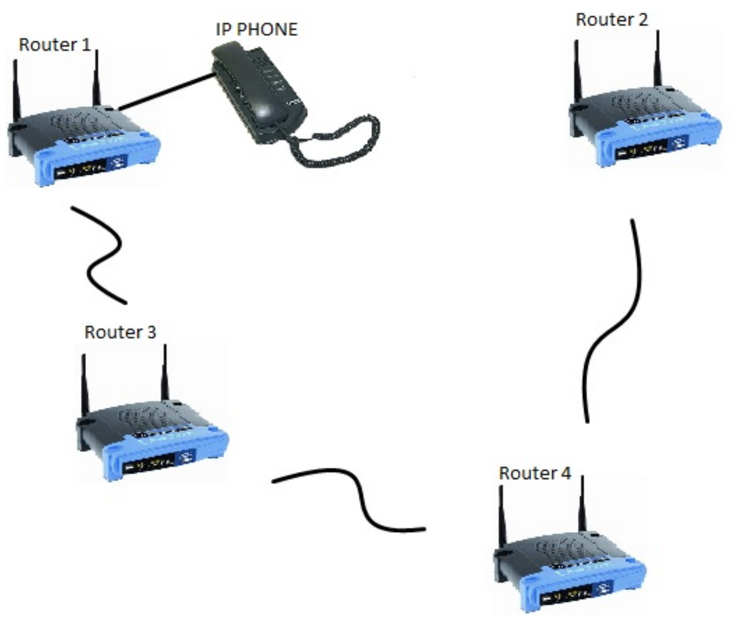


Figure 3.2: IP Phone Registration

In contrast to this, if the IP phone were to register with the router it is plugged into it would always be able to register without sending traffic across the wireless connections and if the phone moved to any router it would simply register there.

The implementation of the two different methods described above vary greatly and the supported method needed to be decided on very early in the project. This process in making this decision required that many situations are taken into account. Each method provides an advantage over and for this reason, the process of selecting the method for the project was complicated even further.

The main differences that presented themselves whilst considering the methods are listed below:

- Network traffic caused by each of the methods.
- The use of nodes that are not running the Asterisk software.
- The configuration of wired and wireless interfaces of the router.

The network traffic generated by the two methods may vary greatly depending on the layout of the network and the devices that are operating in the WMN. There are two types of telephone network participants that need to be considered and they are:

- Phones that connect to the wireless interface of the WMN.
- Phones that are plugged into the wired interface of the mesh router.

Devices that connect to the mesh router via the wired interface can register directly with the router that they are connected to or register with any other mesh router in the network. The first method does not transmit data across the wireless network but communicates with the router (more specifically the Asterisk server) directly. When the phone registers with a remote server wireless traffic is sent and also the problem of data backtracking can come into effect (see figure 3.3). As this project has the wired and wireless interfaces separated there is also problems faced with NAT and firewalls when attempting to send data to a device that is behind these security devices (see section 3.1.3). These issues affect the case where a phone is able to register to a remote Asterisk server.

With these considerations in place, the decision to force the phone to register with the mesh router it is plugged into will be implemented. It should be noted that this does not stop a user from connecting to a remote Asterisk server but this would have to configured manually using the phone settings. By implementing the system in this way, the automated configuration of the Asterisk servers and phones can be implemented.

Devices that connect to the wireless interface of the router are able to move freely throughout the WMN and as such may move to a position in the WMN that would cause the audio to back track through the network in order to reach the destination (this

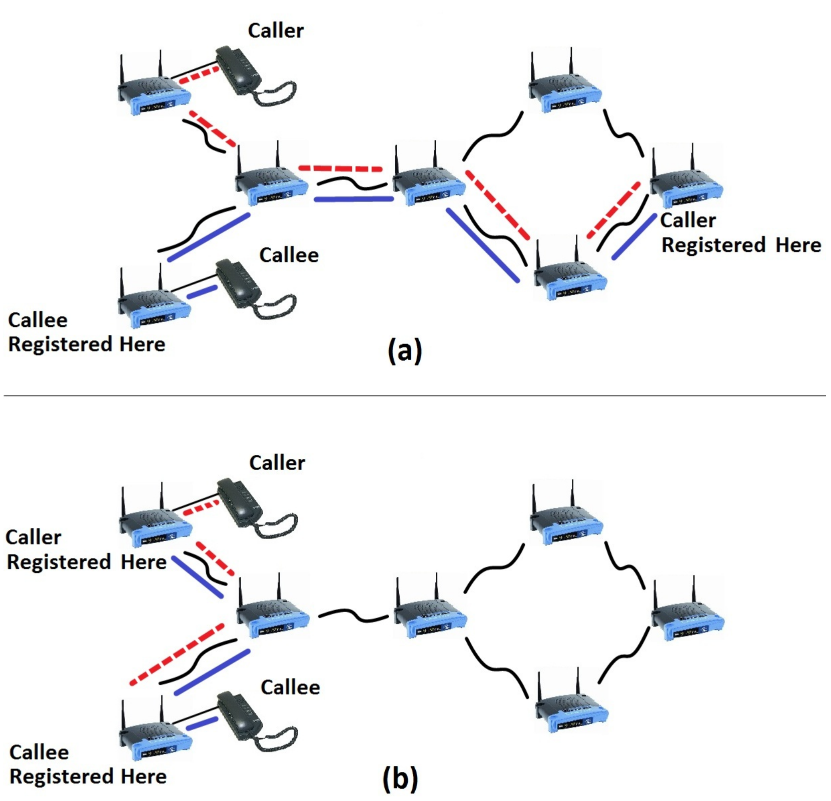


Figure 3.3: Audio data traversal for different configurations

is shown in Figure 3.5). Obviously this is not an acceptable use of network bandwidth and the system would need to be designed in a way that would not occur.

In order to allow for the data traversal to be minimised, the system should be designed to allow the mobile device to constantly change the router that it is registered with whenever the device finds a closer mesh router. This would effectively eliminate the problem but would require more advanced network management and in turn more processing power. As the main consideration is to keep the wireless bandwidth optimised, allowing the mobile device to change Asterisk servers should be implemented. As this project does not incorporate the use of wireless participants (due to time limitations) this consideration will not be included into the system.

3.1.3 Routing of Audio Data

Asterisk provides two options for the management of the VoIP conversations. The first allows the two end users to manage their connection directly (after the connection has been made by Asterisk) while the second forces the data to be routed through the Asterisk servers at each end. In order to better understand this, refer to Figure 3.4.

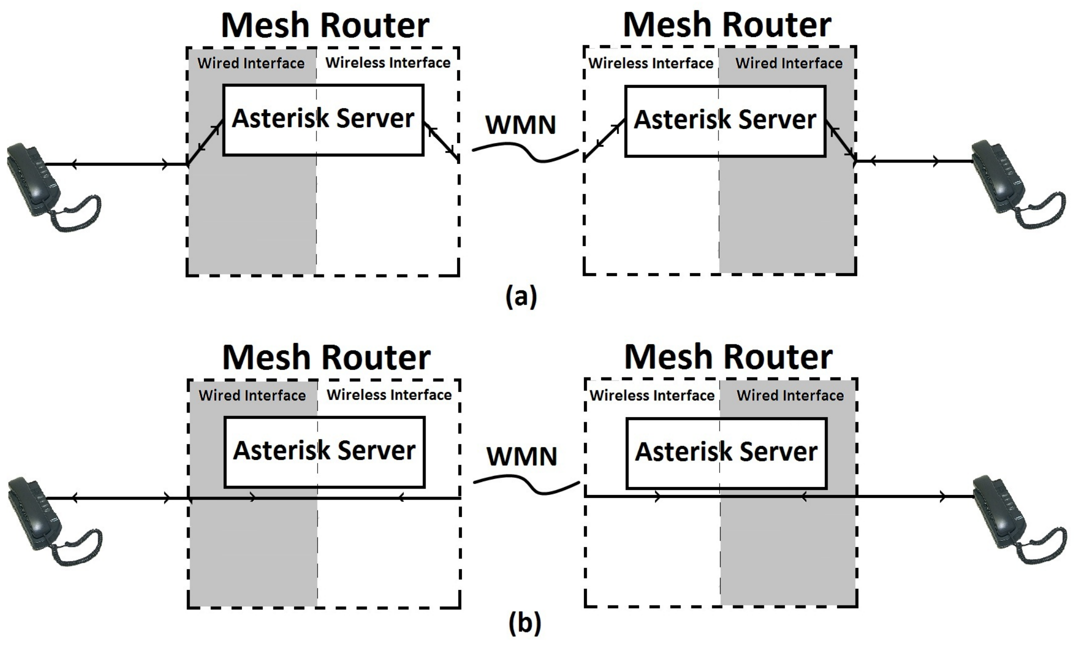


Figure 3.4: Options for the routing of VoIP traffic

It can be seen that when Asterisk is configured to force the audio to be sent through Asterisk server (Figure 3.4(a)) that the bi-directional data transfer is occurring from each of the IP phones. Alternatively, when Asterisk allows for the data to be sent directly between the phones (Figure 3.4(b)) it can be seen that the data bypasses the Asterisk servers on each of the routers.

While this seems to have little effect in the figure, there are serious ramifications of selecting between the two options. If the wired and wireless interfaces are not bridged issues such as Network Address Translation (NAT) and firewalls can hinder the ability for the phones to connect to each other and in most cases there is no audio received at each end. Therefore, if the interfaces of the router are to be separated (as is the case in this project) then the audio traversal must be forced to use the Asterisk server that resides on the router in order to bypass the NAT and firewall issues.

If the system were to incorporate mobile devices that are able to interact with the Asterisk server wirelessly (see section 3.5) it would be beneficial that these devices were able to create a direct link to other phones on the network. This is due to the fact that the mobile devices may actually be closer to the destination than the Asterisk server that they are registered with and this would lead to unnecessary packet forwarding. To understand this better refer to figure 3.5.

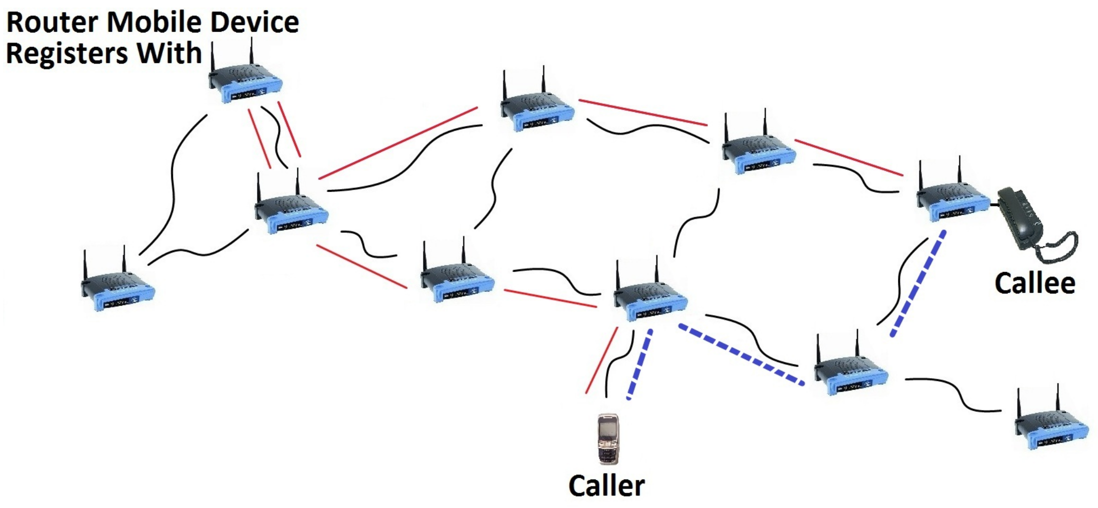


Figure 3.5: Traffic route for alternate configurations

In the figure it can be seen that the data must traverse a lot further when forced to go through Asterisk (thin solid lines) servers as opposed to when it does not (thick segmented lines). It is a major aspect to allow mobile devices to connect directly when designing this system. To implement this, additional programming would be needed to be added to the programs that connect Asterisk servers. This will not be included in this paper but is highlighted to allow for a optimised system to be created in the future.

3.2 Firmware

Institute of Electrical and Electronics Engineers (IEEE) Standard Glossary of Software Engineering Terminology, Std 610.12-1990, defines firmware as:“The combination of a hardware device and computer instructions and data that reside as read-only software on that device”

In the case of this project, it refers to the software that is employed to control the operation of the router and allow a user to modify and add programmability to the device. It is essential that a Linux compatible firmware be installed to allow all software such as Asterisk to run on the router. Many firmware distributions are available for the WRT54GL and are discussed in the following sections.

3.2.1 Variations

There are multiple firmware images available for the Linksys WRT54GL wireless router. These firmware images allow one to alter the operation of the router, more specifically, they allow a small installation of Linux to become the operating system for the router. The compatible distributions include, but are not limited to:

- Open-WRT (*Official Open-WRT Website* 2009)
- DD-WRT (*Official DD-WRT Website* 2009)
- Sveasoft (*Official Sveasoft Website* 2009)
- FreeWRT (*Official Free-WRT Website* 2009)
- Tomato (*Official Tomato Website* 2009)

These distributions can be loaded onto the router to replace the supplied firmware and allow a lot of flexibility and configurability. Of the above listed firmwares, Open-WRT and DD-WRT were the best choices for the specific application that is being proposed in this paper. The reason for this is that the aforementioned distributions have a good developer base and are free to use. They both also have good community support for troubleshooting and guides.

Open-WRT

Open-WRT was originally developed for the WRT54G series of routers as a replacement for the stock firmware for the router. Open-WRT uses a Linux based firmware and has allowed for the addition of many features that are not available with the stock firmware. Some of the notable extensions are as follows:

- Static Dynamic Host Configuration Protocol (DHCP) server.
- Quality of Service (QoS) mechanisms for certain data types.
- Highly configurable firewall and other networking tools.
- Printer server and sharing abilities (with USB support).

The main reason for the success of the Open-WRT firmware is that it is a community based project that is freely available for use. Many people have contributed to the firmware with many other people revising the code constantly leading towards a very stable base for the router to operate from.

DD-WRT

DD-WRT is also a Linux based firmware that is compatible for the WRT54GL wireless router. As with Open-WRT, this firmware is a community based project and has had much success in creating a very flexible firmware. All of the functionality that is available from Open-WRT is included in the DD-WRT variation.

3.2.2 Selection

The selection of the firmware for the router was achieved by heavily researching available options. Table 3.1 shows the key points of each of the systems:

Firmware Distribution	Focus	Costs	Support
Open-WRT	Enthusiasts	Free	Very good
DD-WRT	Enthusiasts	Free	Very good
Sveasoft	Professional Market	Costs involved	Poor
Free-WRT	Professional Market	Free	Good
Tomato	Enthusiasts	Free	Good

Table 3.1: Comparison of Firmware

Of these Open-WRT and DD-WRT both provide similar services and are quality distributions. After considering the two firmware distributions' suitability and also the feedback from many users of both, the following items were found of Open-WRT:

- Large community of contributors.
- Regularly updated.
- Many guides and "how-to's" available.
- Reported success with Asterisk installations.

Some issues regarding licensing of elements of DD-WRT have also been reported which are of concern. Additionally, security issues have been identified that allowed the remote execution of commands on the router (that have since been rectified) were also of concern. For the above reasons, Open-WRT was chosen as the firmware that will be used for this project. It provides all of the required functionality as well as little resource use which allows the elements of the project enough resources to run comfortably.

3.3 Software

Specialized software was needed to get a functioning telephone exchange running on the wireless routers. This software has very specific jobs to perform within the system and the design of the system relies heavily on these jobs being completed properly. These software aspects of the system are:

- Asterisk is responsible for the call setup and maintenance of the network.
- The DHCP server exist to aid in the ability for devices to automatically join the network.
- The OLSR routing protocol allows the wireless nodes to form a WMN.
- The Asterisk Client Information Sharing (ACIS) programs allow the telephone network to automatically configure itself.

Also, configuration of the DHCP server was required to maintain connections between the router and the IP phones that are plugged into the router. Service advertisement methods were also coded and installed in order to implement automated configuration of the Asterisk servers. The routing protocol that is responsible for the setup and maintenance of the WMN is discussed in section 4.1.3.

3.3.1 Asterisk

Asterisk is an free and open source project that was created to provide an alternative to conventional telephony systems. Since its' inception in 1999 by Mark Spencer, Asterisk has enjoyed the support of many members of the community. This community support was available due to the fact that Asterisk has dual licenses including GNU General Public License (GPL) and a proprietary software license to permit licensees to distribute proprietary, unpublished system components. With the open source development, many program bugs and additions were made to Asterisk enabling it to become a very robust and efficient telephony system.

Although Asterisk incorporates many features such as call conferencing, video calls, mailbox creation and call forwarding, these features were not implemented in the set up of the project. During the course of the project, only very basic settings for Asterisk were used which allow call creation/termination of audio conversations. The main reasons for the exclusion of the additional features included:

- Keeping the file system usage to a minimum.
- Reducing the complexity of the file maintenance.
- The capacity of a basic telephone system is able to be determined.

Asterisk is an free and open source project that was created to allow the creation of telephone systems. At the time of writing, the supported Asterisk release is 1.0.10-1 for white Russian distribution of Open-WRT(*Official Open-WRT Website* 2009). This version of Asterisk has multiple available versions for Open-WRT and of these, the *asterisk-mini_1.0.10-1_mipsel.ipk* package was chosen for the reason that this distribution has the smallest install size while still providing all of the required functions that are needed for the project.

Simple Asterisk setups are configured with the modification of two files (in most cases). These two files control the phone connections ("sip.conf" or "iax.conf") and what steps are taken when a phone dials a number ("extensions.conf"). With very little modification, a very basic VoIP telephone network is obtainable and calls are able to be made.

3.3.2 DHCP Server

Open-WRT provides an open source DHCP (Dynamic Host Configuration Protocol) server that can be run on the router to allow clients connected on the ethernet ports of the router to lease an IP address from the router and connect to the SIP server (which is part of Asterisk). This is available as a standard component in the firmware but also is available as a package at *http://downloads.openwrt.org/whiterussian/packages/* and is called dnsmasq.

This package is described as a lightweight, easy to configure DNS forwarder and DHCP server. Although the DNS forwarder will not be considered for this project it is quite useful in many networks. To enable the DCHP server to handle any IP phones that are plugged into the router the configuration files for dnsmasq must be configured correctly. This configuration will be shown in section 4.1.4.

3.3.3 OLSR Routing Protocol

Optimized Linked State Routing (OLSR) protocol is a proactive routing protocol (see section 2.6) that allows for the interconnection of multiple nodes in a wired or wireless network. OLSR attempts to maintain a constantly updated view of the network that is a part of and does this by sending messages to its' neighbours. a link state algorithm, the information reagarding the node' information (such as link quality) and location is flooded throughout the network to give a good idea of the nodes in the network, OLSR optimises the information that is transmitted to ensure that this information does not cripple the network.

The variation of the OLSR routing protocol that is used in this project is called OLSRd and was originally written by Andreas Tønnesen. Since the initial protocol was released the code has been modified and the operation of the routing protocol optimised. Early on there were issues that limited the number of nodes catered for to around a couple of hundred but since the optimisation of code, the number of nodes that can be supported is now within the thousands.

This routing protocol will be used in this project as it has been shown to perform very efficiently in large networks and allows for the proposed system to remain connected at a small cost to resources and bandwidth of the router and the network.

3.4 Asterisk Client Information Sharing (ACIS)

Because of the way that Asterisk is designed there is no way for it to automatically discover and connect to other Asterisk servers. It should be noted that this is not a flaw of Asterisk but has been carefully designed to operate this way. The reason for this is that Asterisk is typically configured to connect to other services by an administrator and is modified rarely (if at all) due to the static nature of wired telephones.

The ability for Asterisk to become aware of other Asterisk servers that are running around them is a fundamental aspect to this project. A major success of the project hinges on the way that this task is completed. There are two considerations for this aspect and they are to ensure that the network is up-to-date and that the amount of data that is being transmitted over the network is kept to a minimum.

In order to successfully implement a system that attempts to optimise these two principles, many scenarios were considered. The data contained in the messages sent between Asterisk servers was kept to a minimum by using data contained in the headers for the sent packets (seders' IP address). Without carefully planning for when information is needed to be sent and reducing the size of the packets this aspect of the project has the ability to cripple the network with bloated information on the status of phones on the network.

To enable the Asterisk servers to maintain a view of the network, client updates are sent at regular intervals. The actual time interval used is very subjective and will vary between applications. A very static phone network would not need to be updated as often as a network where users are constantly being added and dropped.

The information contained in the data sent was reduced to strings containing the extensions[1]. In order to reduce the transmissions to this, the structure of the files "extensions.conf" and "sip.conf" were hard-coded into the programs that controlled the transmissions. In this way simple loops were able to be used to rewrite the files in a suitable format.

[1]Telephone identification numbers (Telephone Numbers)

It was also decided that when a phone disconnects that this information would not be shared with all of the Asterisk servers in the network but rather that the details about the phone (extension and location) are removed after not receiving an up date from the phone after a certain time period. This also helped to reduce network traffic.

3.4.1 Transport Protocol for Messages

There are two protocols that transport data through IP networks. Each have advantages over the other and to ensure that the proper method is chosen both were studied. The two protocols are: User Datagram Protocol (UDP) and Transmission Control Protocol (TCP).

Transport Protocol	User Datagram Protocol	Transmission Control Protocol
Guaranteed delivery	No	Yes
Bandwidth Used	Least	Most
Connection	One to many / one to one	One to one

Table 3.2: Comparison of Transport Protocols

From Table 3.2 it can be seen that TCP ensures that the sent data is received in tact and in the correct order that is was sent in. UDP makes no guarantees about the delivery of the packet at all. UDP packets are typically used in real time applications where it is redundant to have the sender retransmit when a packet is lost. TCP connections are used where data integrity takes precedence over timeliness. In order to keep the Asterisk servers up-to-date, time is an important aspect but it is more important that the information is received correctly.

It is also important that the most efficient method is used in a wireless environment. TCP data integrity comes at a price of extra overhead. When little data is being sent between servers, this overhead may actually incur more data than is being sent. In the case of this project this is especially true, with most transmissions containing four telephone numbers. This fact makes the UDP transport protocol a more attractive option.

Finally, it can be seen that one to many connections can be made with UDP connections. This is especially important in the proposed system as it is assumed that multiple routers will be present in the WMN and being able to send one message to all other routers will reduce the total bandwidth consumed for this aspect of the system.

For the reasons mentioned UDP broadcast messages are used in the implementation of the Asterisk Client Information Sharing (ACIS) programs.

3.4.2 Detection of New Phones

From time to time it can be assumed that a new client will want to connect a new phone into one of the Asterisk servers (Temporarily or permanently). This phone new phone needs to be recognised and added to the two files "extensions.conf" and "sip.conf". In order to complete this, the Asterisk debug information needs to be regularly checked for phones attempting to connect to the server[2]. When a phone is found, its' information is added to the system and Asterisk is told to reload the configuration files.

This does mean that until the phone attempts to reconnect, it will remain disconnected. This could not be avoided in this project as developing an interface with the SIP server would involve vast amounts of time. It may be beneficial to create an interface (or an entirely new SIP server) that would allow for the automatic addition of phones.

[2]When a phone attempts to connect and is not in the "sip.conf" file, the connection is refused and this information is saved to file.

3.5 Adding Support for Mobile Radio Devices

In today's market there exists many powerful mobile radio devices available (with many in use). One good example is the smart phone or mobile phone. With many mobile phones being able to participate in wireless networks already, the ability to connect to a WMN similar to the one described in this paper and use the services available on the network.

It is possible to incorporate a mobile phone into the proposed system with the addition of service discovery. Service discovery is a concept whereby a node in a network can advertise its' services and resources to other nodes in the network. A service or resource that can be advertised may include printing services, networked projectors, e-mail servers and in this case, Asterisk. Many service discovery protocols exist and are implemented in many situations - these include Service Location Protocol (SLP), Jini (for Java), Salutation, XMPP, UPnP. All have strengths and weaknesses and are suited to specific applications.

In addition to this, the phone needs to be capable of participating in the network (transmitting in IEEE 802.11b/g) and be able to process service advertisements and use SIP to register on the network. If these requirements are met, the phone would be able to connect to an Asterisk server in the same way that the SPA901 IP phone does. This would allow the phone to roam within the WMN and make / receive phone calls as long as it has registered with an Asterisk server.

To enable this feature the following service discovery aspects would need to be investigated and implemented on the mesh routers:

- The selection of a suitable service discovery algorithm (XMPP, SLP).
- The supported connection protocols (SIP, IAX or H.323).
- What is advertised to be considered enough information (IP address, available bandwidth, physical location etc).
- The supported audio codecs (possibly phone specific).

This ability will not be implemented in this paper but is included to show the robustness of the proposed system. It is envisioned that this feature will be implemented in a system that would be deployed in a real application. There would be certain design consideration that may need to be altered to ensure a level of quality if these mobile devices were to be implemented. These would include:

- Changing the routing protocol to another that would better suit highly mobile devices.
- Altering the way the connections are made with Asterisk server (allowing hand-offs between servers or forwarding packets).
- Adding information to Inter Asterisk Client Information Sharing to allow specifying codecs, passwords and message banks.

This also would require much testing of the system under new and varying conditions in order to gain an understanding of the potential of the system.

3.6 Combined System

After combining all of the concepts and components described in this chapter, Figure 3.6 gives a graphical representation of how the system will operate on each of the wireless routers.

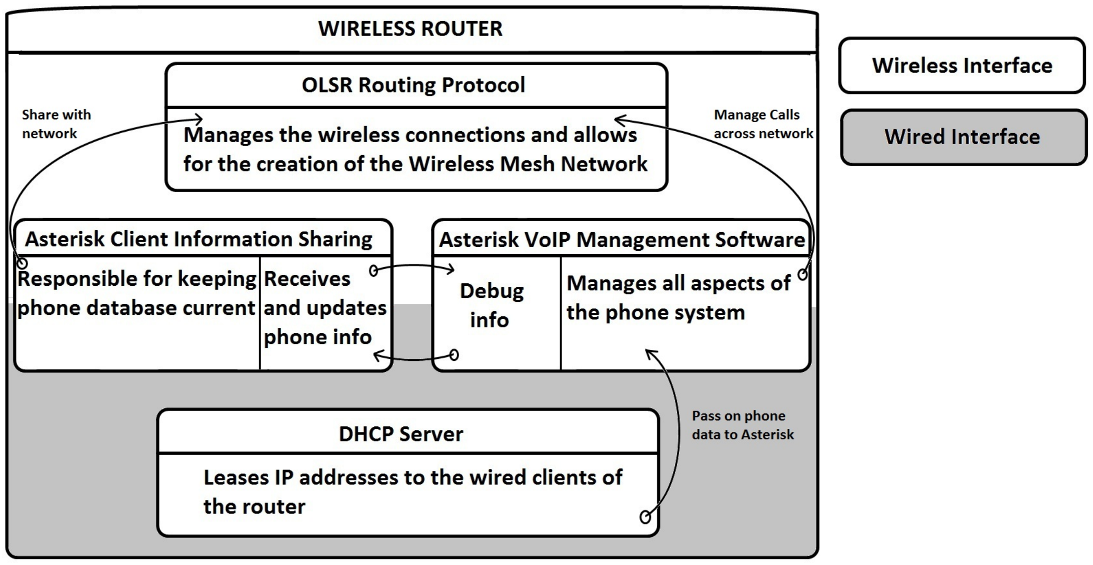

WIRELESS ROUTER
Wireless Interface
Wired Interface
OLSR Routing Protocol
Manages the wireless connections and allows for the creation of the Wireless Mesh Network
Share with network
Manage Calls across network
Asterisk Client Information Sharing
Responsible for keeping phone database current
Receives and updates phone info
Asterisk VoIP Management Software
Debug info
Manages all aspects of the phone system
Pass on phone data to Asterisk
DHCP Server
Leases IP addresses to the wired clients of the router

Figure 3.6: Figure showing how the various components combine.

It can be seen that the components combine and allow for telephone network to exist on the wireless router. Asterisk can be viewed as a gateway that connects the wired and wireless interfaces of the router and is positioned in the figure overlapping the two. This concept was discussed in Section 3.1.2.

Also of interest is the close relationship between the Asterisk server and the ACIS module. In a sense, these two separate entities could be combined as they are both performing functions essential to the telephone network. They have been left separated in this figure to illustrate that ACIS concept is novel and is not a standard feature of Asterisk[3].

[3]The SIP server however is a standard feature and has been included into the Asterisk server for this figure

3.7 Chapter Summary

This chapter has provided information about how the components of the proposed system will operate as well as some of the future extensions and the considerations that are required for their implementation. Throughout this chapter, an effort has been made to emphasise how versatile the system is and also goes towards the various configurations that are able to be implemented in the system.

The decisions and assumptions contained within this chapter are based upon much community feedback as well as expert opinion. While the configuration of various aspects of this system are suited to this application, it should be noted that this may not be the ideal structure for all applications. Where appropriate, an alternative method for configuration, suited to differing deployment environments is provided and it is hoped that this paper provides a good basis for their design and implementation.

Chapter 4

Configuration of Proposed System

This chapter covers the configuration of the network, the devices in the network and how they combine to offer a robust telephone system. The key components of the system include:

- The wireless router.
- The firmware for the wireless router.
- Software aspects that are responsible for the maintenance of the system.
- The IP phone.

In order for the proposed system to operate as intended, it is important that the router is properly configured. Throughout this chapter a Microsoft Windows computer was used to interact with the router and as such, most of the software programs mentioned during this section of the paper specific to Microsoft Windows operating systems.

4.1 Open-WRT Settings and Configuration

The correct configuration of the WRT54GL wireless router is fundamental to the overall success of the proposed system. In order to implement the designs from chapter 3 many changes to the Open-WRT firmware need to be made.

Before being able to configure the router, the custom firmware must first be loaded onto the router and this process is described in section 4.1.1. After successfully loading the new firmware, additional software is to be loaded onto the router and these segments can be seen in sections 4.1.3 through 4.2.

4.1.1 Loading Open-WRT on to the WRT54GL

Replacing the firmware on the WRT54GL wireless router can be completed in two ways. The first and easiest is to use the tool provided on the web interface of the firmware that the router comes with. The alternative method involves sending the firmware image to the router in the boot process. The second method is commonly used when changing from a third party firmware to another. Both of these methods are outlined in the next two sections:

Web Interface

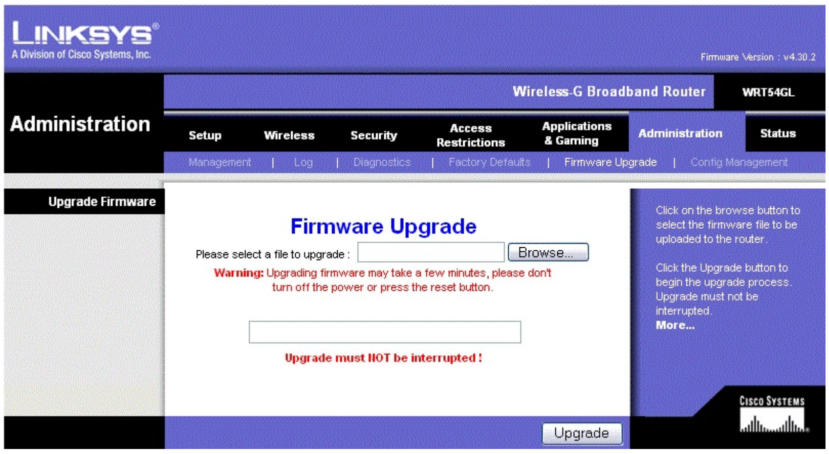
In order to load the custom firmware onto the router using the web interface the router must be connected to a computer using an RJ-45 cable. Once connected and the router can be accessed, the following set of steps need to be followed:

1. Navigate to the web interface (usually 192.168.1.1) using an internet browser.
2. Log onto the routers' web interface with the following default account credentials.
   - Username: <blank>
   - Password: root
3. Click on *Administration* link.

4. Click on *Upgrade Firmware* link. See figure 4.1.

5. Click *Browse* and select desired firmware file.

6. Click *Upgrade*.



Please select a file to upgrade :

Warning: Upgrading firmware may take a few minutes, please don't turn off the power or press the reset button.

Upgrade must NOT be interrupted !

Click on the browse button to select the firmware file to be uploaded to the router.

Click the Upgrade button to begin the upgrade process. Upgrade must not be interrupted.

Figure 4.1: LinkSys Firmware Upgrade Page

Once the above steps have been completed, the custom firmware should be loaded and once the router has finished rebooting, Open-WRT should be running.

Direct Transfer

Direct transfer of the firmware image allows the replacement of the firmware on the router without the use of the web interface. In order to prepare the router for the transfer, there is a command that needs to be set in order to give a 2 second time window (during the boot process) for the transfer to occur. The setting that needs to be set is called *boot_wait* and the value needs to be set to *1*. This can be set in two different ways depending on the firmware that is currently on the router. If the original firmware is on the router, then the web page can be used to change the value as follows:

1. Navigate to the web interface (usually 192.168.1.1).

2. Log onto the routers' web interface with the following default account credentials.

   - Username: <blank>

    - Password: root

3. Click on *Administration* link.

4. Click on *Diagnostics* link.

5. Locate *Ping* and click on the link once after entering each of the following commands:

    - `;cp${IFS}*/*/nvram${IFS}/tmp/n`
    - `;*/n${IFS}set${IFS}boot_wait=on`
    - `;*/n${IFS}commit`
    - `;*/n${IFS}show>tmp/ping.log`

6. After entering the last command and clicking the *Ping* button, a list of output should bee seen with the line "*boot_wait=on*" present.

Alternatively, if there is a custom firmware already loaded onto the router, access to the command line interface is required (see section 4.1.2). If the firmware that is on the router is an Open-WRT distribution then the following command is needed to be entered at the command line:

```
nvram set boot_wait=1
```

Both of the above methods set the router to wait during the boot sequence. Once this has been completed, the actual firmware transfer is able to be completed. In order to successfully do this, the following instructions need to be followed:

1. Open Microsoft Windows command prompt (open run program and type *cmd.exe*).

2. type the following command but do not enter the command.

    - `tftp -i 192.168.1.1 PUT openwrt-wrt54gs-squashfs.bin`

3. Unplug power to the router.

4. Apply power to the router and enter the command from Windows command prompt.

5. If the transfer was successful, a similar output to the Windows command prompt should be seen.

   - `Transfer successful: 154931 bytes in 4 seconds, 38732 bytes/sec`

It should be noted that when *boot_wait* is enabled and the router is booting the IP address is always 192.168.1.1 and that when the process is completed, the IP address will be 192.168.1.1 as well.

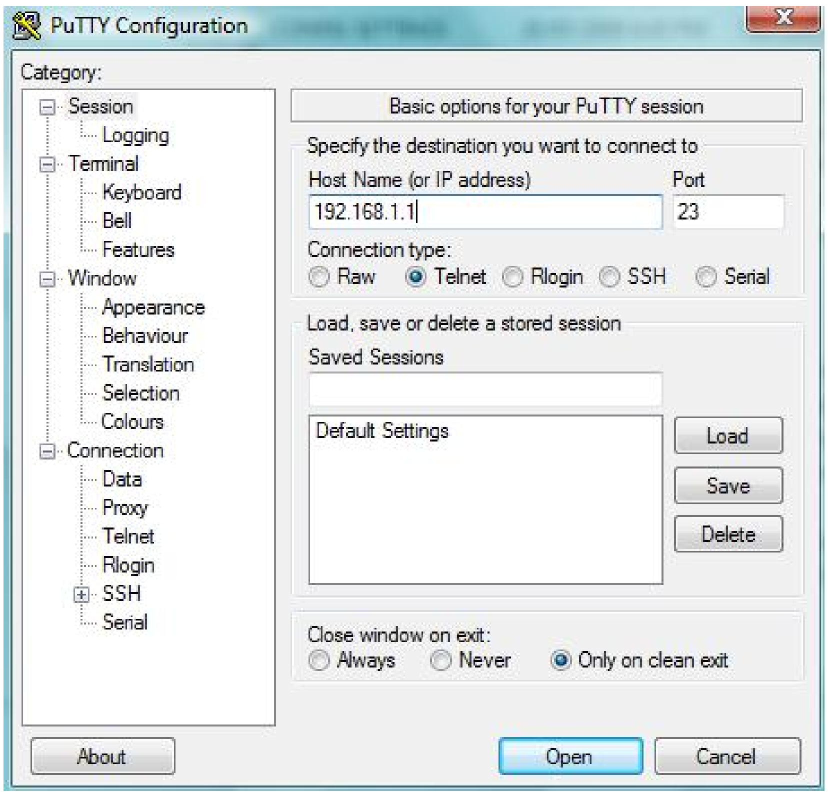
4.1.2 Gaining Access to the Command Line Interface

In order to load software and alter settings on the router, access to the command line interface must first be established. This requires a connection to the router (preferably via an RJ-45 connection) and the use of a communication protocol to allow bidirectional communication between the computer and the router. The software used during the project is called PuTTY (*PuTTY Website* 2009) and uses Telnet (Teletype network) and SSH (Secure SHell) protocols to create connections to the router.

In order to gain access to the command line interface of the router, a connection must be established using the address 192.168.1.1 and the preferred connection type. The first time that the connection is made to the router, Telnet is used and a screen similar to figure 4.2 will be displayed.

It is not necessary, but recommended to create a more secure connection to the router. In order to do this, the command *"passwd"* is entered into the command line and a password is set on the router. This also terminates connections to the router using Telnet.

Once the password has been entered into the command line, an SSH connection needs to be made. To configure PuTTY to use a SSH connection and connect to the router selecting SSH as the connection type is required (refer to figure 4.2). Upon initiating a

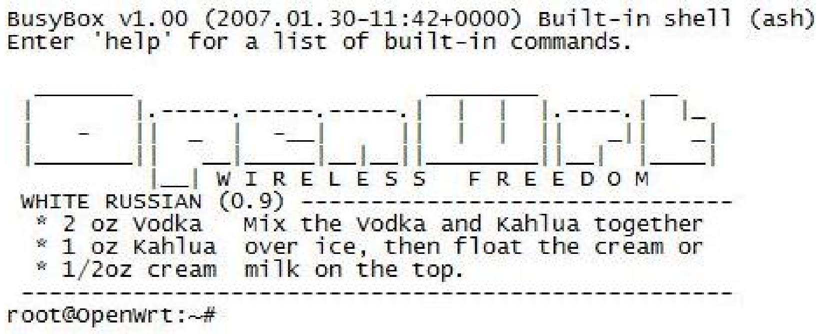


Specify the destination you want to connect to
Host Name (or IP address): 192.168.1.1
Port: 23
Connection type: Raw [✓ Telnet] Rlogin SSH Serial
Load, save or delete a stored session
Saved Sessions
Default Settings
Close window on exit: Always Never [✓ Only on clean exit]

Figure 4.2: Logging onto the WRT54GL using PuTTY with Telnet protocol

connection to the router using the SSH protocol, the username and password needs to be entered (created using *passwd* command).

A screen similar to the one shown in figure 4.3 will be seen and the configuration of the router is able to be completed from this command line interface.



```
BusyBox v1.00 (2007.01.30-11:42+0000) Built-in shell (ash)
Enter 'help' for a list of built-in commands.

  _______                     ________        __
 |       |.-----.-----.-----.|  |  |  |.----.|  |_
 |   -   ||  _  |  -__|     ||  |  |  ||   _||   _|
 |_______||   __|_____|__|__||________||__|  |____|
          |__| W I R E L E S S   F R E E D O M
 WHITE RUSSIAN (0.9) -------------------------------
  * 2 oz Vodka   Mix the Vodka and Kahlua together
  * 1 oz Kahlua  over ice, then float the cream or
  * 1/2oz cream  milk on the top.
 ---------------------------------------------------
root@OpenWrt:~#
```

Figure 4.3: Command Line Interface of Open-WRT

4.1.3 Routing Protocol Implementation

As discussed in section 3.3.3 OLSR routing protocol is being used in this project. There is an implementation for this routing protocol available as a package for Open-WRT. This implementation is named OLSRd and is available as *olsrd_0.4.10-1_mipsel.ipk* (*Official Open-WRT Website* 2009). This routing protocol can be installed onto the router using the following command (assuming access to the internet is available from the router):

- ipkg install olsrd_0.4.10−1_mipsel.ipk

After downloading and installing the package, the routing protocol needs to be configured in order to operate correctly. The following excerpt shows the changes needed in the file *"/tmp/olsrd.conf"* (default path) in order to correctly configure the server (additions include specifying the IP and netmask as well as the LAN interface).

```
...
Hna4 {
        192.168.1.120 255.255.255.248
}
...
Interface "eth1"
```

Once these changes have been made and the router has been rebooted, OLSRd should be running and if other routers (configured in the same way) are available, they should automatically connect to one another.

4.1.4 Network Configuration

Network configuration is a key aspect of the overall design of the project. There were two options that would be available for the WRT54GL and that this would affect the way in which the system would operate. When dealing with the Open-WRT firmware, the LAN section (which refers to the physical RJ45 ports on the router) and the WIFI section (which refers to the wireless interface of the router) can operate in bridged mode or can be separated. By bridging the interfaces, devices on the LAN interface interact directly with the WIFI clients. Whereas if the two interfaces remain separated, packets sent from the LAN interface are routed through to the WIFI interface (If they are destined for a client on that network).

A key difference here is that when the interfaces are bridged, LAN and WIFI are in the same subnet . This is not true for the case where the two interfaces are separate, each interface resides on different subnets. for a graphical representation of the case where the two interfaces are separated see Figure 4.4.

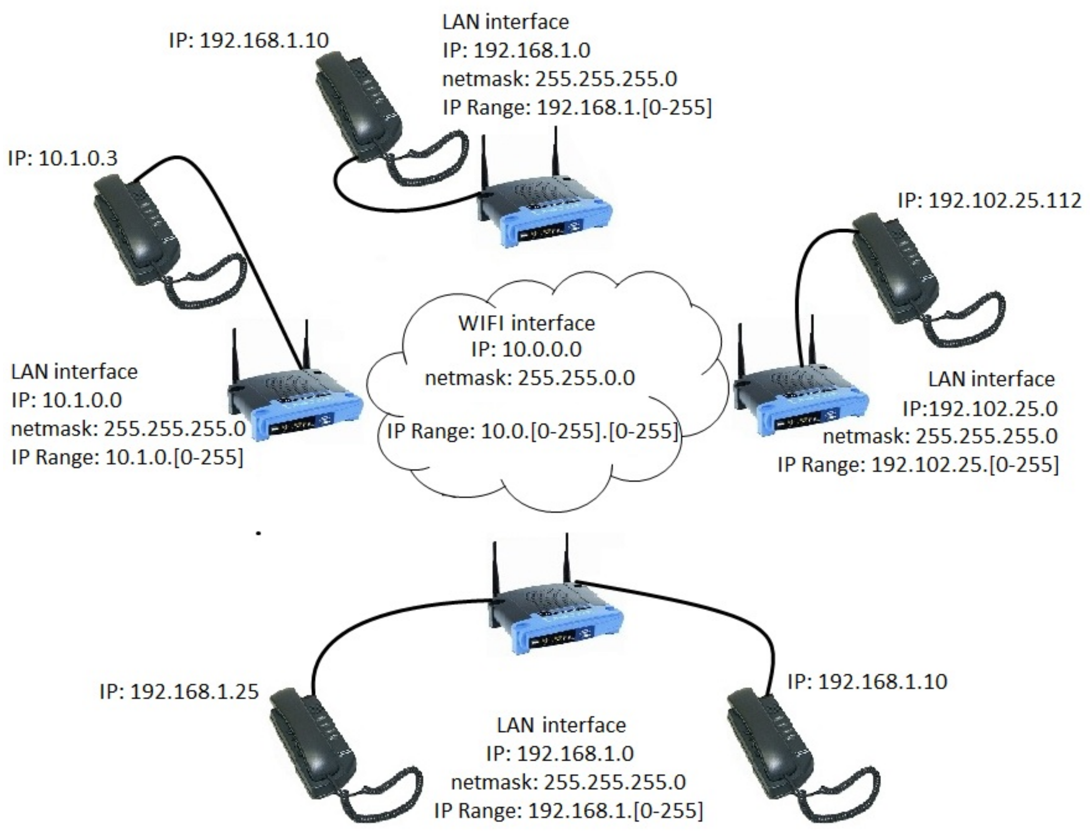


Figure 4.4: Example network setup with separate LAN and WIFI

It should be noted that when the two interfaces are not bridged, that security and routing protocols such as firewalls and Network Address Translation (NAT) traversal can cause issues. Telephone conversations **MUST** be forced to go through Asterisk servers. This is completed with the line "canreinvite=no" in sip.conf.

LAN Configuration

The LAN ports on the WRT54GL are important to correctly configure so that the operation of the WMN operates without issue. The hard and soft phones connected directly to the router through the LAN interface of the router need to be able to establish a connection to the Asterisk server that resides on the router itself.

The IP subnet can be in any range other than that of the WIFI subnet. For the purpose of this project, it will be assumed that the subnet of the LAN interface is 192.168.1.[0-255] With the routers' IP address being 192.168.1.120.

Effectively the devices are allowed 254 IP addresses and these are managed by a DHCP server (see section 4.1.4) and assuming that the connected devices are configured to run a DHCP client running (or have an available IP address in this range), they are able to automatically lease an IP address from the router.

WIFI Configuration

One of the fundamental issues of the project is the architecture of the network. This governs the way in which the various software features interact with each other and also dictates the way that the hardware is to be configured. When determining the proper structure of the network many considerations need to be taken into account. Some of these considerations include: the network topology, mobility and

Since this application requires that wireless routers form a network that may be mobile, it is important that the correct implementation of the routing protocols be achieved.

In order to allow the Wireless Meshed Network (WMN) to operate successfully each of the wireless nodes need to be transmitting on the same channel and all need to be implementing the same routing protocol. As discussed in section 3.3.3, the particular routing protocol selected for this project was an implementation of the popular OLSRd which is available as the package OLSRd.

DHCP server on LAN Interface

In order to create a DHCP server to manage the clients on the LAN interface of the router, the server must be installed and configured on the router. This server is available as a package on Open-WRT called *dnsmasq_2.35-1_mipsel.ipk* (*Official Open-WRT Website* 2009)

In order to install this package, the following command needs to be issued in the command line interface of the router:

- ipkg install dnsmasq_2.35−1_mipsel.ipk

After downloading the package and installing the package the DHCP server needs to be configured. The file *"/etc/dnsmasq.conf"* needs to be altered and the DHCP server range set.

4.2 Configuring Asterisk

The configuration of the Asterisk server is a straight forward process as the maintenance files that allow the Asterisk servers to connect to one another is handled by the Asterisk Client Information Sharing (Section 4.3). In order to load Asterisk onto the router, the following commands are needed to be issued in the command line interface of the router:

- ipkg install asterisk-mini_1.0.10-1_mipsel.ipk

Once the installation of Asterisk is complete, the router should be rebooted and the Asterisk server should be running after booting has completed.

4.2.1 SPA901 Configuration

To allow the SPA901 IP Phones to connect to the Asterisk server that is residing on the router that the phones are plugged into, some alterations need to be made. The easiest way to configure the phones is by using the web interface of the phones.

As the router should be running a DHCP server, the phone needs to be configured to connect to this server and lease an IP address from it. There are keypad commands that allow this to be done and they are as follows:

1. Pick up the handset.
2. Enter "****" into the keypad to enter the voice menu.
3. Enter "101#" into the keypad to enable / disable the DHCP client.
4. Enter "1#" to enable the DHCP client.
5. Enter "1" to confirm the change.
6. Hung up the handset to allow the changes to take effect.

After the above commands have been issued, the IP phone should be able to connect to the router (or computer that is running a DHCP server) and the web interface is able to be accessed. In the configuration page of the phone, the following settings need to be made:

- The user ID in the "Subscriber Information" on the "Ext 1" tab of the page is the phone number (extension) of the phone and a unique number needs to be set here.
- The port used for SIP is set to "5060" (default).

After completing these changes and saving them, the IP phone is ready to be used in the system.

4.3 Configuration of Asterisk Client Information Sharing

In order to allow for the design discussed in Section 3.4 two programs were written in the C programming language. Each of the programs look after different aspect of ACIS and tasks are discussed in the following sections.

4.3.1 ACIS Send

This program is responsible for three aspects of ACIS. these three functions are:

- Checking for phones attempting to connect and reloading Asterisks database when found.
- File maintenance (“sip.conf” and “extensions.conf”).
- Sending of data at a predefined time interval (defined in a configuration file).

The Figure 4.5) shows the basic operation and flow of the program. It can be seen that the program never finishes executing and is constantly checking for new phones and sending data.

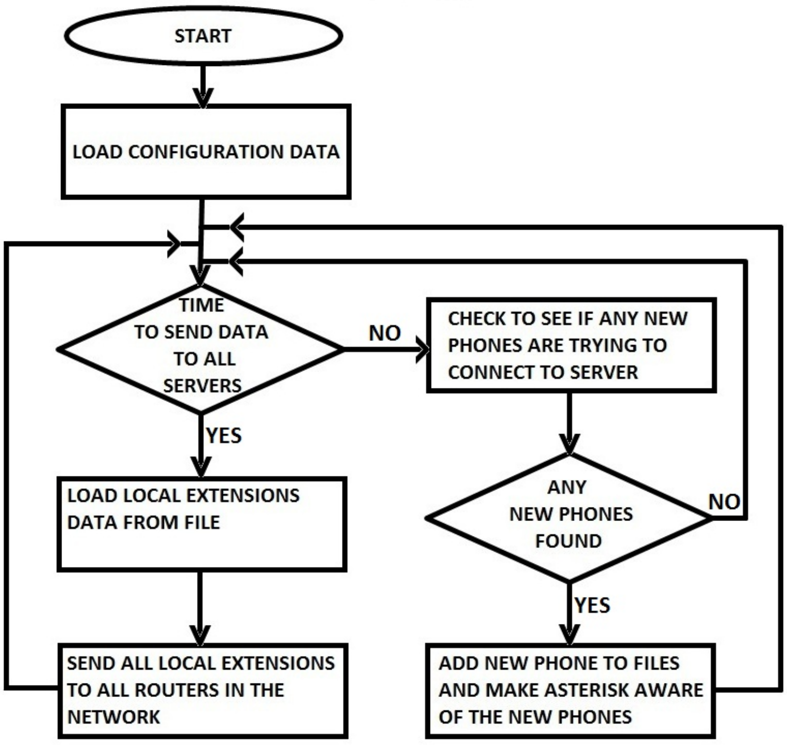

START
LOAD CONFIGURATION DATA
TIME TO SEND DATA TO ALL SERVERS
NO
CHECK TO SEE IF ANY NEW PHONES ARE TRYING TO CONNECT TO SERVER
YES
LOAD LOCAL EXTENSIONS DATA FROM FILE
ANY NEW PHONES FOUND
NO
YES
SEND ALL LOCAL EXTENSIONS TO ALL ROUTERS IN THE NETWORK
ADD NEW PHONE TO FILES AND MAKE ASTERISK AWARE OF THE NEW PHONES

Figure 4.5: Program flow diagram of ACIS Send

4.3.2 ACIS Listen

This program has two main functions. These functions relate to receiving data and then performing file manipulation. Figure 4.6 shows the flow of the program and the functions it performs.

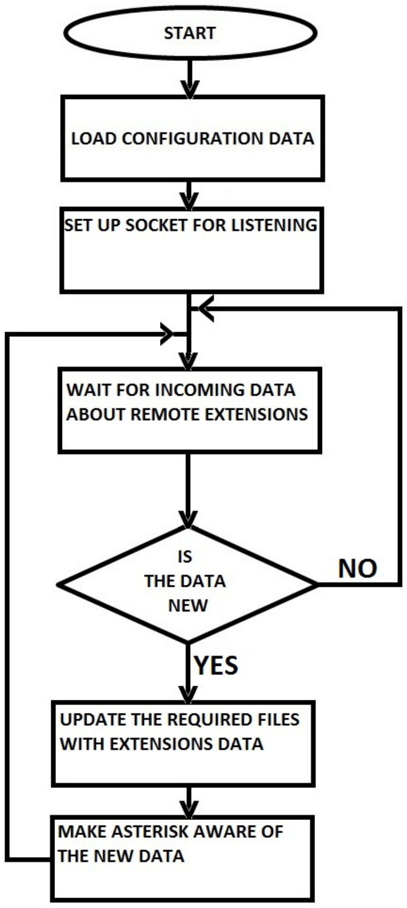

START
LOAD CONFIGURATION DATA
SET UP SOCKET FOR LISTENING
WAIT FOR INCOMING DATA ABOUT REMOTE EXTENSIONS
IS THE DATA NEW
NO
YES
UPDATE THE REQUIRED FILES WITH EXTENSIONS DATA
MAKE ASTERISK AWARE OF THE NEW DATA

Figure 4.6: Program flow diagram of ACIS Listen

As is the case in ACIS Send, it can be seen that the program never exits but runs constantly to ensure that the network is always up-to-date

The ACIS part of this system is concerned with the transfer of data between the Asterisk servers at critical times. These critical times include, when a new server boots and when a phone is connected to the system. Additionally, data will be transmitted at timed regular intervals. There are four programs running on each of the routers and they are:

This aspect of the system requires more specific instructions as compared to the other components that are discussed in this chapter. This is due to the fact that this aspect is not supported or available from Open-WRT repositories but is a custom set of programs.

Each of the parts of this aspect of the system has been coded in the C programming language and compiled in order to be compatible with the router. The source code for the two individual programs used to carry out this function are listed in appendix C

4.3.3 Preparing the Packages

In order to enable the programs to execute on the router, a process of building a package was needed to be completed. Open-WRT povides a Software Development Kit (SDK) for their firmware. This SDK was placed on a Linux computer and the programs need to be compiled from within the SDK with specialised make makefiles[1].

The steps involved to create these packages included:

- "acissend" directory is created in the OpenWrt-SDK-Linux-i686-1/package/ directory and make file (Appendix C.5) copied here.
- "src" directory is created in the OpenWrt-SDK-Linux-i686-1/package/acissend directory and make file(Appendix C.6) as well as acissend.c(Appendix C.4)copied here.

Once the files are in the correct folders, the command "make" needs to be issued from the OpenWrt-SDK-Linux-i686-1/ directory.

Once compiled, the programs need to be transferred to the router and stored on the local file system. This was achieved using the Unix cutility "scp"

```
scp acissend_1_mipsel.ipk root@192.168.1.110:
```

[1]one to set the rules for compilation in Linux environment and one to create a package suitable for installation on the WRT54GL

This process was repeated for ACIS Listen and when both packages were copied to the router, there were installed onto the router using the inbuilt “ipkg” package manager.

After copying and configuring the programs the system is ready to be tested and this will be discussed in the next chapter.

Chapter 5

Testing of the Proposed System

This chapter investigates the performance of the system. To evaluate the network multiple measures are used. These include:

- The operation of ACIS.
- Call capacity over the wireless links.
- Call quality.
- Delay between end users.
- Jitter.
- Packet loss.

To get an accurate idea of how the system performs, the above criteria are to be measured during a series of different testing conditions. These will be explained and the results provided in this chapter.

5.1 Testing Environment

The text environment is especially important for reliable testing of the proposed system. Unfortunately, the testing environment was poor and this resulted in some aspects of the system being hard to measure properly. Effort was used to gain results thats were indicative of a real deployment.

5.1.1 Layout of Testing Area

Four routers were used in the testing of the system. Each was arranged in a way that forced data to be passed from router to router. Figure 5.1 shows this arrangement and it can be seen that in order for data to be sent from router1 to router4, the data must be passed on by router2 and router3.

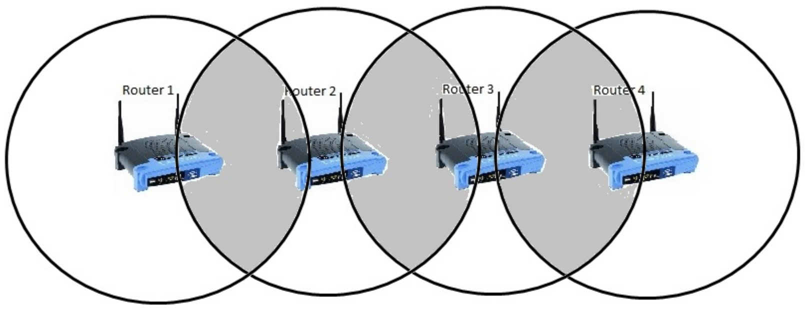


Figure 5.1: Router layout for testing

Testing conditions were not ideal and certain modifications were required. Testing was conducted in a confined space where the routers were all within transmission range of one another. To overcome this situation modifications needed to be made. There were a number of options that were considered to reproduce the situation displayed in Figure 5.1.

There were a number of solution which were identified to alleviate the problems being associated with the close proximity of the routers. Firstly, configuring the firewalls on each of the routers to only accept traffic from certain routers. Secondly, the transmission power of each router could be altered to reduce the transmission range to a small distance. Thirdly, the antennas on the routers could be modified so that the transmissions are controlled and shielded.

Ideally, configuring the firewall would have been implemented but due to time constraints this was not able to be completed. Setting the transmission power to a lower level also was not able to be completed as the configuration of the transmission power did not alter the actual transmission power of the routers. By shielding the routers' antennas, the network could be configured allowing the arrangement discussed.

5.1.2 Isolating Routers' Transmissions

In order to prevent transmissions from being received by all other routers, a crude isolation device was designed. This device consisted of a cardboard box wrapped in aluminium paper and a small wire connecting the antenna to the next. Thus controlling the distribution of the signal. This can be seen in Figure 5.2 and Figure 5.3.

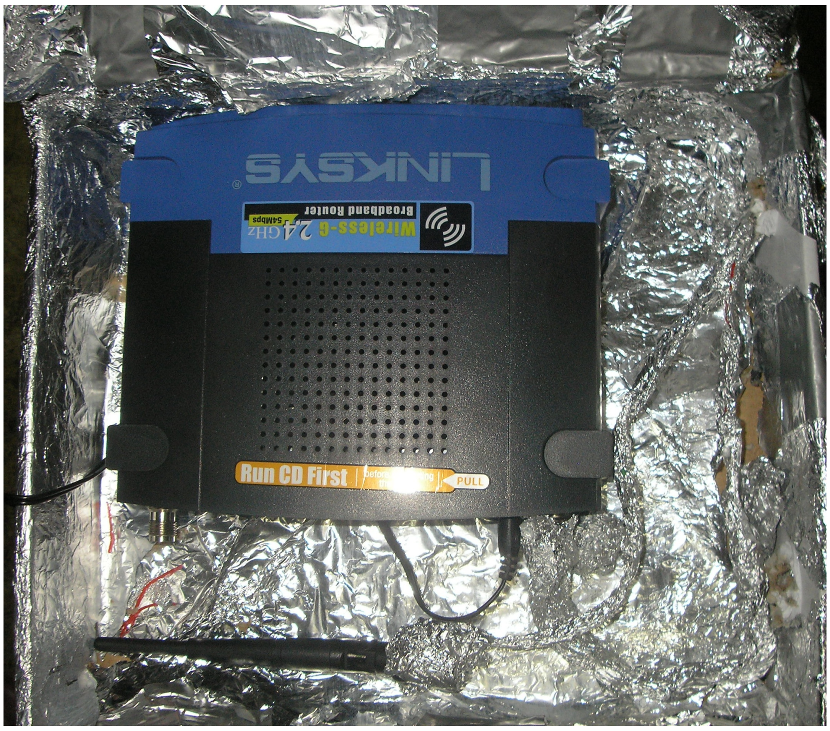


Figure 5.2: Photo showing the inside of the box housing the router

The actual link tests proved to be good and the output of the "ping" and "traceroute" commands are displayed in Appendix B. The link quality was excellent and the average round trip time was 1.7ms, 2.7ms and 3.8 for single, two and three hops respectively.

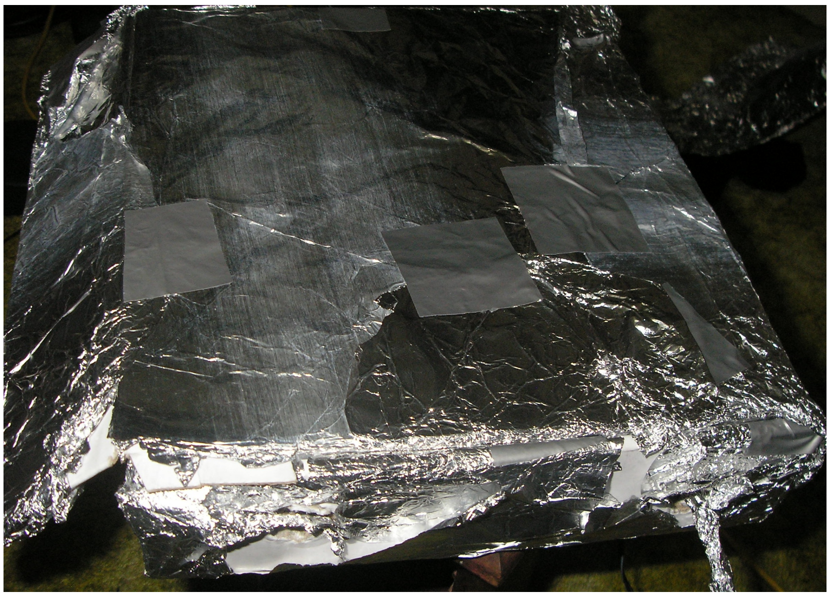


Figure 5.3: Photo showing the outside of the box housing the routers

5.2 VoIP Testing Metrics

Throughout the testing of the system, many differing methods are used to measure the performance of the system. These metrics attempt to measure the quality of wireless links, voice quality and stability. Throughout the following sections these performance indicators will be described and explained.

5.2.1 Delay

Delay is the time taken for packets to traverse from sender to receiver. For the purpose of this project, the delay will be the time taken for voice traffic to travel from the source to the destination.

The delay will be measured for a varying number of wireless hops as well as for varying traffic conditions. As a combination of hops and traffic will affect the delay, an effort to identify the limits on the system will be made.

5.2.2 Jitter

Jitter is an undesirable effect of a combination of factors. Jitter, in this application, refers to the arrival of voice packets outside of an acceptable time period.

In order for the voice to be properly reconstructed at the receiver, the packets need to arrive at a constant rate. In the case of this project (G.771 and G.729) packets are expected 50 times a second or every 2 milliseconds. When these packets are sent over IP networks, there are often variances in the times that the packets arrive.

This may be due to congestion, incorrect clock frequencies and packetisation variances. As this is a known problem there are mechanisms in place to allow for small variations and this is called a jitter buffer. This works by delaying the playing of the audio for a set time (usually 10 milliseconds) to allow for late packets.

When a packet arrives later than is allowed by the buffer, the packet is assumed lost (discarded when it arrives) and the audio is played without the information. This is obviously unacceptable as this reduces the quality of the conversation.

5.2.3 Packet Loss

Packet loss on the WMN is able to be measured with JPerf (see section 5.3.1) and reported on. It is important that packet loss is kept to a minimum for reasons of data integrity. Packet loss affects the quality of audio as well as the integrity of data sent across the network.

5.2.4 Resource Use

The resource use for this project is focused on the Central Processing Unit (CPU) and Random Access Memory (RAM) of the router. These resources are needed to be monitored in order to make sure that router does not run out of resources and affect the VoIP services.

These resources will be monitored for varying conditions of the system and reported on. To check the status of these resources, the software application TOP (see section 5.3.2) will be viewed using the command line interface of the router.

5.3 Testing Suite

In the process of testing the system, multiple software analysing tools are used. These tools and their purpose are explained in the remainder of this section.

5.3.1 JPerf

JPerf is a Graphical User Interface (GUI) for IPerf, giving a user friendly interface. IPerf is a tool to measure characteristics of network links such as bandwidth, jitter (latency variation) and packet loss. In order to test a connection, two instances of JPerf must be running at each end of the link. One of the Jperf instances runs in client mode and sends data over the network to the other JPerf program which is running in server mode. The server reports and responds back to the client (to model two way traffic). Refer to figure 5.4 for a grahpical representation.

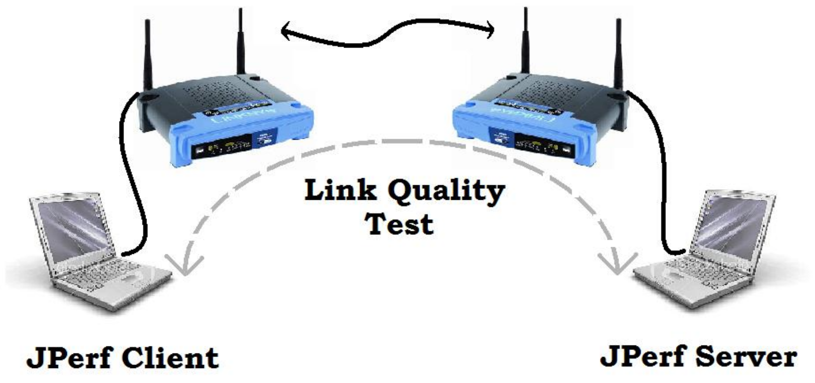


Figure 5.4: Setup of JPerf Testing

By altering the characteristics of the data sent across the network typical voice traffic can be modeled and using the output of the run the results can be interpreted. This allows for the limits in regard to the number of simultaneous two way conversations that can be achieved across the network.

In order to represent the typical traffic that needs to be sent across the network, a number of calculations need to be determined. For G.711 codec the typical payload for the packets is 160 bytes and 50 are sent a second, therefore, to determine the bandwidth for any number of two way conversations is governed by the equation:

```
Bandwidth = (number of calls) x (packets per second) x (packet payload)
```

For the G.729 codec, the payload for each of the packets (50 per second) is 20 bytes. As an example, the settings for 20 calls using the G.711 audio codec over the network are configured as follows:

- UDP Packet Size = 160 bytes
- UDP Bandwidth = (20 x 50 x 160) = 160 000 bytes/sec

To determine the capacity of the simulations each of the outputs of the JPerf program are used. The actual limits of the various outputs are listed below:

- Delays over 350 - 400 milliseconds are considered unacceptable.
- Packet loss is generally to be kept under 1% of the total packets sent.
- Jitter is to be kept within the limits of the jitter buffer.

5.3.2 TOP

TOP is a linux tool that shows the processes running on the system. It reports the Central Processing Unit (CPU) usage per program as well as the system memory usage. It is a useful tool to monitor the resource use on the router and allows for the limitations of the router to be viewed.

The application is run from the command line interface of the router and a typical output of the program can be seen below:

```
Mem: 11448K used, 2860K free, 0K shrd, 784K buff, 4380K cached
Load average: 0.00, 0.00, 0.00    (State: S=sleeping R=running, W=waiting)

  PID USER     STATUS   RSS  PPID %CPU %MEM COMMAND
  525 root     R        400   522  0.7  2.7 top
  521 root     S        588   368  0.3  4.1 dropbear
  402 root     S       1420   394  0.0  9.9 asterisk
  389 root     S       1420     1  0.0  9.9 asterisk
  401 root     S       1420   394  0.0  9.9 asterisk
  394 root     S       1420   389  0.0  9.9 asterisk
  399 root     S       1420   394  0.0  9.9 asterisk
  400 root     S       1420   394  0.0  9.9 asterisk
  450 root     S        596   368  0.0  4.1 dropbear
  440 root     S        532     1  0.0  3.7 olsrd
  451 root     S        524   450  0.0  3.6 ash
  522 root     S        440   521  0.0  3.0 ash
  420 nobody   S        424     1  0.0  2.9 dnsmasq
  368 root     S        392     1  0.0  2.7 dropbear
  315 root     S        368     1  0.0  2.5 udhcpc
  376 root     S        364     1  0.0  2.5 httpd
    1 root     S        356     0  0.0  2.4 init
   97 root     S        356     1  0.0  2.4 init
   98 root     S        348     1  0.0  2.4 syslogd
```

5.4 Results Obtained

One hop tests were carried out using JPerf and monitoring of the resource use of the router with TOP. The tests were run multiple times and the averages were taken. Both the G.711 and G.729 audio codecs were simulated using data contained in Table 2.1 in Section 2.3. These tests were recorded and placed in graph that can be seen in Figure 5.5.

As only one hop tests were conducted during the testing phase of this Thesis. This was due to time constraints. But round trip times were gathered (using "ping") and can be seen in Appendix B.

The one hop test provides the best case for the proposed system. While a good idea of the capacity of the system is gained from these results, much more testing is needed to be carried out on the call capacity of the system.

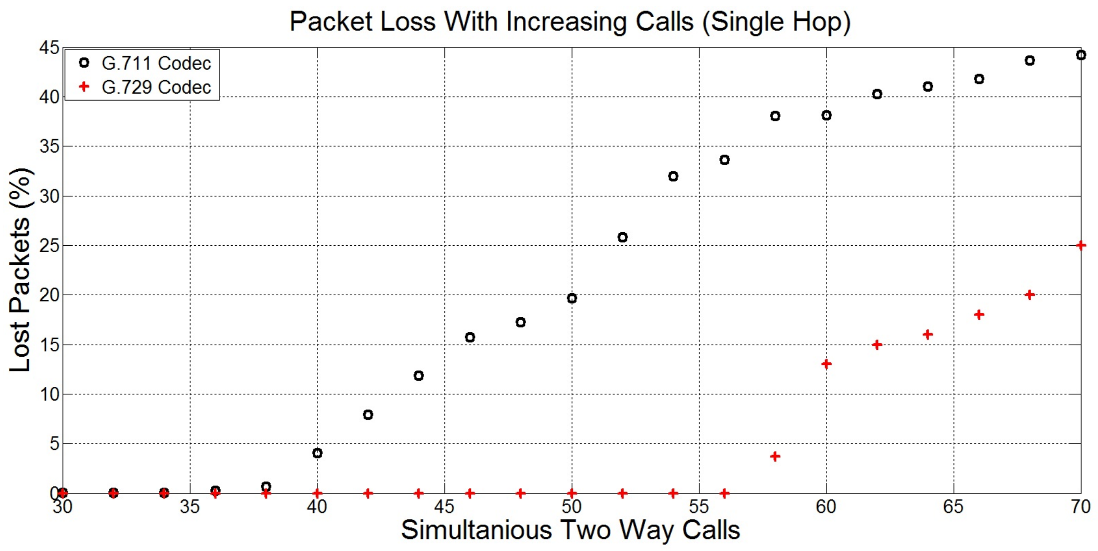


Figure 5.5: Single Hop Results

From Figure 5.5 it can be seen that the G.711 audio codec was able at attain a capacity of 36 simultaneous two way audio calls before packet loss exceeded 1%[1] of the total packets sent. Under these circumstances each of the routers had to transmit 36*64 kilo bits/s = 2304 kilo bits/s or 288 kilo Bytes/s to the remote router while the remote router had to transfer the same amount of data back. This resulted in a total transfer rate of 576 kilo Bytes/s. While this is far from the theoretical transfer rate of an 802.11g device, the sheer amount of packets transferred across the wireless network equates to 50*36*2 = 3600 packets.

The graph also shows the number of simultaneous calls achieved with the G.729 codec. This codec was able to achieve 56 simultaneous two way audio calls before the packet loss exceeded 1%. This resulted in a total transfer rate of 56*8 kilo bits/s = 448 kilo bits/s or 56 kilo Bytes/s for each way. This equates to a total transfer rate of 112 kilo

[1] 1% is generally accepted as the upper limit of acceptable losses

Bytes/s which is 38% of the total that was obtained with the G.711 codec. Taking into account the amount of packets sent across the network with the G.711 sending 3600 and the G.729 sending 5600 it can be seen that while 38% less data was sent over the network 155% more packets were sent.

Jitter is not shown in Figure 5.5 as it remained well within acceptable limits. This was expected as jitter increases with the addition of more hops. Jitter has the potential to cripple the proposed system if not carefully planned for. Jitter problems can be alleviated to a point by increasing the size of the jitter buffer. Although this has the undesirable effect of delaying the audio playback. When the buffer is increased, the audio remains in the buffer for a longer period of time.

5.5 Performance of ACIS Programs

In order to evaluate the performance of the ACIS programs, each router was placed at a distance apart that forced the creation of a chain arrangement[2]. This arrangement is illustrated in Figure 5.1.

To ensure that each of the routers were receiving the updates, all routers were simultaneously logged into via a computer using "PuTTY". Once all routers had been accessed, and were being monitered, a phone was plugged into one of the routers while the ACIS programs were executing.

Once the phone had been detected (from the Asterisk debugging file), all routers were checked to see if the appropriate files had been updated as well as the Asterisk server reloaded. To ensure that the system worked properly, the test was repeated with two phones plugged into various routers and test calls across the wireless connections were attempted.

5.5.1 Results of ACIS Tests

As expected, the program correctly added the information to the local files, reloaded Asterisk with the correct information and distributed these updates to all other routers across the network. In addition to this, at no time were the routers underload while these tasks were being completed. While running both programs simultaneously, the peak cpu usage was never found to be above 5% and the memory usage was very minimal.

Most importantly, phone calls were able to be made across the network after being detected by the ACIS system. This is the key performance criteria and it performed extremely well.

[2] the end routers were only able to communicate with one router while the middle routers were able to communicate with the two routers around them

5.5.2 Observations and Known Issues

The design of the ACIS programs is previously described in Section 3.4. In the previous section, it is stated that ACIS Send is responsible for the removal of phones that are unplugged. Due to time constraints this was not implemented into the final program. While this is unfortunate and needs to be implemented for future applications, this did not prevent the concept from being proven to work.

During the process of debugging the ACIS Send program, another issue became apparent. This problem presented itself when an Asterisk server failed to register with a remote Asterisk server. Upon failing to properly register, an entry is put into the Asterisk log file that is extremely similar to that of a failed phone registration.

To prevent this incorrect entry, a limit of eight digits for the phone extensions were put in place[3]. While all extensions used in the testing of this system were under eight digits long, practical application may need to exceed this length. Therefore, this would need to be rectified.

[3]The entry that appears in the Asterisk log when a failed server registration occurs is the IP address of the server. For example 10.0.0.10

5.6 Chapter Summary

This chapter has provided a basis for testing and has shown the results of stressing the proposed system. It can be seen that the system is able to handle a reasonable amount of telephone conversations before becoming overwhelmed.

Chapter 6

Conclusions and Further Work

The main aim of this project was to design and implement a self configuring and optimised telephone network that is able to operate over wireless channels. Research into the underlying principles of the system was conducted and the design and implementation of these components was carried out. The results obtained as part of the testing of the system are provided in Chapter 5 are discussed in further detail within this chapter.

Also further work that would only build upon the system created in this paper are presented towards the end of this chapter. For this system to reach its' full potential, the additions mentioned in this chapter need to be implemented in the future work done in this area.

6.1 Achievement of Project Objectives

The Project Specification (Appendix A) lists the work that was to be completed as part of this Thesis. With the exception of the non-overlapping numbering system (part of Point 4 in the list) that the core programme units have been completed. The non-overlapping number scheme was given a lot of consideration but proved too difficult without the having a centralised management system for the WMN.

Additionally, some aspects of point 7 5.4 were completed. Point 8 is also mentioned in Section 3.5 and a good design basis provided.

6.2 Discussion of Results

The results obtained as part of the testing of the system are promising. This Thesis has provided a good basis for further work to be done in this field. Currently, the hardware is able to handle creating and maintaining a small to medium sized telephone network residing entirely on cheap hardware and freely available software components.

The key limitation of the system produced during the course of the project pertain to the bandwidth available in wireless networks. With the limited testing it is already evident that with more hops introduced that the number of simultaneous will fall (Possibly dramatically). With wireless technology constantly improving[1] and new methods for increasing bandwidth constantly being researched, it is envisioned that systems similar to the system proposed here will be implemented.

The shift towards cheaply available, high bandwidth, low latency wireless networks are certain to gain momentum in the future. These developments will soon enable the creation of larger, more reliable telephone networks to provided in Wireless Mesh Networks.

[1]The 802.11n draft is currently in use and boast speeds up to twice that of the 802.11g standard used in this Thesis

6.3 Further Work

In order for this project to be deployed in a real situation there is much more work needed. While the systems is extremely promising it falls short of being a complete system. Some of the work that needs to be done can be seen in the following sections.

6.3.1 Quality of Service

Quality of Service (QoS) mechanisms would allow for the prioritising of voice data over the network. While this would not have affected the results obtained in this paper, in real applications there would be more traffic on the network such as Internet use and file transfers. If the voice traffic was not allowed priority over data packets, the quality of the conversations would reduce rapidly.

6.3.2 Use of Compressed Header Information

The use of header compression would be of great importance. This could reduce the overhead on all voice data on the network from around 58bytes down to 24bytes on each packet. This has obvious benefits even if only reducing the bandwidth used.

6.3.3 Added Functionality

This project has primarily focused on allowing two way telephone conversations to be created over the network. This allows a basic telephone network to be established but lacks some key features. Some of the most popular features are:

- Message banks[2].
- Call conferencing.
- Call waiting.

[2]This has potential to be completed with the addition of an SD memory card. Refer to Section 2.8.1

While these features are not included in this project, they would almost certainly be needed in a real application. All of these features - and more - are able to be provided by Asterisk, and with some modifications to aspects of this project, could be incorporated into the system.

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Appendix A

Project Specification

Place your project specification here.

Appendix B

WMN Muliple Hop Tests

This is the raw output of the “ping” and “traceroute” commands. The output shows the round trip time and the hops needed to get to the destination. These tests were completed from within one of the routers (10.0.0.40) and the results show increasing lengths of time with the addition of more hops.

Loop back test

```
root@OpenWrt:~# ping 10.0.0.40
PING 10.0.0.40 (10.0.0.40): 56 data bytes
64 bytes from 10.0.0.40: icmp_seq=0 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=1 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=2 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=3 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=4 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=5 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=6 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=7 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=8 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=9 ttl=64 time=0.7 ms
64 bytes from 10.0.0.40: icmp_seq=10 ttl=64 time=0.7 ms

--- 10.0.0.40 ping statistics ---
11 packets transmitted, 11 packets received, 0% packet loss
round-trip min/avg/max = 0.7/0.7/0.7 ms
```

Single hop test

```
root@OpenWrt:~# ping 10.0.0.30
PING 10.0.0.30 (10.0.0.30): 56 data bytes
64 bytes from 10.0.0.30: icmp_seq=0 ttl=64 time=3.8 ms
64 bytes from 10.0.0.30: icmp_seq=1 ttl=64 time=1.5 ms
64 bytes from 10.0.0.30: icmp_seq=2 ttl=64 time=1.5 ms
64 bytes from 10.0.0.30: icmp_seq=3 ttl=64 time=1.4 ms
```

```
64 bytes from 10.0.0.30: icmp_seq=4 ttl=64 time=1.5 ms
64 bytes from 10.0.0.30: icmp_seq=5 ttl=64 time=1.5 ms
64 bytes from 10.0.0.30: icmp_seq=6 ttl=64 time=1.4 ms
64 bytes from 10.0.0.30: icmp_seq=7 ttl=64 time=1.5 ms
64 bytes from 10.0.0.30: icmp_seq=8 ttl=64 time=1.5 ms
64 bytes from 10.0.0.30: icmp_seq=9 ttl=64 time=1.4 ms
64 bytes from 10.0.0.30: icmp_seq=10 ttl=64 time=2.5 ms

--- 10.0.0.30 ping statistics ---
11 packets transmitted, 11 packets received, 0% packet loss
round-trip min/avg/max = 1.4/1.7/3.8 ms

root@OpenWrt:~# traceroute 10.0.0.30
traceroute to 10.0.0.30 (10.0.0.30), 30 hops max, 40 byte packets
 1  10.0.0.30 (10.0.0.30)  4.38 ms  3.352 ms  3.652 ms
```

Two hop test

```
root@OpenWrt:~# ping 10.0.0.20
PING 10.0.0.20 (10.0.0.20): 56 data bytes
64 bytes from 10.0.0.20: icmp_seq=0 ttl=63 time=4.5 ms
64 bytes from 10.0.0.20: icmp_seq=1 ttl=63 time=2.7 ms
64 bytes from 10.0.0.20: icmp_seq=2 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=3 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=4 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=5 ttl=63 time=2.6 ms
64 bytes from 10.0.0.20: icmp_seq=6 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=7 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=8 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=9 ttl=63 time=2.5 ms
64 bytes from 10.0.0.20: icmp_seq=10 ttl=63 time=2.5 ms
```

```
--- 10.0.0.20 ping statistics ---
11 packets transmitted, 11 packets received, 0% packet loss
round-trip min/avg/max = 2.5/2.7/4.5 ms

root@OpenWrt:~# traceroute 10.0.0.20
traceroute to 10.0.0.20 (10.0.0.20), 30 hops max, 40 byte packets
 1  10.0.0.30 (10.0.0.30)  3.979 ms  4.161 ms  3.572 ms
 2  10.0.0.20 (10.0.0.20)  4.791 ms  3.648 ms  2.318 ms
```

Three hop test

```
root@OpenWrt:~# ping 10.0.0.10
PING 10.0.0.10 (10.0.0.10): 56 data bytes
64 bytes from 10.0.0.10: icmp_seq=0 ttl=62 time=3.8 ms
64 bytes from 10.0.0.10: icmp_seq=1 ttl=62 time=3.6 ms
64 bytes from 10.0.0.10: icmp_seq=2 ttl=62 time=3.5 ms
64 bytes from 10.0.0.10: icmp_seq=3 ttl=62 time=3.6 ms
64 bytes from 10.0.0.10: icmp_seq=4 ttl=62 time=3.6 ms
64 bytes from 10.0.0.10: icmp_seq=5 ttl=62 time=3.8 ms
64 bytes from 10.0.0.10: icmp_seq=6 ttl=62 time=3.5 ms
64 bytes from 10.0.0.10: icmp_seq=7 ttl=62 time=5.4 ms
64 bytes from 10.0.0.10: icmp_seq=8 ttl=62 time=4.2 ms
64 bytes from 10.0.0.10: icmp_seq=9 ttl=62 time=3.5 ms
64 bytes from 10.0.0.10: icmp_seq=10 ttl=62 time=4.1 ms

--- 10.0.0.10 ping statistics ---
11 packets transmitted, 11 packets received, 0% packet loss
round-trip min/avg/max = 3.5/3.8/5.4 ms

root@OpenWrt:~# traceroute 10.0.0.10
traceroute to 10.0.0.10 (10.0.0.10), 30 hops max, 40 byte packets
 1  10.0.0.30 (10.0.0.30)  1.734 ms  1.337 ms  1.309 ms
```

```
2  10.0.0.20 (10.0.0.20)  2.503 ms  3.771 ms  3.494 ms
3  10.0.0.10 (10.0.0.10)  4.018 ms  3.506 ms  3.824 ms
```

Appendix C

Asterisk Client Information Sharing Source Code

C.1 The acislisten.c C Code

```c
/*****************************************************
C CODE for acislisten (Asterisk Client Information
Sharing - listen)

acislisten is a program that has been developed to wait
endlessly for data about remote Asterisk servers. Once
data has been received, this data is processed to
determine if the information is new.

If the data received contains information about a new
Asterisk server, the data is added to the local
Asterisk server and the Asterisk server is reloaded
to enable calls to the remote Asterisk server.

This program is written to specifically for the
WRT54GL (ver 1.1) wireless router running Open-WRT
(White Russian v0.9) Firmware.

Author: Adam Jones
Date: October 2009
*****************************************************/
#include <sys/socket.h>
#include <sys/types.h>
#include <netinet/in.h>
#include <netdb.h>
#include <stdio.h>
#include <string.h>
#include <stdlib.h>
#include <unistd.h>
#include <errno.h>
#include <arpa/inet.h>
#include <limits.h>
#include <stdbool.h>

/*****************************************************
This structure is to hold virtual files
in memory as well as the extensions that
are to be stored into the files, "sip.conf"
and "extensions.conf".
*****************************************************/
struct list
{
    char str[128];
    struct list *next;
};

typedef struct list LST;
typedef LST *LIST;
/*****************************************************/

// Function Prototypes
LIST ReadFileToMem(char *);
LIST SearchList(LIST , char *);
void DeleteList(LIST);
void AddToList(LIST*, char *);
void AddEXTDataLocal(LIST, char *, char *);
void AddSIPDataLocal(LIST, char *, char *);
void SetLocalSettings();
void WaitForInfo();
void PrintDataToFile(LIST, char *);
void ProcessData(char *, char *);

// Global Variables
char LOCALIP[16];
char CODEC[20];
char EXT[50];
char MESSAGES[50];
char SIP[50];

/****************************************************
The main function is responsible for
retrieving the contents of the global
variables and also for starting the
waiting process.
INPUT:
        (none)
OUTPUT:
        (none)
****************************************************/
int main()
{
        SetLocalSettings();
        WaitForInfo();
}
/****************************************************/

/****************************************************
This function creates a socket for
listening. Upon receiving a data item
the function "ProcessData()" is called
and the data is processed and added to
the appropriate files. This function
will loop indefinately waiting for
information.
INPUT:
        (none)
OUTPUT:
        (none)
****************************************************/
void WaitForInfo()
{
        // Local variable declaration
        char buf[40];
        int sock, length, fromlen, n, received=0;
        struct sockaddr_in server;
        struct sockaddr_in from;

        // Get socket for listening
        sock=socket(AF_INET, SOCK_DGRAM, 0);
        if (sock<0)     error("Opening socket");

        length = sizeof(server);
        bzero(&server,length);
        fromlen = sizeof(struct sockaddr_in);

        // Set the paramters of the socket.
        server.sin_family=AF_INET;
        server.sin_addr.s_addr=INADDR_ANY;
        server.sin_port=htons(32130);
        // Atempt to bind the socket to the port
        if (bind(sock,(struct sockaddr *)&server,length)<0){
        error("binding");
        }

        // Loop forever to receive updates
        while(1)
        {
                // Wait for incoming packet
                n = recvfrom(sock,buf,sizeof(buf),0,
                        (struct sockaddr *)&from,&fromlen);
                if (n < 0)      error("recvfrom");

                // Once a packet has been received, send the
                // information contained in the packet as
                // well as the IP address of the sender of
                // the packet to function "ProcessData()".
                ProcessData(buf,
                        (char *)inet_ntoa(from.sin_addr));
        }
}
/****************************************************/

/****************************************************
This function checks that the information
does not already exist in the files
"extensions.conf" and "sip.conf". If the
information is new, it is added to the file/s.
INPUT:
        buf -   String that contains an extension for
                a telephone on the network.

        IP -    String that is the IP address of the
                sender of 'buf'. This is used to add
                the phone to the Asterisk network
OUTPUT:
        (none)
****************************************************/
void ProcessData(char *buf, char *IP)
{
        // Local variable declarations
        char newstr[40], tempstr[40];
        LIST sipptr = malloc(sizeof(LST));
        LIST extptr = malloc(sizeof(LST));

        // Put the extension into a format that represents
        // how is appears in the file "sip.conf"
        sprintf(tempstr,"[%s]",IP);

        // Read "sip.conf into memory
        sipptr=ReadFileToMem(SIP);

        // Check to see if the extension already exists
        // in "sip.conf"
        if(SearchList(sipptr,tempstr)==NULL)
        {
                // If the extension doesn't exist
                // already, add it
                AddSIPDataLocal(sipptr,IP,buf);

                // Save to updated "sip.conf" back to disk
                PrintDataToFile(sipptr, SIP);

                // system command to tell asterisk
                // to reload "sip.conf"
                system("asterisk -rx 'sip reload'");

        }
        // Read "extensions.conf" into memory
        extptr=ReadFileToMem(EXT);

        // Put the extension into a format that
        // represents how is appears in the file
        // "extensions.conf"
        sprintf(tempstr,">%s,",buf);

        // Check to see if the extension already
        // exists in "extensions.conf"
        if(SearchList(extptr,tempstr)==NULL)
        {
                // If the extension doesn't exist
                // already, add it
                AddEXTDataLocal(extptr,IP,buf);

                // Save to updated "extensions.conf"
                // back to disk
                PrintDataToFile(extptr, EXT);

                // system command to tell asterisk to
                // reload "extensions.conf"
                system("asterisk -rx 'extensions reload'");

        }
        // Free the memory that holds "extensions.conf"
        // and "sip.conf"
        DeleteList(sipptr);
        DeleteList(extptr);
}
/*****************************************************/

/*****************************************************
This function read a file and stores it
into memory.
INPUT:
        String -         contains the filename to be
                         read from.
OUTPUT:
        LIST pointer -   contains the address to the
                                 first element of the file
                                 (in memory)
*****************************************************/
LIST ReadFileToMem(char *fn)
{
        // Local variable declarations
        FILE *fileptr;
        LIST p = NULL;
        char line[LINE_MAX];
        int length;
        // Open the file for reading
        fileptr=fopen(fn,"r");
    if (!fileptr) {error("Opening File");}

        // Loop to get the contents of the file line-by-line
        // and store the contents into a list
        while(fgets(line, sizeof(line), fileptr) != NULL )
        {
                // If the line ends with a new line, change it
                // to a null terminator
                length = strlen(line) - 1;
                if(line[length] == '\n') {line[length] = 0;}

                // Add the line to the list
                AddToList(&p, line);
        }
        // Close the file
        fclose(fileptr);

        // Return the pointer to the start of the list.
        return p;
}
/***********************************************/

/***********************************************
This function searches for a match
between a list and a string and returns
the matching element.
INPUT:
        LIST pointer -  Contains a pointer to a list
                        that is to be searched for a
                        match.
        string -        The string that is to be
                        searched for within the list.
OUTPUT:
        LIST pointer -  Upon finding the string within
                        the list, the pointer to that
                        element is returned. If the
                        string is not found, NULL is
                        returned by the function.
***********************************************/
LIST SearchList(LIST ptr, char *searchstr)
{
        while(ptr!=NULL)
        {
                if(strstr(ptr->str,searchstr))
                        return ptr;
                ptr = ptr->next;
        }
        return NULL;
}
/***********************************************/

/***********************************************
This function deltes the contents of the
list that is passed to it from the point
of the element that is passed to it.
INPUT:
        LIST pointer -  Contains a pointer to a list
```

```
                                   that is to be destroyed.
OUTPUT:
        (none)
*****************************************************/
void DeleteList(LIST ptr)
{
        while(ptr!=NULL)
        {
                LIST next = ptr->next;
                free(ptr);
                ptr = next;
        }
}
/*****************************************************/

/*****************************************************
This function adds an entry to the end of a list.
INPUT:
        LIST pointer -  This is a pointer to a pointer
                                that contains a list.
        String -        This string contains the
                        information that is to be added
                        to the end of the list.
OUTPUT:
        (none)
*****************************************************/
void AddToList(LIST* pptr, char *str)
{
        // Create a pointer that will contina the address
        // of the list that is to be added to.
        LIST ptr = *pptr;
        LIST p = ptr;

        // If the list not empty.
        if(p)
        {
                // Go to the end of the list.
                while(p->next)
                {
                        p=p->next;
                }
                // Create a new element and move to it.
                p->next = malloc(sizeof(LST));
                p=p->next;
        }
        // If the list is empty.
        else
        {
                // Create the first element.
                p = malloc(sizeof(LST));
                *pptr = p;
        }
        // Set the next pointer to point nowhere (the end
        // of list)
        p->next = NULL;

        // Copy the string (passed into function) to the
        // element.
        strcpy(p->str, str);
```

```
}
/**************************************************/

/**************************************************
This function adds a extension to the
"extensions.conf" file (held in memory).
INPUT:
        LIST pointer -  This pointer points to a list
                                that contains "extensions.conf"
        String -        This string contains the
                        extension that is to be added
                        to "extensions.conf"
OUTPUT:
        (none)
**************************************************/
void AddEXTDataLocal(LIST ptr, char *IP, char *str)
{
        // Local variable declarations.
        LIST pstart, tempptr, newentry=malloc(sizeof(LST));
        char tempstr[60];

        // Find the line that holds "[remote]'
        ptr=SearchList(ptr, "[remote]");

        // Record the start position of the new entries.
        pstart=ptr;

        // Set the pointers so that tmpstr is
        // inserted 'here'.
        newentry->next = ptr->next;
        ptr->next = newentry;
        ptr = ptr->next;

        // Copy the string into the newly allocated space.
        sprintf(tempstr,"exten=>%s,1,NoOp()", str);
        strcpy(newentry->str, tempstr);

        // Set the pointers so that tmpstr is
        // inserted 'here'.
        newentry = malloc(sizeof(LST));
        newentry->next = ptr->next;
        ptr->next = newentry;
        ptr = ptr->next;

        // Copy the string into the newly allocated space.
        sprintf(tempstr,"exten=>%s,2,Dial(SIP/%s/${EXTEN})",
                        str, IP);
        strcpy(newentry->str, tempstr);

        // Set the pointers so that tmpstr is
        // inserted 'here'.
        newentry = malloc(sizeof(LST));
        newentry->next = ptr->next;
        ptr->next = newentry;
        ptr = ptr->next;

        // Copy the string into the newly allocated space.
        sprintf(tempstr,"exten=>%s,3,HangUp()\n", str);
        strcpy(newentry->str, tempstr);

        newentry = malloc(sizeof(LST));
        // Find the line that holds "[remote]'
        while(!strstr(ptr->next->str, "[phones]"))
        {
                ptr=ptr->next;
        }

        // Create a string to test for the Asterisk
        // servers presence.
        sprintf(tempstr,"[%s_incoming]", IP);
        // If there is not already an entry for this
        //asterisk server.
        if (SearchList(pstart,tempstr)==NULL)
        {
                // Add information 'here' for the server.
                tempptr = malloc(sizeof(LST));
                tempptr->next = ptr->next;
                sprintf(tempstr,"[%s_incoming]", IP);
                strcpy(tempptr->str, tempstr);
                ptr->next = tempptr;
                ptr = ptr->next;

                tempptr = malloc(sizeof(LST));
                tempptr->next = ptr->next;
                sprintf(tempstr,"include=>internal\n");
                strcpy(tempptr->str, tempstr);
                ptr->next = tempptr;
                ptr = ptr->next;
        }
}
/*************************************************/

/*************************************************
This function adds a extension to the "sip.conf"
file (held in memory).
INPUT:
        LIST pointer -  This pointer points to a list
                                that contains "sip.conf"
        String -        This string contains the
                        extension that is to be added
                        to "sip.conf"
OUTPUT:
        (none)
*************************************************/
void AddSIPDataLocal(LIST ptr, char *IP, char *str)
{
        // Local variable declarations.
        char tempstr[60];
        LIST newentry, startptr=ptr;
        bool alreadyExists=false;

        // Find the position of register
        while(ptr=SearchList(ptr, "register"))
        {
                // If the IP address is registered.
                if(strstr(ptr->str,IP))
                {
                        // Record that the registration has already
                        // been entered
                        alreadyExists=true;
                }
                // Move to the next list item.
                ptr=ptr->next;
        }
        // If there is no registration information in
        // the file.
        if (!alreadyExists)
        {
                // Add registration details.
                newentry = malloc(sizeof(LST));
                sprintf(tempstr,"register=>%s:welcome@%s/%s",
                        LOCALIP, IP, IP);

                strcpy(newentry->str, tempstr);
                newentry->next = ptr->next;
                ptr->next = newentry;
        }
        // Add entries to the end of the file.
        sprintf(tempstr,"\n[%s]", IP);
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"type=friend");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"secret=welcome");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"context=%s_incoming", IP);
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"host=dynamic");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"disallow=all");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"allow=%s",CODEC);
        AddToList(&startptr, tempstr);
}
/***************************************************/

/***************************************************
This function loads global variables
from a configuration file that is
stored in the router in the directory /bin/
INPUT:
        (none)
OUTPUT:
        (none)
***************************************************/
void SetLocalSettings()
{
        // Local variable declarations
        FILE *fp;

        // Open the file for reading
        fp=fopen("/bin/LOCALIP.ROUTER","r");
    if (!fp)
        error("Opening File");

        // Obtain the IP address of the wireless interface
        // of the router (used for unique name for
        // Asterisk server).
        fgets(LOCALIP, sizeof(LOCALIP), fp);
        int len = strlen(LOCALIP) - 1;
        if(LOCALIP[len] == '\n')
                LOCALIP[len] = 0;

        // Obtain the codec that is being allowed to be
        // used by the Asterisk server (Used in "sip.conf")
        fgets(CODEC, sizeof(CODEC), fp);
        len = strlen(CODEC) - 1;
        if(CODEC[len] == '\n')
                CODEC[len] = 0;

        // Location of "sip.conf" (for fopen() calls)
        fgets(SIP, sizeof(SIP), fp);
        len = strlen(SIP) - 1;
        if(SIP[len] == '\n')
                SIP[len] = 0;

        // Location of "extensions.conf" (for fopen() calls)
        fgets(EXT, sizeof(EXT), fp);
        len = strlen(EXT) - 1;
        if(EXT[len] == '\n')
                EXT[len] = 0;

        // Location of Asterisk log file (for finding
        // new phones)
        fgets(MESSAGES, sizeof(MESSAGES), fp);
        len = strlen(MESSAGES) - 1;
        if(MESSAGES[len] == '\n')
                MESSAGES[len] = 0;

        // Close the file.
        fclose(fp);
}
/**************************************************/

/**************************************************
This function stores the conents of the
lists in memory back to file.

INPUT:
        list pointer -  List to be saved to file
        String -        Name of the file that the
                        memory contents are to be
                        written to.
OUTPUT:
        (none)
**************************************************/
void PrintDataToFile(LIST ptr, char *filename)
{
        // Local variable declaration
        FILE *fp;

        // Open the file for writing
        fp=fopen(filename,"w");
    if (!fp)
        error("Opening File");

        // Loop through the list, printing each string in
        // the list as a new line in the file.
        while(ptr != NULL) {
                fprintf(fp,"%s\n",ptr->str);
                ptr = ptr->next;
        }
        // Close the file.
        fclose(fp);
}
/*****************************************************/
```

C.2 The `makefile` for Open-WRT SDK

Make file for acislisten.c residing in the OpenWrt-SDK-Linux-i686-1/package/ directory.

```
include $(TOPDIR)/rules.mk

PKG_NAME:=acislisten
PKG_RELEASE:=1

PKG_BUILD_DIR:=$(BUILD_DIR)/$(PKG_NAME)

include $(INCLUDE_DIR)/package.mk

define Package/acislisten
        SECTION:=utils
        CATEGORY:=Utilities
        TITLE:=acislisten --- Receives large update
        DESCRIPTION:=\
        Gets a list of all IP adresses \\\
        and the extensions associated \\\
        with them and records that data.
endef

define Build/Prepare
        mkdir -p $(PKG_BUILD_DIR)
        $(CP) ./src/* $(PKG_BUILD_DIR)/
endef

define Package/acislisten/install
        $(INSTALL_DIR) $(1)/bin
        $(INSTALL_BIN) $(PKG_BUILD_DIR)/acislisten $(1)/bin
endef

$(eval $(call BuildPackage,acislisten))
```

C.3 The `makefile` for Linux

Make file for acissend.c residing in the OpenWrt-SDK-Linux-i686-1/package/acislisten directory.

```
acislisten: acislisten.o
        $(CC) $(LDFLAGS) acislisten.o -o acislisten
acislisten.o: acislisten.c
        $(CC) $(CFLAGS) -c acislisten.c

# remove object files and executable
clean:
        rm *.o acislisten
```

C.4 The acissend.c C Code

```
/*****************************************************
C CODE for acislisten (Asterisk Client Information
Sharing - listen)

acislisten is a program that has been developed to wait
endlessly for data about remote Asterisk servers. Once
data has been received, this data is processed to
determine if the information is new.

If the data received contains information about a new
Asterisk server, the data is added to the local
Asterisk server and the Asterisk server is reloaded
to enable calls to the remote Asterisk server.

This program is written to specifically for the
WRT54GL (ver 1.1) wireless router running Open-WRT
(White Russian v0.9) Firmware.

Author: Adam Jones
Date: October 2009
*****************************************************/
#include <sys/socket.h>
#include <sys/types.h>
#include <netinet/in.h>
#include <netdb.h>
#include <stdio.h>
#include <string.h>
#include <stdlib.h>
#include <unistd.h>
#include <errno.h>
#include <arpa/inet.h>
#include <limits.h>
#include <stdbool.h>

/*****************************************************
This structure is to hold virtual files
in memory as well as the extensions that
are to be stored into the files, "sip.conf"
and "extensions.conf".
*****************************************************/
struct list
{
    char str[128];
    struct list *next;
};

typedef struct list LST;
typedef LST *LIST;
/*****************************************************/

// Function Prototypes
LIST ReadFileToMem(char *);
LIST SearchList(LIST , char *);
void DeleteList(LIST);
void AddToList(LIST*, char *);
void AddEXTDataLocal(LIST, char *, char *);
void AddSIPDataLocal(LIST, char *, char *);
void SetLocalSettings();
void WaitForInfo();
void PrintDataToFile(LIST, char *);
void ProcessData(char *, char *);

// Global Variables
char LOCALIP[16];
char CODEC[20];
char EXT[50];
char MESSAGES[50];
char SIP[50];

/***************************************************
The main function is responsible for
retrieving the contents of the global
variables and also for starting the
waiting process.
INPUT:
        (none)
OUTPUT:
        (none)
***************************************************/
int main()
{
        SetLocalSettings();
        WaitForInfo();
}
/***************************************************/

/***************************************************
This function creates a socket for
listening. Upon receiving a data item
the function "ProcessData()" is called
and the data is processed and added to
the appropriate files. This function
will loop indefinately waiting for
information.
INPUT:
        (none)
OUTPUT:
        (none)
***************************************************/
void WaitForInfo()
{
        // Local variable declaration
        char buf[40];
        int sock, length, fromlen, n, received=0;
        struct sockaddr_in server;
        struct sockaddr_in from;

        // Get socket for listening
        sock=socket(AF_INET, SOCK_DGRAM, 0);
        if (sock<0)     error("Opening socket");

        length = sizeof(server);
        bzero(&server, length);
        fromlen = sizeof(struct sockaddr_in);

        // Set the paramters of the socket.
        server.sin_family=AF_INET;
        server.sin_addr.s_addr=INADDR_ANY;
        server.sin_port=htons(32130);
        // Atempt to bind the socket to the port
        if (bind(sock,(struct sockaddr *)&server,length)<0){
        error("binding");
        }

        // Loop forever to receive updates
        while(1)
        {
                // Wait for incoming packet
                n = recvfrom(sock,buf,sizeof(buf),0,
                        (struct sockaddr *)&from,&fromlen);
                if (n < 0)      error("recvfrom");

                // Once a packet has been received, send the
                // information contained in the packet as
                // well as the IP address of the sender of
                // the packet to function "ProcessData()".
                ProcessData(buf,
                        (char *)inet_ntoa(from.sin_addr));
        }
}
/**************************************************/

/**************************************************
This function checks that the information
does not already exist in the files
"extensions.conf" and "sip.conf". If the
information is new, it is added to the file/s.
INPUT:
        buf -   String that contains an extension for
                a telephone on the network.

        IP -    String that is the IP address of the
                sender of 'buf'. This is used to add
                the phone to the Asterisk network
OUTPUT:
        (none)
**************************************************/
void ProcessData(char *buf, char *IP)
{
        // Local variable declarations
        char newstr[40], tempstr[40];
        LIST sipptr = malloc(sizeof(LST));
        LIST extptr = malloc(sizeof(LST));

        // Put the extension into a format that represents
        // how is appears in the file "sip.conf"
        sprintf(tempstr,"[%s]",IP);

        // Read "sip.conf into memory
        sipptr=ReadFileToMem(SIP);

        // Check to see if the extension already exists
        // in "sip.conf"
        if(SearchList(sipptr,tempstr)==NULL)
        {
                // If the extension doesn't exist
                // already, add it
                AddSIPDataLocal(sipptr,IP,buf);

                // Save to updated "sip.conf" back to disk
                PrintDataToFile(sipptr, SIP);

                // system command to tell asterisk
                // to reload "sip.conf"
                system("asterisk -rx 'sip reload'");

        }
        // Read "extensions.conf" into memory
        extptr=ReadFileToMem(EXT);

        // Put the extension into a format that
        // represents how is appears in the file
        // "extensions.conf"
        sprintf(tempstr,">%s,",buf);

        // Check to see if the extension already
        // exists in "extensions.conf"
        if(SearchList(extptr,tempstr)==NULL)
        {
                // If the extension doesn't exist
                // already, add it
                AddEXTDataLocal(extptr,IP,buf);

                // Save to updated "extensions.conf"
                // back to disk
                PrintDataToFile(extptr, EXT);

                // system command to tell asterisk to
                // reload "extensions.conf"
                system("asterisk -rx 'extensions reload'");

        }
        // Free the memory that holds "extensions.conf"
        // and "sip.conf"
        DeleteList(sipptr);
        DeleteList(extptr);
}
/*****************************************************/

/*****************************************************
This function read a file and stores it
into memory.
INPUT:
        String -          contains the filename to be
                          read from.
OUTPUT:
        LIST pointer -    contains the address to the
                                   first element of the file
                                   (in memory)
*****************************************************/
LIST ReadFileToMem(char *fn)
{
        // Local variable declarations
        FILE *fileptr;
        LIST p = NULL;
        char line[LINE_MAX];
        int length;
        // Open the file for reading
        fileptr=fopen(fn,"r");
    if (!fileptr) {error("Opening File");}

        // Loop to get the contents of the file line-by-line
        // and store the contents into a list
        while(fgets(line, sizeof(line), fileptr) != NULL )
        {
                // If the line ends with a new line, change it
                // to a null terminator
                length = strlen(line) - 1;
                if(line[length] == '\n') {line[length] = 0;}

                // Add the line to the list
                AddToList(&p, line);
        }
        // Close the file
        fclose(fileptr);

        // Return the pointer to the start of the list.
        return p;
}
/**************************************************/

/**************************************************
This function searches for a match
between a list and a string and returns
the matching element.
INPUT:
        LIST pointer -  Contains a pointer to a list
                                that is to be searched for a
                                match.
        string -        The string that is to be
                        searched for within the list.
OUTPUT:
        LIST pointer -  Upon finding the string within
                                the list, the pointer to that
                                element is returned. If the
                                string is not found, NULL is
                                returned by the function.
**************************************************/
LIST SearchList(LIST ptr, char *searchstr)
{
        while(ptr!=NULL)
        {
                if(strstr(ptr->str,searchstr))
                        return ptr;
                ptr = ptr->next;
        }
        return NULL;
}
/**************************************************/

/**************************************************
This function deltes the contents of the
list that is passed to it from the point
of the element that is passed to it.
INPUT:
        LIST pointer -  Contains a pointer to a list
```

```
                                  that is to be destroyed.
OUTPUT:
        (none)
****************************************************/
void DeleteList(LIST ptr)
{
        while(ptr!=NULL)
        {
                LIST next = ptr->next;
                free(ptr);
                ptr = next;
        }
}
/****************************************************/

/****************************************************
 This function adds an entry to the end of a list.
INPUT:
        LIST pointer -  This is a pointer to a pointer
                                that contains a list.
        String -        This string contains the
                        information that is to be added
                        to the end of the list.
OUTPUT:
        (none)
****************************************************/
void AddToList(LIST* pptr, char *str)
{
        // Create a pointer that will contina the address
        // of the list that is to be added to.
        LIST ptr = *pptr;
        LIST p = ptr;

        // If the list not empty.
        if(p)
        {
                // Go to the end of the list.
                while(p->next)
                {
                        p=p->next;
                }
                // Create a new element and move to it.
                p->next = malloc(sizeof(LST));
                p=p->next;
        }
        // If the list is empty.
        else
        {
                // Create the first element.
                p = malloc(sizeof(LST));
                *pptr = p;
        }
        // Set the next pointer to point nowhere (the end
        // of list)
        p->next = NULL;

        // Copy the string (passed into function) to the
        // element.
        strcpy(p->str, str);
```

```
}
/***************************************************/

/***************************************************
This function adds a extension to the
"extensions.conf" file (held in memory).
INPUT:
        LIST pointer -  This pointer points to a list
                                that contains "extensions.conf"
        String -        This string contains the
                        extension that is to be added
                        to "extensions.conf"
OUTPUT:
        (none)
***************************************************/
void AddEXTDataLocal(LIST ptr, char *IP, char *str)
{
        // Local variable declarations.
        LIST pstart, tempptr, newentry=malloc(sizeof(LST));
        char tempstr[60];

        // Find the line that holds "[remote]'
        ptr=SearchList(ptr, "[remote]");

        // Record the start position of the new entries.
        pstart=ptr;

        // Set the pointers so that tmpstr is
        // inserted 'here'.
        newentry->next = ptr->next;
        ptr->next = newentry;
        ptr = ptr->next;

        // Copy the string into the newly allocated space.
        sprintf(tempstr,"exten=>%s,1,NoOp()", str);
        strcpy(newentry->str, tempstr);

        // Set the pointers so that tmpstr is
        // inserted 'here'.
        newentry = malloc(sizeof(LST));
        newentry->next = ptr->next;
        ptr->next = newentry;
        ptr = ptr->next;

        // Copy the string into the newly allocated space.
        sprintf(tempstr,"exten=>%s,2,Dial(SIP/%s/${EXTEN})",
                        str, IP);
        strcpy(newentry->str, tempstr);

        // Set the pointers so that tmpstr is
        // inserted 'here'.
        newentry = malloc(sizeof(LST));
        newentry->next = ptr->next;
        ptr->next = newentry;
        ptr = ptr->next;

        // Copy the string into the newly allocated space.
        sprintf(tempstr,"exten=>%s,3,HangUp()\n", str);
        strcpy(newentry->str, tempstr);

        newentry = malloc(sizeof(LST));
        // Find the line that holds "[remote]'
        while(!strstr(ptr->next->str, "[phones]"))
        {
                ptr=ptr->next;
        }

        // Create a string to test for the Asterisk
        // servers presence.
        sprintf(tempstr,"[%s_incoming]", IP);
        // If there is not already an entry for this
        //asterisk server.
        if (SearchList(pstart,tempstr)==NULL)
        {
                // Add information 'here' for the server.
                tempptr = malloc(sizeof(LST));
                tempptr->next = ptr->next;
                sprintf(tempstr,"[%s_incoming]", IP);
                strcpy(tempptr->str, tempstr);
                ptr->next = tempptr;
                ptr = ptr->next;

                tempptr = malloc(sizeof(LST));
                tempptr->next = ptr->next;
                sprintf(tempstr,"include=>internal\n");
                strcpy(tempptr->str, tempstr);
                ptr->next = tempptr;
                ptr = ptr->next;
        }
}
/*************************************************/

/*************************************************
This function adds a extension to the "sip.conf"
file (held in memory).
INPUT:
        LIST pointer -  This pointer points to a list
                                that contains "sip.conf"
        String -        This string contains the
                        extension that is to be added
                        to "sip.conf"
OUTPUT:
        (none)
*************************************************/
void AddSIPDataLocal(LIST ptr, char *IP, char *str)
{
        // Local variable declarations.
        char tempstr[60];
        LIST newentry, startptr=ptr;
        bool alreadyExists=false;

        // Find the position of register
        while(ptr=SearchList(ptr, "register"))
        {
                // If the IP address is registered.
                if(strstr(ptr->str,IP))
                {
                        // Record that the registration has already
                        // been entered
                        alreadyExists=true;
                }
                // Move to the next list item.
                ptr=ptr->next;
        }
        // If there is no registration information in
        // the file.
        if (!alreadyExists)
        {
                // Add registration details.
                newentry = malloc(sizeof(LST));
                sprintf(tempstr,"register=>%s:welcome@%s/%s",
                        LOCALIP, IP, IP);

                strcpy(newentry->str, tempstr);
                newentry->next = ptr->next;
                ptr->next = newentry;
        }
        // Add entries to the end of the file.
        sprintf(tempstr,"\n[%s]", IP);
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"type=friend");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"secret=welcome");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"context=%s_incoming", IP);
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"host=dynamic");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"disallow=all");
        AddToList(&startptr, tempstr);
        sprintf(tempstr,"allow=%s",CODEC);
        AddToList(&startptr, tempstr);
}
/**************************************************/

/**************************************************
This function loads global variables
from a configuration file that is
stored in the router in the directory /bin/
INPUT:
        (none)
OUTPUT:
        (none)
**************************************************/
void SetLocalSettings()
{
        // Local variable declarations
        FILE *fp;

        // Open the file for reading
        fp=fopen("/bin/LOCALIP.ROUTER","r");
    if (!fp)
        error("Opening File");

        // Obtain the IP address of the wireless interface
        // of the router (used for unique name for
        // Asterisk server).
        fgets(LOCALIP, sizeof(LOCALIP), fp);
        int len = strlen(LOCALIP) - 1;
        if(LOCALIP[len] == '\n')
                LOCALIP[len] = 0;

        // Obtain the codec that is being allowed to be
        // used by the Asterisk server (Used in "sip.conf")
        fgets(CODEC, sizeof(CODEC), fp);
        len = strlen(CODEC) - 1;
        if(CODEC[len] == '\n')
                CODEC[len] = 0;

        // Location of "sip.conf" (for fopen() calls)
        fgets(SIP, sizeof(SIP), fp);
        len = strlen(SIP) - 1;
        if(SIP[len] == '\n')
                SIP[len] = 0;

        // Location of "extensions.conf" (for fopen() calls)
        fgets(EXT, sizeof(EXT), fp);
        len = strlen(EXT) - 1;
        if(EXT[len] == '\n')
                EXT[len] = 0;

        // Location of Asterisk log file (for finding
        // new phones)
        fgets(MESSAGES, sizeof(MESSAGES), fp);
        len = strlen(MESSAGES) - 1;
        if(MESSAGES[len] == '\n')
                MESSAGES[len] = 0;

        // Close the file.
        fclose(fp);
}
/***************************************************/

/***************************************************
This function stores the conents of the
lists in memory back to file.

INPUT:
        list pointer -  List to be saved to file
        String -        Name of the file that the
                        memory contents are to be
                        written to.
OUTPUT:
        (none)
***************************************************/
void PrintDataToFile(LIST ptr, char *filename)
{
        // Local variable declaration
        FILE *fp;

        // Open the file for writing
        fp=fopen(filename,"w");
    if (!fp)
        error("Opening File");

        // Loop through the list, printing each string in
        // the list as a new line in the file.
        while(ptr != NULL) {
                fprintf(fp,"%s\n",ptr->str);
                ptr = ptr->next;
        }
        // Close the file.
        fclose(fp);
}
/*****************************************************/
```

C.5 The makefile for Open-WRT SDK

Make file for acissend.c residing in the OpenWrt-SDK-Linux-i686-1/package/ directory.

```
include $(TOPDIR)/rules.mk

PKG_NAME:=acissend
PKG_RELEASE:=1

PKG_BUILD_DIR:=$(BUILD_DIR)/$(PKG_NAME)

include $(INCLUDE_DIR)/package.mk

define Package/acissend
        SECTION:=utils
        CATEGORY:=Utilities
        TITLE:=acissend --- Receives large update
        DESCRIPTION:=\
        Gets a list of all IP adresses \\\
        and the extensions associated \\\
        with them and records that data.
endef

define Build/Prepare
        mkdir -p $(PKG_BUILD_DIR)
        $(CP) ./src/* $(PKG_BUILD_DIR)/
endef

define Package/acissend/install
        $(INSTALL_DIR) $(1)/bin
        $(INSTALL_BIN) $(PKG_BUILD_DIR)/acissend $(1)/bin
endef

$(eval $(call BuildPackage,acissend))
```

C.6 The `makefile` for Linux

Make file for acissend.c residing in the OpenWrt-SDK-Linux-i686-1/package/acissend directory.

```
acissend: acissend.o
        $(CC) $(LDFLAGS) acissend.o -o acissend
acissend.o: acissend.c
        $(CC) $(CFLAGS) -c acissend.c

# remove object files and executable
clean:
        rm *.o acissend
```