An Architecture for Mobile Communications in Hazardous Situations and Physical Disasters.

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Abbreviations

802.11	WIRELESS NETWORK TECHNOLOGY
108G/54G	DATA RATE OF WLAN EQUIPMENT
3G	3D GENERATION
AC	AUTHENTICATION CENTRE
ACK	ACKNOWLEDGE
AODV	AD HOC ON-DEMAND DISTANCE VECTOR ROUTING
AP	ACCESS POINT
BSC	BASE STATION CONTROLLER
BSS	BASIC SERVICE SET
BTS	BASE STATION TRANSCEIVER
CBR	CONSTANT BIT RATE
CDMA	CODE DIVISION MULTIPLE ACCESS
CODEC	COMPRESSOR-DECOMPRESSOR
CPU	COMPUTER PROCESSOR UNIT
D-AMPS	DIGITAL-ADVANCED MOBILE PHONE SERVICE
DARPA	DEFENCE ADVANCED RESEARCH PROJECTS AGENCY
DSDV	DESTINATION-SEQUENCED DISTANCE VECTOR
DSP	DIGITAL SIGNAL PROCESSING
DSR	DYNAMIC SOURCE ROUTING (PROTOCOL)
DSSS	DIRECT-SEQUENCE SPREAD SPECTRUM
E2E	END TO END (DELAY) OR ETE
EDGE	ENHANCED DATA RATES FOR GSM EVOLUTION
ETSI	EUROPEAN TELECOMMUNICATIONS STANDARD INSTITUTE

FDMA	FREQUENCY DIVISION MULTIPLE ACCESS
FHSS	FREQUENCY-HOPPING SPREAD SPECTRUM
FTP	FILE TRANSPORT PROTOCOL
GB	GIGABYTE
GHz	GIGAHERTZ
GMSC	GATEWAY MOBILE SWITCHING CENTRE
GPRS	GENERAL PACKET RADIO SERVICE
GSM	GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS
HLR	HOME LOCATION REGISTER
ID	IDENTIFICATION (NUMBER)
IEEE	INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS
IETF	THE INTERNET ENGINEERING TASK FORCE
IP	INTERNET PROTOCOL
IPv4	INTERNET PROTOCOL VERSION 4
IPv6	INTERNET PROTOCOL VERSION 6
ISDN	INTEGRATED SERVICE DIGITAL NETWORK
ITU	INTERNATIONAL TELECOMMUNICATIONS UNION
LAN	LOCAL ACCESS NETWORK
MAC	MEDIUM ACCESS CONTROL
Mbit	MEGABIT
MH	MOBILE HOST
MNC	MOBILE NETWORK CODE
MS	MOBILE STATION
MSIN	MOBILE SUBSCRIBER IDENTIFICATION NUMBER
MSSC	MOBILE SERVICES SWITCHING CENTRE

OMC	OPERATIONS MAINTENANCE CENTRE
OSI	OPEN SYSTEMS INTERCONNECTION REFERENCE
PCM	PULSE CODE MODULATION
PHY	PHYSICAL (LAYER)
PRNET	PACKET RADIO NETWORK
PSTN	PUBLIC SWITCHED TELEPHONE NETWORK
RAM	RANDOM ACCESS MEMORY
RF	RADIO FREQUENCY
RREP	ROUTE REPLY
RREQ	ROUTE REQUEST
RTS/CTS	REQUEST TO SEND / CLEAR TO SEND
SIM	SUBSCRIBERS IDENTIFICATION MODULE
SMS	SORT MESSAGE SERVICE
SSID	SERVICE SET IDENTIFIER
SURAN	SURVIVABLE RADIO NETWORK
TCP/IP	TRANSMISSION CONTROL PROTOCOL/INTERNET PROTOCOL
TORA	TEMPORALLY-ORDERED ROUTING ALGORITHM
TX/RX	TRANSMIT RECEIVE
UMTS	UNIVERSAL MOBILE TELECOMMUNICATIONS SYSTEM
UTRAN	UNIVERSAL TERRESTRIAL RADIO ACCESS NETWORK
VLR	VISITOR LOCATION REGISTER
VOIP	VOICE OVER INTERNET PROTOCOL
WEP	WIRED EQUIVALENT PRIVACY
WI-FI	WIRELESS FIDELITY
WLAN	WIRELESS LOCAL ACCESS NETWORK

WTC	WORLD TRADE CENTRE

Abstract

Hazardous environmental conditions have always been a threat to human lives around the globe. Human society has seen some of the worst disasters due to accidents, physical phenomena or even cases that humans have created on purpose. The existing infrastructure can guarantee that there are hospitals, markets, mass transportation networks, sophisticated communications networks, and many more to cover all possible needs from a home user to an enterprise company. Unfortunately, the infrastructure has been proven unstable due to rapid environmental changes. The sophisticated networks, as well as the support buildings, can become useless in seconds in the event of a physical phenomenon such as an earthquake, a fire or a flood or even worse in the event of a well organized terrorist attack.

The major problems identified are associated with inadequate capacity of the network, equipment vulnerable to physical phenomena and methodologies of disaster recovery that require time and work force to be applied. Modern telecommunication systems are designed in a cost effective way, to support as many users as they can, by using minimum equipment, but they cannot support users in hazardous environments.

As a response to this situation we present the development of a novel architecture, which is based on an fast deployed network, infrastructure independent. The proposed network is capable of providing mobile subscribers with messaging and voice services in hazardous environments at the time of the event. Similar studies are

based on infrastructure as they are in the need of extra hardware deployment. The novelty of our research is that we combine 802.11 and GSM in order to form a fast deployed network, infrastructure independent. The proposed architecture has two modes of operation: messages only or voice system. This solution benefits from the advantages of a deployed, infrastructure independent Ad Hoc network. This network is able to recover quickly from errors and can survive in hazardous dynamic environments. In addition we benefit from GSM technology using already implemented functions such as encoding/decoding for voice transmission. Combining those two technologies we can deploy a network which satisfies the challenges previously mentioned. While 802.11 handles connectivity and data transfers, GSM is responsible for bit error correction of voice calls and a number of other functions such as messaging and identification.

The proposed architecture has been designed and simulated in order to evaluate the network. The evaluation has been separated in two phases. Messaging and voice capabilities of the network have been tested to investigate their performance. In the evaluation we check the factors affecting the network in a hazardous environment and we compare it to other approaches and similar networks. The results prove that the concept of messaging service is valid as the system can operate in hazardous environments. Voice capabilities of the system have been proven to work but further work is needed for maximising the performance and the reliability of the network. The new architecture can form the basis for the next generation emergency telecommunication services.

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CHAPTER 1. Introduction

The introduction of wireless networking and mobile communications has advanced through the last three decades. The demand for faster telecommunication networks has led to a new era of voice communications including the Global System for Mobiles (GSM) [3] technology and new wireless services like 802.11 [6],[7],[12]. In most cases, wireless networks have two basic characteristics. They offer wireless services through mobile devices and they use wired infrastructure as the base of handling and controlling traffic between the devices and the main network. As humans tend to live together in populated areas, they have created a very sophisticated infrastructure to meet their demands and make life easier. Infrastructure nowadays in big cities has become a crucial need. Fast networks have been deployed for serving society's basic needs, for example in the health sector or in telecommunications. Infrastructure today can help people to be more productive and complete easily and quickly even in the most demanding tasks.

The need for wireless environments, real time communications and the large volume of data being transferred through networks, has made telecom and wireless networks even more sophisticated. A new challenge, though, is the demand for reliability, capacity and speed. Those demands have been increased in the last decade as wireless technology has been expanded and became widely popular. A challenge in GSM networks today is forwarding traffic from wireless mobile phones, through wireless antennas and other equipment, to centralized wired servers. This creates a problem to real time networks, as they have to support a large number of simultaneous users calling at the same time and consuming even more bandwidth [20]. An additional challenge is that messages and especially digitized voice (in the form of packets), must reach their destination within acceptable delay limits. While users talk to each other, long waiting times on calls, or even worse, disconnections, would make the system unreliable. To avoid that, telecom providers have established a powerful infrastructure by placing antennas (or cell sites as they are called), in strategic locations across a city in order to optimize the performance of the GSM network in a cost effective way. However, it has been found that the supporting infrastructure of GSM has major weaknesses in cases of major disasters.

As companies assume that only a fraction of their users will call simultaneously and the fact that the cell sites have been placed relatively far apart using a cost effective strategy, GSM has been found to be unreliable in emergency situations. It is a fact that telecom providers are using a minimum number of cell sites to cover the optimal range and the user capacity of an area. This may lead the system to collapse under heavy traffic conditions, as it has not been designed to support all the subscribers at the same time. When such a network fails, the communication between users is impossible. In hazardous situations where communication is vital for human lives, the service does not exist. Therefore, this thesis is focused on the design of a new network architecture that provides infrastructure independent mobile communications in hazardous environments.

1.1. Mobile Networks and Environments

In voice communications the telephony network [19] has evolved in the last few years. Old wired analogue telephone networks were replaced by digital ones. The next step in telephony evolution is wireless telephony, which provides users with the capability of using a mobile phone whilst on the move.

Currently there is a very sophisticated network in which subscribers can talk and send short messages, or multimedia messages, including pictures or even videos. Obviously those demands required the devices to be more sophisticated with large amounts of memory and greater CPU power. The first analogue mobile network has evolved to a digital one, ready to serve even the most demanding users. Today there are many different types of mobile networks. Some of them are in a transition from the third generation GSM networks [7], [11], [3] to faster and better ones (3G, UMTS) [6], [23]. The latter have become popular as the new challenges to be met in telecommunications are increased reliability and a greater number of services offered.

In the beginning of radio engineering, a great achievement was the establishment of a simple link connecting a transmitter and a receiver. The first attempt for a one-way communication link was also the first step for mobile communications. The next challenge was to establish a two-way link. This step of the development had a great advantage compared to the previous. One-way links had no future, as establishing a call to a mobile phone did not have a confirmation tone. Using two-way links the situation has improved. Yet the service was limited to a certain area that could be reached by using a main transmitter, or a number of smaller ones, and by allocating the available channel frequencies of the particular site. Obviously the callers could only speak to others only within the range of the site. Today we call this area a cell, and it is represented by a hexagonal region [11].

At the very early stages of development it was impossible to connect two cells because the infrastructure was inadequate. The major problem was to select the frequency of the transmitters and receivers in such a way that they would not interfere with each other. A small set of the frequencies/channels was not enough to cover a large area. Furthermore, even though the transmitters were powerful enough they were operating in frequencies that could not be reused for hundreds of kilometres. It is obvious that this limits the network's capacity. We may agree that capacity was not a problem at this early development as mobile equipment was expensive for the public. In 1960s - 1970s market price of the mobile equipment dropped in value dramatically and the capacity threshold was broken. There was extensive research work started in order to find an alternative way to allocate more frequencies, otherwise the system had no future [11].

In 1982, in the Conference of the European Posts and Telegraphs also known as (CEPT) [11], a new group was formed under the name Group Special Mobile (GSM), in order to develop a Pan-European public mobile system. Their main scope was to meet requirements such as good speech quality, low call costs, support for international roaming and others. In 1989 GSM development was transferred to the European Telecommunication Standards Institute (ETSI) [7]. A year later ETSI published the first GSM specification, and in 1991 they initiated the first commercial service. In 1993, more than 30 GSM networks were created in 22 counties. Since then GSM networks have become popular all over the world. A year later there were 1.3 million subscribers worldwide. In 1997, GSM was popular as its number of subscribers increased to 55 millions. North America made a much delayed entry in GSM technology and introduced a derivative of GSM, called PCS1900 [11]. Today GSM exists all around the world and the acronym GSM stands for the well known Global System for Mobile Communication.

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1.2. Evolution of Mobile Telecommunication Networks

As the developers of the 1st generation of GSM were trying to improve capacity and mobility in an analogue network a new evolution in microprocessors led the world industry to smaller, lighter and more sophisticated mobiles.

Figure 1. Evolution of Cellular Phones [13].

Figure 1 illustrates the generations of mobiles and their main properties. Starting from the left hand side the first mobile had only voice and messaging capabilities. The development of mobile phones has evolved from first generation to third. Mobiles were developed so that today they are capable of sending videos and offer a great number of services, through different wireless networks. In 1990-7 the development of 2G cellular systems improved the capacity, the coverage and speech/transmission quality. New semiconductor processors and microwave devices brought digital broadcasting to mobile communications.

Figure 2. Evolution of 2G Cellular Phones and Bandwidth Consumption [13].

In figure 2 we can see the 2G main mobile connectivity which has been used for several years in many places and is still in use by many cities. This architecture is obviously better than the 1G because it is improved in every aspect.

The first objective of GSM development was to establish a link between mobiles for voice communication. As the network evolved to the second-generation, new challenges were created. New services such as fax, short messaging and data communications were needed. Rapidly those services were becoming fixed in mobile communications and new systems were born to support those services as Digital AMPS (D-AMPS), Code Division Multiple Access (CDMA) and Personal Digital Communication (PDC).

Since 1990 several services have been defined as data transmission in 9.6kbps but only a few were offered. As a result, GSM was enhanced in 1995 in order to add new services as delivered in ISDN, which could be considered comparable to the services of fixed telephone network. In 1996 ETSI announced a new phase of development in GSM (Phase +2) and 3G was introduced to the market.

Figure 3. Evolution of 3G Cellular Phones and Bandwidth Consumption [13].

Figure 3 illustrates the changes in mobile trends in order to build and support new networks. As the requirement from mobile operators has changed while the GSM telecom has evolved from 2G to 3G, old conventional hardware has been replaced by new powerful equipment in terms of bandwidth, services and the number of users it can support. 3G capabilities and their features are well known today as they involve intelligent networking, high data rate services, enhanced speech compression / decompression (CODEC), General Packet Radio Service (GPRS) and enhanced data rates for GSM (EDGE). Nowadays UMTS is a 3G GSM popular standard, fully compatible with GSM as it is using GSM Phase +2 enhanced core network [11],[12].

Figure 4. Evolution of 3G Cellular Phones and Networks [13].

Figure 4 illustrates the expandability of the 3G network and its relationship to other wireless networks through a number of different services, like IP telephony, that already existed in 2G mobiles and internet services.

As those pictures include a number of important terms, and our research at present is focused on the GSM systems, we will introduce its basic structure and functionality through a survey in chapter 2 which highlights the basic components of GSM architecture. From this chapter it is clear that most of the research that has been done on GSM networks is focused on optimization and maximization of the network capacity, speed and performance as well as connectivity to similar wireless technologies.

1.3. GSM Applications in Hazardous Situations.

GSM development objectives were always clear and focused on several requirements, for example voice quality. After many years of development and research, the mobile telecommunications industry is still expanding year after year [Appendix 1], and can provide voice communication with a great percentage of coverage along mega-cities and populated areas [8], [20].

GSM is a powerful wireless technology using the air as the medium between the network and the mobile phones. The mobiles communicate with antennas which are connected to mobile base stations and exchange data with the GSM core system. The connections between the antennas and the rest of the GSM system are based on wired connections. Mobile companies face a great challenge as the wired equipment cannot always support mobile users; in many cases there is a big risk of a network breakdown. Hazardous environments, physical phenomena and disasters can force the network to total failure in seconds as it has not been designed to support situations in which the environment can change rapidly. It is a fact that many mobile companies assume that only a fraction of their users will call simultaneously. History has proven that in cases where the demand is higher than the capacity of the network, it is inevitable that the GSM network will fail, as for example during the earthquake in Greece in 2001. The overloaded network blocked all outgoing calls and then disconnected all inbound ones, as it was overloaded by 400%. It is clear that a GSM network is weak in situations where the capacity is increased as it cannot support all the subscribers simultaneously.

In addition, during the last decade, new challenges have been added to the telecom industry. Accidents or physical disasters and other scenarios that can place human life in danger are unavoidable and require a dramatic improvement in how we respond to them. The new problem is to provide communication services to people suffering from a disaster at the time of the incident. In this field of research many companies have already provided recovery disaster solutions as for example Verizon [17], which provided mobile antennas for the areas of disaster trying to recover telecom services. This approach was successful but also a slow process as it requires time to set up the equipment. A different approach is the replacement of destroyed hardware and cables, which has also been a successful procedure but again it needs a couple of days to restore the basic telecom services. Although there have been proposals and applied techniques that were successful, none of the solutions managed to help victims in the first minutes or hours of the disaster. All of the methods focused in disaster recovery. In other words, those methodologies have one main purpose; to recover infrastructure as soon as possible. In the case of an emergency situation where the network fails there is no fast recovery of the network. As the network is based on infrastructure it will fail in any event that can physically affect the cell sites. Obviously, the failure cannot only appear when the antennas are completely destroyed. A power failure or a local flood can easily disrupt communications as the wired equipment is susceptible to all hazardous conditions

In the future, new network architectures will have to be developed in order to provide mobile users services like communication in places where there is no signal coverage. Furthermore, companies will have to consider fast deployment of such a network, in the affected area which suffers from a hazardous situation at the time of the event, since the first hours after the event are important for human life. There have been many instances in which victims died in collapsed structures, after waiting a long time, because they were not able to ask for help.

These developments will also result in many changes of GSM networking as today mobile phones can communicate only with local GSM antennas. A new challenge is the design or modification of the network. Mobile phones must become compatible with the new architecture in such a way that they can be used in areas with no GSM services due to a failed network. Furthermore, the development of a fast deployed network should support a network environment which is not depended on infrastructure and antennas. History has shown that infrastructure appears to be weak and cannot support subscribers in case of a hazardous event.

To summarise the challenges that modern mobile telecommunication networks face, they lack emergency services in the case of a hazardous event. The network is vulnerable to physical phenomena or disasters as all hardware wired components can fail. A disaster can disrupt communications or even worse can force the network to fail. Victims in need of asking for help are not supported in hazardous environments. Current disaster recovery approaches are based on fast replacement of the infrastructure. This fact makes them weak inside a hazardous environment. Broken networks cannot be recovered quickly and loss of human lives is likely to happen during the first hours of the disaster. All the effort in the telecom industry is focused on cost effective systems and minimization. In general, companies design their networks in such way they cannot support more than a fraction of their total number of subscribers calling simultaneously. Recovery disaster solutions are slow and people must be sent into the hazardous environment to restore it.

1.4. Thesis Aims and Objectives

The aim of this thesis is to provide a new network architecture which involves mobile communications in places where GSM services have failed. For this purpose, a new architecture is needed in order to design such a network. This architecture will give the users the ability to use their mobile phones in areas that the GSM infrastructure is not operational due to a hazardous event. The network is considered as an Ad Hoc one and is based on mobile phones only as it is infrastructure independent. We will investigate the main properties of a hazardous environment and study a combination of two technologies in order to design a network with GSM services, but also with 802.11 Ad Hoc capabilities. Before looking for a solution, we have to consider some issues:

1. What are the requirements for building such an architecture? Challenges of the current architecture have to be investigated and answered as the new architecture will borrow elements from the current GSM and 802.11 technologies. For example, the need of Ad Hoc environment would consider devices that can communicate without infrastructure support and can provide communication services in heavy traffic conditions. The network is dynamic as there are no fixed routes; it has added mobility and other parameters that can affect any wireless dynamic network as the environment changes rapidly and continuously.

2. An investigation is needed regarding the existing technologies and standards that are best suited for this network architecture. Additionally, further research on the mixture of the two technologies is needed, in order to identify their main properties and components. This will help us to understand how the technologies work and how they can be combined. Furthermore, it will provide us with the knowledge of any weaknesses or limitations of the system. Before making design decisions the related survey will present current approaches that have been applied in many cases, and will give us an understanding of what methods were used and their weaknesses.

3. Having the knowledge of the basic properties of GSM and 802.11 technologies, of the related work, and of the challenges, our next aim is to understand and evaluate the environmental conditions and the demands of a hazardous environment. In such a way we will point out the main properties and parameters that affect the network's performance. Physical limitations, parameters like bandwidth, delays and interference will be investigated for the deployed network. Decision making will follow, concerning the best technologies and techniques or methods to be used.

4. The next objective is to find a way of using a mobile phone in hazardous environments, without service support and use it to get help. As no service is available in the area and the mobile reads "no signal" our aim is to find a way of communication with other mobile phones in the form of text messages or voice communication. In order to achieve that, we will have to choose suitable technologies for connecting the devices and find ways of transferring voice data.

5. Deploying an network quickly in the affected area will give the capability of messaging and voice communication to users. The operation of the mobile phone will be the same as when the GSM network operates normally. A suitable technology, or more than one, must be chosen for the design of a dynamic network which is able to recover from packet loss and bit errors. Then a simulation of a fast deployed text and voice network will follow to establish reliable communication between users. Finally, an evaluation of the network will take place, proving and validating the operation under heavy traffic conditions in hazardous environments.

The design of the deployed network should allow direct communication between the devices, as in hazardous environments we are not depending on infrastructure. The implementation of the architecture should help victims in the suffering area. The design is based on mobile devices that can be found anywhere. All of the challenges and problems must be solved in order to design a reliable network. Finally, the proposed architecture will be evaluated in a case study scenario in order to analyse the reliability and the speed of the deployed network in the affected area.

1.5. Novelty Aspect of the Thesis

In this thesis, we present an architecture of mobile communications, for using mobile phones in hazardous environments. We have proposed a fast deployed network, which is infrastructure independent, for data and voice connectivity between mobile phones in areas that suffer from physical disasters or hazards. In order to build the network two major technologies are combined. GSM and 802.11 will allow us to develop a fast deployed network. The various components can provide several services and functionalities in order to provide a GSM emergency network in areas that suffer a hazardous event. In contrast with other approaches, we propose switching to a different network and forwarding data through it, so that both technologies cooperate to form such a network. In the thesis, two major phases have been designed and tested. Each of those contributes to a specific problem and its solution from the messaging functionality of the network, to the voice communication using mobile phones in areas that there is no GSM service. The contributions are as follows:

- The first contribution surveys the architecture of several components and several approaches have been deployed in order to recover communication in places that have been partially or completely destroyed by accidents or physical disasters. Additionally we investigate and present next generation support services and the requirements for building a fast deployed emergency network as well as its main properties and characteristics. Finally, in the survey we highlight the issues and problems that may arise while designing such a network, which is followed by the analysis of all major parameters that affect telecommunication networks in hazardous environments.

- The next contribution is the taxonomy of the requirements and challenges, for the proposed network in terms of environmental conditions as the proposed network is a dynamic one. Several services that are included in GSM networking, such as authentication, cannot be used in Ad Hoc networking due to the lack of infrastructure support and wired equipment. Furthermore, an analytical presentation of the model and the network functionality is discussed. The survey will give us the knowledge in order to understand and solve any issues or challenges. Finally, a survey that includes routing algorithms and other components of the network proves that is possible to combine GSM and 802.11 technologies and build such a network.

- In this research we have proposed an architecture of communication between mobile phones in hazardous environments. As the devices can only have direct communication with antennas or cell sites as they are called, we have presented a method for direct connectivity between the mobile phones without using the GSM network infrastructure. The novelty of our approach is that in case of GSM loss of service, instead of switching to a different network, two technologies are being used for the communication of the devices. 802.11 routing is being used to overcome connectivity problems and transferring data across the links while at the same time bit errors are handled by the GSM hardware. Using this architecture we have managed to establish fast and reliable communications with bit error control that results in good voice quality between users, within the Ad Hoc network, which is not dependent on GSM cell sites.

- The proposed architecture allows fast deployment of the emergency communication network. This is a challenge for the next generation emergency services and in our research this has been achieved as demonstrated from our results. Simulations of the proposed network, each corresponding to a different aspect, prove that the proposed network meets the demands and the requirements that have been established from the related survey for an emergency network. As the results indicate, this research can be used to build a new collaborative architecture, which will integrate emergency services in current telecommunication networks. The novelty of this research is not trying to modify current devices or protocols, but using them to achieve our goal.

- The results of our research have been well evaluated using a well known simulator tool which is called Opnet 11.5. The network has been analysed, discussed and implemented in simulations in order to develop a prototype network with messaging capability, and another one with voice. Those prototypes have been evaluated in order to investigate the way that the proposed network will behave in hazardous environments. Major parameters like mobility have been implemented in the scenarios in order to build a global network which is close to reality and can fulfil the requirements of the emergency network as discussed in the thesis. Our results prove that this network meets the challenges for the next generation of the emergency networks. It is compatible with the current demands and standards of the existing technology as we will see in the relevant chapters. Finally, the contributed network can be further expanded as it consists of the two most popular wireless technologies, and can be integrated or bridged with other protocols.

1.6. Thesis Structure

The thesis is structured into the following chapters:

Chapter 1. The introduction to the thesis is presented. Mobile telecommunication networking and its evolution are discussed, highlighting the challenges for providing a fully mobile network. The basic concepts of a very well structured network known as GSM is presented.

In this section, we clarify that the design of the network has always been based on clear phases that focused on providing a new benefit based on a new technical innovation. As the GSM network evolved and has become the most popular network across the planet, we have identified the next generation support system for it, as well as the missing components and weakness. The final sections present the thesis aims, the objectives of the thesis, the novelty and contribution of this research.

Chapter 2. This chapter surveys the current technologies involved in our research. The description of GSM architecture is followed by the 802.11 architecture. Ad Hoc networking is presented next, followed by related work on voice communications in order to highlight the problems of the existing technologies that form the motivation for our research. Finally a survey on related work and different approaches is presented in order to explore the methods and identify weaknesses and problems.

Chapter 3. A presentation of architecture of a fast deployed network designed for disastrous environments is presented. In this chapter a detailed analysis of the components is presented. Our deployed network is explained in terms of message and voice transmission between mobile devices in affected areas with no GSM service support. This is followed by an extensive analysis on hazardous environments. The main properties of a dynamic environment in an affected area are presented and a detailed description is given about disastrous/hazardous environments. Next we identify weaknesses and parameters in order to overcome problems and make decisions on how to build the basic model of the proposed network. The basic model is then presented in order to explain how it works and make decisions on several issues. The technical information involving routing algorithms and decisions are discussed in order to complete the design of the network in such a way that will meet the requirements.

Chapter 4. This chapter investigates messaging capabilities in the proposed architecture. A prototype model is presented. A simulation is included as well as a presentation and an evaluation of the results. The chapter finishes with the findings of the research, the results and a discussion for this component of the network.

Chapter 5. This chapter is related to the voice capabilities of the network. A few sections are related to parameters and components of voice communications like encoding techniques, codecs and delays which may affect the voice network. Then the network is presented in two phases. After building the prototype, the global network is tested with all added parameters. The chapter finishes with a discussion on various aspects such as evaluated results, limitations and the performance of the voice network.

Chapter 6. This chapter consists of the conclusion of our research. In addition, future work proposals can be found for further development on various aspects, in order to expand the network and its functionality and to connect it with other known technologies.

CHAPTER 2. Survey of Related Work

2.1. Introduction

Chapter 2 presents the architectures of existing wireless technologies, which are related to our research. Additionally this chapter contains information about Ad Hoc networks and related routing algorithms. The chapter finishes with a related survey in networking for voice communications and disaster recovery techniques in mobile communications.

2.2. GSM Architecture

The Global System for Mobile Communication or GSM as it is called is a set of ETSI [7] standards specifying the infrastructure of digital cellular services. The standard is being used in more than 210 countries worldwide. GSM is a system that connects mobile communications such as mobile phones with the rest of the known technologies as the Public Switched Telephone Network (PSTN) [10], [19]. Today, GSM technology is in use by more than one fifth of the world's population. By March 2006 there were over 1.7 billion GSM subscribers, representing approximately 77% of the world's cellular market [8]. The growth of GSM continues with almost 400 million new customers in the last 12 months as new infrastructure is being built in many countries [9].

The GSM network consists of many components. Using different functions and interfaces the network can be divided into three parts; the mobile station, the base station subsystem, and the network subsystem. The network subsystem has two different roles. It provides services such as SMS [24], and handles calls between users through the Mobile Service Switching Centre (MSSC). It also connects mobile users to the fixed network (PSTN, ISDN). The MSSC handles handovers, allocates channels for calls and provides a sophisticated cost and mobility management system. The MSSC is connected to the Base Station subsystem. The latter controls the radio link between Base Station Controller (BSC) and the MSSC. Each Base Station Controller is connected to many Base Transceiver Stations (BTS), which are connected through the air to the Mobile Hosts (MH), (mobile phones). Whenever a call is placed, users are connected wirelessly to the local BTS (cell site) and the call is handled through the rest of the network, as each network part handles different functions [3], [11].

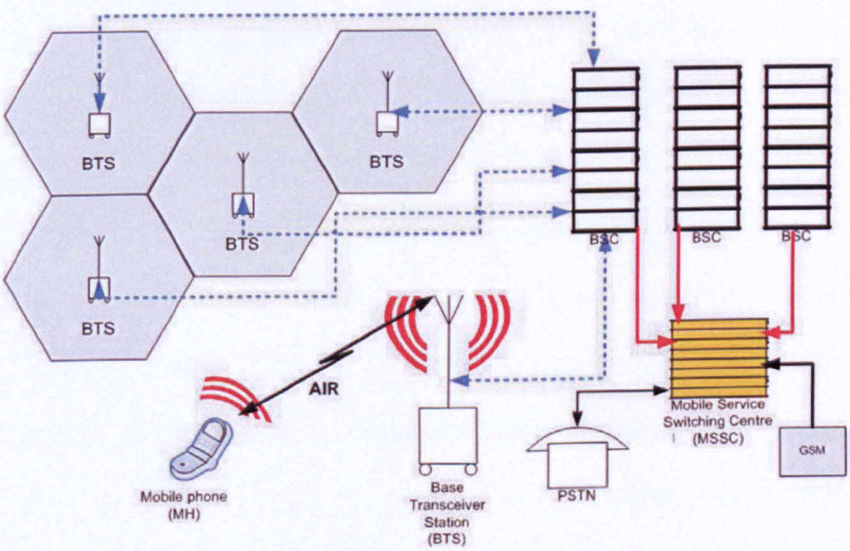

Figure 5. The GSM Architecture

The GSM network is connected by wires to its main components, except for the mobile phones. The medium of communication in this case is air. The mobile phones or mobile hosts as they are called have sophisticated hardware, which supports services such as localization and authentication. The local cell sites are nowadays customized by operators for maximum efficiency and many techniques are being used for maximizing the capacity and the voice quality. Furthermore, a method called cell sectorization [106] is being used in order to increase the capacity of the subscribers in the network. Sectorization improves the cellular coverage for users, and provides increased capacity. This method provides coverage across populated areas while at the same time, it reduces interference. Additionally the GSM network supports functions for frequency reuse, which plays a major role in the capacity of the network and handover methodologies to provide continuous communication for the users on the move.

Comparing GSM with similar wireless networks, many methods and parameters are found to have the same characteristics and operations. Today, in the telecommunication industry a big field of research is focused on new techniques for maximizing the performance of the network like Caution++ [15], [21], [22]. A different field of research and development is concerned with expanding the current networks and minimising bandwidth and costs by using sophisticated software like Astrix [14]. GSM is expanding and evolving to a new era of telecommunications which is called 3G. Nowadays new approaches and methods are being researched for additional connectivity with the rest of the known technologies as well as for providing emergency services for the network [22].

2.3. Wireless Networks and the 802.11 Architecture

Wireless networks today are associated with many usual actions during everyday life. Wireless technology can solve many complex problems or help in situations where wires are not convenient to use. In other words, a wireless network offers the freedom not to use cables in many cases. It is obvious that not all networks or connections can be replaced by wireless, like the power networks. Today most of the people using electronic devices demand to be cable free. According to them they prefer devices that do not have cables. The reason is very simple; freedom of movement is important nowadays in the office, in the car or even in the house.

Behind the friendly interface and the colourful devices there is a sophisticated hardware circuit, programmed with complex routing algorithms and mechanisms that guarantee connectivity, reliability of the connections and a stable operating behaviour of the device. Wireless networking market involves two large fields of research: hardware research, which deals with the electronic components, and software research that deals with the applications that the devices will support.

Dealing with the concept of wireless networking we refer to a number of different technologies and protocols combined in many different ways in order to satisfy a number of different applications and demands. The advantages of a wireless network, including the freedom of movement as described earlier, make wireless technology easy to use. The installation is always quick and effortless. Most of the installations are based on friendly graphical user interfaces. The user can provide information about the characteristics of the network as well as security, which is optional in home networks. In addition, a wireless network can be easily expanded or altered according to the needs of the scenario. Outside the home, wireless networking is available in hotspots at coffee shops, businesses, and airports - great when someone is on the move and needs to get some work done [25].

One of the most popular technologies in wireless networking is called the 802.11 protocol which uses radio frequencies for transmitting data. Local area networks are being used by thousands of companies and users around the world as wireless networks are well known for their advantages. The 802.11x [7], [25], [26], specification allows transmission of approximately 108 Mbps of raw data at distances up to a few hundred feet over the 2.4 GHz unlicensed band. 802.11 has many variations and can be used in different networks depending on the design and the requirement [Appendix 2]. The coverage area depends on obstacles, materials, the environment and the line of sight.

802.11 networks consist of different physical components in order to form a fully working and expandable network. Stations or mobile host as they are called are the actual computing devices having wireless network interfaces and offering wireless communication between other stations or access points. A mobile host can typically be a laptop computer, a PDA or even a mobile phone. It is obvious that desktop computers can be considered as stations as long as they are connected wirelessly in the network. Access points are the devices, which connect stations together and can also perform a wireless to wired interconnection. Today's access points can perform a number of different functions, as they exist on the market in multi devices including

ADSL modems, routers, firewalls and other components. The wireless medium is used to transfer data between stations and access points. The architecture of 802.11 allows multiple physical layers to be developed for supporting the 802.11 MAC. Two radio frequency (RF) layers have been standardised for communication. Distribution systems are the components used to forward the data frames to their destination. This is very useful in situations where many access points have been connected to form a large covered area. There is a need for access points to communicate with each other in order to track movements of the stations as they move.

2.3.1. 802.11 Architecture

The basic architecture of the 802.11 wireless system is quite easy to understand. It has been designed in such a way that can accommodate even the most demanding installations. A great issue is that a mobile station is able to communicate with any other mobile or wired station transparently, which means that above the MAC layer, 802.11 [27] appears like any other 802.x LAN and offers comparable services. A major advantage for our research is that the services are compatible with Ad Hoc and infrastructure modes.

Infrastructure mode is a communication method that requires a wireless access point. The latter is used for handling all communications between mobile nodes in the same area. If one mobile station needs to communicate with the second one the communication must take two hops. The sender station will transmit data to the access point and the latter will transfer the data to the second mobile station.

On the other hand, in Ad Hoc mode many wireless nodes may communicate directly with each other, without a wireless access point (Figure 6). Deploying such a network can create a fast deployed, short duration, small size network. This method will be discussed more in the thesis as in hazardous situations we are not relying on infrastructure.

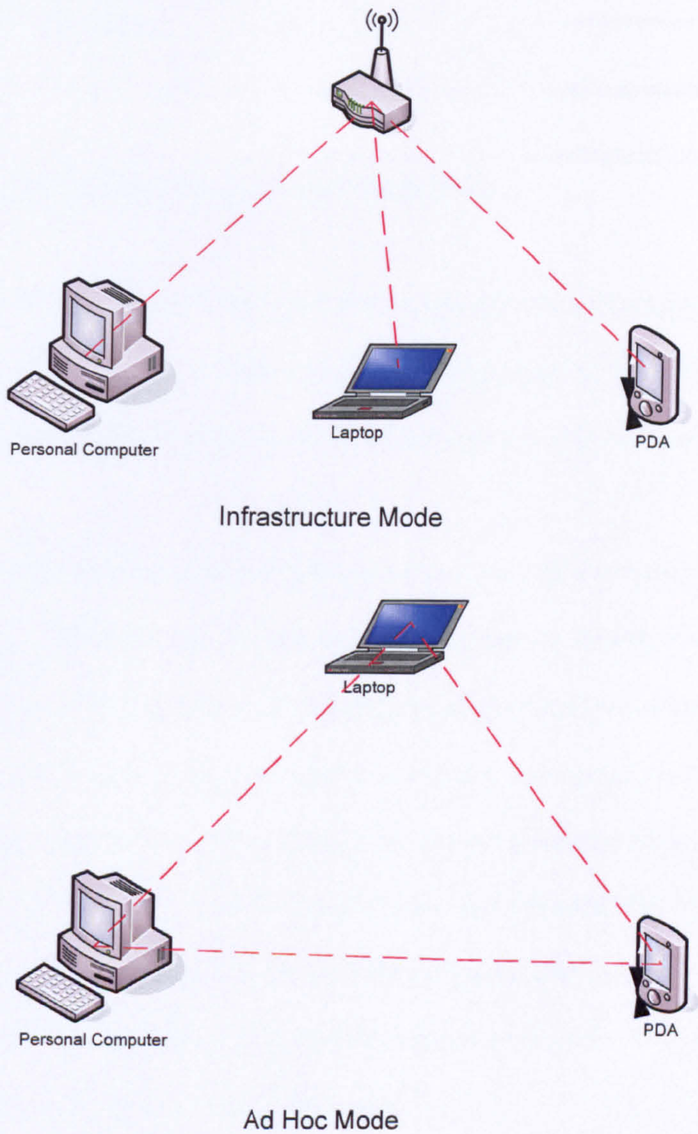

Figure 6. Infrastructure and Ad Hoc Modes in 802.11 Wireless Networks.

Both types of networks have advantages and disadvantages depending of the design implementation. Using an access point for a multihop transmission uses more network bandwidth but also allows no restriction of the distance between stations within the range of the access point. Access point can assist stations to save power. Ad Hoc networks on the other hand, have the great advantage of being infrastructure independent [25], [27].

2.4. Ad Hoc Environments and Networks

A part of our research is related to wireless networks and especially to Ad Hoc networks. The reason for this choice will be discussed later on, as at present we would like to present some information on Ad Hoc networks and environments.

A mobile Ad Hoc network is formed from a number of stations. We call the stations mobile nodes as they are not necessary laptops or desktop computers. They might be simple hardware devices that have low memory and power consumption. An Ad Hoc network can be deployed without using any infrastructure such as routers, access points, switches or hubs. Such a network is considered as infrastructure independent. It is clear that Ad Hoc mode is more cost effective than using a network with several components and is very efficient at places the infrastructure is lost or cannot be deployed. Applications on Ad Hoc networks can be found in military use or even everyday simple forms of communication.

2.4.1. Ad Hoc Networks Applications and their Evolution

Historically Ad Hoc networking was started as a military project back in 1972 by the Defence Advanced Research Projects Agency (DARPA) [28]. Its first name was PRNET, which stands for Packet Radio Network. A few years later by 1987 PRNET was a reliable, fully working network involving routing protocols and was designed for military use in combat environments. The components of the network could be easily flash updated by using a serial interface. The nodes were using a radio frequency for data exchange and they had an interface for monitoring and further development. Later on, after further implementations the physical data link and network layer (OSI model) [7],[Appendix 3], were introduced in the system as well as a number of enhancements like error handling , Cyclic Redundancy Checksum CRC and radio frequency selection within 20 available channels between 1718.4MHz and 1840.0MHz. The final network was fast, reliable and easy to deploy.

Meanwhile, in 1980 till 1993 the project was improved by the SURAN (Survivable Adaptive Radio Networks) program. The progress of this investigation made the network even faster and more reliable, as better algorithms, introduced for improving the performance, made the product invulnerable to electronic attacks. In the mean-time, the scientific community realised that this technology had potential to become very popular for commercial use. The first laptops introduced on the market and IEEE 802.11 subcommittee had adopted the term "Ad Hoc networks".

In 1994-6 Internet Engineering Task Force (IETF) [29], launched a work group on the topic called MANET (Mobile Ad Hoc Networking) in order to standardise routing protocols for Ad Hoc networks and the IEEE 802.11 subcommittee standardised a medium access protocol. Collision avoidance and tolerated hidden terminals were the base of the protocol as its main purpose was to make it usable for building mobile Ad Hoc networks' prototype devices for laptop computers. Today there is a variety of 802.11 PCI/PCMCIA cards. Bluetooth is one technology which is a similar to Ad Hoc networks that borrowed many features from Ad Hoc networking and benefited from it.

802.11 devices are becoming cheaper and more efficient day by day as the third generation of wireless networking has expanded dramatically. Today many office environments are based on wireless technology for various reasons, as explained previously and the technology can be used in civilian applications, business, entertainment and many other environments.

2.4.2. Popular Ad Hoc Routing Algorithms

Direct communication among neighbouring devices can be achieved easily on a wireless network. Communication though for non-neighbouring nodes requires a routing algorithm. A routing algorithm is actually a sophisticated method of connecting nodes with each other, using several optimizations in order to achieve reliable and fast communication. A lot of work has been done for routing techniques and protocols, as without them Ad Hoc networks cannot operate optimally. Routing protocols vary but they follow a similar technique for route discovery, maintenance and transfers. Sending data from one node to another requires a certain procedure which is called routing. Routing is responsible for many operations within the nodes and is of major importance as it provides information needed for a reliable and successful communication. In order to send data from one node to another, the routing algorithm will first initiate an operation which identifies neighbouring nodes. Each node broadcasts an identification name which is the IP address in our case. Additionally it keeps information about other nodes located nearby. This data is stored in what is called a routing table. Whenever a user is sending data, the routing algorithm will first establish a route between two or more nodes, depending on their location and range, and then it will send this data. A routing algorithm is also responsible for checking that the link between nodes is active. If a broken link is found, the algorithm will try to find an alternative route. This is a very brief description of the procedure. It is obvious that each routing algorithm uses a slightly different method of routing for various functions.

Currently in this field of research there are many algorithms. Some of them are very basic and others are hybrids. For our research, four classic popular algorithms have been studied. Destination Sequenced Distance Vector (DSDV) [46], the Ad Hoc On-demand Distance Vector Routing (AODV) [31], [43], the Dynamic Source Routing (DSR) [52], [54], and Temporary Ordered Routing Algorithm (TORA) [56]. Before going into more details, we will review the types of routing algorithms as mentioned previously.

2.4.2.1. Destination Sequenced Distance Vector

Destination Sequenced Distance Vector (DSDV) is called a table-driven routing protocol. Each node keeps one or more storing tables for routing information such as destination IP and next hop. The tables are always updated periodically, by propagating updates through the network, in order to store a consistent network view. This is easy to achieve because each node periodically broadcasts its routing table. It is clear that a broken link can be detected if no broadcast has been received for a while from surrounding nodes. Because in DSDV the nodes periodically send packets with routing table information [48], it is obvious that in large networks, the bandwidth consumption is increased as more information needs to be broadcast about routing. This is a major disadvantage compared to other protocols as in a large scale networks bandwidth is important and should be reserved for data or voice transmissions. Especially in cases when a network is stressed due to large overhead information it is clear that if any new nodes try to join the already congested network then they will fail as they will not be able to exchange routing information with neighbouring ones. In a scenario that a node represents a user trying to send a message fast while in the need of help, if the user cannot join the network to send the message immediately then we can consider that as a disadvantage. Thus, for fast communications we have to use a protocol which is fast at filling the routing table with the routing information. Convergence is a term used to define the time a node needs to settle down, and know the routing information of the surrounding environment. DSDV has a period of convergence. This period may cause delays in the network. Delays may cause packet loss and a message can be lost because if there is no information on routing, the packets will be dropped. The value of the convergence time is not easy to determined [51] and to adjusted, which causes more delays and more bandwidth consumption. Moreover, there are two similar disadvantages for using DSDV compared with other routing protocols. According to H. Bakht [50], DSDV performs worst when the mobility of the nodes is high especially in large networks. It is inevitable that in scenarios like emergencies, high movement speed is expected as well as massive movement of nodes in random directions. DSDV uses a technique for choosing the shortest path for the route, which is very handy for our scenario at first sight. This procedure though is complex and can cause delays when a link fails (broken link), [46]. DSDV can delay a node re-entering the network as it takes some time to recalculate the routing tables, discard old routes and update the sequence number on each node. Speaking of the sequence number, we can consider it as the state of current routing information written in a table. When the routing information is updated, the sequence number is increased and the old one is discarded. In terms of security, DSDV assumes that all nodes forming the network are trust-worthy. If a malicious node wants to attack the network it will be easy as there is nothing to stop the node from sending data on the network. In case a designer considers security, this parameter must be investigated and if it is needed to have a secure network this is not the appropriate protocol to choose.

As a conclusion to DSDV we may consider this routing algorithm as a fairly good algorithm for small networks with low mobility and low bandwidth requirements.

DSR and AODV can be described as source-initiated on-demand routing protocols. Obviously, this type of protocol creates routes only when a source node's packet needs routing. In comparison with DSDV, this is better as it can save bandwidth and perform optimally in terms of how fast the routes are discovered.

2.4.2.2. Dynamic Source Routing

In the DSR [50], [52], [54] protocol, the source first determines the complete route for sending the packet. It determines the routing path and updates the routing table of the nodes while the packet is travelling through the nodes. DSR uses a technique also known as hop by hop route to destination. Data packets are being sent to this destination from one node to another. Each packet always includes information in the header about the sender. The protocol involves two main techniques for route discovery and route maintenance. The route discovery operates when the sender has no prior information on the recipient. The procedure is repeated from all nodes sending request packets (RREQ) to surrounding nodes until the destination has been found. The recipient sends back a reply (RREP) packet which finally arrives back to the sender. The information is stored at the sender for future use. This procedure is typical for this type of protocol when it is trying to perform a route discovery.

The route maintenance procedure checks if there is any broken link. (e.g a node is out of range) and notifies the sender. If a broken link is found, the sender is aware of it and removes it from the route cache. When a new node is introduced, the network checks immediately if it can route it via the newly introduced node. If this node discovers a shorter route, it will transfer the packet itself to the next node. It is obvious that casting reduces the overhead bandwidth at the expense of memory and CPU. DSR has a better mechanism for routing as it keeps information about the route aggressively in a cache. The fact that the route is discovered on demand may cause some performance problems, especially in large networks, as the route is aggressively maintained. In other words there might be a case of uncontrolled messages flooding the network as the node requests flood the whole network. Therefore a larger network will become congested eventually as the number of control and data messages is increased. DSR performs better than DSDV in terms of data delivery, routing, delays and mobility and in some cases better than AODV's routing discovery procedure [55].

2.4.2.3. Temporary Ordered Routing Algorithm

Continuing the review of the routing algorithms TORA, [56], [57], [58], has been found to be fast when establishing connections. Additionally, a form of prioritization exists as it has a mechanism to handle the long distance routes first and then the short ones. This is a very good approach as it uses less bandwidth than other routing algorithms. TORA is an on demand protocol. For the route discovery, it uses a similar technique to DSR, but it also creates a copy of all destinations in each node. In terms of delays TORA is fast as each node has the required information stored about the destinations. TORA assumes that all nodes behave as synchronized clocks and its mechanism for link failure discovery is based on time. TORA has been found to perform satisfactory for static network topologies. In large heavily loaded networks and especially when mobility is introduced, the performance is poor and unreliable.

2.4.2.4. Ad Hoc On-demand Distance Vector Routing

Ad Hoc On-demand Distance Vector Routing or AODV [31]-[45] borrows elements from DSR [52], and DSDV [46]. It has a similar mechanism for route discovery as the DSR, and uses the hop by hop method from DSDV. When a source node needs a route for a specific destination, it sends a message (RREQ) [31], to all surrounding nodes. This message is broadcast until it reaches the destination node. The recipient sends back to the source node a reply message along the reverse route. This message contains information about the forward (original) route and it sends the information to all nodes involved in the transmission. Each node only keeps the next hop and not the complete route. In the AODV protocol, the routing table is only available for active nodes. We call a node active when it is related with one or more transmitted packets. Broken links are detected via transmitted messages or link layer detection. A broken link will be marked unreachable from the surrounding nodes that are involved in the transmission of the packet.

Route discovery is essential before establishing a communication link. Nodes are sending a request message to all neighbours. Surrounding nodes will forward the message until it reaches the recipient. When the last one receives the message, it broadcasts back an acknowledged (ACK) [34] message to the original sender, through the active path. The active path is actually the path that has already been established between the sender and the recipient. If the path is broken the ACK message is discarded, and an error message is sent back to the original sender. Then the procedure starts again. When the source node receives an ACK message then it knows that a path has been established and the transmission of the packets is starting immediately. A great feature is that it can discover broken links fast. AODV uses "hello" messages in order to check the status of surrounding nodes. If a node fails to reply "hello" within a predefined period of time, it is considered a broken link. Sometimes this can cause overhead bandwidth locally. AODV has been designed for networks in which mobility, speed, and many other parameters are changing continuously.

AODV has been found to use relatively small amounts of memory and less CPU power. It can discover broken links fast which means that we will have less delays due to retransmissions. Moreover it can provide both multicast and unicast connectivity through the nodes which is quite important as we will discuss later in the section reviewing voice communication.

2.5. Encoding and Decoding Techniques

Wireless environments rely on digital data flow transferred through the air, by using radio frequencies. When a user is speaking through a phone to another one, the analogue sound wave (voice) coming out of his/her mouth is digitised by a sophisticated piece of hardware. In other words, the analogue signal is converted to digital one and travels through the pipe (communication channel) in the form of digital packets. This is known as pulse code modulation (PCM) [12], which is used to convert analogue signals to digital ones by using a special algorithm called a codec. The analogue voice is converted to PCM samples and is passed to the compression algorithm, which compresses them. The data is travelling through the network in the form of packets (packetisation). This is very useful in terms of transmission as the packets have predefined size and when received can be put back together in order to be converted again to an analogue sound that the human ear can interpret and the human brain can understand. Additionally this technique is used as small packets can travel fast and reliably. The two way communication is also known as End-to-End Voice Flow. (Figure .7)

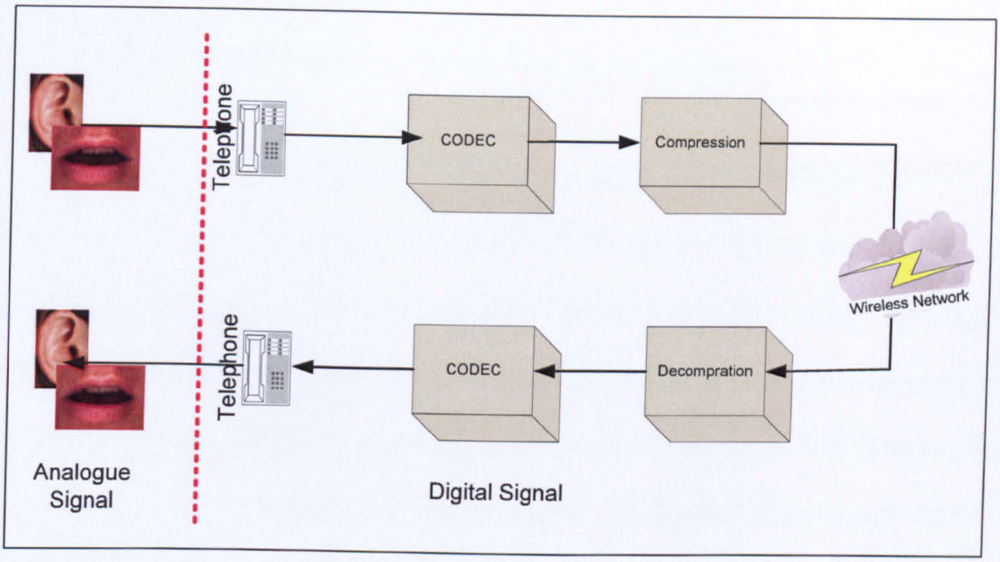

Figure 7. End-to-End Voice Flow

A classic problem during this kind of transmission occurs when delayed packets are delivered to the destination. As we will see later in the case when the packets are not there within a predefined time, one following another, there is a problem. The specific predefined delay time can cause a major problem. While there is an established voice call between two users, if a delay occurs speakers will not be able to understand each other. This happens because the final sound wave will arrive at the headphone distorted including noise and gaps. Obviously, this happens because the human brain cannot interpret discrete signals in the form of digital samples. The human brain is able to interpret analogue sound signals, such as sound waves (voice) only. Later on, we will describe this kind of delay as well as present a draft overview of how it can be minimized. Voice compression has been standardised and many codecs [12] are being used nowadays. A classic one is the G.729 [12]. Considering that those codecs are being used for infrastructure dependent networks, it is clear that in wireless environments, delays may affect the overall performance dramatically.

2.6. Mobile Networking in Voice Communications

Today modern mobile networks combine different technologies in order to have optimal performance as they have to handle real-time traffic without delays and problems in voice quality. Theoretically, we could say that such networks have been already built in such a way that they can handle all of the subscribers calling at the same time. This is partially true as users can call other mobiles or land lines at any time and can also send messages fast to anyone. It is clear though, that mobile companies provide those services also for economic reasons. Today all GSM services are under pre-paid or subscription contracts, as making a profit is the motive to build and expand such a company. Nowadays companies assume that only a fraction of their users will call simultaneously. Obviously if a company has a hundred users, it will not build the infrastructure in such a way that can accommodate all of them as in real life this is very unlikely to happen except in special situations. A company could not afford to design a network with such a huge capacity for obvious economical reasons. Many techniques and methods have been designed and applied in order to minimise the capacity and to use fewer stations placed in strategic geographical locations. Sectorization techniques [21], [22], the Caution++ project [15] and ASTIX [14] development platform for building mobile radio networks are a few of techniques that are being used in order to optimize traffic and network behaviour. This research of course has continued as the demand of traffic has increased.

The GSM network today operates fairly satisfactorily in normal situations and cases that can be overloaded. We can define as normal operation the everyday usage of the network. Peak times can be considered as the time periods that the traffic becomes heavier and occasional peak times when the network is stressed. We have a peak time for example in a city centre during a lunch break (12pm-1pm) when all people are going out of the office for a short break to have a snack and possibly to call someone. In this case the network can reconfigure itself in a way that can accept more callers. Occasional peak times can be considered at public holidays such as Christmas or New Years Eve. In the United Kingdom for example on New Year's Day 2005, the total number of text messages sent reached 133 million, which is the highest recorded daily total [30], and 92 million text messages were sent by mobile subscribers on Valentine's Day 2005.

The network behaviour is highly dependent on the dynamic conditions such as the traffic and mobility. It is very easy for the GSM to be congested or even to fail under certain circumstances. In order to understand the problem we will present the type of failures we expect in different conditions. The network failure can be described as partial or total failure depending on the area of affect. It is obvious that a partial failure (locally) is more often to happen, as a local hazardous situation is more likely to happen in a local area than to the whole country. We can also characterize the failure as a break-down in the wired or the wireless part of the network. In the following diagram, (figure 8) star 1 indicates a wired failure between the cell site and the base station controller. Star 2 is related with failures regarding the wireless environment between the host (mobile phone) and the cell site. Star 3 shows a break down in the Mobile Services Switching Centre. Finally star 4 represents a failure in the network routing the GSM calls to land lines through the Public Switched Telephone Network (PSTN). A conclusion to this section is that even the most sophisticated network around the world can collapse as it is based on infrastructure, which is vulnerable to physical phenomena, disasters or other hazards.

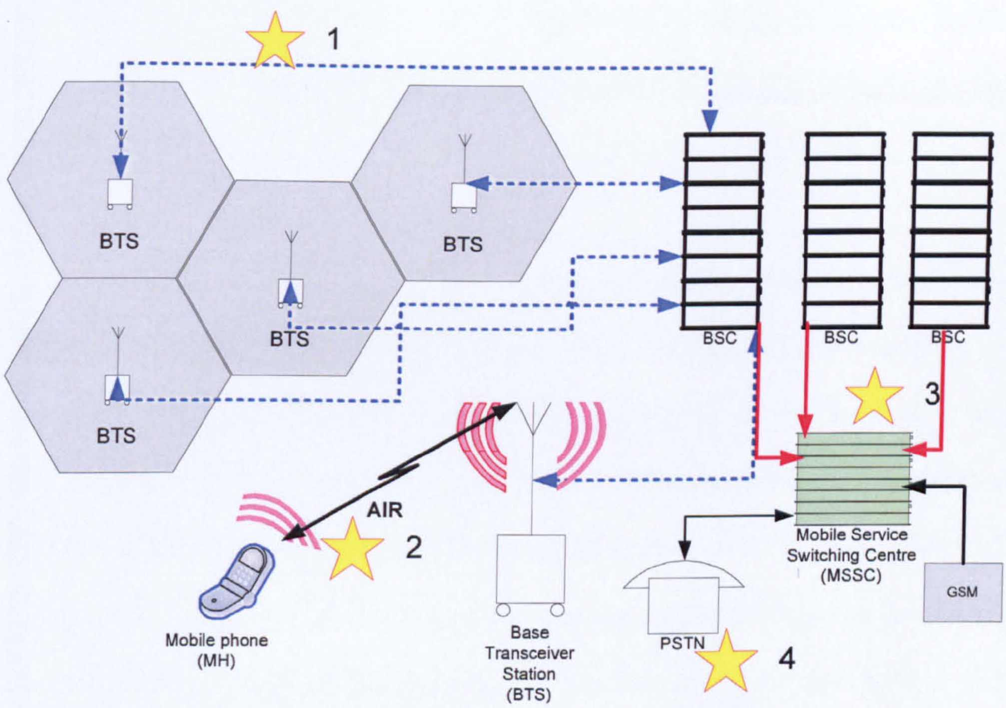

Figure 8. GSM Network and Failures.

Our investigation as we will see in the next sections describes the reasons a network collapses during the event of a hazard as well as related solutions that have been successfully applied and criticisms of each of these.

2.7. Disaster Recovery Techniques in Mobile Communications

Different organizations and funded research teams are investigating every day disastrous environments with the common goal of finding the optimal solution in disaster recovery in terms of speed, number of recovered services and reconstruction of the destroyed infrastructure. It is a fact that most of those projects have been a success but it is also a fact that many human lives were lost because of the lack of speed. In figure 12, we can observe that in the first days after the WTC terrorist attack the network recovery could not support users as the network was restored ten days after the destruction.

A classic network failure due to congestion has happened before several times. On the 7^{th} of September 1999 - (Earthquake Athens Greece) the GSM network was congested and overloaded a few minutes after the event. The traffic demand was increased up to 400 percent and the network was overloaded and failed. The network first blocked all incoming calls as it was designed to do in order to keep current calls active but a few minutes later interrupted all ongoing calls. The result was that mobile phone subscribers could not call or be called even four hours after the event, as they where trying to call relatives constantly. SMS messages could not be delivered as the network was stressed over the limits and in some cases, short messages were delivered eight hours later. Athens was left for several hours without GSM services in most of the areas covered by the cellular company. We cannot provide references as companies refused to publish them. As this is an extreme case but not the worst case scenario, looking at other events such as when Los Angeles was cut off from the rest of the world on the 17^{th} of January 1994 because of a massive equipment failure. After the event the telecom company developed an improved network including a prioritisation system for handling incoming, outgoing calls and a number of other operations [68].

The problem of congested networks though is not the only one. Disruption of service due to inadequate infrastructure support has been proven to be vital for the network. There have been situations in which areas suffered from electrical failures. Back in 2003 in NE United States, a massive power failure due to an earthquake forced the cell sites to work with backup batteries. The blackout last for more than 12 hours. The antennas had sufficient batteries for up to 6 hours. As soon as the backup batteries were drained, cellular services were disrupted immediately. Calls could not be rerouted and handover could not be done as most of the cells were inactive [68]. SMS could not be delivered as many users were within the affected areas.

Physical destruction of hardware components (network modules or cables/fibres) has been proven to be the most common reason for disruption as it is very easy for the infrastructure to collapse. Verizon introduced an agenda [17], in this report-meeting, in which wireless support to response and recovery efforts was one of the major subjects. Vulnerability of wireless services, critical issues and the future of emergency of wireless communications were among the subjects that were discussed. For the scenario presented in the meeting, the wireless operational concept is the following.

A mobile host initiates a call. The call is transmitted to an antenna which is a nearby mobile cell site. From the cell the call is routed to a mobile switch and then to another mobile user wirelessly or to a landline. During the 11^{th} of September 2001 World Trade Centre (WTC) destruction, many mobile cell sites (trucks) were placed

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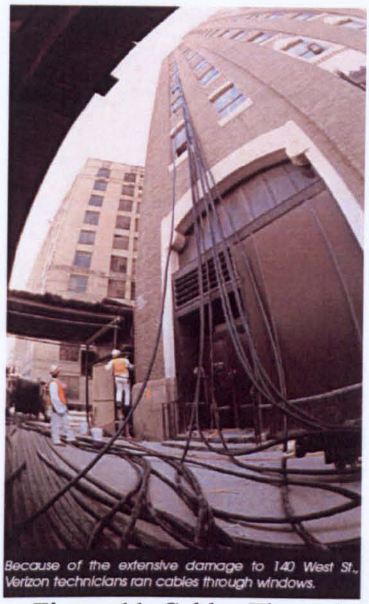
in strategic areas replacing the actual congested or destroyed cell sites. Those trucks were able to accept calls and route GSM data to stations. [17]

Figure 9. Verizon's Deployed Solution [17]

While the attacks damaged or disrupted some 200,000 voice lines, 100,000 business lines, 3.6 million data circuits and 10 cellular towers, Verizon and Verizon Wireless re-built much of the telecommunications network. Furthermore, they managed to recover everything back to normal in lower Manhattan within a few days.

Figure 10. Mobile Telecom Equipment After the First Days of WTC destruction [70].



Because of the extensive damage to 140 West St., Verizon technicians ran cables through windows.

Figure 11. Cables Through Windows

A different approach in order to recover physical damage was to reroute more than 500 massive copper and fiberoptic cables that either originated or were routed through the windows of West Street building [70].

The project was a huge success for the company but still there was no support during the first minutes or the first couple of hours after the WTC attacks.

The following diagram illustrates the progress of the recovery of the communications in terms of restored sites, mobile cells and radio frequency.

Figure 12. Restoration of Cellular Telecommunications [70].

2.8. The Future of the Next Generation Cellular Mobile Networks

As cellular mobile networks evolve, new achievements are presented day by day involving integration of mobile networks with wireless local access networks. University of Athens [92] has published a paper in 2005 for the next generation of mobile communications (referred to 4g), which will be based on infrastructure involving wireless and wired access. Their scope is to keep the network connected even when there is a local failure. Their implementation is based on UMTS/WLAN and their primary scope is to use and implement any cellular, satellite, WLAN-type systems, short range connectivity or even wired systems like ADSL in order to create a common platform in which the cellular network will always be connected. Their model is based on infrastructure but it has an advantage compared to the previous cases. They have a range of networks to choose from, in order to keep communications alive. This makes their proposed solution evolutionary but as this approach is based on infrastructure a physical phenomenon like an earthquake could still destroy the basic part of the network (access points, BTS, BSS and UTRAN) and cause corruption of communication or even worse a network failure.

University of Louisville published a similar research [93], which presents an architecture for a heterogeneous multi-hop network. The objective of their work is related to the integration of cellular networks with WLAN's and MANET's. The architecture is also based on infrastructure, which may result in failures, as discussed above. In this paper, the authors have also presented a number of different approaches, which are infrastructure dependent.

Obviously wireless technology is promising for many different applications and a huge scientific force is investing time and effort to use it as it has many advantages and can be used in cellular networks. Recent studies have focused on wireless mesh technologies [94] involving numerous different protocols and network types. As the need for communication is increasing, more researchers are motivated to study in this field of research.

In the previous sections, we presented the survey and investigation that has been done related to our thesis. GSM technology has been briefly presented to understand the basic components that are mentioned in the thesis. Similarly we have presented a survey on related wireless technology, focused on 802.xx standard as this is the one we will use. The main characteristics as well as the properties of voice communications have been presented. Finally, we have discussed real scenarios that have been applied in real life situations. All of the scenarios we presented in this thesis have been successful solutions, in order to recover infrastructure, although all of them took time and a huge work force to be accomplished. Our concerns though are different. The main aim of our research is not to develop one more disaster recovery system, method or a solution in order to recover infrastructure fast. We are interested in a different approach. What we will present in the next chapters is an fast deployed network independent of infrastructure. The idea behind it is that in time of need, people will immediately be able to communicate using mobile phones in the hazardous environment during a disastrous scenario when the infrastructure has collapsed.

2.9. Summary

This chapter summarizes the survey of the related work through our studies. The chapter presents the main components of the GSM architecture in order to give a basic understanding of the main components of the technology to the reader. An introduction of wireless networks and the main concepts of the 802.11 technology are described, as it is relevant to our field of research. Ad Hoc networking is then presented describing their applications, main properties and their evolution. Routing algorithms follow as they are of major importance for wireless networks. The chapter continues by presenting related sections in voice communications, and their parameters as encoding/decoding techniques. The chapter finishes with a survey on related work on disaster recovery in mobile communications as well as the evolution of new cellular networks.

CHAPTER 3. An Architecture of an Fast Deployed Network in Hazardous Environments

In this chapter, we will review the main properties of a hazardous environment and the conditions that affect telecommunication networks. Furthermore, we will discuss the challenges which are associated with such a network and we will present the basic model of an fast deployed network.

3.1. Introduction

This research focuses on the behaviour and the properties of an fast deployed network in hazardous environments. This network has specific characteristics, as it is not dependent on infrastructure. Mobility is one of the parameters affecting this network. People in the affected area most probably will be on the move away from it, while they are asking for help. Some of them might not be able to get away, as for example, if there is a fire on the ground floor of a 7 floored building not allowing access to exits. Having a closer look at hazardous environments, it is clear that we suffer many losses at the time of the event as well as a few minutes or hours after the event has started. Fires for example may take a couple of hours to burn everything; people trapped in buildings have limited oxygen supply. The question that arises is: what if those victims in affected areas had a way to communicate with others and ask for help even if there was no telecommunications infrastructure?

In the next chapters we will present our work in steps starting from thoughts on how to simulate the disaster environment using a simulator tool. The operation of the network, the basic properties and components of the network will be presented next. Then we will establish basic models of communication between mobile hosts (nodes). Furthermore, we will simulate two networks; one for sending messages between users and a second one for voice communication. Finally, we will discuss the main parameters affecting such a network and we will point out important parameters that will be included in simulations within the next two chapters.

3.2. Hazardous Environments and Conditions Affecting Telecommunication Networks

During the last two decades, telecommunication networks have been proved to be vulnerable to physical disasters. Partial or even complete loss of telecommunication services has been proven to lead to massive damages in properties but most importantly to preventable loss of human lives. In many cases, those accidents have been described as the worst-case scenarios with whole cities cut off the rest of the world without communication services.

Observing the characteristics of those disasters we can easily define three types of failures: network failures due to congestions, disruption of service due to inadequate infrastructure support and physical destruction of hardware components (network modules or cables/fibres). Additionally in the case of a hazardous event, we must take into consideration the fact that the dynamic environment changes rapidly. The following figure illustrates the WTC attack and indicates with different colours the damage the area suffered.

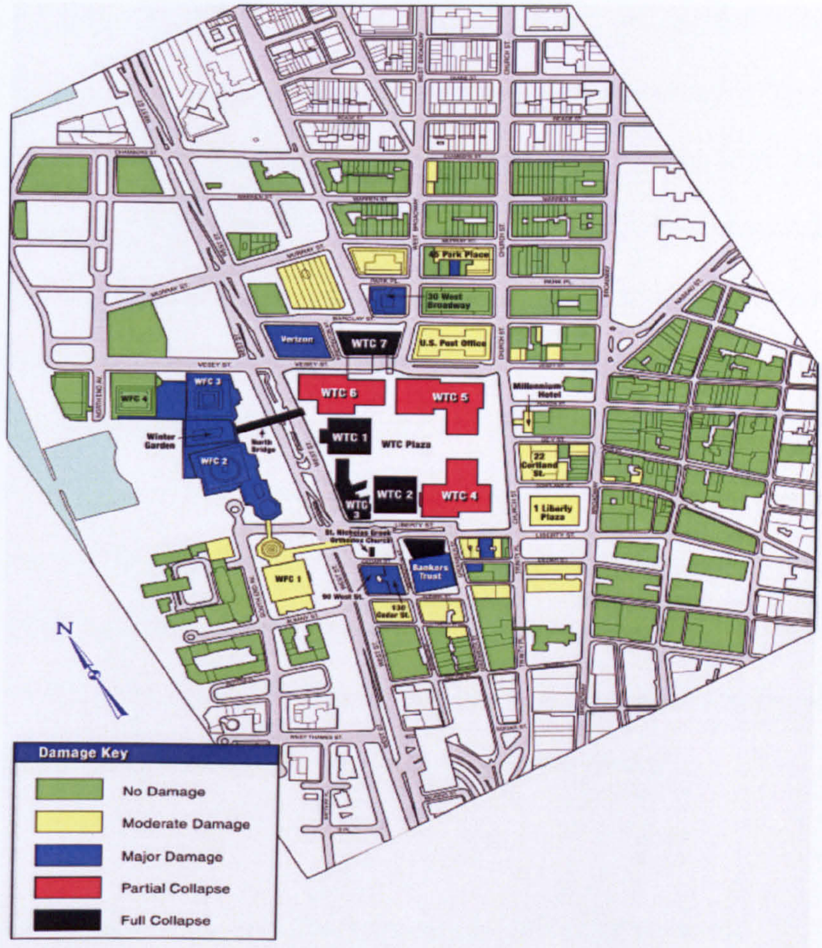


Figure 13. WTC Attack – Damages

Three types of disaster can be easily identified on the previous figure as in areas noted by black colour the infrastructure has collapsed. Avoiding the risk of repeating ourselves here, we present quickly a case of disruption of service, which may occur in green areas. Most of the circuits, data and power lines destroyed were passing under or inside the WTCs buildings. Congestion occured in yellow areas (suffering moderate damage), if any of the cells survived the attack, as all traffic from mobile phones within the area, was redirected through the remaining cells overloading the network. It is obvious that the capacity of the network is important in these situations.

In scenarios like the previous one, mobility is the parameter that changes dramatically. It has been observed that a panicked crowd tends to run in big groups away from the hazardous areas. This fact leads to a massive fast movement of mobile hosts in different random directions. This has been implemented in our simulation. A random start position has been assigned for each mobile host for initialization. Random mobility paths have been assigned to people nearby the area as well as random speeds (5-10m/s simulating walking or running). No specialized algorithm or mathematics, has been investigated for initial positions or movement or people moving around the map. The simulator used during the research can be adjusted from the interface providing the user with a variety of options and profiles about random mobility and many other parameters. It is a fact that the more devices introduced in the network within the affected area, the bigger the risk of congestion in the network, due to interference, retransmission and broken links.

3.3. An Architecture in Mobile Communications Recovery

The idea of the proposed network is based on the following fact. If the infrastructure collapses, users will use their mobile phone to communicate no matter if there is a GSM service or not. When mobile phones are not able to communicate with the surrounding cell sites they form a fast deployed network with each other, as might happen in the event of a hazard.

The architecture will provide an Ad Hoc network which is overlaid on the damaged infrastructure to enable two modes of emergency communication. The proposed network will provide the users a messaging service or a more complete voice communication solution. This architecture exploits the inherent ability of GSM messaging infrastructure and through the use of 802.11 technology from computers and adequate PDAs the ability to build an Ad Hoc network that is capable of voice communication. The integration of those two components gives the hazardous area a two mode communication system.

The benefits of this novel approach are the following: 802.11 can fast form an Ad Hoc network, which can be used without any infrastructure support. The network can handle routings on demand and send or receive data to any host. In the case when a middle link in the path is switched off or goes out of range, the network will recover the transfer by using a different route. Those unique characteristics answer the challenges of having a fast deployed network, which provides fast communication in hazardous environments at the time of the event. Additionally, by using a suitable routing algorithm it is possible to recover links and keep the network connected as the 802.11 architecture can recover from connectivity errors. Finally, the network can be deployed without the need of engineers working in the affected area. This obviously minimises the risk of loss of human lives.

At the early stages of this research, we were trying to find a suitable way to form the proposed network by using only GSM devices. Unfortunately, GSM mobile phones cannot communicate directly with each other. They only transmit and receive data through cell sites. The first proposed idea was to modify the core software of the mobile but this causes a new problem. Obviously modifying the software of a mobile phone is not a problem. The actual challenge is that changing the software of the phone would also require us to modify basic tuning circuits. Furthermore, changing the tuning module of the mobile means a further modification on the cell sites. Eventually, base station controllers and mobile switching centres have to be changed as well. Finally, vendors will have to agree to a new protocol for communication, change the entire infrastructure, and of course, subscribers will have to replace their handsets. We do not say that this cannot be happen, but this is not our aim. In this research we are not trying to modify devices. It is not our goal to design a full product as a new solution, which replaces current technologies. Our scope is to benefit from existing technologies and use them to form a new network, which can operate in an emergency. Furthermore, we are trying to combine two technologies to achieve communication instead of just switching to one or another depending on the current status of the network. Finally, it is our aim to deploy a network fast without the need of restoring infrastructure or using an engineering work-force. The next figure presents the proposed architecture including all details about the related technologies involved.

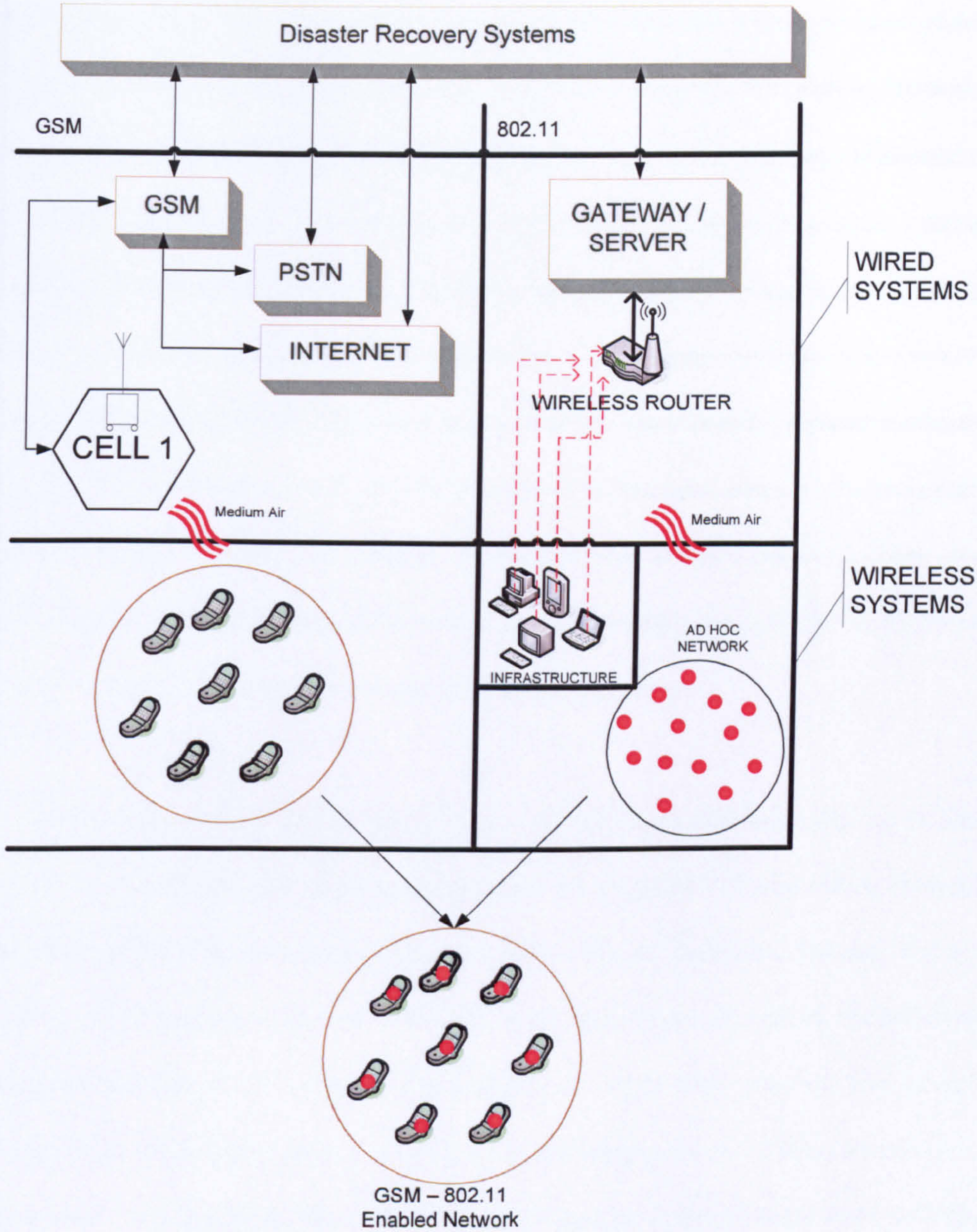


Figure 14. Architecture for Mobile Networking in Hazardous Environments.

Continuing the investigation on current technologies, we found many similar technologies in the field of wireless networking. The new challenge was to choose a suitable technology, which can be combined with GSM in order to form such a network. The solution came through our studies in wireless networking. In the early stages of course, it was difficult to choose one of the existing network technologies but everything became clear at the time we understood the main properties advantages and disadvantages of each one. Bluetooth was a trend for the market at this period, but it lacks speed, network deployment, and range. It has a big battery consumption and will cause a problem if we try to connect for example a thousand people. Bluetooth [65], [66], [67] has been designed for quick communication in handheld devices for exchanging files. Infrared technology was also examined but it lacks one major parameter, it needs direct optical communication. In other words if the LED's are not aligned the communication may be disrupted or become slower. Even worse if an obstruction passes through the sensors, communication breaks occur between the two devices. Finally, after an investigation on wireless technologies the one that was agreed to be used was the 802.11 standard as it was best suited for our scenario.

The structure of the architecture (figure. 14), combines GSM and 802.11. In the case of a network loss the devices become transceivers by using 802.11 and GSM technologies. All the routing traffic as well as messages and voice calls are redirected through 802.11. The network is no longer dependent on GSM cell sites. It is deployed fast as an infrastructure independent network at the time of any hazardous event. This network has advantages compared to existing ones, as any hazardous event cannot harm mobile communications between users. The benefit of the network is that it can be deployed anywhere, quickly and can provide users with messaging and voice capabilities. In the next few sections, we will discuss the components of the network, its operation and configuration in order to build it and evaluate it.

3.4. Components of the Proposed Network

The proposed network is formed from mobile devices only. Such a network in wireless terminology is called an Ad Hoc network. An Ad Hoc network is a group of wireless mobile nodes that can form a network "on-the-fly" which is changing dynamically. Because of its nature, the proposed network is operating in a dynamic environment which is independent of infrastructure. Furthermore, it lacks the power of servers and wired connections, which offer significantly more resources in terms of bandwidth and data transfer than a wireless network, which is based on mobile devices only. For the proposed architecture, the AODV routing protocol will be used since it has been found to be the most suitable routing protocol from the survey conducted in the previous section because it is based on routing on demand and uses less bandwidth for the proposed network. AODV can operate well and has many advantages compared with the other routing protocols as discussed earlier on the survey. Further discussion for this choice will be presented in the next section, in which we will analyse the reasons for choosing AODV as the most suitable routing protocol for the proposed network.

It is obvious that in this environment each mobile node operates not only as a host, but also as a router which is able to forward data packets to other nodes in the network. In the next few sections we will analyse the various components of the proposed architecture in terms of hardware components, operation of the network, experimental phases, implementations and a discussion on decision making for choosing parameters of the network.

3.4.1. A Network for the Proposed Architecture

The diagram below illustrates the operation of the proposed network. The first phase of our research is focusing on message exchange between users. The second phase is a bit more complex as it involves voice communication between the mobiles. The idea of transmitting a short message over 802.11 between GSM phones is not an easy task. 802.11 networks not only lack specific GSM services such as localization and authentication, but also lack a specific protocol for handling communication between GSM and 802.11 and between neighbour nodes.

Figure 15. Communication in Areas Suffer Loss of GSM Service.

In the case of loss of GSM service, the mobile phones forward the SMS message to the 802.11 network. The messages then travel through, until they reach their final destination. When a message is received by the 802.11 module, the latter forwards it back to GSM device. The user will use the phone the same way he used to use it for sending or receiving an SMS message or calling, as the process will be completely transparent to the user.

3.4.2. Routing for the Proposed Network Architecture

Based on our survey and our knowledge of the properties, advantages and disadvantages of the routing algorithms, we have chosen to use AODV. AODV has been designed for networks in which mobility, speed, and many other parameters are changing continuously as its focus is primarily on recovering transmission errors fast. DSDV is not the optimal algorithm for our demands as it is designed for small networks with low mobility and low bandwidth requirements. DSR has a better mechanism for routing as it keeps information about the route aggressively in cache. The route is discovered on demand but this will cause performance problems as the route is aggressively maintained. DSDV is a table-driven routing protocol and broadcasts messages periodically in order to maintain the routes. Node requests may cause delays because of the uncontrolled messages flooding the whole network. Therefore, a larger network will become congested eventually as the number of control and data messages is increased. Obviously, DSR performs better than DSDV in terms of data delivery, routing, delays and mobility. TORA requires a large amount of memory and this is not desired for our case as we are trying to combine two technologies in one device without stressing it. Stressing the CPU will degrade the performance of the nodes in the case of a large network topology. TORA has been found to perform satisfactory for static network topologies. However, in large heavy load networks and especially when mobility is introduced the performance has been found to be poor and the performance unreliable. In a fast deployed network with high mobility as all nodes are in a hazardous environment TORA is not the right choice as it lacks in mobility performance. Considering the device's memory this might be a problem as low memory devices can affect performance especially if they are transmitting voice. Considering an emergency situation, things might get worse, because in a hazardous environment users might need to contact others extensively. In this case the network can be congested for a short time (during initialization) because of the high volume of nodes trying to communicate but will recover in a short time.

We may agree that AODV is the best choice according to our research as we do not intend to compare protocols by simulating them, or to implement a new one for simulating our case study.

3.4.3. Mobile Hosts and Network Setup

In real life a host or node or a wireless device as it is called can be a laptop, a desktop personal computer, a palmtop equipped with wireless interface or any device that can communicate with wireless networks. As has been mentioned in the survey the 802.xx services are transparent to the user which means that, regardless of the device type, the services for all 802.xx devices will be the same. Such devices that combine GSM and 802.xx services currently exist in the market. One of them for instance, could be the I-mate™ K-Jam. This device has a sufficient processor (200MHz) and memory as well as Ad Hoc capabilities and it could be used for a real experiment of the proposed architecture. The proposed architecture will not be tested using real equipment for practical and technical reasons, though it will be implemented on a simulator called Opnet using the Opnet modeller 11.5 [78]. This simulator tool will be presented analytically in the next section. For now we will mention that any mobile host can be implemented in any environment as Opnet has the option to choose the model of the host and its characteristics.

Considering a host with two interfaces GSM and 802.xx we assume for the proposed network that the GSM and wireless hardware module are interconnected through an interface. This interface is not going to be explained in terms of its electronic functionality, as this is not our purpose. We assume that the interface is able to use the data encoding/decoding and exchange information between 802.11 and GSM domain as shown on figure 16.

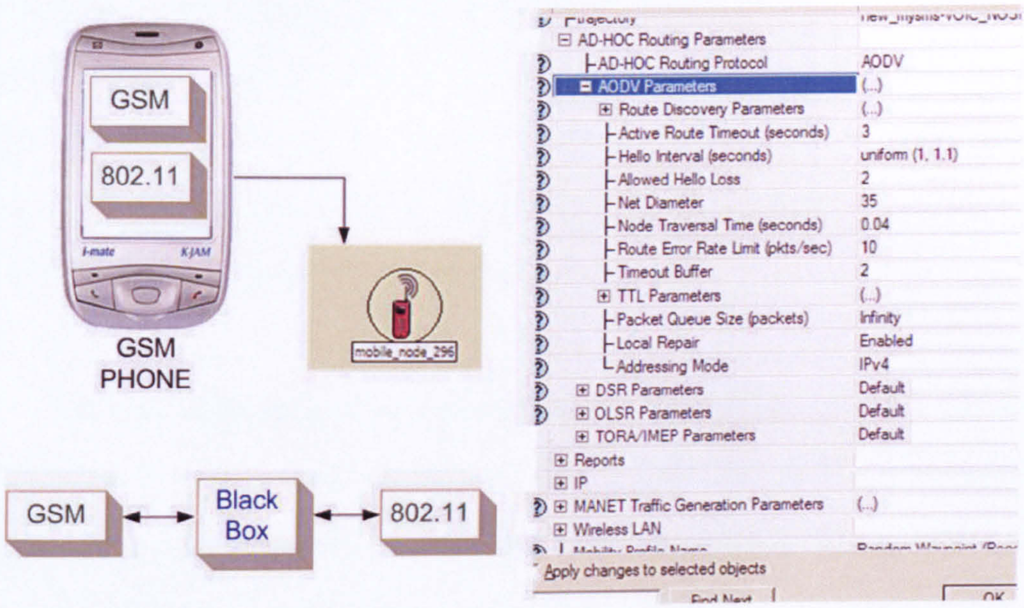


Figure 16. Mobile Node for Proposed Network.

The information which is transferred from one domain to the other can be translated between the two domains. 802.xx hardware is able to access all other modules such as the SIM card, phone books, display functions and many more. It is important to mention that if the device was a final product for emergency networks the only important thing is that the user must be able to use the phone, send SMS messages or speak, without worrying about connectivity technologies and parameters. Finally, it is worth noticing that the node can be customized according to the needs of the researcher (Figure 16 right hand side), and can be monitored and perform many operations.

3.4.4. Implementation of the Architecture and Network Configuration

In order to test the proposed architecture and evaluate the network, a simulated environment has to be created. As has been already discussed, the environment in hazardous situations changes dynamically. The optimal setup for a simulation involving hazardous environments could be considered one that could force a device to fail for example due to fire because the temperature is too high. Unfortunately, even though there is an option for a node to fail at random time in Opnet it is not possible to simulate fire, smoke, dust conditions that could affect transmissions realistically. Nevertheless, there are many other major parameters that will be configured for the implementations.

The place of the incident has been defined as a standard map of 3Km x 3 Km The simulation takes place in a real map of a part of Liverpool's city centre downloaded from Google Earth and the hazardous event takes place in Byrom street in our school building. For this simulation we assume that people in the school for some reason, cannot access the entrances and are asking for help from people outside in the nearby area who are friends and rescue services. The landline telephones are not operating and there is no GSM coverage in the area. The only way of communication is our proposed method.

Figure 17. Environmental Map – Green Lines Representing Mobility and Blue Lines the Communications Between Nodes.

In figure 17, we present the environmental map in which people have been placed randomly in the hazardous area as well as away from the event. Random mobility and speed profiles have been assigned to them. Victims have been limited to move within the school area. The crowd can move freely anywhere, as in many cases some people run away and others are moving close to watch the event. Rescue services relatives and friends of the victims can move freely in the map but the final point of their movement is close to the area of the event. For the implementation of the network 300 people (nodes) have been placed randomly in the map. 10 victims have been marked with a yellow colour and their IP's are monitored by probes. 10 random nodes have been marked with red colour representing friends and relatives. Finally, special icons have been used for the rescue services for visual representation in future developments.

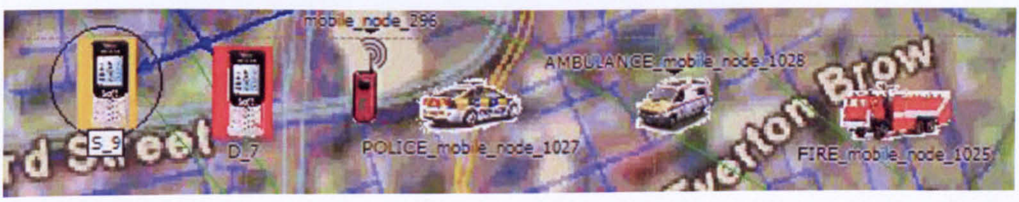


Figure 18. Different Nodes Used For Identification Purposes

In real life situations, we have to take into consideration that the medium of communication is air, which is not the best one as there is interference everywhere and other conditions may affect humans and electronics such as dust, smoke or high temperature. In this scenario, we assume that the 10 victims are alive and able to send a message or speak through a mobile phone and the simulation lasts 5 minutes. Later in the evaluation chapter we will discuss the results more analytically. The reason for setting up only 300 people on the map is technical as well as the simulation time. The more events the simulator has to "run" the more time and capacity of a hard drive it needs.

Our intension is to fast deploy a network and see how it behaves during the first few minutes of the event. More details on the simulation will be given in the following chapters. Additionally, we will present the results and the evaluation later and we will not talk more about the parameters now. In the next two sections, we will present the two different phases of the research, which involve message and voice communications.

3.4.5. A Message Based Network in Hazardous Environment

The network involves messaging, and has scope for fast and reliable message delivery. Message delivery needs fast communication between the nodes. The procedure for message transmission is to allocate a channel, transmit the message quickly and release the channel for other users. As we will discuss analytically later in the respective chapter, this can be achieved by transmitting a burst containing the message. In wireless networks, the more transmissions and data exchange we have in the network the bigger is the risk of interference. By sending messages fast and releasing the channel immediately, we minimize the number of uncontrolled messages flooding the whole network as well as the interference. Furthermore, we minimize the risk of delay in the network, which is caused by collisions. It is worth noticing that in wireless networks retransmissions may be observed when two nodes are transmitting simultaneously.

3.4.6. A Voice Based Network in Hazardous Environment

Voice communication on the other hand is much more complex. The idea is similar to the message transmission, but this time the data that will be sent through the links will keep two channels forming a bidirectional link established for as long as the call lasts. It is obvious that for voice communications more parameters such as delay will be evaluated. In voice communications, packets must arrive at the destination within a predefined time. This will ensure that the listeners will be able to understand each other. The quality of the voice will be dependent on a specific codec that will be used according to the GSM specifications. In other words the codec that will be used is already implemented and tested in most of the GSM mobile phones on the market.

For the scenario, all discussed parameters will be evaluated and the collected statistics will be presented and criticized. As this is an emergency network, before presenting you the networks or results, we will first introduce the basic idea on how the emergency network can be deployed and initialized. This is an Ad Hoc network and some of the services existing in infrastructure networks will not exist in the proposed one. This is not considered a disadvantage.

3.4.7. Emergency Mode Operation for the Network.

As we have mentioned previously, we define an "Emergency Mode Operation" as the procedure in which a mobile phone switches from normal GSM operation to the 802.11/GSM. This procedure is transparent to the user. The system works as presented in the following diagram (figure 19). The mobile periodically probes other mobiles in the area asking if they have a GSM service. This is done through the 802.11 network. As soon as nearby mobiles give a positive reply back there is no need for changing from one mode to another. If a device has no signal and receives a negative feedback from other mobiles in the area (that they also have no GSM service), the device initiates the emergency mode operation. Then it switches to 802.11/GSM technologies, which will be used to transfer all data to and from other mobiles. The probes can be active while the GSM hardware is on or can be initiated as soon as the device "senses" loss of GSM service. This option actually depends on the developers of any future vendor who would like to implement the routines. From our knowledge though, we propose that the extended use of wireless hardware is not healthy for the battery. It is a fact that 802.11 has high battery consumption. Researchers are investigating the issues of power consumption and battery life. At the moment they have already announced new battery types able to provide 60% more duration than the batteries that mobile devices are using today. Battery life and power consumption are important for the devices. The uncontrolled usage of the wireless hardware module of the device will drain the battery quickly. This will cause the device to switch off. If this happens, people in need of help will not be able to communicate with others.

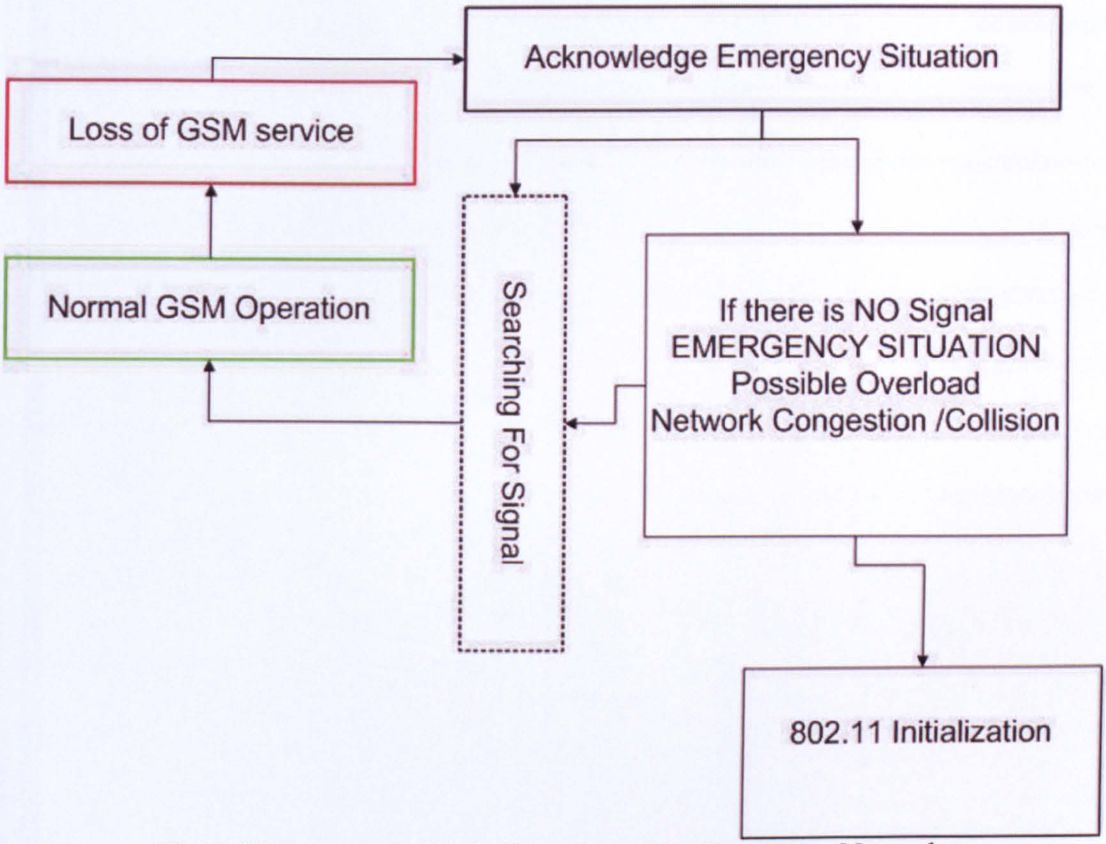


Figure 19. Emergency Mode Operation of the Emergency Network

This schematic in figure 19 explains the basic scheme of how to initiate the emergency mode or go back to normal GSM operation. As we will discuss later on the future work chapter, all implementations are related to GSM and 802.11 technologies.

3.5. Summary

Presenting the related survey of the components and the properties of voice communications, we examined the elements that can coexist in order to present an architecture of an emergency network that meets the challenges and can use different components in order to become a fast, reliable and stable network. In the first sections of the chapter, an analytical discussion surveys the main properties of a hazardous environment to identify problems and conditions. Then the architecture and its novel operation are presented. The proposed network is explained in terms of its main components as well as its advantages. The basic network model is also presented and a detailed presentation of its operation is followed by a survey on choosing appropriate routing algorithms. The decision making section is followed by the components of the network, the emergency mode operation and other considerations.

CHAPTER 4. A Message Based Network in Hazardous Environments

4.1. Introduction

In this chapter, we will present the global network operation in hazardous environments. The network will be deployed in a short period of time, providing the users with messaging capabilities. This will allow the users to exchange messages when the GSM network is not operational. The architecture of the network provides the users with both messaging and voice capabilities as discussed before. It can be deployed when there is no infrastructure, as it consists of 802.11 nodes forming an Ad Hoc network. In this chapter, we will explore and analyse the messaging component of the architecture.

The architecture of the network will be implemented in a simulator for testing and evaluation purposes. This will allow us to collect statistics about the network behaviour. In the first few sections of the chapter, we will look into a prototype model which will give us knowledge and valuable information about the communication between the nodes. Then we will present and investigate the global network with added messaging capabilities. For the implementation of the scenario we will use Opnet 11.5 in which we can parameterize many parameters affecting wireless networks such as random mobility, random speed of the nodes and many more. The evaluation of the both prototype and the global network will be presented next, followed by the discussion and a comparison with systems from similar studies.

4.2. Criteria required for a Succesfully Working Network under Hazardous Environments

In the section we summarize the criteria required for a successfully working network providing messages or voice communication, under hazardous situations. Each criterion will be explained within the next two chapters, by analyzing it and demonstrating it through the experimental procedure. The criteria are the following :

i. The deployed network must be able to support users in hazardous environments instantly after the deployment. Victims must be able to send messages immediately after the beginning of a hazard to relatives, friends or rescue services.

ii. The devices forming the deployed network must be able to communicate directly without using the infrastructure. The latter has a high risk of being physically destroyed in a hazardous situation.

iii. The deployed network must be able to provide messaging or voice services without using the GSM infrastructure and services.

iv. The maximum delay of the arrival time of a message (delivered to the recipient), must be same or smaller compared to the GSM's network delivery time.

v. The node capacity of a single node must be same or better than the capacity of the GSM network. By the term node capacity we mean the number of messages a node can handle within a predefined period of time (No of messages / second). The deployed network is formed of 802.xx enabled nodes.

Each node must be able to process the same number of messages such as a GSM mobile phone can handle.

vi. The capacity of the channel between two nodes must be sufficient in order to have message transmission from sender to recipient. Additional capacity in terms of bandwidth must be free for other messages that might be forwarded through these nodes.

vii. The network must provide connectivity for voice calls. Voice calls must be established between two users and the nodes must provide good voice quality.

viii. The devices involved in voice calls must be able to transfer data through 802.11 network without stressing the CPU. If this happens, we are expecting a degradation of service, which will result in bad voice quality, delays or even worse a corrupted voice call.

ix. In case of a local congestion, due to heavy traffic, interference or collisions, the voice network must recover quickly and restore communications.

4.3. Prototype Model of the Message Based Network

The main operation of the network has three main objectives: Connectivity on demand between the users, messaging capabilities and voice calls. As has been explained earlier the concept of the messaging service can be performed by allocating a channel quickly, sending a message and immediately releasing the channel as soon as the transmission ends. This will result in faster channel reuse by different users in a predefined time. The more channels available in the network, the more connections can be established between users, thus the more messages we can send. The prototype model will be evaluated through a scenario in which we will establish communication between three nodes. The scope is to illustrate the communication between two nodes via a third one to achieve fast and reliable communication between them and to evaluate if the type of communication is suitable for deploying the global network.

4.3.1. Experimental Procedure for the Prototype Model

Before setting up the experimental procedure for the prototype, we assume that the GSM service is lost, the infrastructure has failed and the nodes are aware of the emergency situation. For the implementation of the traffic between nodes in the simulated scenarios different distributions have been used. Distributions are used in order to generate specific transfer starting times for data packets, their sizes, the repetition of their transmission and their periodicity. Details about full list of distributions available in Opnet can be found on the manual chapter 25, [110]. For our scenarios, uniform and constant distributions have been used in order to model different application profiles depending to each node. In order to simulate different attributes in the implementation uniform distributions are used for starting time to make sure that all devices are not starting at the same time. In order to simulate a situation where all devices are starting at the same time, we use a constant distribution. For inter-repetitions of transmitted packets a constant distribution has been used only for monitored (marked) nodes while random traffic has been assigned to the rest of the nodes forming the network.

The prototype (figure. 20) consists of three nodes in a small geographical location. Each node is represented by a red mobile phone. The scope of this scenario is to achieve successful communication between node A and node C via node B. The reason for choosing this topology is that in such a network nodes are acting as transceivers. Their role is not only to send and receive messages related to them, but also to forward messages for different destinations. Additionally the nodes exchange routing information whenever it is needed (on demand).

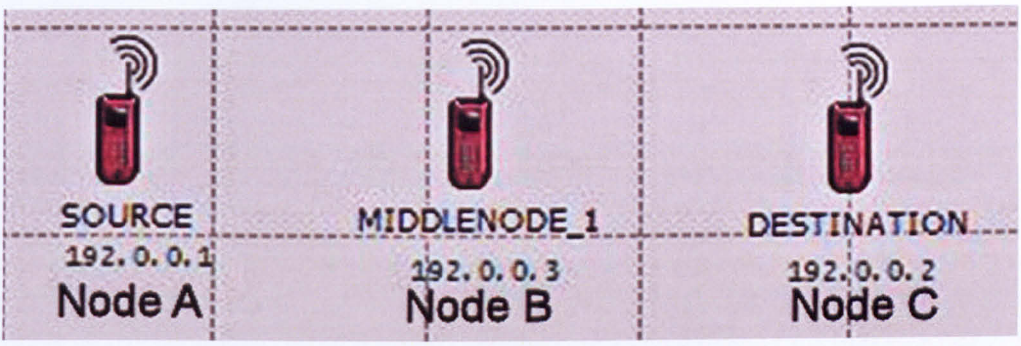


Figure 20. Prototype Model of the Message Enabled Network

The scenario is the following: Two users (node A and node C) will try to send SMS messages within a sort time to each other via a middle one (node B). Simulation time has been set to end in 60 seconds. As we will see later in the global network 60 seconds is not a sufficient time, as the nodes require time to stabilize their routing tables. In this case the predefined time is sufficient as the stabilization is only needed in case we have many nodes in the network and have added mobility. This time has been agreed as the scenario simulates a conversation between two users within the prototype model.

More analytically, Source node A, is sending an SMS message to the destination node C (via middle node B) every twenty seconds starting from approximately t=0.1sec. The destination node replies to source (via middle node) in 30 seconds starting from t=28sec. The message exchange continues simulating two users sending messages to each other (texting). The starting time of the transmissions presented in this paragraph is an approximation, as each node will start transmission randomly.

By assigning periodical timing of transmissions for the sender and the receiver we can simulate a situation in which user A is sending a short message to user C. In real life humans would have to spend a short time 5-40 seconds to read the message and reply with the word "ok" which may be considered as a quick reply confirmation SMS. A good approximation of the average time of replying to an SMS is considered to be 20 seconds (μ: average message sending time). The timings of the message transmission of the prototype have been set this way in order to simulate the scenario for the prototype only. As we will see later in the global network all nodes have random timings for simulating random message transmission between users. It is worth-while to notice that in the global messaging network marked nodes for evaluation purposes have been set to transmit messages periodically starting at a random timing and all other nodes exchanging messages are transmitting randomly to random locations in order to create background traffic and stretch the network as it happens in real life situations.

IP addresses have been assigned manually for all the devices in the experiment. User A has been assigned with the IP address 192.0.0.1, user B has 192.0.0.2 and so on. Later on in the global network, IP addresses will be assigned randomly using one of the features of the simulator for random profiling.

The type of transmission has that been used is unicast. Obviously we are not using multicasting as it consumes more bandwidth. The message transfer will be a fast constant bit rate (CBR) burst. The bandwidth of the channel has been set to 2 Mbit/sec. The data rate has been chosen and from the survey, we see that it provides a greater range compared to faster data rates. Furthermore, the chosen data rate (2 Mbits/sec) allows the node to have a sufficient range of communication. The higher the data rate is, the smaller the communication range of the node becomes. Same principle applies with battery consumption. A high data rate decreases the communication range and increases the battery consumption of the device. As we will explain later in our analysis this data rate is more than sufficient for handling much more SMS messages than a human could send in a predefined time.

Considering the fact that we have to transmit short messages means that we have decided that the transmission must be short and fast. In other words in a hazardous environment we should not keep a channel opened for a long time. The requirement is to allocate a channel fast, send the message fast, close the channel and allow someone else to use it immediately as we finish with the transmission of the message. We must not forget that the scenario is referred to as a hazardous environment. Any extra overhead bandwidth or slow channel allocations will lead to congestion or a possible breakdown due to flooded or repeated messages. This must be avoided as this network relies on the mobile hosts only.

As this is the prototype for the message based global network we have not added mobility yet to the nodes and there is no need to let the nodes and routing tables settle down before we start transmission. What we expect to see from the plots is spikes during transmission and reception. Each spike in the graph represents a transmitted/received message, in both traffic and routing plots. The message's size has been set to be 300 bits. The reason for choosing this size is that in the transmitted message there is space for including the original SMS message as it was meant to be sent from a GSM device. Additionally apart form the 160 alphanumeric characters there is enough space to include even more headers with information for authentication or other reasons. This has been done to prepare the network for future development and expandability.

4.3.2. Proofing the Prototype Model

The simulation of the prototype model was implemented and simulated successfully and various statistics were collected. From the four graphs in figure 21 we may observe a functional graph of message transmission and reception for destination and source nodes.

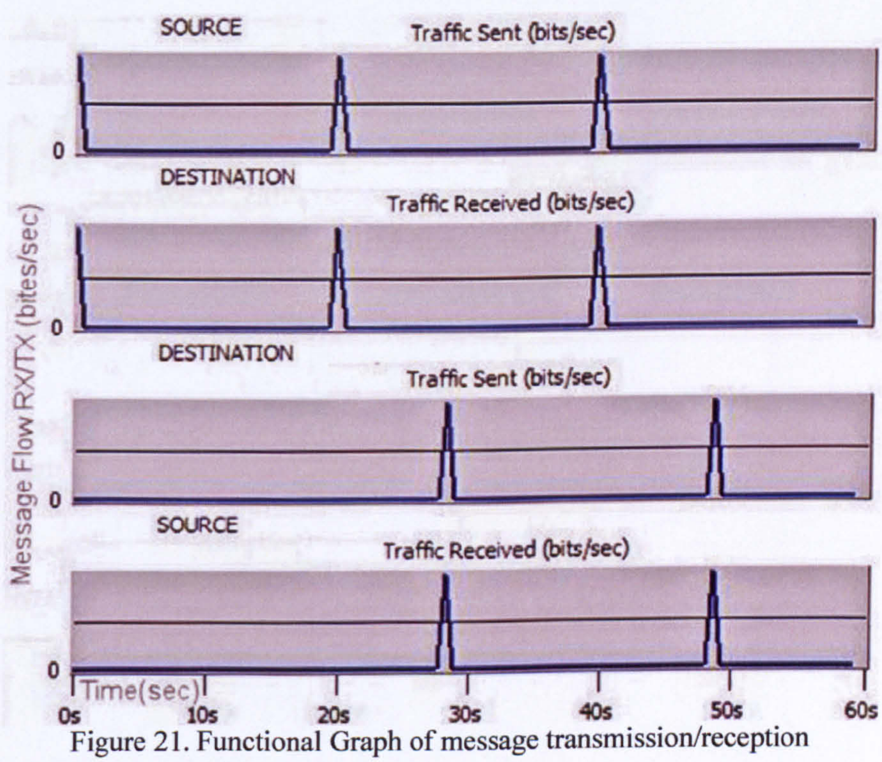


Figure 21. Functional Graph of message transmission/reception

The top two plots illustrate message transmission and reception from source node to destination and the bottom two show message transmission and reception from destination to source node. According to the AODV protocol, the source node requested routing information about the recipient in specific timings in order to send a message. The recipient node has acknowledged the request and replied that it is ready to accept data. Then the data transfer has successfully completed. In real life, this could be a case of a victim asking for help and a friend responds that he is aware and is going to help. The victim then confirms back that he is waiting for rescue.

Wireless Lan Delay (sec)

Sort By Node	Sorted By Average	Sort By Peak
SOURCE_0	0,0025015	0,0038624
Middle Node	0,002483	0,0025006
DESTINATION_0	0,0024829	0,0024908

Figure 22. Wireless Lan Delay (sec)

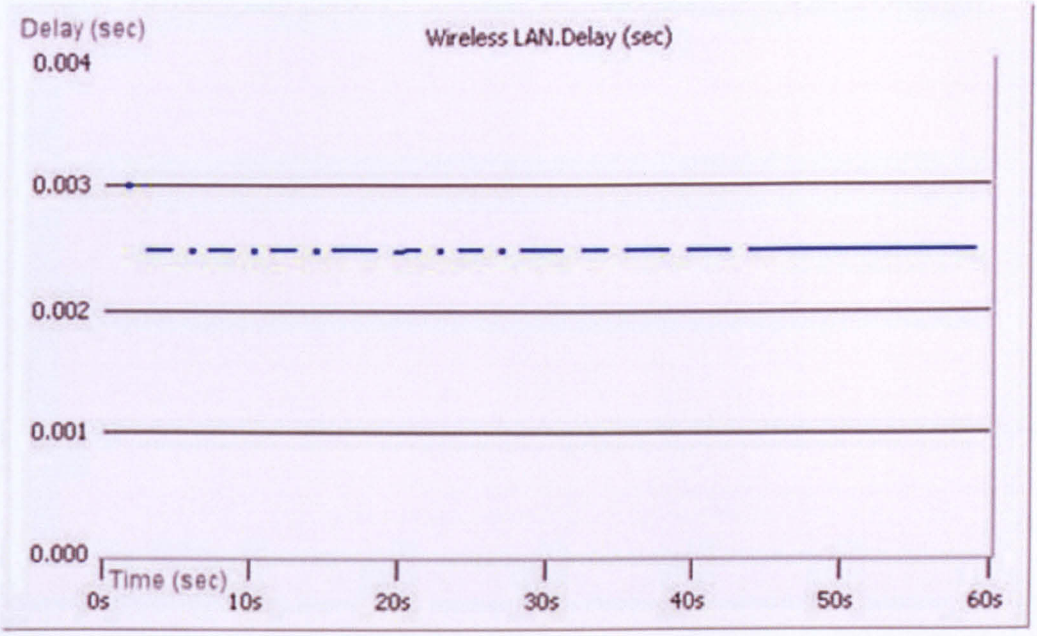


Figure 23. Wireless Lan Delay Plot

In figure 22 we present the collected statistics of the delay of the transmission /reception process and in figure 23 the plot for the simulated scenario. In order to verify that the transmission was fast as described in criterion 1 we will compare this transmission to the average delivery time of an SMS arriving at destination. In TELECOMWORLDWIRE(2005) ANACOM reveal results of Portuguese SMS study [109].

DIAGRAM
ON THIS PAGE
EXCLUDED
UNDER
INSTRUCTION
FROM THE
UNIVERSITY

Figure 24. A study from AVACOM [109] for the sms average delivery time

According to figure 24 and APPENDIX.4B the average sms delay for an sms reaching the recipient is 15 seconds as described from AVACOM, [109]. The delivery message obviously is related to a fully operating GSM network which involves many connected devices. Such a behaviour from the GSM network of delivering an SMS with a delay time of 15 seconds is acceptable for the GSM industry. Obviously if there was no infrastructure this delay time could be reduced. Currently the message has to transferred from mobile phone (through air) to BTS then to core GSM network (wired subnetwork). Afterwards it is redirected to another cell site and reaches the destination. In the case of the scenario described above, the process of sending and receiving messages for the source and destination node (via a middle node) has been successfully completed in an average of 2,4-2,5ms. This comparison could be considered as not fair because we compare a system involving transmission of a few SMS between 3 devices. Avacom's study was tested and evaluated for 3 different GSM networks providers involving many mobiles and a number of about 17000 SMS messages sent through the networks [APPENDIX.4B]. For this reason we will compare again the results in figure 24 with a global network deployed as it will be described in the next section. More collected statistics and plots regarding the prototype model can be found in APPENDIX.4A

4.3.3. Messaging Capabilities of a GSM Network

For the prototype model, we have assumed that GSM service is lost, the nodes are in range and people reply to an SMS in average every 20 seconds, (3 SMS per minute). Considering the criteria for a successfully working network the prototype model satisfies all of the conditions, which are related with message capabilities. The deployed network can provide communication capabilities to the users shortly after it has been initialized. Direct connectivity between the devices is ensured by using 802.11. Another criterion is met while the devices communicate directly, without using the infrastructure because they operate in Ad Hoc mode. The fact that the devices are able to exchange messages satisfies the condition for a successfully working network to provide messaging capabilities in a hazardous environment.

According to other researches and designs, a GSM network is limited in terms of the number of SMS messages that it can deliver within a predefined period [107], [108], [109]. Providers limit the number of the SMS messages that can be delivered in a mobile phone from one to three per second. The reason for applying this limitation is that GSM providers implement various techniques in their network to avoid congestion and attacks due to bad use of the service such as bulk advertising through SMS. In an SMS related paper [108] the writer states that in an area of $68m^2$ 240msgs/sec can be delivered by the existing GSM equipment. Assuming that we place one person in every $1m^2$ then the GSM equipment can deliver 3 SMS/sec to each person in the area. Obviously, our proposed network performs better than the GSM one, because it can deliver significantly more than one to three messages per second. Theoretically and for the reason that the message's size is 300bits we could transmit more than 6000 messages per second across a 2Mbps link. This number though, is dependent of the topology, the number of nodes involved in the network, the mobility, the processing delay, the buffer size and the type of traffic is used. Comparing the proposed prototype with the current GSM network the message delivery that the nodes can process every second is better in every aspect; Speed, reliability and average time of message delivery as we will demonstrate in the global network. The statistics from a functional test shows that the delivery time of a message/sec between two nodes is 2.2 ms while 10 messages/sec will result in an 8 ms delay time. In the scenario, we assume that messages are being sent by humans (1msg/20secs for marked nodes and random traffic with limit of 1 message/sec per node) and not by using bulk mechanisms which are able to send several messages per second. The conclusion of this section is that we may observe that increasing the number of messages delivered every second to one node will increase the delivery delay time a few ms. For the messaging service this has no effect as the messages will be delivered shortly after they have been sent. We may also observe that there is no need for requiring the nodes to deliver more than one message per second as people will spend time to read it and reply (20 seconds in average). Even in large-scale networks, the delivery time will not be affected because the channel capacity is large (2Mbps) and can support extremely heavy background traffic.

The prototype has been used for testing the main scheme of communications (figure. 21) between source and destination nodes. From the results we see that AODV routing traffic as well as the sender's and recipient's communication through the middle node were successful, fast and reliable. For this scenario no packets were dropped as expected. As this will form the global network though many questions arise, as in a hazardous environment things get more complicated. Many important parameters such as interference, mobility, dynamic changes on the topology and heavy traffic can affect the network performance. It is obvious that for answering all these questions and to study the network behaviour, it is necessary to evaluate the global network within a more realistic environment. For this reason, the global network will be tested in a scenario with added parameters and this follows in the next section.

4.3.4. A Global Message Based Network in Hazardous Environments.

In the next scenario, we will analyze, and evaluate the global network in order to investigate its behaviour under heavy traffic conditions inside a dynamic environment with added random mobility, trajectories and timings. The size of the map has been set to 3km x 3km. Instead of using a single middle node we have placed 300 nodes randomly on Liverpool's city centre map, taken from Google Earth [91]. The following picture in figure 25 illustrates the map scenario.

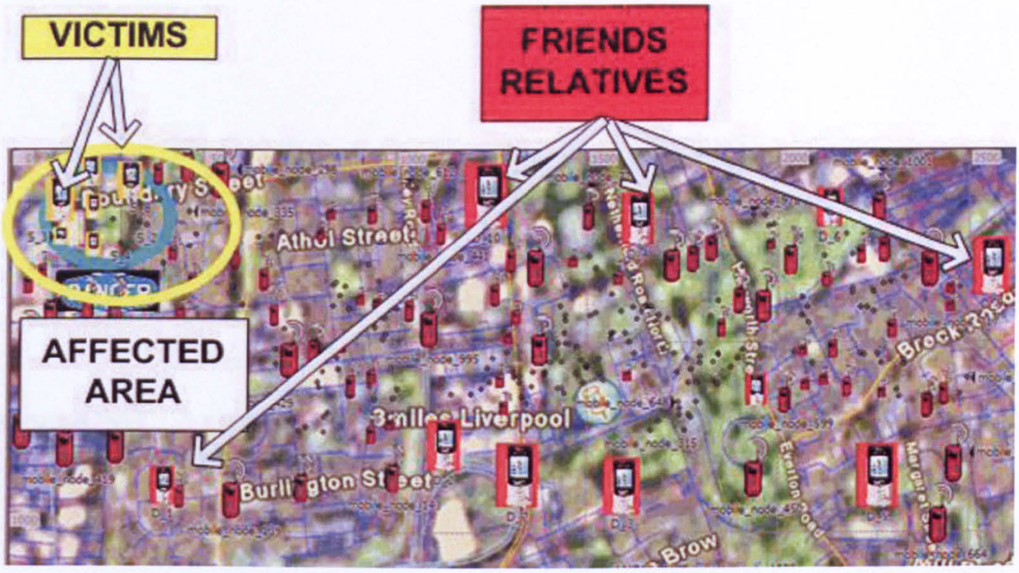


Figure 25. Global Network in a Hazardous Environment

Trying to make the simulation more realistic, which will result in better accuracy, we have created and simulated a global network involving 300 people connected in a scenario map which takes place in Liverpool's city centre. The area of disaster is located on the left upper corner. Within the affected area there are 10 trapped people trying to get in touch with relatives or friends. These nodes have been marked with yellow colour and their IP's have been marked for monitoring. The nodes representing relative and friends of the victims have been placed randomly on the scenario map (figure 25) (red colour) and will be also monitored in this scenario. The main purpose in this scenario is to investigate if all 20 (out of 300) monitored devices managed to communicate successfully. Random traffic has been assigned to additional 80 random nodes in the scenario in order to simulate an environment with many users sending messages outside the hazardous area stressing the network. 100 out of 300 nodes will exchange messages. A random mobility profile has been set up in Opnet, which allows the nodes to move with a speed of 5m/sec to 10m/sec. This is an average speed selected in order to simulate slow movement, walking or running through the area. Mobility of all nodes allows free random movement and speed but the victims have been limited to move in the hazardous area as in this scenario they are trapped and they are asking for help.

It is important to say that this network can be scaled as much as we desire with one limitation. The more nodes and connections the greater the memory and CPU power needed. Our equipment consists of a Pentium 2.00Ghz CPU and 1 GB memory. The simulation time has been set to five minutes. The reason that we did not choose to end the simulation in 60 seconds is that there is a predefined time delay of 150 seconds at the start of the simulation, which is the default time for the nodes to settle down and to stabilize their routing tables. As you will notice later on the plots at the beginning of the simulation, the traffic is heavier. As time passes (up to 150 seconds), the traffic is decreased. This behaviour is expected as the nodes are establishing connections at the beginning with surrounding environment. The default value of the delay is 150 seconds as defined by the simulator's manual.

Random MAC and IP addresses have been set randomly for all the nodes. In order to observe the behaviour of the 20 nodes as mentioned above, we marked their IP addresses for identification purposes. Access Point Capability id is obviously disabled as this is an Ad Hoc network. As we have seen before, the requirement for this model is not based on infrastructure, as in the case of a disaster situation we cannot rely on it.

4.4. Evaluation of the Global Network in Hazardous Environments.

The evaluation of the network will be presented in three sections. Communication of the global network is related to the message delivery as well as other related parameters in hazardous environments. In this section we will present and discuss how the network operates. Additionally we will present the related results and highlight some interesting points. Routing and traffic will be presented and analyzed next, followed by the last section of the evaluation which is related to the performance of the global network.

4.4.1. Message Delivery and Evaluation

In this section, we will evaluate the global network by presenting the results of the simulation. Additionally we will compare important statistics with others that have been found from similar studies.

Start from the graph in figure 26 illustrating a functional graph of a random pair of connected nodes exchanging messages. Considering the fact that the network converges at t=150 seconds we may observe that the messages are transmitted immediately after the simulation starts and before this time form spikes at the 30th, 50th, 70th, and so on up too the 150th second of the simulated time. Convergence is in progress while devices are starting immediately sending their messages. This happens because the AODV protocol is characterized as an on demand protocol. When data is available for transmission, the nodes will start the process of routing and sending immediately. At the same time the network will continue its convergence until the 150 second. Convergence time is the elapsed time for devices to complete the exchange of reachability information. According to [110] and [111] routing configurations are complex, it is difficult to estimate convergence times analytically even when a simulator supports such procedures. A delay at the start of simulation before sending background traffic may be needed to allow routing tables time to stabilize. The default value for this background delay time is defined as 150 seconds. Respecting this value, we will let the network stabilise at 150 seconds. Allowing stabilization of the routing tables we will make sure that all devices have information about the surrounding environment. Changing the default value is not recommended as it may lead to inaccurate results due to miscalculated nodes. The graph of convergence can be characterized as an exponentially decreasing to zero plot. As we get closer to zero, there is a greater probability the results have converged. As we approach zero, the simulation runtimes will get longer. In real life situations, it is obvious that when nodes know their neighbouring devices and routings will transfer data and execute routing requests faster as they have prior information stored on their routing tables. This results in fewer collisions, change of frequencies and delays in large congested topologies. In order to avoid repeating ourselves we will illustrate the convergence graph in the next section (figure 31) which deals with the network traffic and other parameters.

It is worth noticing that we could force the nodes starting at the same time (eg. t=0 sec) but we would observe a collision at the beginning of the simulation, and an immediate retransmission. Even if we simulate such a condition, the collision at the beginning cannot be considered as an error because from the network's architecture a node cannot receive and transmit at the same moment. What will happen in this case is that we will observe one or more retransmissions as the node will try to resend the message in case of a collision. The repetition of transmitted messages between users simulates a continuous text based conversation between senders and recipients.

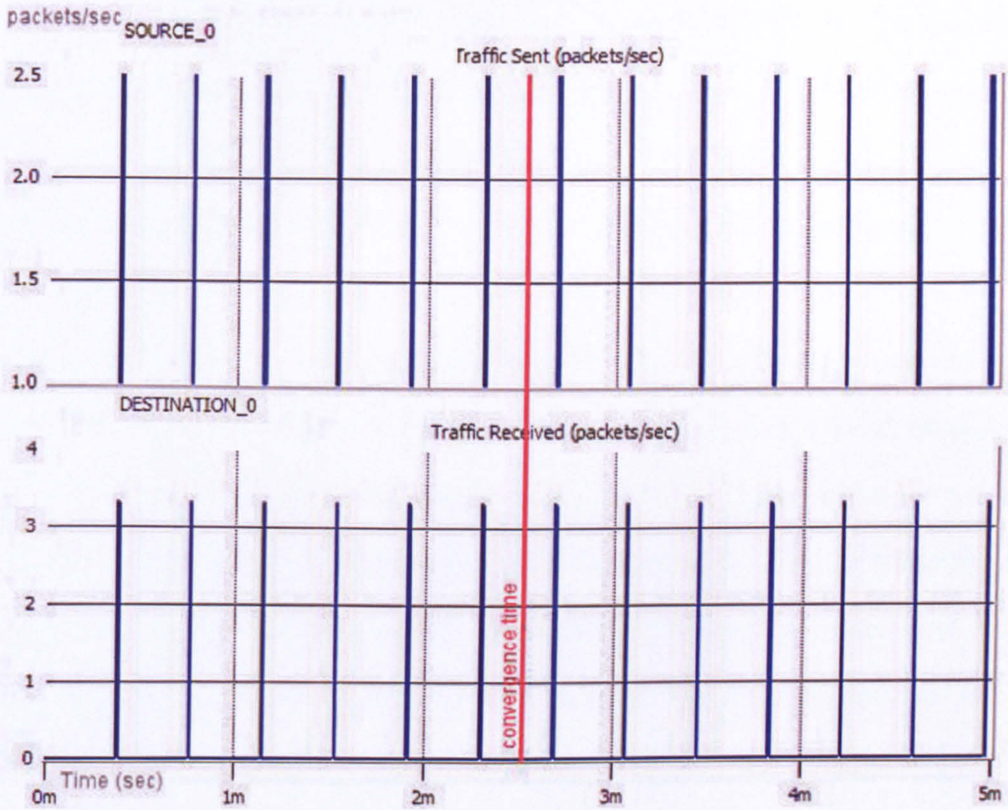


Figure 26. Successful Communication of a Random Pair of Nodes.

The function test graph on figure 26 illustrates the successful communication between a pair of nodes (victim/rescuer). The messages are delivered from sender to recipient quickly without any losses. As described earlier, this scenario takes place within the area of the disaster. Considering the very noisy and disturbed environment we have already discussed, the best way to send messages through the 802.11 network, is to allocate a channel, transmit as fast as possible and release the channel immediately. In our case a fast burst has been used, to send the data rapidly from one node to another and then release the allocated channel for other users trying to transmit. By using constant bit rate (CBR) traffic, it is possible to send a rapid burst for a small amount of time through the physical layer.

Scalability of the network introduces some interesting behaviour and limitations in some cases. It is unavoidable that lost packets and dropped messages may occur for several reasons. The more users in the area map, the more links are available, but also more interference and other factors may affect the network. The most common reason for a failure of delivery is a broken link in areas that are not crowded. Depending on the random mobility profile, as the nodes move constantly, some of them may move far away from the others. Clearly if a node is out of range, it cannot exchange information with surrounding nodes. Examining such a scenario from logs and raw data we are able to understand the reason a node is not transmitting or not accepting any messages (e.g. due to interference, range or other reasons). A message might get lost or received after a while, in the network in the case that the routing tables are not stabilized, which means that all the nodes are not yet known to the network and to each other. In this case, the messages will not be sent on time as if the route to destination is not known and the actual transmission will not begin. In this case the message will not be lost. Always depending on the configuration, there will be a number of retries for retransmission and the node will try to retransmit periodically. There is a trade off between retransmission retries and the condition of a message to get lost. The more retransmissions of the message there are, the greater the chance that it will be delivered, but also the more the interference and collisions will be caused.

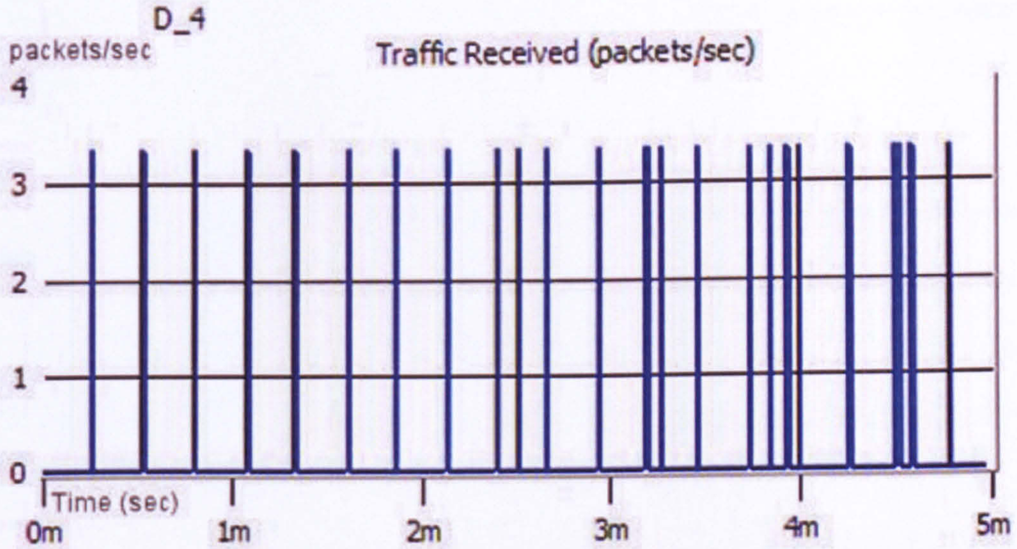


Figure 27. Random Traffic – Messages Delivered from Multiple Senders.

Figure 27 illustrates a case where a node receives messages not only from a relative/friend but also from other nodes as random nodes have been set up to send random messages to random destination nodes. Observing this graph and considering the scenario as the worst-case scenario, we can observe that the node can accept even more messages than the previous case (1 message every 20 seconds). On this plot, we may see that some spikes are repeated constantly every 20 seconds. These represent the messages that are being sent from marked nodes (constant distribution). The rest of the randomly delivered messages are received from other nodes that transmit randomly as a random profile has been applied in order to create realistic traffic in the network. It is obvious that at the same time each node that routes traffic and performs a number of operations through the physical layers, is also executing requests for routing, transmission purposes and a number of other operations. In the scenario assuming that all of the 299 nodes (in range) would transmit messages to a single node at different random time, the messages would be received. Inevitably, in this case we would expect to see increased bandwidth, as the routing information would increase as well.

In the next section of the thesis we will present more statistics for the behaviour of the network in order to explain, evaluate and compare the global network with similar systems. The parameters that will be presented are related to delays and routing. We will also compare the messaged based network in hazardous environments with similar studies in order to discuss its performance and behaviour.

4.4.2. Routing and Traffic.

The global network statistics have appeared to be very good as we had minimum losses as seen in the statistics. Before going into details about various parameters that affect the network we will introduce a table in which we present the most important collected statistics from the simulation.

Statistic	Average	Maximum	Minimum
AODV Number of Hops per Route	1.0252	3.3295	1.0000
AODV Route Discovery Time	0.3058	9.6098	0.0023
AODV Routing Traffic Received (bits/sec)	25,026,776	69,007,360	0
AODV Routing Traffic Received (pkts/sec)	65,150	179,797	0
AODV Routing Traffic Sent (bits/sec)	128,119	1,154,560	0
AODV Routing Traffic Sent (pkts/sec)	335.8	2,903.3	0.0
AODV Total Cached Replies Sent	128.40	304.00	1.00
AODV Total Packets Dropped	1	1	1
AODV Total Replies Sent from Destination	12381	4.0000	1.0000
AODV Total Route Errors Sent	42.68	590.00	1.00
AODV Total Route Replies Sent	100.46	304.00	1.00
AODV Total Route Requests Sent	1.689	21.000	1.000

Figure 28. Global Network – AODV Collected Statistics

Statistic	Average	Maximum	Minimum
Global Net Delay (secs)	0.1432	9.6150	0.0015
Global Net Traffic Received (bits/sec)	2,816	56,000	0
Global Net Traffic Received (packets/sec)	1.173	23.333	0.000
Global Net Traffic Sent (bits/sec)	2,928	168,000	0
Global Net Traffic Sent (packets/sec)	1.220	70.000	0.000

Figure 29. Global Network – Network Collected Statistics

Figures 28 and 29 illustrate the collected statistics from the global network regarding the most important parameters that may affect the network. Through the next few sections we will discuss the network performance and behaviour.

Using the next sets of graphs we will discuss the delay time, which is the time a message needs to get delivered. Furthermore, we will present results for dropped packets, routing considerations and mobility, which are the main factors that are affecting any wireless network and its performance. Delay is known as one of the most important parameters in such networks. The less the delay the better the performance is. Considering the third and fourth criteria for an emergency network and by the SMS delivery survey on GSM systems, we know that current GSM networks are able to transmit one to three SMS's per second to a mobile phone. We also know that the delivery time for an SMS to arrive to destinations has an average of 15 seconds as seen from survey. The first question arises is how our novel architecture is compared to current GSM networks in terms of delays. How fast is the proposed approach and how reliable? In the next plot (figure 30 and tables in figure 28, 29) we may observe that the delay time in average has found to be 0.1432 seconds.

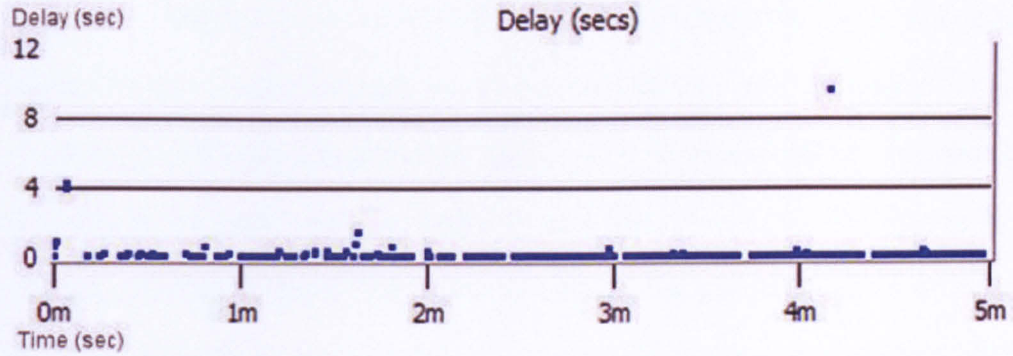


Figure 30. Time of Arrival for Global Network (Delay).

On the graph above (figure 30), we present the delay of the message transmission. The average value is 0.1432 seconds, the maximum time for a transmission is 9.6150 seconds and the minimum 0.0015 seconds (see tables of figure 28, 29). The maximum value was 9.6 seconds for only one transmitted message. This is not considered as a bad result as this might happen in cases like broken links or when the nodes are out of range. In our scenario, one of the nodes was out of range from the rest, and the message could not be delivered. The main reason that created the dropped packet was the mobility of the node. As has been presented earlier, the mobility profile introduces random movement and a speed range of 5m/sec to 10 m/sec simulating this way of human movement, (walking or running).

A similar study entitled "A Load a Ware Routing (LWR) Based on Local Information" [89] introduces one of the results, which is the plot of a 100 nodes transmitting data. The transmission type is similar to ours as well as the terrain and other parameters as it appears in the paper. Obviously, the study of the paper is not the same and their plots are comparing two routings. From their graph though, we may understand that delay is increased when speed is increased. Eventually if the nodes are moving too fast they will pass near others that may use same transmission frequencies. This will force them to hop in to a new frequency. It is clear that interference from nodes will introduce more delay in the network. Our simulation has a mobility profile with varying random speeds from 5m/sec to 10m/sec. Comparing the graph in the paper with our results obtained from the global network and by observing the graph in paper [89] we may see that for the speed range of 5ms - 10ms their delay time varies from 0.3-0.5ms. The average delay time we obtained is 0.1432 seconds, which is a fairly good result and very close to their study.

Within collected results, the message transmission delay is satisfactory for the global network in hazardous situations. Our results introduce an end-to-end delay of 0.1423 seconds which appears to be similar to a wireless networking study. Furthermore the criterion for the maximum delay time of the delivery of a message is satisfied through our proposed messaging network.

Four graphs are presented in figure 31. The routing traffic is illustrated in those 4 plots in bits/sec and packets/sec. Received traffic is large compared to sent traffic, as each node is exchanging data with its surrounding environment. In this graph it is interesting that the traffic is decreasing as time passes. We expect this behaviour as the default value for routing tables time to stabilize is 150 seconds. As it can be observed after t=150 seconds, the overhead bandwidth is reduced as the nodes have stabilized routing tables and less routing information is needed.

The performance of the global network is very good as connectivity and message delivery have been achieved successfully in this scenario. We must keep in mind that in large topologies including million of users the system may introduce some delays before stabilization. This is not considered as an error as for this type of network dynamic conditions as range of the node, or an unknown route may introduce delay in the system, which results in a delayed a message transmission.

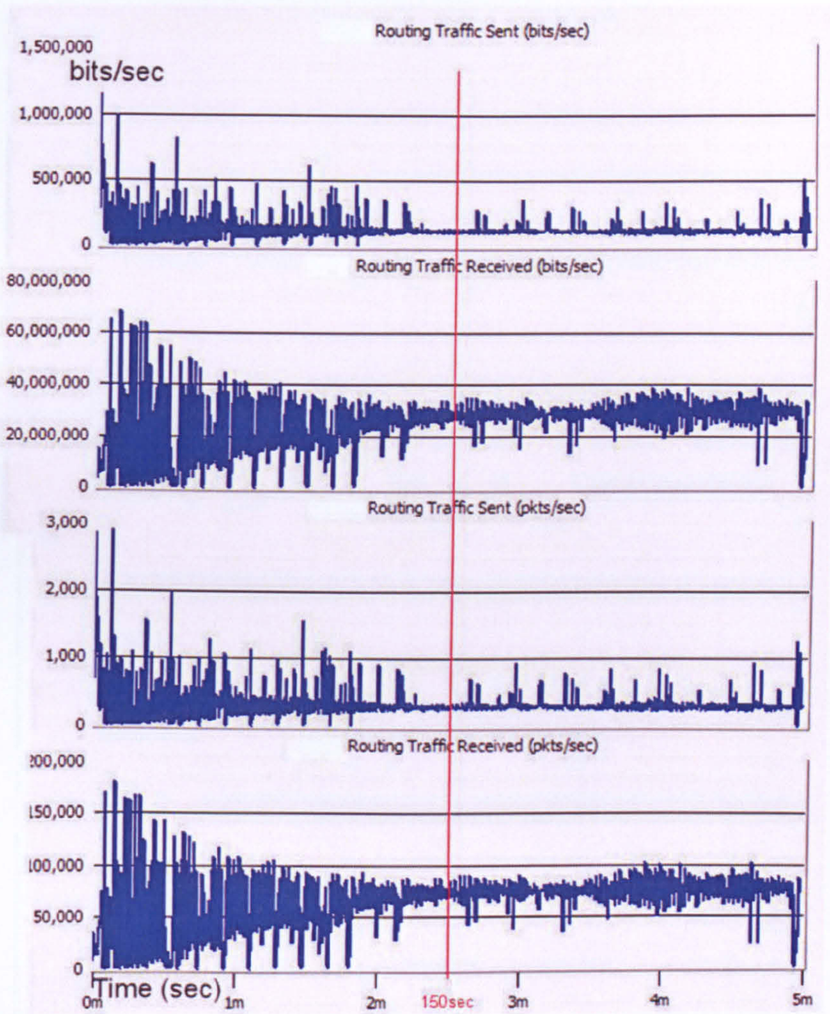


Figure 31. Routing Traffic and Stabilization for the Global Network.

Exploring more the behaviour of the nodes in terms of connectivity with each other, we may observe the importance of stabilization time for the network. Figure 32, on the next page, illustrates the total route requests sent to nodes, the total replies sent, the total route errors sent, the replies sent from destination nodes and the cached replies sent. Total requests of the network and replies are decreased as the time reaches t=150. This is happening because as time passes, the nodes have more information about the nodes of the surrounding environment. In a scaled network, the overhead bandwidth will be increased but the number of nodes will be increased as well; thus, the number of links between the nodes will be increased. The network will be able to operate normally as the bandwidth consumption before stabilization will not affect the network performance as the more nodes are introduced in the network the more powerful the network becomes.

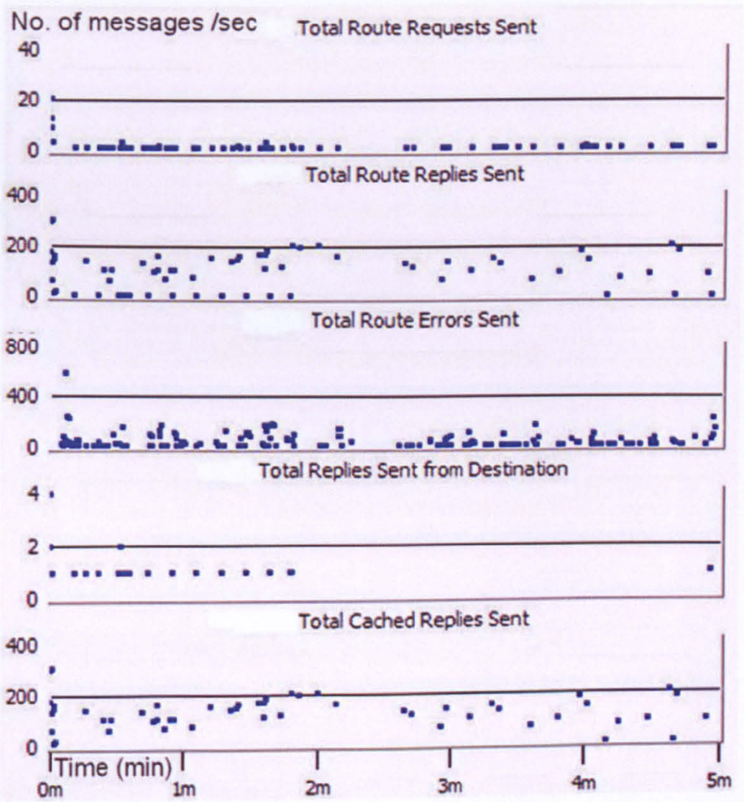


Figure 32. Routing Statistics for the Node Communication in Global Network.

In figure 32 we may observe that route errors reduced dramatically after the routing tables have been stabilized. The fact we still get errors is due the random mobility or due to tx/rx errors. Tx/Rx (collision) errors can occur often when a node tries to transmit and receive at the same time. In this case the node makes a timeout and tries to retransmit after a few milliseconds depending on the configuration and the surrounding environment (Statistics figure 28 and 29). Considering the fact that the network is changing dynamically we get this error often if the node trying to retransmit finds a broken link or is interfering with one or more nodes. This behaviour appears very often when the destinations are not known. The same situation applies for recipients sending an acknowledgement back to sender before the transfer. Middle nodes of the route are affected more often as they may become broken links as the movement of nodes (people) is random in such situations.

Cached replies are simple data requests being reused when the same route, for example, is being requested for use from different nodes. Cached replies are decreased after t=150 seconds, because the network is stabilized and the nodes calculate fewer different routes in order to exchange data. Of course this is always depending on the mobility. Cached replies are more often to appear in networks using static routes only rather than in a dynamic network.

4.4.3. Performance of the Global Network.

The performance of the network depends on the successful transmission and reception of the messages in the network. To measure performance we may observe several parameters from statistics and make a judgement on the quality of the network. Factors that are often investigated for measuring the performance (always depending to the case scenario), are the packets dropped, route discovery time, hops per route and delivery ratio. Studying these and comparing with similar systems we will be able to present how well or badly the system performs.

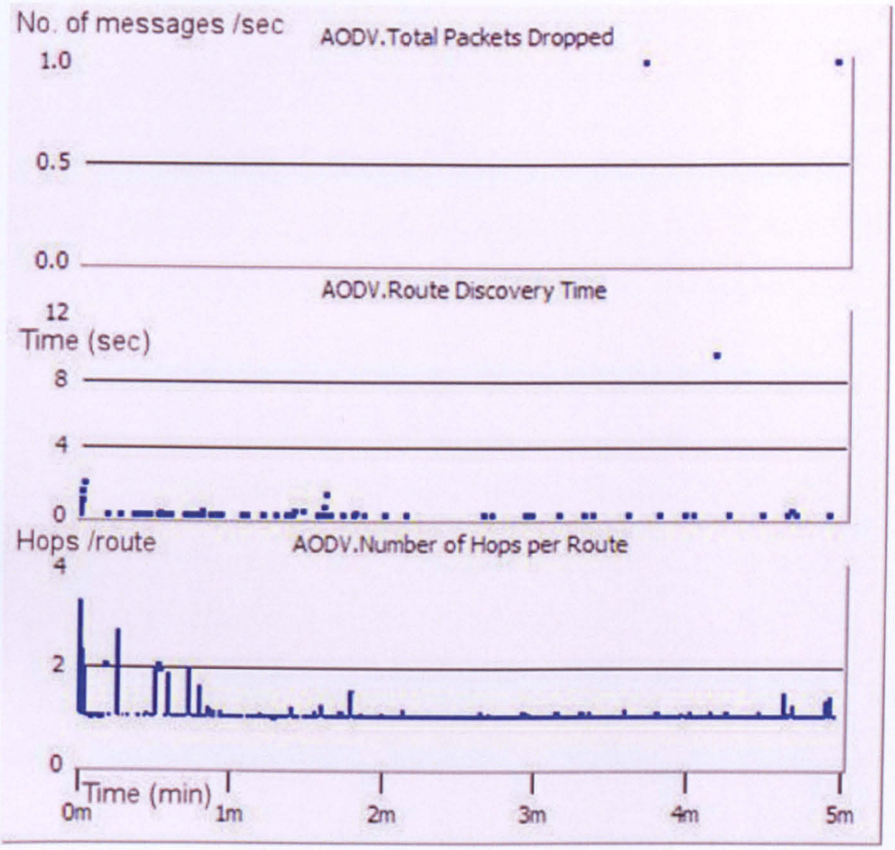


Figure 33. Performance of the Global Network.

In figure 33 the total packets dropped depends on the network's setup as well as the conditions and parameters affecting the network itself. Mobility, interference and bad reception conditions are the parameters most affecting communication. The fact that we allow the network to stabilize and we give it the time to find the routes first, creates a big load on traffic as the network is initialized, because it is exchanging information for routing purposes. Figure 33 illustrates the total packets dropped in the network. This plot is related to the delivery ratio as the fewer dropped packets in the network the better is the delivery ratio. According to the simulator our delivery ratio is greater than 99 %. Later in this section, we will present two comparisons with two different but similar systems and discuss analytically the delivery ratio of the global network.

The route discovery has a major role on the message delivery. If the discovery time is big the system introduces delays which results in message delivery delays or even in message delivery failure, if the sender exceeds the maximum retransmissions of a message. Inevitably, dropped packets may be introduced in the case that nodes are out of range. This is a physical limitation, and is not considered as an error. If the nodes are out of range the message delivery cannot be completed as the radios of the devices cannot communicate.

Next hop represents the information a device has about a particular destination. This information is stored in devices. It includes the details about the next device in which a message will be forwarded in order to reach the final destination. The criterion of choosing of the next device (next hop) is that an incoming message must follow the optimal route to the destination. When a device receives a packet it checks the destination address and tries to associate this address with the next hop. From the relate graph (figure 33) we can clearly see that the number of hops decreases as the network stabilizes. This is normal as at the beginning all the nodes transmit and receive in order to get answers for their routing requests. This appears more in areas with too many nodes together trying to exchange data. In dynamic networks and especially in a hazardous situation this is expected as the environment changes rapidly. Obviously, the number of hops is decreased as time passes.

Delivery ratio is often represented by a percentage of values from zero, as the worst case scenario, to an optimal value of one and is the ratio which characterizes how well or badly the system performs. The delivery ratio depends on many parameters like routing protocol, mobility and delay. Our delivery ratio is greater than 99%. This percentage characterizes a good delivery ratio, a stable network which can deliver messages quickly and with reliability in a hazardous environment. It is obvious that the delivery ratio is related to other factors like mobility. Generally the faster the speed of a node travelling the bigger the possibility of losing a message because there is not enough time for transmission or due to a collision. Movement of the node though is also an advantage as a message could be delivered over a large distance through cars are travelling on roads. The great advantage though of our network is the rapid transmission of a message as described earlier. Our delivery ratio can be characterized as very good as is near to 1.

Similar studies have shown that AODV can deliver messages with good delivery ratios. A particular case is characterized in a paper related to a similar routing algorithm, entitled as "A Load a Ware Routing (LWR) Based on Local Information" [89]. In this paper the writers propose a new algorithm which performs better than AODV and they present a plot of the mobility versus the delivery ratio. This plot illustrates that in their research the AODV performs fairly well from a 0.55 to 0.45 (for 5m/sec to 10m/sec). Their algorithm performs better than AODV for this speed range. The difference in delivery ratios compared to our results is the type of transmission. Clearly major effect on the ratio are the topology, the area and the size of the transmitted packets. At this point we will not comment more about their setup and results which can be found in their paper [89]. Our point is to verify that the network can deliver fast, reliably and make sure that it can satisfy all the criteria of an emergency network as described earlier. For justification reasons though we will refer to a different paper which shows the delivery ratio to be near to one. A study which is entitled "Infrastructured Ad Hoc Networks" [90] has a similar delivery ratio (near 1) compared to our results. Looking at this study, we may comment that they have a similar setup to ours and their parameters are similar. Clearly, it is very difficult to find identical simulations in this field of study but the referred papers can be characterized as being close and related to our field of research. In their research there is a graph illustrating the packet delivery versus a varying traffic load. From their plot it is obvious that the delivery ratio remains constant (near to one) for different protocols and for a range of 0-20 CBR flows. Taking into consideration that our marked nodes only created the same flows in our network we could say that we have introduced a similar delivery ratio in our network.

This situation of having different delivery ratios is often introduced as the delivery ratio is also dependent on the network setup. For example if we deploy a network with the same characteristics in different maps we will observe that if the area is large and the number of nodes is small, the delivery ratio may be affected as nodes may go out range. Finally, it is worth noticing that the routing protocol plays also a major role as it has been presented in the survey.

In the same paper [90], there is also a different plot, which presents the total number of routing messages per second versus the number of the CBR flows. In their plot it is obvious that the total number of routing messages (per second) increases as the number of CBR increases. Our comment here is that in our network we face the same situation with a little difference. In the next graph, we present our next result through a similar plot.

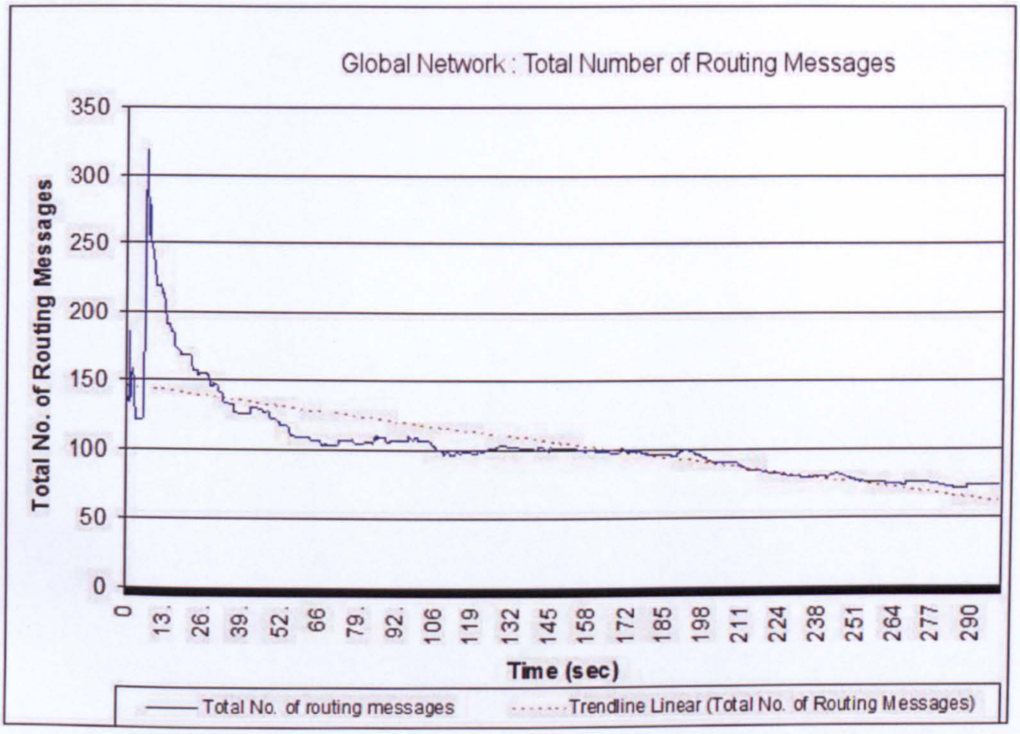


Figure 34. Total Number of Routing Messages for the Global Network

In figure 34 we present a graph of the total number of routing messages over time. The blue line represents the number of messages and the red line is a trend line (Trend/Regression Linear Type) that can be used as a way of visually depicting the relationship between the independent (x) and dependent (y) variables in the graph. A straight line depicts a linear trend in the data. In this graph we may observe that the trend line has a value of 150 to 100 messages up to 150 seconds (the convergence time) at which the routing tables are stabilized, and then decreases from 100 to 60. Comparing the result from our scenario with the results of [90] we may observe a similarity on the behaviour of the two scenarios as in both cases we may observe that the overhead bandwidth is increasing as the number of traffic flows increases. The difference in our case is that the total number of routing messages is decreasing as time passes, while the number of nodes and traffic characteristics remains constant. That means the proposed network can operate in hazardous environments without introducing congestion due to routing messages (flood) and that the network will be able to deliver the messages quickly at any time.

The case study proves that the proposed network can provide communication in hazardous environments as has been discussed in the objectives of the thesis. Additionally by completing the experimental procedure of the network involving messaging capabilities, we complete some of the thesis objectives as have been introduced in the thesis aims and objectives section. As this research has been motivated from creating an emergency network which can operate in areas that suffer from disasters (and especially in the first conference that the idea and the first paper introduced in PGNET 2004, the audience asked repeatedly that we should consider voice communication and implemented it in the fast deployed network.) This was the spark for continuing the research as we will see in the next phase that will be presented in the next chapter

4.5. Summary

In this chapter, we have presented the global network with added messaging capabilities. The chapter first presents analytically the prototype of the network, which consists of a few nodes. The prototype network is associated with communication of the nodes and illustrates the communication between nodes. The evaluation of the prototype model was successful. Communication has been achieved. Then the global network is presented for further testing and evaluation. In the evaluation we discuss the network performance and present analytically the results of our research. The network operates very well in the hazardous environment. The challenges are met as the proposed architecture can overcome the problems as described in the criteria for an emergency network and the proposal. The results obtained were compared with other approaches and studies. It is worthy to notice that the performance of such a network is affected by dynamic conditions as the surrounding environment introduces rapid changes. Mobility, interference and broken links are the most important parameters affecting the network performance where implemented in global network. The deployment of the global network was successful. Its performance is considered very good as from the comparison we have achieved a delivery ratio near to 1. Similar studies and approach where compared to our proposed network for evaluation purposes. The objectives of the thesis regarding a messaged enabled network were achieved as both the prototype and global network can operate in hazardous environments fast and reliable. The architecture can provide communication within a hazardous environment, 150 seconds after the time of the event, while the network is introducing stability before and after convergence time (stabilization). The deployment of the network (150 seconds) can be considered faster than the conventional ways of disaster recovery, which require a week for fully recovering the destroyed infrastructure. The Criteria for the emergency network regarding the messaging component where fully met. The network introduces a delivery time of an average of 1-2ms while the conventional GSM networks has a delay 15 seconds. Furthermore the delivery ratio or the proposed network exceeds

99% and is similar with the compared study of AVACOM which shows the same results for 3 different GSM networks.

CHAPTER 5. A Voice Based Network in Hazardous Environments.

5.1. Introduction

In this chapter we will explore the voice enabled network in hazardous environments and we will present the results of our study. The chapter starts by presenting additional information and properties of voice communications related to our research. In the first sections, we will analyse related technologies, methods and a few challenges often met in voice communications. Furthermore, we will present the decisions made in order to overcome problems, the prototype and the global voice network. Then we will introduce the prototype network including voice capabilities, which will be able to operate without any support from infrastructure. Additionally we will present the global voice enabled network and we will analyse its operation. Using a simulated environment, we will evaluate the prototype, the global network and highlight any interesting points. The evaluation and a discussion will follow addressing different issues and comparing our results with related systems.

5.2. Voice Communications.

Voice communication today is one of the greatest fields of research in communications. Many methods and techniques are being currently used in devices at both experimental level and products. New techniques and modifications are being achieved every day in order to optimize the performance of the cellular network. Our research borrows various elements from current technologies. It is well known that the majority of wireless voice systems are based on a wired subsystem. This means that even when the medium of communication is air, between the host and the antenna, powerful-wired equipment is being used for call routing and other network operations. Wired servers can handle even the most demanding applications and services because of their CPU power the memory and the bandwidth resources. The proposed network though lacks infrastructure. For this reason we have to find alternatives and address some problems in order to build a robust, stable and reliable network. Before presenting the configuration of the network, we will present some information on voice communications. The main idea of the message network is to transmit a short burst and release the channel quickly. In this phase the channel must be kept open as the (voice) data flow must be continuous and uninterrupted. The situation is much more challenging as voice communications have a significantly greater degree of complexity than the message enabled network.

5.2.1. Voice Communications and the Full Duplex Problem

The objective of this voice wireless network is to keep constant communication in any channel, as continuous data flow is needed to transmit voice. No infrastructure exists to support the transmission of routing or voice and additionally we are dependant only on the wireless links between mobile hosts. We must ensure that the devices are not stressed; otherwise, degradation of the performance may rise. Quality of voice is a trade off as callers must be able to understand each other, but the bandwidth is limited. Furthermore, we must consider that the 802.11 network recovers quickly from broken links, but it cannot recover bit errors. Errors in voice transmissions are handled by the GSM hardware.

Mobile phones use sophisticated methods for handling voice quality and ensure continuous conversation between GSM subscribers. These methods are not investigated in our thesis. Without loss of generality we consider those techniques as buffers that are able to handle the packet flow from one mobile to another and control their delivery time in such a way that the voice communication is reliable and has very good quality. These techniques (encoding/decoding and jitter) will be discussed within the next few sections as they offer a great advantage. There is a fairly good trade off in terms of CPU, memory, available bandwidth and voice quality. The better the voice quality is the more CPU, memory and bandwidth is needed. Though, in our case this works in our favour. The architecture combines GSM and 802.11 technologies. While GSM handles voice quality, 802.11 handles errors in lines/channels.

For this case study, we are not using any method like echo cancellation [11], or other techniques, which are currently being used in similar systems. It is not our purpose to enhance the network, or evaluate the performance of the mobile phones, but to investigate if the prototype model and the global voice enabled network, will operate under certain circumstances and heavy traffic conditions in hazardous environments.

The first challenge we encounter is that the channel capacity is limited. A good example to understand channel capacity is the following. Imagine a water pipe transferring water from one place to another (Shannon's Theory) [112]. As soon as we supply the pipe with an adequate quantity of water the transfer will be successful. If we try to feed the same pipe with 10x the original quantity of water at the same time, then we are expecting an overfilled pipe with spilled water on one side. The same happens with network links. In our case we have many different pipes to use as there are many devices in the area. It is obvious that the bandwidth consumption between two callers is not the only challenge in wireless networks. Furthermore, the network must satisfy the criteria for an emergency voice network as described in our previous section 4.2 - criteria 7, 8 and 9. On the next section, we will present other factors, affecting wireless voice networks. For the rest of this section we will focus on bandwidth considerations, traffic type and transmission.

Using a suitable type of connection and parameters we can achieve voice communication between two users. The problem though is how to maintain "real time" communication between the two speakers and how to transmit over other connections in case communication is needed through the same channel (link). In order to maintain real time communication we need a real time channel between two speakers. The proposed network is not able to support transmission and reception (full duplex) on the same channel. When a node is sending out data, it cannot receive at the same time and visa versa. Furthermore creating a single pseudo-channel with full duplex properties, it will create more overhead bandwidth traffic as when the user A stops speaking and B wants to reply, we have to reverse the traffic direction of the channel from B to A; then request routing, allocate channel and so on. Considering that this is a complex procedure, involving different control messages between the devices it is possible that it can result in a definite breakdown as the congestion (due to interference and delays) will be increased every time we switch the channel from transmit to receive.

Our approach is different. Instead of using one channel between the two nodes, we are using two. Between user A and user B we are using two channels one constantly transmitting (Tx), from A to B and one constantly receiving (Rx) from B to A. As we have mentioned before we cannot force a node to transmit and receive at the same time. Instead, we can maintain a "real time" speech by allocating the channels and sending the voice in both lines in smaller parts (packets), which is also convenient for obvious reasons like encoding/decoding. Using this method of splitting the data into smaller packets it is possible to transmit, receive and maintain a pseudo "real-time" conversation. It is obvious that the voice is not real-time but the time between of actual hearing and speaking is so small that the brain can perceive it as a normal conversation. This method of transmission is related to modern VoIP, 802.11, and other coding technologies but is using simple transmission between nodes as we do not want to stress the network with complex protocols and routing algorithms.

Before introducing the prototype model and the global network, we will present related technologies to our research as well as considerations about the configuration that will be used in order to create and simulate the voice network.

5.2.2. VoIP – Networks Related to VoIP and 802.11

In the section, it is not our intention to cover VoIP in details as it can be found in many textbooks, but to outline VoIP elements that are related to our research. The basic concept of using voice over IP was initially to cover areas where no public switching telephone network (PSTN) existed. As VoIP evolved with exponential rate, it is now used for long distance calls, for decreasing traffic in telephone networks for connectivity in big enterprise offices and many other applications. As VoIP migrates to 802.11 [12] networks we can bypass the copper wires and form faster, more intelligent voice networks depending always on the current design.

Three variations of 802.11x networks [APPENDIX 2], are suitable for VoIP through 802.11; 802.11a, 802.11b and 802.11g. It is obvious that the trade-off between these protocols is between bandwidth, range frequency and penetrations. The IEEE 802.11b specification [1], allows 11 Mbps transmission within a range of 300 feet for indoor use and a maximum of 20 Km for outdoor under certain circumstances (point-to point). The protocol also supports slower speeds as 5.5 Mbps or 2 Mbps. The trade-off in this case is that increasing the speed results in smaller range of transmission. 802.11b uses 2.4 GHz band. For message transmission, a suitable range with enough bandwidth is 2 Mbps as we can achieve a greater range of communication between the devices while having sufficient bandwidth. 802.11a on the other hand side offers 5.1 GHz for indoor use and 5.7 GHz for outdoor use. RF interference in this case is much less than 2 GHz, because 5 GHz bands are less crowded than 2 GHz. Interference is a very tricky part as we have to keep in mind that 802.11 also operates near 2 GHz. 802.11a also supports speeds up too 54 Mbps and is more suitable for voice applications. Finally, 802.11g variant is an extension of 802.11b and operates at 2.4 GHz. It is more compatible to 802.11b devices and is the optimal solution for indoor applications as it can also achieve a 54 Mbps speed although it is more susceptible to interference as it works in 2.4 GHz band. For our scenario, we will not choose 802.11g as it has even smaller range than 802.11a operating at 11 Mbps. 802.11a also operates at 54 Mbps but consumes batteries quick as high frequencies and high data rates require more energy. Considering a hazardous situation we give the user the privilege of having a device with a descent battery time before it drains. 802.11b may be considered as the optimal solution for our fast deployed network as it has the best trade-off in terms of speed bandwidth, penetrations, frequencies and power consumption. The reason of this choice is that delays or collisions affect the performance of any wireless network. We are trying to maximize the available bandwidth, available channels, minimize CPU usage and battery life of the device. Additionally we minimize the risk of congestion due to delays or interference. This will benefit our proposed network to introduce a stable and reliable performance. After presenting this short decision-making section based on survey, we propose that for this network, 802.11b is the best option as we can use 1-2 Mbps for message transmission and 11 Mbps for voice communications. Finally, the range of transmission and life of battery is greater as seen from this survey.

5.2.3. Codecs Designed for Vo802.11 Networks

All modern devices nowadays use codecs to transmit and receive data. The most often and common codecs used in IP telephony, GSM and wireless today are G.711, G729 and G.723.1 [25], [12] . Initially all of these codecs were designed for circuit-switched telephony. Most of the codecs that are used in mobile telephony are a recent development based on the G.711, G729 and G.723.1. Some textbooks mention that because of the fact those codecs were designed only for circuit-switched networks, they are not suitable for packet-switched networks as they focus more on handling bit errors rather than packet losses. In our case though we are not worried about that because we have a mixed network involving 802.11 and AODV which has a priority to recover links for securing reliable communication. As for the text books which emphasize that the original codecs are not suitable, they also explain techniques for enhanced speech modes which is actually a modified improved version of the codec, based of the original ones (G.711, G729 and G.723.1). This modification actually is already implemented in every device as this problem was introduced as the mobile phones evolved. Concluding this section it is worthy to mention a few of these products which can be found in speech processing codecs (DSP chips) also included in all devices. Some of those are the Adaptive jitter buffer, the enhanced G.711 and the acoustic voice cancellation [12]. The proposed network will use a low data rate codec in order to decrease packet loss robustness and consume low bandwidth.

Avoiding the risk of repetition, we will present later on the communication scheme, which is based on the existing devices and technologies.

5.2.4. Encoding for Voice Comuunications

The main concept of a voice call is to establish communication between two mobile hosts in a wireless environment and make sure that the two speakers are able to understand each other. Since this is a wireless network, a low bandwidth codec is needed. Additionally the system must be able to recover from errors fast in order to achieve an acceptable quality of voice during a call. Also in wireless networks we have to always keep in mind other factors that may affect the conversation as for example if an intermediate node connecting two speakers switches off or moves out of range.

According to our survey, an Ad Hoc AODV algorithm is suitable for this system, as it recovers from errors faster than other routing protocols. For the proposed network, G.723 will be used as it has low bandwidth consumption at 12kbit/sec. This codec offers a fairly good voice quality and is implemented in many devices. It is also well established and performs well compared to other codecs [95]. G.723 is being used in many variations. Very often is being used with a data rate of 5.3 and 6.3 kbit/sec, though there are many cases that it can be used with data rate of 12kbit/sec especially in voice applications [99]. Then in Electronic Engineering Times [100] reports that combination of protocol VoAAL2 and G.723 is slightly more efficient than VoIP because the packets have smaller headers. Additionally G.723 is being used with data rate of 12kbit/sec as it is more close to the GSM-EFR codec that is being used in mobile cellular networks [101].

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Figure 35. Survey in Codec Quality [95].

As GSM mobiles already have codecs for speech transmission we can easily use one of the pre-existing in the device. In this way, we can ensure that the codec used is reliable as it has been tested before, and is compatible with the mobile core software. In other words we can use the phone processor to encode voice. It is clear that it is not our interest to modify a mobile phone or any of the existing technologies but to use the device for encoding/decoding purposes only and then use the wireless network for the transmission.

5.2.5. Delays Affecting the Overall Network Performance

A very important problem in voice networks is the delay that may be introduced on packet transmission. Delays may affect the voice quality as in voice networks there are several reasons that may cause a delay, which leads to a congested network or even worse a network breakdown. As the proposed network is based on mobile phones it is clear that those devices have limited processing power as well as memory. In terms of design we have considered a suitable configuration of the network which requires not only small amounts of memory but also low CPU processing power.

The International Telecommunications Union (ITU) [16], has already commented G.114 for defining a delay specification for transmissions. Delays between 0-150 ms are considered as acceptable for most user applications. 150 – 400 ms are also accepted for specialised applications provided that the administrators of the network are aware about the transmission delay in order to control the impact of the application to other network applications. 400ms and above is not acceptable for common applications however in very special cases this limit can be exceeded. Echo cancellers are used nowadays when the delays exceed 25ms [97]. In most of the voice networks, we can distinguish delays between fixed and variable ones. Fixed delay components usually are due to the hardware parts such as mobile phones or routers and variable delays are due to queuing, buffer sizes, and connections through a wireless network. In our case as the network is an Ad Hoc on demand type and we will have to consider that delays may arise due to broken links, interference, as the network is dynamic with added mobility. In the evaluation section we will present analytically the type of delays that are important in such networks as well as our results. We will not investigate any fixed delays, which are related with the mobile devices, as our scope is to study and evaluate the wireless network behaviour and characteristics, and not the delays that are caused by the mobile phones' hardware.

5.3. A Prototype Voice Model for Fast Deployed Networks

In the previous sections we have seen major factors that are affecting the network as well as decisions for the network setup. As has been discussed previously the parameters of the proposed network are related with the dynamic environment as the network will face rapid changes in the case of a hazardous event. All of the parameters of both the prototype and the global voice enabled network have been chosen after a survey. The criterion for those choices is to build a low consumption bandwidth network, which can handle traffic data under extreme conditions, recover from errors and minimize the risk of congestion.

In this section we will present the model of the prototype network. This model will be used to evaluate the communication between the nodes. Furthermore, we will use it to test properties of the nodes, performance and delays before deploying the global voice network.

The configuration of the prototype network consists of two nodes. The objective is to establish voice communications as described earlier by implementing all the previous parameters in the simulator. Two channels will be allocated and used simultaneously between the selected nodes. Both channels will be sending data in opposite directions. This will emulate the effect of having a real time conversation, as both speakers will be able to speak and hear continuously. The configuration of the network has different types of traffic flows than those for messages. In the message network the objective was to allocate a channel, transmit fast and release it immediately. For voice communication, the channels will be kept open for the duration of the call. They will be released only when the call is finished. This will ensure that the speakers have a priority and the preallocated channels will be used only for their conversation.

The data rate has been set to 11Mbit as this will allow us to maintain a channel able to accommodate voice calls, with message forwarding and routing data transmissions through the same link simultaneously. The key point in achieving various simultaneous transmissions is that different layers can be used for messaging, control messages and voice. Those can be managed by physical layer, while application layer can be used for voice calls.

For both prototype and global network we will assume that GSM service has failed, thus the infrastructure is not operational. The Scenario takes place in Liverpool city centre in Byrom Street and the map in use has dimensions of 3km x 3km. For the prototype model we will use only 2 nodes for evaluating the performance as well as the connectivity and behaviour of the network. The codec will be used is the G.723 as it consumes low bandwidth of 12 Kbit/sec (32 packets/sec) which is a well-established codec in mobile communications. If we were using other codecs as for example GSM or G.711, the voice quality would be similar or a bit better but the bandwidth consumption would be increased up too 50 Kbit/sec depending on the characteristics of the codec. The type of traffic flows that will be used is UDP traffic unicast transmission for each channel and the stabilization of the routing tables is 150sec. This is the default value provided by the simulator, which allows the routing tables to settle down before the initiation of the calls.

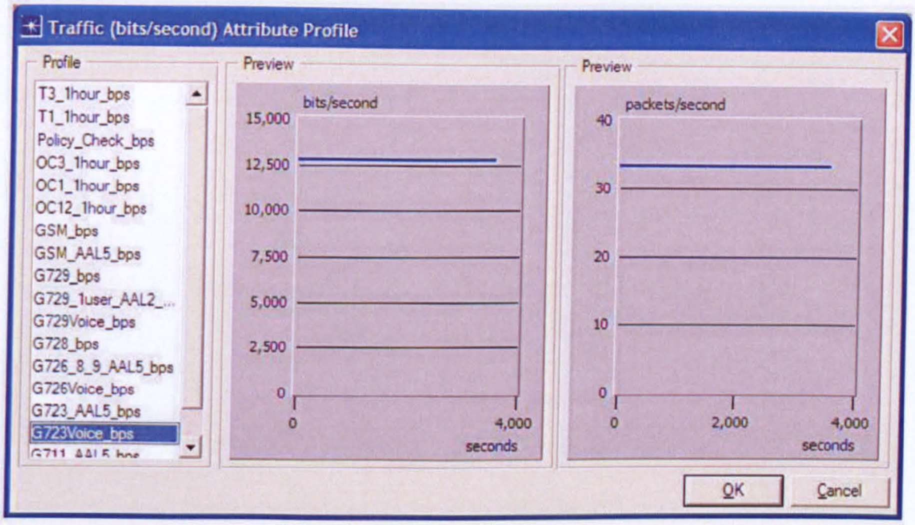

Figure 36. Traffic Profile and Codec G.723 for Voice Transmissions.

Profiles for speed and mobility will be applied later on the global voice network, as this is a prototype model and its purpose is to help us evaluate the communications, the delays and other factors, which may cause degradation of voice quality. The result of the prototype will be presented in the first section of the evaluation in this chapter including plots, statistics and comments.

5.4. A Global Voice Network in Hazardous Environments.

The global network will be deployed in the same map (3km x 3km). The scenario consists of 300 nodes. 20 of them have been marked as they will be monitored for evaluation. 10 marked nodes have been placed in the area of affect (yellow colour) and are limited to move only in the hazardous area as we assume they are trapped. Rest 10 marked nodes have been randomly selected as the relatives or friends of the victims (red colour), and are been placed randomly in the map. In figure 37 we present the simulated environment of the global voice network. Green lines represent the trajectories of the nodes. The node mobility profile has been set to random and the speed varies from 5-10m /sec. Finally thick blue lines represent the connections between nodes (figure. 37).

Figure 37. Global Voice Enabled Network - Scenario Map

As this network is independent of infrastructure, it is clear that the transmission relies in wireless Ad Hoc nodes only. No gateways or routers are been used as in VoIP. We must also take into consideration that the mobile device does not have the same CPU power and memory as a server. There is a big risk in this network for delays, which may result in distorted or broken communication between the users. For this reason it has been decided to investigate the processing delay times as well. Device settings have been applied to the node configuration using a processing speed of 200MHz and having an amount of 64MB of RAM. Those values have been chosen as they represent the average of existing devices currently on the market.

The simulation of the global network will last five minutes as this can be considered sufficient time to evaluate its performance. Stabilization time has been set to 150sec. The calls will be initiated immediately after the stabilization and will end when simulation finishes. In real life, the calls could have different time durations and consume less bandwidth as the speakers tend to speak in mobiles a few minutes or even hours every day. The possibility of using a mobile phone twenty-four hours a day is very small or non-existent. In order to stress the network we will force fully loaded bidirectional links and all calls will not be ended until the end of the simulation. This will help us to simulate a situation in which the callers are speaking continuously, consuming the bandwidth resources and stressing the network.

5.5. Evaluation of the voice network

In the evaluation we will present the results of the prototype and discuss the behaviour of the network. Additionally we will present the global voice enabled network in terms of successful transmissions, jitter delay, end to end delay, processing delay and utilization of links. During the evaluation sections, we will also explain how various factors affect the network, as for example the end-to-end delay. The results will help us decide if the network satisfies the criteria (7-9 voice) of an emergency network in hazardous environments. Finally, we will compare the proposed network with similar systems and highlight any similarities or differences between the compared results.

5.5.1. Evaluation of the Prototype Model

The prototype has been designed for testing the communication of a single call between two nodes in the network. As this is the prototype model we expect a smooth behaviour of the network, without congestion or delays due to interference or broken links. Random mobility has not been applied to the prototype as the main scope is to check if the transmission is successful, by using the traffic characteristics as described in the previous sections.

From the prototype we have obtained the following plots as presented on figure 38. Traffic started after the 150^{th} second of the simulation. The transferred data has been successfully received from recipient without any delays or problems. The traffic type has been set to CBR and the traffic flows using UDP. The functional test graph in figure 38 illustrates the successful transmission between the two nodes. The plots include the transmitted and received traffic in both directions, in bits per second and packets per second. The simulation reported no packet loss and the transmissions were complete at 5 minutes with a constant data rate of 33 packets/ sec at 12.5 Kbit/sec.

The fact that the transmission has been set to use UDP decreases the bandwidth consumption as UDP has smaller headers in size. Additionally by using UDP traffic we can avoid interference due to collisions and retransmissions as if a packet is dropped the network will not try to recover it. It is obvious that lost packets may reduce the quality of the voice in transmission but it is more important to avoid retransmissions as this may cause great delays. The fact that we are not retransmitting lost packets will also benefit the hardware as its buffering queue will not be filled and the processor of the hardware will keep operating normally. This is actually help to keep the devices sending packets without stretching the CPU. We have already seen in the survey that the quality of the voice can be affected and delays will occur depending on the degradation of the service.

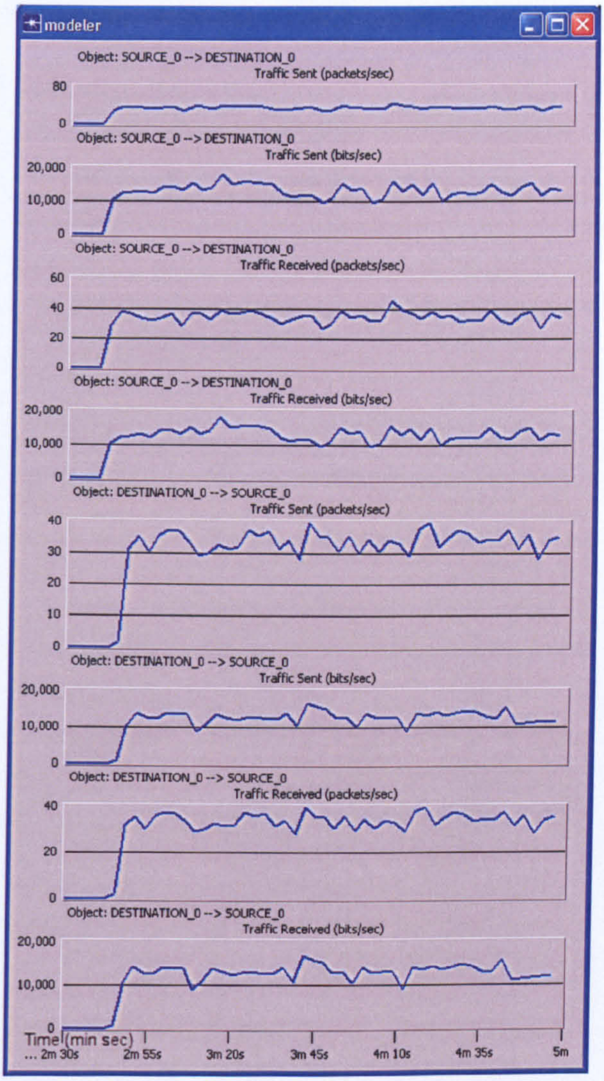

Figure 38. Functional Graph - Successful Voice Transmission of the Prototype Model.

Figure 38 illustrates a functional test graph for the prototype model. On this graph, we see that the traffic has started immediately after the 150th second as described. The top four plots represent the traffic sent from source to destination (node A – node B) in packets and bits /sec. The bottom four plots represent traffic from destination to source (node B – node A). By presenting this functional graph we justify that the criterion 7 that states that the network must provide connectivity for voice calls is met. Both nodes have initiated a successful call, which had no interrupts during the simulation. The second part of the criterion states that the established call must have a good voice quality. This part was also met for the prototype model as will be seen in the next few sections.

In order to achieve good voice quality we have to measure the delay or End-to-End delay (ETE) as it called, between the devices. Jitter delay will show us the time needed for the mechanism (jitter) to handle the delivery of the packets flowing in the link. Finally, the processing delay shows the time the processor needs to handle the data packets of the voice call. For the prototype model we will present our results next. In the next section though we will discuss more about those parameters while evaluating the global network under more realistic environment (implemented mobility, etc...)

Examining the prototype, we have also obtained plots of the end-to-end and jitter delay (figure 39), (for statistics see APPENDIX 4C – figure 56).

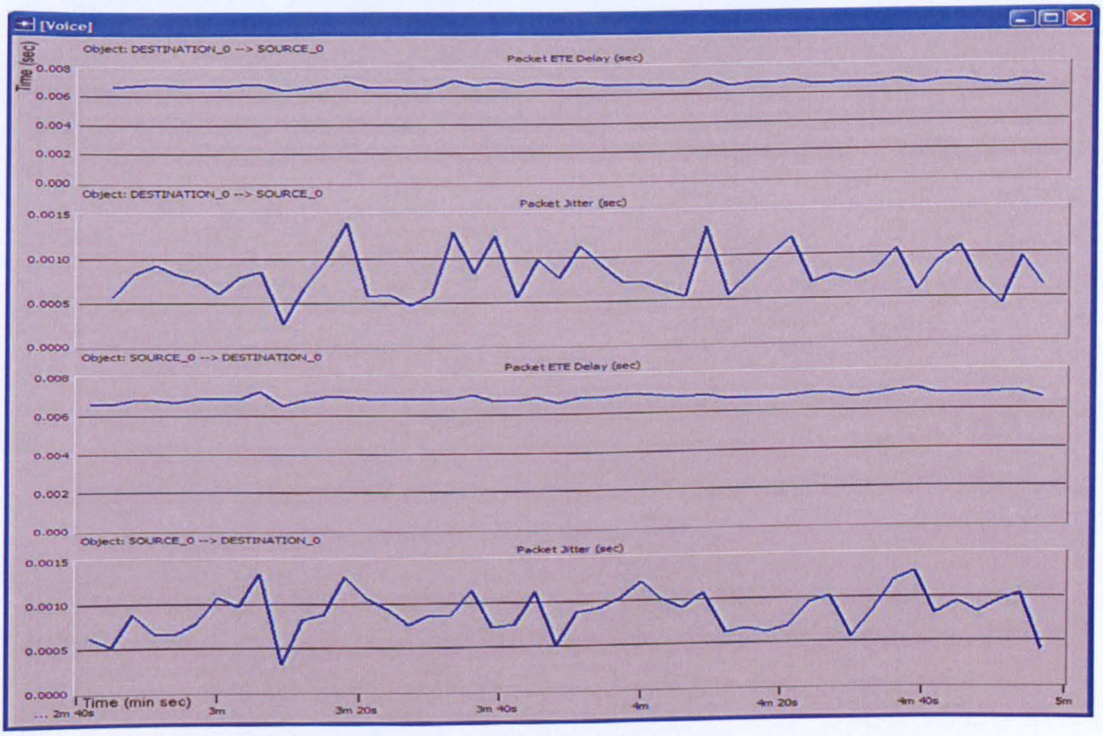

Figure 39. ETE and Jitter Delay for the Prototype Model.

The ETE delay for 2 flows is 7 ms and the jitter delay is 1 ms. For the evaluation of the prototype two different studies [102],[103], related to voice over IP and enhancing TCP performance of Ad Hoc networks, introduce similar characteristics as close to our scenarios. In these papers, there are similarities in their setup compared to ours. The objective of the first research is different but through their results, we observe that the end-to-end and jitter delay as well as mobility characteristics are similar to our. Avoiding repetition, we will present those analytically in the next section as the global network will be also compared to those. For the time being, we mention that the end-to-end delay in paper [103] is varying from 10 ms to 0.2 sec for 10 CBR flows while AODV-SWAN protocol performs better than DSR-SWAN and DSR-INSIGNIA. AODV as described in survey is designed for this type of networks and has an advantage when is used in dynamic networks. Average jitter delay for AODV in paper [103], has been found to vary from 0.01 sec to a maximum of 0.15 sec for AODV-SWAN and 10 CBR flows. Furthermore, in study [102] the average delay for the range of 5 m/sec – 10 m/sec has values of 0.09-0.013 sec. For the prototype, no mobility profile has been implemented yet. The global network, which will be discussed and evaluated in the next section, has a random mobility profile with varying speeds of 5-10 m/sec. There we will discuss the results more analytically.

The fact that the delay is low in the prototype model is not only related since there is no much interference, retries, broken links or other factors that increase overall delays. One of the aims of building the prototype is to check also how the CPU of the device will perform. As it has been previously discussed degradation of service, (which causes delays) often occurs from stretched devices or buffering queues. In our prototype model the results shows that the processor is operating normally (figure 40).

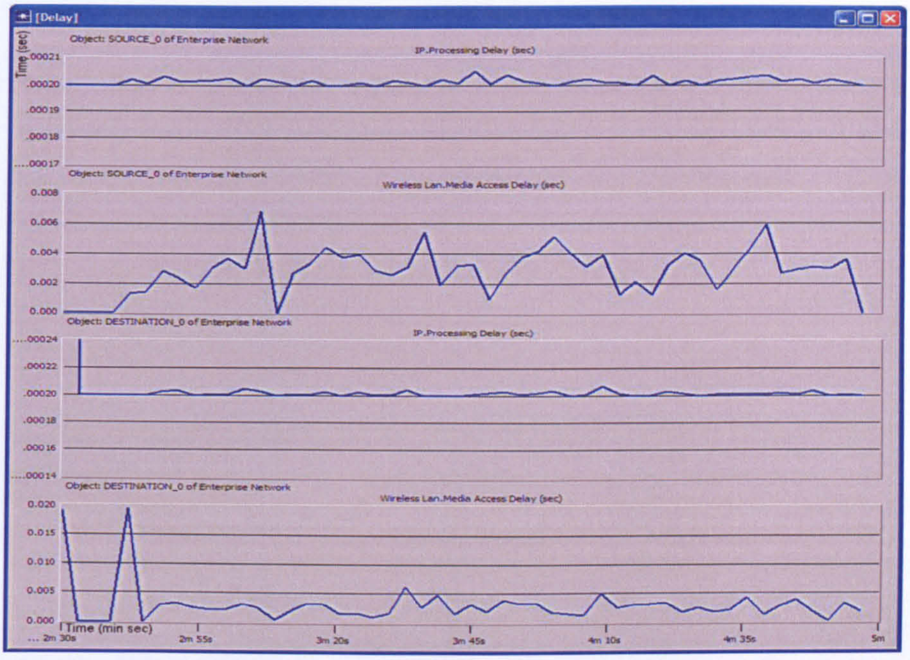

Figure 40. Delays of the Prototype Model.

Figure 40 presents the processing delay and the wireless lan media access delay for the two nodes. The network's delay is 3ms, medium access delay from 1ms to 4ms. The results of the prototype model shows that the prototype model can operate without degrading the service in local disasters. The processor is sufficient to handle voice calls. From this we may agree that any delays that may arise will be due to other factors such as interference but not from the devices. Finally, a conclusion from this section is that criterion 8 (regarding CPU and degradation of service) of a network that operates normally in hazardous environments has been met for the prototype.

According to Nortel Networks [101], G.723 performs satisfactorily at a factor 3.8/5. The factor categorizes the measurement of the quality of voice for a codec and takes values from 1 (bad) to 5 (excellent).We conclude that the quality of the voice is acceptable. Often this codec is used at data rates of 5.3/6.3kbit/sec. Operating it though in 12kbit/sec, will improve the voice quality as there is more available bandwidth for voice data which may lead to even better results. This opinion is also supported in [99] and [100], in which the authors write that G.723 has been found to behave better compared to other codecs in many cases and using different protocols. Using a rate of 12 kbit/sec instead of 5.3/6.3 will allow us to have the same bandwidth for GSM-EFR that is being used for GSM telecommunications. Finally G.723 introduces a total delay of 67.5ms when implemented in DSP chipsets. Though we are not investigating the performance of different hardware it is worth noticing that even in the case when we implement and use the prototype model in real hardware we are expecting a maximum delay of 7 ms (ETE) + 67.5 ms (Codec) +1 ms (jitter) which results in a 75ms ETE delay. According to table 3 of [101] the maximum acceptable delay is 150ms, while up to 400 ms "care is required to assure user satisfaction". The prototype model operates within these limits. More details about speech codecs can be found in [101], and in many other textbooks and papers regarding codecs, their use, advantages, characteristics and delays.

5.6. A Global Voice Network Operating in Hazardous Networks

Compared to the related research about issues in integrating cellular networks [93], the proposed network model has the advantage that is not trying to use alternative hardwired routes to interconnect mobile phones. Instead of using MANET stations or other techniques the proposed architecture is based on the devices only. Additionally the proposed global network can be deployed after 150 seconds in areas suffering from a hazardous event. This is an advantage, compared to the related applications in the survey section.

Evaluation of the global voice network will be presented next. Before presenting the results we will resume some points, as in this scenario we evaluate a dynamic infrastructureless network. It is obvious that heavy traffic is expected to be seen as the network will consume more network resources and more bandwidth. Interference may appear because the nodes are close to each other and have to process more data, for routing and transfer purposes. Furthermore, random mobility has been assigned, which means that there is a possibility of one or more nodes moving out of range. In this case we will observe a situation of retransmission as the AODV will try to recover a broken link. It is very interesting for this case to investigate how the network will behave. 20 nodes have been marked (10 "victims", 10 "relatives or friends") for this scenario and they will be monitored in order to collect statistics on the network. The simulation time has been set to 5 minutes in order to investigate the behaviour of the network at the time of the incident.

In the next section we will present all the statistics that have been collected for various parameters. For the global voice network 300 nodes have been placed on the map. 20 of them are observed. A traffic profile for voice of 12.5kbit/sec has been applied to them. Additionally all of the nodes are exchanging data.

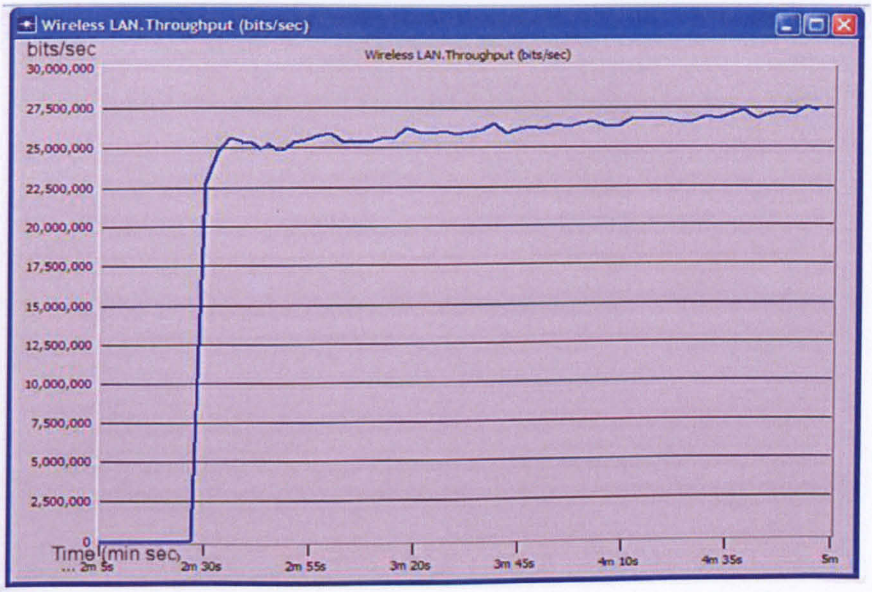

Figure 41. Throughput of Network.

Figure 41 shows that the throughput of the global network after the stabilization time has been found to be 25-27 Mbits/sec. Generally, in such networks retransmissions are not desirable especially when the links transfer voice data as this can cause delays. Furthermore, heavy traffic may cause collisions or even disrupt the communication if packet loss is high or broken links appear. The proposed network introduces small delays. The throughput of the global network is high (due to overhead and voice calls data) and the delay values are generally small, of the order of 7ms. We see that the network benefits from AODV, which offers fast recovery from errors and performs well in dynamic networks under heavy traffic conditions.

5.6.1. Delays Affecting the Network and Comparison

In this section, we evaluate the end-to-end delay for the global network. The end-to-end or ETE delay as it is called, is the time between the packet creation at the source node and its reception at the destination node. The plots obtained from the global voice enabled network introduce delays of an average of 7 ms (figure 42, 43). The peak in the beginning of the transmissions (0.01 s) appears, because at this time all nodes are starting the transmission simultaneously. This causes a short delay until the routes have been established and the channels are allocated to each node. This is often to be seen when all nodes are requesting routing information at the same time as they need some time to establish the route. In a hazardous environment, for example, a local disaster, this behaviour of the network may appear in areas where people are trapped in the area of affect in the case that their mobile hosts are requesting routing information. It is worth noticing that in a real life scenarios the effect of this will be smaller as in the simulation all the nodes requested data at exactly the same time. This is unlikely to happen in real life situation as callers are usually trying to establish calls at different times. In the next figure, we present the end-to-end delay (ETE) for the global voice enabled network, which will be followed by a comparison with related studies.

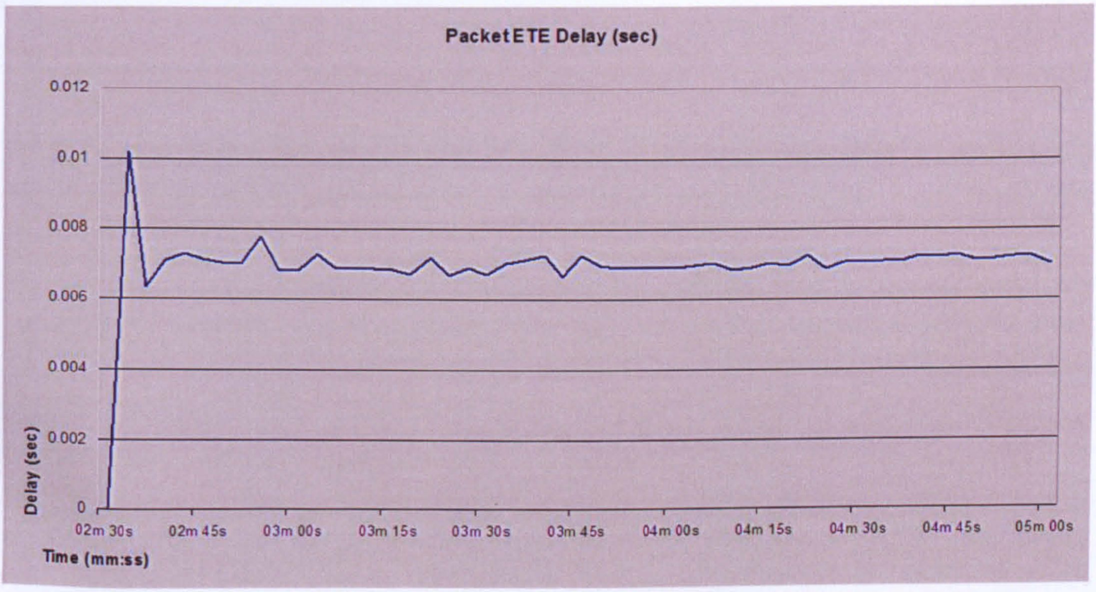

Figure 42. End to End Delay for the Global Voice Network

Wireless LAN

Statistic	Average	Maximum	Minimum
Packet ETE Delay (sec)	0.0071034	0.0102801	0.0002863
Wireless LAN Load (bits/sec)	126,682	265,891	0
Wireless LAN Media Access Delay (sec)	0.0033167	0.0430840	0.0000391

Figure 43. Wireless Las Statistics

In this section, we compare the ETE delay results of the proposed voice network with three similar studies. The first one is related with a presentation of Fabricio Lira Figueiredo entitled "Voice over wireless Ad Hoc networks" [103]. The compared network is an Ad Hoc one, with data rate of 11Mbps, involves real time traffic and voice/data capabilities. Their traffic rate is 64kbps (G.711)+20% overhead =100kbps as they do support data services. Their scenario map has a coverage area of 10km and in their results the present the ETE delay over the number of 10 flows in the network, [103]. Their obtained results are introducing an ETE delay of 10 ms for two flows, which is increased to 0.1 sec for 10 flows. At the same time our network (figure 42, 43 and Appendix 4C) introduces an average ETE delay of 7 ms for up to 20 flows. Comparing those two we can conclude that both are fairly good results. The fact that their result is higher than ours is due to the different size of transmitted traffic. At this point it is fair to mention that it is very difficult to find and compare identical scenarios, because of the nature of the study of Ad Hoc networks. DSR algorithms on the same paper are found to introduce bigger delays of up too 0.8 sec as the number of flows is increased in the network. Clearly, AODV introduces a smaller delay from DSR and the network has greater delays than the proposed one and proves the case that AODV can "behave" and adapt easily to dynamic environment.

Concluding this section we have examined the end-to-end delay for the proposed network and made a comparison to a similar study as described above. The behaviour of the proposed network with the compared one can be characterized as good because it introduces a small delay of 7 ms while it enables voice capabilities. AODV is able to handle communications better than DSR as seen from other case studies while the network meets criteria 6 and 7. More statistics for this section can be found in Appendix 4C.

5.6.2. Mobility Affecting the delays in a Hazardous Environment

The next comparison of the global voice enabled network is related with the mobility of the nodes in the hazardous environment. As it has been seen previously the mobility of the nodes affects the network performance. Out of range nodes are considered as broken links and paths recovered from the network by alternative routes. The nodes which are in range and moving around the dynamic wireless network, also introduce delays.

The mobility profile that has been applied to the proposed network varies from 5-10m/sec (walking/running speed), and the delay for this range has an average of 7ms (figure 43). A similar research has been found entitled "Enhancing TCP Performance in Mobile Ad Hoc Networks" [102]. In this paper TCP performance is enhanced and evaluated using four different protocols (ECN, Newreno, Sack and E-Newreno). In their study, a plot of their work can be found, related to delay vs speed of the moving nodes in a wireless network. Their network involves 20 mobile nodes with FTP traffic assigned and random mobility. The simulation time is 500 seconds and AODV been used as the routing protocol. Comparing the plot with the proposed global voice enabled network, our network has a constant ETE delay of 7ms on average. The compared network has 0.9-0.15 seconds for the speed range of 5-10 m/sec. The use of UDP/CBR traffic has smaller headers and consumes less bandwidth and introduces fewer delays due to retransmissions. Furthermore, the use of a low data-rate codec which is used in our network results to low bandwidth consumption. The global voice enabled network introduces delays of 7ms and can provide communications for victims trapped in a hazardous area.

5.6.3. Evaluating the Scalability of the Global Voice Network

Scalability of the network is important as in real life such a network will include thousands or even millions of nodes. The last comparison of the network is related with the scalability. A similar study entitled "A multi-hop MANET demonstrator tested on real-time applications" involves PDA devices, in order to test real time applications as VoIP calls or streaming [104]. Some of their results involve a scenario which analyses the ETE delay for both 20 and 40 nodes transmitting data. For this comparison, the global enabled network has been scaled with 40 nodes communicating in the area. After the end of the simulation of the global voice enabled network including 40 nodes (speakers) we have obtained the following graph.

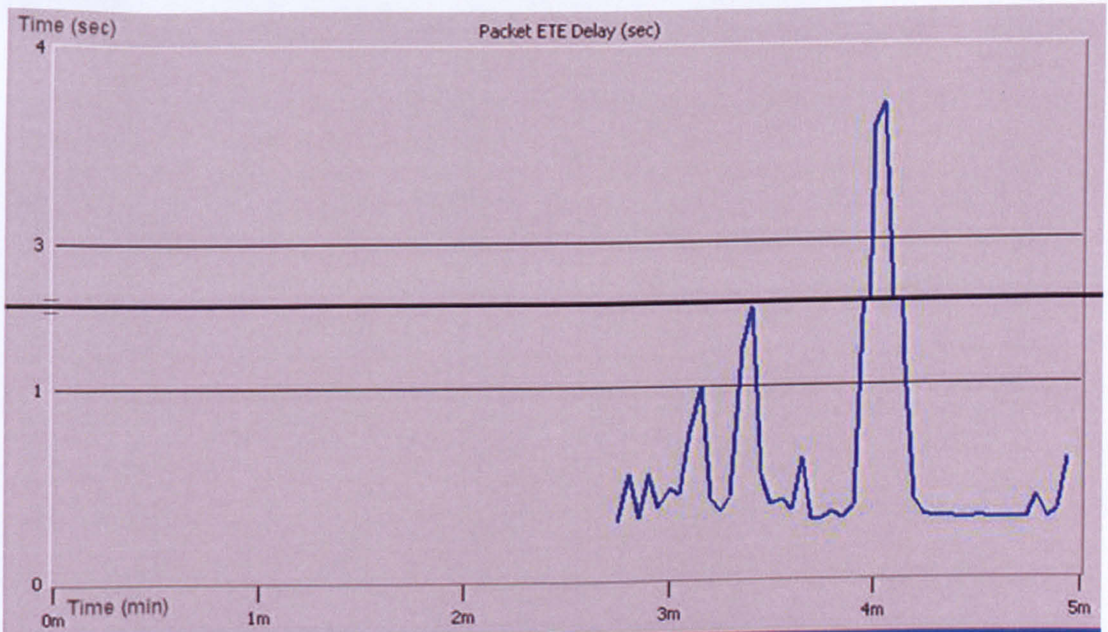

Figure 44. End To End Delay for 40 Nodes for the Global Voice Network

The compared network has inputs of 10, 20 and 40 nodes with speed rate of 0-10 m/sec for 3 different scenarios with different number of nodes. Transmission involves traffic sources with a packet size of 512 bytes, map size of is 1000 x 800 (mxm) and simulation time 900 seconds. Comparing our results to [104], we may observe that the global network has higher delay values than in the compared paper. In our plot we may observe a spike on the graph at t=4mins which appears to be due to a short collision between the nodes. This case is often met in MANET's when the nodes are close together and the available channels are limited. It is also clear that the network also recovers after about 20 seconds as the ETE returns to normal levels again.

Furthermore, it is worth mentioning that the spike appears on the graph means that the system is introducing delays. The fact that the network recovered is related to AODV properties of correcting routes quickly in the case of an error. Though in scaled networks containing millions of users, the same incident could lead the system to a temporary congestion, high data packet loss, broken links and communication disconnections. It is clear that this instability of the system is related with mobility, interference, topology, number of nodes and number of established calls. The fact that network managed to recover from a short congestion means that network satisfies the last criterion for a network in hazardous environments. This is a great advantage of our proposed network because it is able to recover from errors and operate normally after short congestion.

5.7. Discussion

The proposed network has been found to meet the challenges of the next generation emergency service. Furthermore the voice enabled network meets all the criteria for being an emergency network; It provides communication between callers, voice callers and can recover from possible errors quickly. From the obtained results we may agree that the network can be deployed and operate in areas that a hazardous event takes place. Its advantage compared to other approaches is that it can be fast deployed providing communication services to victims. It does not need a human work force in order to be deployed. Additionally, its main advantage is that it is not using the infrastructure which obviously means that even if all the power stations fail due to power failure, destruction due to floods or earthquakes the network will not be affected. The network is working in Ad Hoc mode. Compared to other approaches there is no need for deploying extra hardware, cables or replacing destroyed hardware equipment. As has been discussed in the survey most of the approaches (disaster recovery methods) took at least 2 days to be deployed to restore main communications in hazardous areas. The proposed network can be used after a short period of deployment of the network. When a hazardous situation has been triggered, the network is deployed during the first 150 seconds. The deployment period is needed for identification purposes between the devices and for minimizing the risk of unknown routes at the beginning of the deployment, between the portable mobile devices (mobile phones).

A physical limitation for devices using 802.xx networks is the battery life but this is not of major importance as the network has the purpose to support victims during the first hours of the disaster. Furthermore, a big field of research related with battery life supports the devices by increasing battery life by even 60% [98]. Finally, the proposed network has the potential to be used even under collapsed structures, because it is formed by wireless mobile devices. Wireless devices have a better chance to communicate as long as they are in range, which means that if a victim is in range under certain conditions he/she could call for help, which is not possible to be done using the conventional GSM network. Messaging capability and services are also enabled without using centralized infrastructure. The proposed network also provides a great chance of finding a victim as the rescue team could call the victim's mobile trying to locate (from the ringing sound) the where-abouts of the victim. The proposed fast emergency service has many advantages compared to conventional systems but may introduce a couple of limitations, which we will present in the next section.

In this chapter we have investigated voice communication in a fast deployed network. The network is deployed in 150 seconds. This approach is faster than conventional recoveries in terms of speed of deployment. Conventional approaches may take a week while the proposed network can be deployed within the first 150 seconds of a hazardous event. The network, provides the ability to mobile users to speak through GSM / 802.11 enabled devices even when the GSM network has failed. As it has been proved both prototype model and the Global network performed in a very satisfactory way. They have been found to operate well compared to different approaches. Both models have been designed to operate with low bandwidth consumption and fast recovery through retransmissions.

The proposed network presents two physical limitations. The first one involves the location that is being deployed, as this network has been found to be more suitable for populated and crowded areas. The network will not operate optimally in places like sea or a desert, as physically there are not enough devices in the area to support message transmission, though this is not considered as a problem due to the fact that the original idea is helping people in populated areas suffer from a physical disaster. Obviously if someone is in a desert, even if he manages to send the message calling for help, it will be tough for rescuing services to go there within a reasonable amount of time. As this network works in Ad Hoc mode, infrastructure mode cannot be supported directly. For this reason further investigation is needed in terms of GSM / 802.11 programming that support communication between places where GSM service is active and places suffer GSM service loss.

The second limitation is related with the network architecture. A good hint for a new research objective is new research which will allow our network to establish communication in both areas having GSM service available, with areas suffer from loss of the service. It is clear that in this research we are not building a full product. As it has been discussed the network operates in Ad Hoc mode. This means that it has not been configured to communicate with any kind of infrastructure. It would be a great expansion to make this network operate anywhere independently if the GSM network is operational or not.

In IEEE Communications Magazine (April 2006) Abbas Jamalipour writes that there are many challenges on this kind of networks and time will tell whether this is the beginning of an infrastructureless communication era. We agree that there are challenges as many implementations like new security and routing models may be needed for optimizing and building reliable networks. Such a network cannot be supported only by current devices and technologies. In the near future, we expect that new applications in TCP stack, maybe new or modified protocols and new software, will improve the performance of such devices and networks and enhance its operation.

5.8. Summary

A case study related to voice communications services through a fast-deployed network is presented within a period of 150 seconds, starting at the time a hazardous event has been triggered. The chapter starts with a survey for voice communications for identifying parameters which have been introduced in our research after completing the messaging network. New objectives, requirements and challenges are presented as this network involves voice communications. The "full duplex" problem is examined, followed by codecs and protocols from related technologies like 802.11VoIP. An investigation follows for parameters and factors which may affect the network and the case study scenario of the prototype is presented. The prototype model has been evaluated and the obtained results have been found to be similar to other studies. In terms of bandwidth, delays, time of deployment, call establishment and parameters affecting the network all the objectives are met. The global network is deployed in a simulated local disaster in Liverpool and then it is evaluated. It has been found to introduce similar behaviour with other approaches as lower end-to-end delay results were achieved compared to other approaches. Then a few sections follow, including explanations on testing results and discussion. An observation has been made during the evaluation, that the network may introduce delays. The advantage of the architecture is that the network can recover in a short time. The proposed voice network can operate in hazardous environments and is able to provide communication services and recover while working under heavy traffic conditions. The callers are able to communicate by using the proposed architecture at anytime. The chapter ends by presenting a discussion and few physical limitations have been observed.

CHAPTER 6. Conclusion and Future Work

6.1. Thesis Summary

Chapter 1 introduced the aim of the research. Cellular and wireless technologies were presented, as well as the challenges and the problems. The evolution of telephony was then discussed. All the challenges that arose during the development of mobile systems were presented. Their nature though, is such that they can meet challenges in a cost effective way. During the last decades though most of the challenges were met, but one major challenge was not satisfied. GSM applications in hazardous environments were not investigated. The problem remained as physical disasters and phenomena can harm the GSM network. A new collaborative architecture was introduced in this thesis in order to build such a network that can provide communications in hazardous environments.

Chapter 2 presented an overview for the well known technologies and their components were described. In order to identify and understand the requirements for the various technologies and the new challenges for telecom services, a further analysis on GSM, 802.11 and routing algorithms was presented. Voice communications were outlined while problems were investigated as a telecom network can introduce different type of failures. Furthermore, the progress of real life scenarios and the disaster recovery methodologies was discussed as it introduced new challenges. Finalizing the survey of related work and solutions that have already been deployed, we identified the main disadvantages and weaknesses of the current cellular architecture. GSM has weak infrastructure and its recovery in the case of a failure requires time. We suffer human losses in the hazardous area as the system is not able to provide communication in the time of the event.

Chapter 3 reviewed the properties of a hazardous environment. The provided solution was analysed through a novel architecture. The architecture proposed an Ad Hoc network which is able to be deployed fast in hazardous environments providing communications to subscribers when the GSM infrastructure has failed. The network is overlaid over the damaged GSM infrastructure, providing messages and voice capabilities in areas affected. The proposed network, benefits from GSM and 802.11, as they are operating in parallel to achieve reliable communications. Moreover, the network has the advantage of recovering quickly from connectivity errors, providing communications by using low consumption bandwidth which is crucial when infrastructure is no longer existent. A detailed description of the components of the network, the various details, as the choice of a routing algorithm and its operation were well analysed. The network has been built in such a way it is less vulnerable to interference, offers low consumption in terms of bandwidth and battery consumption and has a great range of communication with neighbouring devices.

Chapter 4 is related with messaging capabilities which were analysed and the messaging component of the system were explained and evaluated. Our results show that the network can operate in hazardous environments, as it introduces similar or better results compared to other approaches and a delivery ratio of messages near to 1. The network is capable of handling messages and performs well in crowded areas such as city centres. Challenges were met for the messaging component, as the objectives were achieved and the problems were solved. The new architecture is able to provide subscribers with an emergency network, which is infrastructure independent, can be deployed fast and provide messaging service at the time of the hazardous event.

Chapter 5 presented the next component of the architecture, which is the voice communication in hazardous environments. A further survey was presented related to voice communications, similar voice techniques, related protocols, encoding / decoding techniques and various parameters that affect voice communication. The survey's scope was not only to inform the reader for the existence of all of the details presented, but also to explore the challenges of the architecture and present a decision making for the global voice network. After understanding the challenges, through the surveys, the network design and its evaluation, this thesis has included a working evaluated model of a network, which can be used in emergency or hazardous situations. The results show that the network can operate in hazardous environments and voice calls can be established, though the system introduces some instability. Short congestion due to heavy traffic, mobility and interference were introduced but the system recovered. Delays were investigated and the results were compared with other approaches. Voice capabilities have been established using a low bandwidth codec but due to the nature of the 802.11 architecture, the system can introduce delays in large topologies. A physical limitation of the network is observed. It cannot operate sufficiently in places like deserts or sea as there are not enough devices in the area to form the network. It performs well in crowded areas.

Chapter 6 summarizes the thesis by presenting the conclusions, the contributions and the future work. A discussion is presented following the key points of the thesis. Furthermore, our conclusions are presented as the proposed novel infrastructure can work for hazardous environments. The proposed network is able to provide communication in hazardous environments. Both messaging and voice components have been found to operate in the affected areas and the evaluation has shown that the network can be deployed successfully in such scenarios. The objectives of the research have been achieved progressively and the experimental procedure has been finished successfully. In the last sections of the thesis, a few key points are presented for further development and future work.

6.2. Contributions

This thesis presents the architecture of an Ad Hoc fast deployed network in areas that suffer a hazardous situation. The proposed architecture presents a fast deployed network with messaging and voice capabilities able to operate in hazardous environments. This network can provide communications for trapped victims in a hazardous environment without using the current infrastructure which introduces weakness in extreme conditions, due to physical phenomena or other hazardous events. The proposed architecture is of major importance as nowadays physical disasters, accidents and other hazardous events occur frequently.

This research provides:

- A new architecture which involves an emergency network. The proposed network can be fast deployed and has both messaging and voice capabilities. This network can be deployed in areas that suffer a physical disaster or any hazardous event without being dependent on infrastructure. Victims in the affected area can use their mobile phones to call for help in the case of an emergency even if the GSM cellular service has failed.

- A survey in telecommunication future in terms of emergency networking was conducted. Furthermore a survey of all related technologies regarding cellular and wireless networking identified key points and challenges for the proposed network as well as weakness of current cellular networks. Additionally an analysis of related work in disaster recovery was presented, highlighting limitations and problems that may rise in cellular networks in hazardous environments.

- A proposal for a network design based on the new architecture was presented. The basic components of the GSM and 802.11 technologies were analysed. Challenges of the current cellular systems as well as their weakness were identified through an investigation on real life scenarios that have been applied in disastrous environments. This investigation has given us the knowledge of how to build such a network, what components to use and how to benefit from them.

- The architecture and the operation of the new proposed network are presented followed by decision making on suitable components and algorithms that can make the network fast and reliable. The novel architecture in the thesis, proposes that the network uses GSM technology for voice communication while 802.11 handles routes and data transfer between the links.

- A global message capable network has been designed, simulated and analysed. The evaluation proves that the prototype is operating and the mobile phones can be connected in a hazardous environment, providing connectivity and communication between users. The global messaged enabled network has been found to operate in hazardous environments and as seen from the evaluation the users in the affected area can exchange messages even when GSM service has failed.

- The deployment of a voice enabled network has been successful. A prototype model has been evaluated in order to ensure communication and connectivity between users. Furthermore a global network has been evaluated in a simulated hazardous environment. The evaluation of the global message network has proved the case that users can use their mobile phones to speak with each other even when the cellular service is not available. This scenario includes a survey of related voice technologies and techniques that are being used, as well as a decision making plan in order to choose suitable components.

6.3. Conclusion

A major problem for a mobile telecommunication network is that it cannot provide communication in the case of a hazardous event. If the infrastructure fails communications are disrupted or fail. The major problem is that the wired sub network components are vulnerable to physical phenomena and disasters. History has proven that we suffer loss of human lives in the first hours of such an event.

The disaster recovery methods have been successful in many cases but they require time and people to work in the hazardous environment. As human life is crucial we are trying to minimise the risk of losses in the affected area and avoid more people to work in it, in order to restore the network. Summarising there is a need for an emergency network which is able to operate in hazardous environments. The proposed architecture provides a network, which operates in Ad Hoc mode in the affected areas, is not affected by infrastructure destruction and can be quickly deployed at the time of the event.

GSM and 802.11 form a network while they operate in parallel in order to provide a messaging and voice enabled network. Most of the approaches are trying to switch to an alternative domain/technology to achieve communication. Our novel architecture combines the two of them to achieve communication. Most of the approaches that have been presented are in need of extra hardware placed in several locations, which can be destroyed in the case of a hazardous event. The proposed network is applied without the need for additional hardware.

The users will be able to communicate in case of emergency using their mobile phones. In the thesis all challenges are met by using the new architecture. In the proposed architecture 802.11 handles routing, connectivity, data transfers and error recovery from broken links while GSM handles encoding/decoding for voice calls, identification for messaging, and the interface of device with the user. Furthermore, challenges like fast deployment, low capacity bandwidth, minimization of the risk for flooding the network and a scheme for initializing the emergency mode operation are solved.

Messaging capabilities are investigated and evaluated. The idea of allocating a channel quickly, transmitting a message and releasing the channel for other users is crucial. The messaging component of the network has been found to operate very well in hazardous environments introducing a delivery ratio near to 1 while other approaches introduce a ratio of 0.6-1. End-to end delay has an average of 0.14 seconds while other approaches introduce higher delays up to 0.6 seconds. Mobility is not affecting the network formed by using AODV routing. In contrast a different approach had an average value of delivery ratio of 0.6, while a different one had similar results with our network. Finally, the total number of routing messages is decreasing as the time passes. As the number of nodes is the same for the time of the experiment this proves that the network will not introduce congestion through flooding.

Voice capabilities have been simulated and evaluated. Voice calls can be made through a bi-directional link between mobile users. The idea of using two channels one for transmission and a second for reception has been proved to work as it minimizes the delays due to route requesting and delays on route establishment. Use of UDP traffic type in combination with a low consumption codec (G.723) results in good speech quality while minimizing the risk of having an explosive behaviour of the networks which can lead in delays and congestion. The evaluation shows that voice calls can be established in the network. Compared to similar approaches the network has performed very well and leads to similar results to other approaches. End-to-end delay has an average value of 7ms while other approaches 0.11-0.6 seconds. The mobility does not affect the performance as the delay times are constant in an average value of 7ms. The network can introduce delays temporarily but it recovers as this is a major advantage of AODV. Interference and heavy traffic conditions have led the system to a short end-to-end delay value of 4 seconds for 15-20seconds while other systems maintained a delay time of 0.6 seconds.

Due to the nature of the proposed architecture as well as the AODV routing algorithm in the network the voice components may introduce delays. The network is able to recover in a short time but future investigation is needed to ensure that the instability that was introduced will be eliminated, as stability of the network can affect the overall performance. In voice global enabled network the results shows that scalability of the network can be affected by a temporary congestion for a short period of time. This is a phenomenon, which occurs often in Ad Hoc networks and this is opinion is supported by other researchers. The advantage though of the proposed network in combination with the AODV algorithm, is that it can adapt easily to dynamic conditions and can recover quickly form errors. The evolution of routing algorithms will solve this instability in the near future, as new algorithms will be discovered. The proposed architecture can be further expanded, implemented in current telecommunication mobile networks and constitute the basis for the next generation emergency networks.

In contrast to current approaches for restoration of telecom services which are based on infrastructure disaster recovery, the methodology and techniques that were used for the design of this network, can be fast and successfully deployed. The network can be considered as an emergency telecom network, which can be used fast whenever it is needed, without requiring a task force to set it up. Its main purpose it to provide communications during the first minutes or hours of any accident, catastrophe or physical phenomenon and can help in avoiding the loss of human lives during the first hours of a hazardous event.

In total this research has been successful in meeting the aims of building an architecture for mobile communications which can operate in hazardous environments. The network was successfully simulated and evaluated in order to meet the set of the thesis objectives. Whilst the network has provided communication for victims within the affected area with their rescuers, the evaluation has proved that the network can also be deployed fast, is not affected by environmental changes and is a survival solution as it can set the base for the next generation emergency service.

6.4. Future Work

In this research, the challenge for a network that can operate in hazardous environment has been satisfied. Yet, there are a few points that can be extended in order to continue this research and optimize the proposed network performance.

An extension of the proposed architecture can be in connecting the network with infrastructure. During a disaster a local area may lack GSM service but surrounding areas may not be affected by the hazardous event. An interconnection between those two areas is needed in order to connect the proposed network with rest of the GSM network and PSTN [90],[93],[94]. This investigation is challenging as an infrastructure independent Ad Hoc network must be connected with infrastructure (routes, access points or other). This will ensure that victims can also call landlines and will expand the capabilities of the proposed architecture.

Further work is needed for the mobile interface. A study related to mobile phones is needed in order to offer the interconnectivity of GSM with 802.11 and make the device able to translate information between two domains. Furthermore, the emergency operation has to be implemented. The most challenging part related to 802.11 though is to design and evaluate a system, which will make the devices capable of localization. This can help forwarding the messages, routing requests and supporting voice calls from sender to recipient more efficiently. Obviously if the device has information about related devices and their location, the routes can be established faster and the risk of flooding or extra delays is minimised.

A prioritization system for handling links, their priorities and the balance of their load across them, could be also researched. Instead of forwarding a second call through the mobile which has already established a call, an alternative routing could be used. This will allow the devices that are not in call to handle already established voice calls and will enhance the performance of the network as well as the performance of the voice quality. In addition, a new modification in the AODV protocol could make it possible to have a real time auto-configured network that can adapt to any environment depending on the traffic condition and the demand for communication services [59]-[62]. Furthermore the network could be evaluated using other routing protocols or modifications of the existing ones. A challenging issue could be to implement functions as extra information on routing tables.

Finally new research of the quality of service can be made in order to evaluate the quality of the calls. It would be interesting to study the performance of the devices as well as the delays that are introduced while they operate in hazardous environments.

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DIAGRAM
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INSTRUCTION
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UNIVERSITY

CHAPTER 8. Appendices

Appendix .1 Cell Sites in Service

Figure 45. GSM Cell sites in Service.

Over the last twenty years GSM telecom service operators add new cells in order to support more users. 1985-2005 [20].

Appendix .2 802.xx Fast Reference

http://searchmobilecomputing.techtarget.com/sDefinition/0,,sid40_gci992311,00.html

The IEEE 802 Standard comprises a family of networking standards that cover the physical layer specifications of technologies from Ethernet to wireless. IEEE 802 is subdivided into 22 parts that cover the physical and data-link aspects of networking. The better known specifications (bold in table below) include 802.3 Ethernet, 802.11 Wi-Fi, 802.15 Bluetooth/ZigBee, and 802.16. The following table lists highlights of the most popular sections of IEEE 802 and has links for additional information:

802	**Overview**	**Basics of physical and logical networking concepts.**
802.1	Bridging	LAN/MAN bridging and management. Covers management and the lower sub-layers of OSI Layer 2, including MAC-based bridging (Media Access Control), virtual LANs and port-based access control.
802.2	Logical Link	Commonly referred to as the LLC or Logical Link Control specification. The LLC is the top sub-layer in the data-link layer, OSI Layer 2. Interfaces with the network Layer 3.
802.3	**Ethernet**	"Grandaddy" of the 802 specifications. Provides asynchronous networking using "carrier sense, multiple access with collision detect" (CSMA/CD) over coax, twisted-pair copper, and fiber media. Current speeds range from 10 Mbps to 10 Gbps. Click for a list of the "hot" 802.3 technologies.
802.4	Token Bus	Disbanded
802.5	**Token Ring**	The original token-passing standard for twisted-pair, shielded copper cables. Supports copper and fiber cabling from 4 Mbps to 100 Mbps. Often called "IBM Token-Ring."
802.6	Distributed queue dual bus (DQDB)	"Superseded **Revision of 802.1D-1990 edition (ISO/IEC 10038). 802.1D incorporates P802.1p and P802.12e. It also incorporates and supersedes published standards 802.1j and 802.6k. Superseded by 802.1D-2004." (See IEEE status page.)
802.7	Broadband LAN Practices	Withdrawn Standard. Withdrawn Date: Feb 07, 2003. No longer endorsed by the IEEE. (See IEEE

		status page.)
802.8	Fiber Optic Practices	Withdrawn PAR. Standards project no longer endorsed by the IEEE. (See IEEE status page.)
802.9	Integrated Services LAN	Withdrawn PAR. Standards project no longer endorsed by the IEEE. (See IEEE status page.)
802.10	Interoperable LAN security	Superseded **Contains: IEEE Std 802.10b-1992. (See IEEE status page.)
802.11	**Wi-Fi**	Wireless LAN Media Access Control and Physical Layer specification. 802.11a,b,g,etc. are amendments to the original 802.11 standard. Products that implement 802.11 standards must pass tests and are referred to as "Wi-Fi certified."
802.11a		• Specifies a PHY that operates in the 5 GHz U-NII band in the US - initially 5.15-5.35 AND 5.725-5.85 - since expanded to additional frequencies • Uses Orthogonal Frequency-Division Multiplexing • Enhanced data speed to 54 Mbps • Ratified after 802.11b
802.11b		• Enhancement to 802.11 that added higher data rate modes to the DSSS (Direct Sequence Spread Spectrum) already defined in the original 802.11 standard • Boosted data speed to 11 Mbps • 22 MHz Bandwidth yields 3 non-overlaping channels in the frequency range of 2.400 GHz to 2.4835 GHz • Beacons at 1 Mbps, falls back to 5.5, 2, or 1 Mbps from 11 Mbps max.
802.11d		• Enhancement to 802.11a and 802.11b that allows for global roaming • Particulars can be set at Media Access Control (MAC) layer
802.11e		• Enhancement to 802.11 that includes quality of service (QoS) features • Facilitates prioritization of data, voice, and video transmissions
802.11g		• Extends the maximum data rate of WLAN devices that operate in the 2.4 GHz band, in

		a fashion that permits interoperation with 802.11b devices • Uses OFDM Modulation (Orthogonal FDM) • Operates at up to 54 megabits per second (Mbps), with fall-back speeds that include the "b" speeds
802.11h		• Enhancement to 802.11a that resolves interference issues • Dynamic frequency selection (DFS) • Transmit power control (TPC)
802.11i		• Enhancement to 802.11 that offers additional security for WLAN applications • Defines more robust encryption, authentication, and key exchange, as well as options for key caching and pre-authentication
802.11j		• Japanese regulatory extensions to 802.11a specification • Frequency range 4.9 GHz to 5.0 GHz
802.11k		• Radio resource measurements for networks using 802.11 family specifications
802.11m		• Maintenance of 802.11 family specifications • Corrections and amendments to existing documentation
802.11n		• Higher-speed standards -- under development • Several competing and non-compatible technologies; often called "pre-n" • Top speeds claimed of 108, 240, and 350+ MHz • Competing proposals come from the groups, EWC, TGn Sync, and WWiSE and are all variations based on MIMO (multiple input, multiple output)
802.11x		• Mis-used "generic" term for 802.11 family specifications

802.12	Demand Priority	Increases Ethernet data rate to 100 Mbps by controlling media utilization.
802.13	Not used	Not used
802.14	Cable modems	Withdrawn PAR. Standards project no longer endorsed by the IEEE.
802.15	**Wireless Personal Area Networks**	Communications specification that was approved in early 2002 by the IEEE for wireless personal area networks (WPANs).
802.15.1	Bluetooth	Short range (10m) wireless technology for cordless mouse, keyboard, and hands-free headset at 2.4 GHz.
802.15.3a	UWB	Short range, high-bandwidth "ultra wideband" link
802.15.4	ZigBee	Short range wireless sensor networks
802.15.5	mesh network	• Extension of network coverage without increasing the transmit power or the receiver sensitivity • Enhanced reliability via route redundancy • Easier network configuration - Better device battery life
802.16	Wireless Metropolitan Area Networks	This family of standards covers Fixed and Mobile Broadband Wireless Access methods used to create Wireless Metropolitan Area Networks (WMANs.) Connects Base Stations to the Internet using OFDM in unlicensed (900 MHz, 2.4, 5.8 GHz) or licensed (700 MHz, 2.5 – 3.6 GHz) frequency bands. Products that implement 802.16 standards can undergo WiMAX certification testing.
802.17	Resilient Packet Ring	IEEE working group description
802.18	Radio Regulatory TAG	IEEE 802.18 standards committee
802.19	Coexistence	IEEE 802.19 Coexistence Technical Advisory Group
802.20	Mobile Broadband Wireless Access	IEEE 802.20 mission and project scope
802.21	Media Independent Handoff	IEEE 802.21 mission and project scope

802.22	Wireless Regional Area Network	IEEE 802.22 mission and project scope

Lisa Phifer and Jim Trulove contributed to this updated guide. (February 2006)

Appendix .3 OSI Model

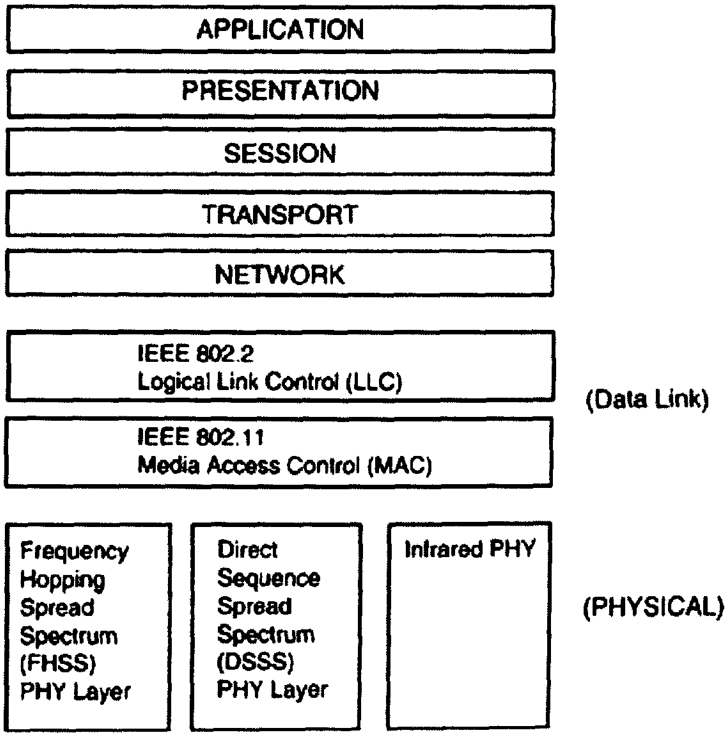

Figure 2-4: IEEE 802.11 standards mapped to the OSI reference model

Figure 46. IEEE 802.11 Standards mapped to the OSI reference model.

Appendix .4A Global Message Enabled Network Collected Statistics

Network Convergence Activity and Duration

Statistic	Average	Maximum	Minimum
IP Network Convergence Activity	0,00185	1	0
IP Network Convergence Duration (sec)	0,11081	0,11081	0,11081
IP Traffic Dropped (packets/sec)	0	0	0

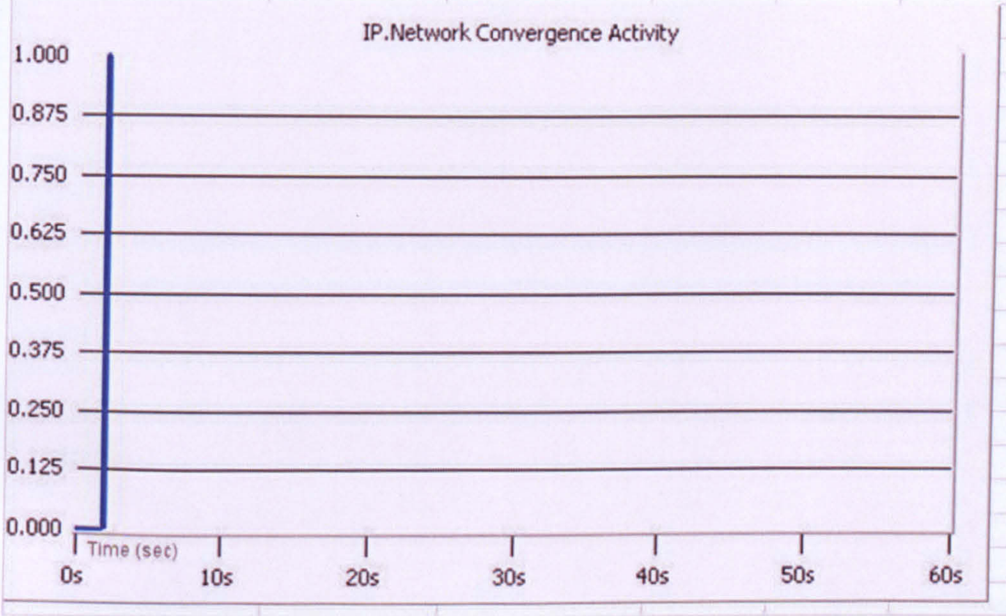

Figure 47. Network Convergence for Message Enabled Network

IP Processing Delay (sec)

Sort By Node	Sorted By Average	Sort By Peak
DESTINATION 0	0,00023774	0,0023133
SOURCE 0	0,0002	0,0002
Middle Node	0,0002	0,0002

Figure 48. IP Processing Delay (sec) for Message Enabled Network

Traffic Dropped (packets/sec)

Statistic	Average	Maximum	Minimum
IP Traffic Dropped (packets/sec)	0	0	0

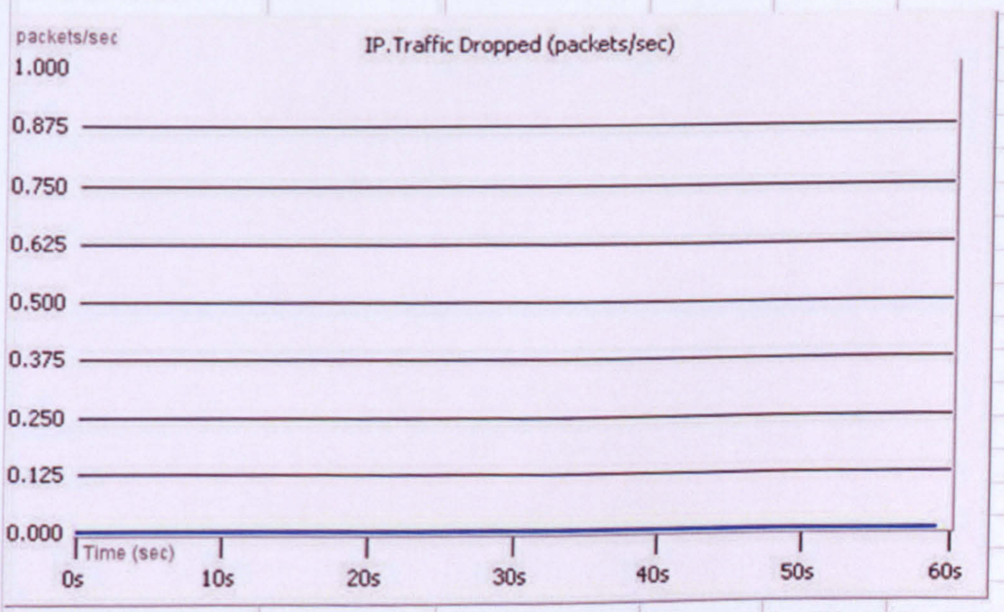

Figure 49. Traffic Dropped for Message Enabled Network

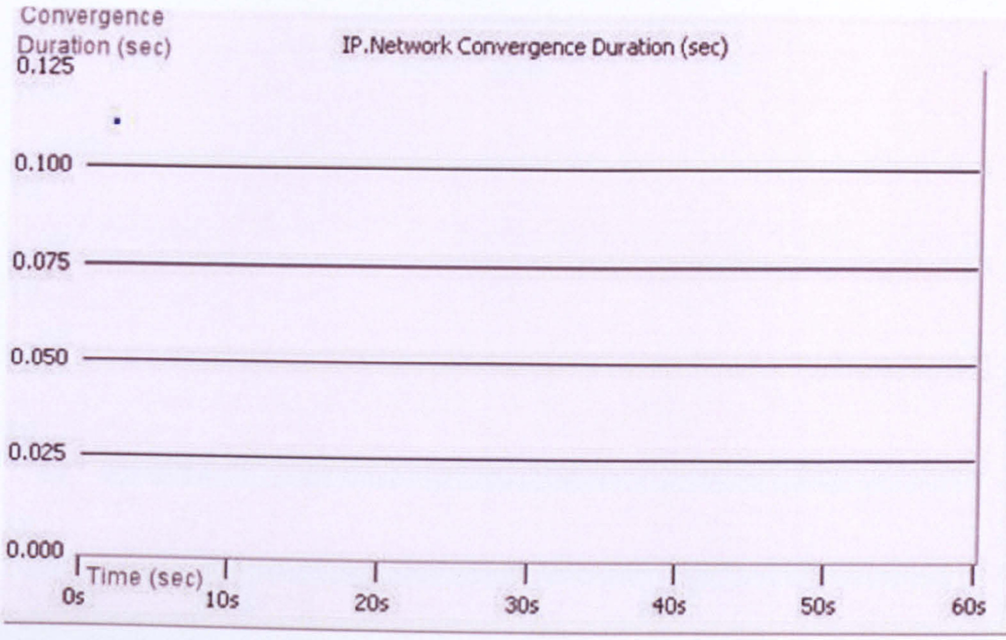

Figure 50. Convergence Duration for Message Enabled Network

AODV Number of Hops per Route

Sort By Node	Sorted By Average	Sort By Peak
DESTINATION 0	1	1
SOURCE 0	1	1

Figure 51. AODV – Hops per Route for Message Enabled Network

Appendix .4B SMS Average Delivery Time

REFERECE: [109]

TELECOMWORLDWIRE-30 September 2005-ANACOM reveal results of Portuguese SMS study(C)1994-2005 M2 COMMUNICATIONS LTD http://www.m2.com

Results of a quality study looking at the accessibility, delivery rate, delivery time and variation in time of the SMS service provided by three networks in Portugal have been revealed by ICP - ANACOM.

The study looked at Optimus, TMN and Vodafone from 30 May to 6 June 2005 in Lisbon and Porto and found that of the 51,538 attempts to send test messages, more than 99.9% were successful. The study found that over 99.7% of the test messages were delivered within the time frame defined by the European Telecommunications Standards Institute (ETSI) of up to 175 seconds. For Vodafone, 99.87% were delivered within ETSI's time frame, error free and to the destination terminal, with TMN attaining 99.8% and Optimus 99.53%.

ANACOM said over 99% of the messages were delivered in under 20 seconds, with the average delivery time of a short message being 12 seconds. In terms of the individual operators, Vodafone was given a 100% success rate in SMS sending, with TMN at 99.9% and Optimus at 99.83%. Average delivery time, delivery rate and service availability were not seen to differ substantially from working days to weekends or at varying times of the day.

http://www.anacom.pt/template12.jsp?categoryId=167302

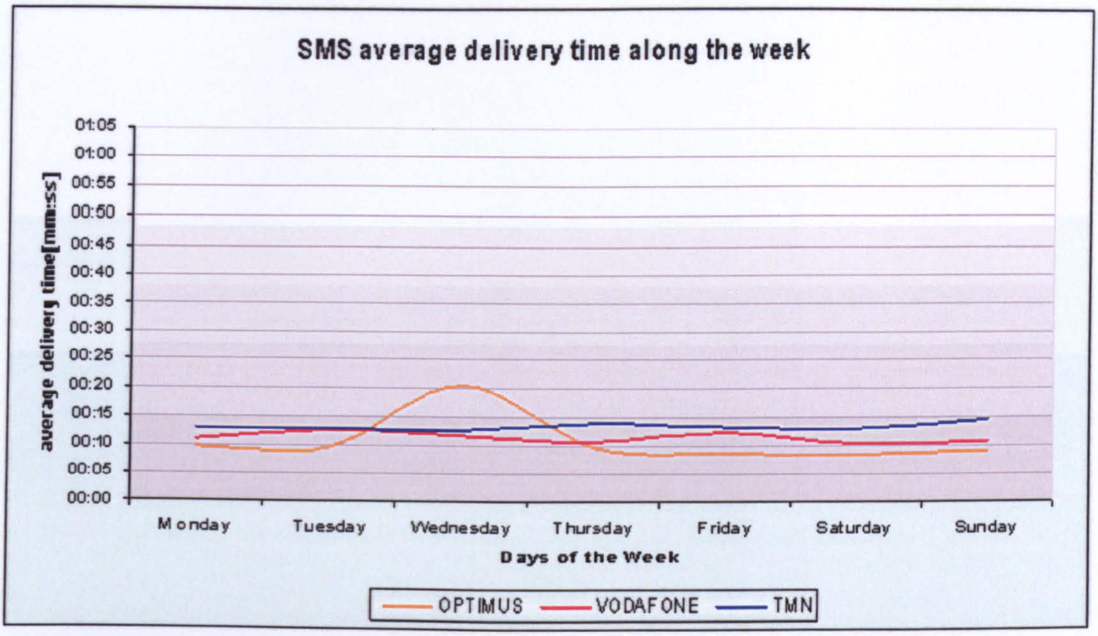

Figure 52. SMS Average Delivery Time Along the Week

SMS		OPTIMUS	VODAFONE	TMN
Sent	Total	17.180	17.180	17.178
		100%	100%	100%
	Received at SMSC	17.151	17.180	17.176
		99.83%	100.00%	99.99%
	Not received at SMSC	29	0	2
		0.17%	0.00%	0.01%
Delivered	Total	17.127	17.163	17.150
		99.69%	99.90%	99.84%
	Error free SMS	17.126	17.161	17.150
		99.69%	99.89%	99.84%
	Error SMS	1	2	0
		0.01%	0.01%	0.00%
Not delivered		53	17	28
		0.31%	0.10%	0.16%
Delivery time above 175 s		26	4	6
		0.15%	0.02%	0.03%
Doubled (received)		0	1	2
		0.00%	0.01%	0.01%
Delivery Time	Minimum	00:00:04,25	00:00:07,12	00:00:07,81
	Average	00:00:10,62	00:00:11,24	00:00:13,21
	Maximum	00:31:01,64	00:05:52,75	00:05:40,59

Figure 53. Collected Statistics From The Survey on SMS Delivery Time

System Performance

The table opposite shows that on average messages arrived at the user's mobile device within 10 seconds of being sent from the SMS message generator. This was considered an acceptable delay and comparable to existing commercial SMS services.

[4] This is less than 100% since some users had their mobile phone switched off for more than 3 consecutive days,
after which time the system stopped attempting to resend the undelivered messages.

Total messages sent	5681
Total messages received by users	5393
Percentage of messages delivered[4]	95%
Average message delivery time (from SMS service provider to user's mobile device)	< 10 seconds

Appendix .4C Global Message Enabled Network Collected Statistics

Wireless Lan Delay (sec)-Average Values

Node	IP Processing Delay (sec)	Wireless Lan Delay (sec)	Wireless Lan Media Access Delay (sec)
D_1	0.00024037	0.0032263	0.00009083
D_8	0.00024307	0.0032070	0.00006787
D_9	0.00050633	0.0052912	0.00006948
S_9	0.00022129	0.0045274	0.00018446
mobile_node_420	0.00020049	0.0033516	0.00003085
mobile_node_439	0.00020781	0.0031873	0.00061817
mobile_node_446	0.00020265	0.0033569	0.00005669
mobile_node_483	0.00020252	0.0032205	0.00040462
mobile_node_533	0.00020817	0.0031910	0.00043480
mobile_node_609	0.00020267	0.0033550	0.00005467
mobile_node_642	0.00020523	0.0032131	0.00050942
mobile_node_869	0.00021472	0.0032291	0.00029120
mobile_node_871	0.00020688	0.0031893	0.00066442

Figure 54. Wireless Lan Delay (sec) – Average Values

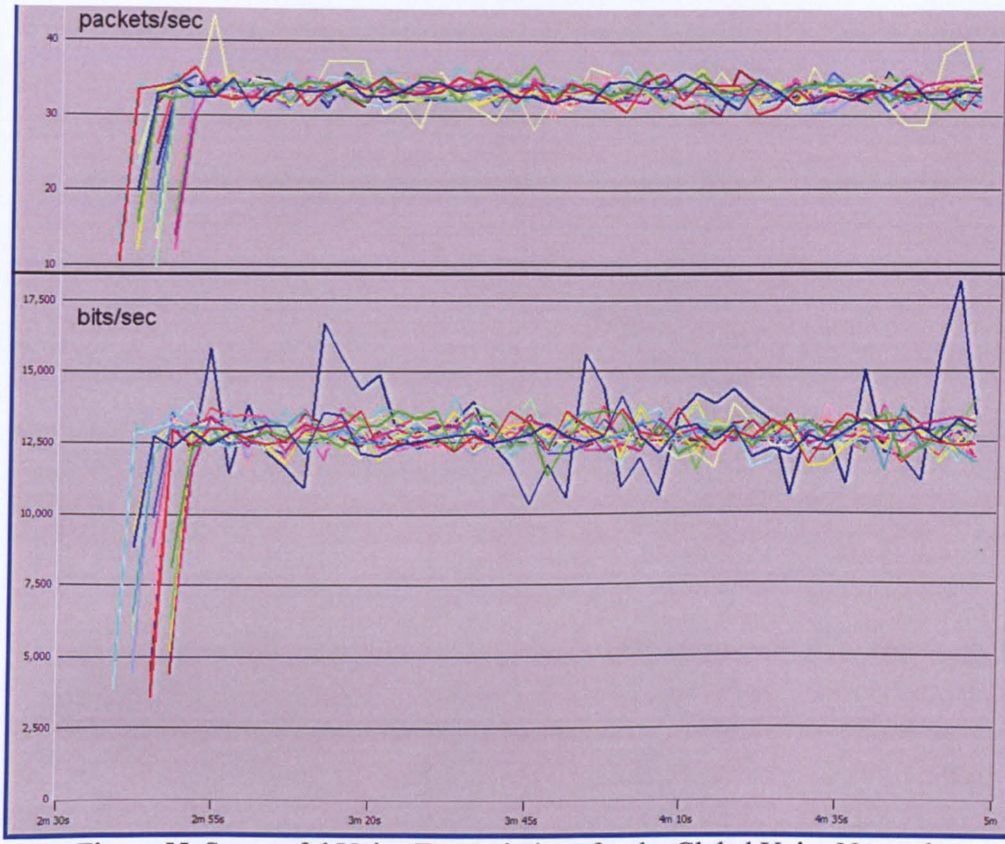

Figure 55. Successful Voice Transmissions for the Global Voice Network

Statistics for the Prototype Voice Network

Statistic	Average	Maximum	Minimum
Packet ETE Delay (sec)	0.0067764	0.0071136	0.0064873
Packet ETE Delay (sec)	0.0068139	0.0072478	0.0065008
Packet Jitter (sec)	0.0008537	0.0013378	0.0003145
Packet Jitter (sec)	0.0007998	0.0013972	0.0002615
Traffic Received (bits/sec)	6,126	18,045	0
Traffic Received (bits/sec)	5,669	16,803	0
Traffic Received (packets/sec)	16.145	45.667	0.000
Traffic Received (packets/sec)	14.897	40.000	0.000
Traffic Sent (bits/sec)	6,126	18,045	0
Traffic Sent (bits/sec)	5,669	16,803	0
Traffic Sent (packets/sec)	16.145	45.667	0.000
Traffic Sent (packets/sec)	14.897	40.000	0.000

Figure 56. Statistics for the Prototype Voice Network