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An Intelligent Personal Communication System

by

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**University of Science & Technology
Kumasi, Ghana**

**A thesis submitted to
the Faculty of Graduate Studies and Research
in partial fulfillment of
the requirements for the degree of**

Master of Engineering

**Ottawa-Carleton Institute for Electrical Engineering
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Abstract

There are two areas of PCS study where Intelligent Network concepts are involved. One is the service creation aspects of IN and the other is the use of IN capabilities in mobility management and tracking of PCS users. This thesis discusses the use of IN capabilities in the creation of services in PCS. A PCS model based on an IN platform, has been developed. The set of IN features required in the personal communication service are identified. With this set of features, the service independent functionality of IN has been used to model the network resource allocation process involved in the provision of personal communication service to subscribers. The modelling process is done in three parts: modelling the service request, the service independent network functional units and the real-time execution of service logic of individual service requests. Simulation results of the network performance with varying PCS traffic loads are presented.

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To My Dear Mother

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List of Abbreviations and Acronyms

AIN	Advanced Intelligent Network
ATM	Asynchronous Transfer Mode
BCP	Basic Call Processing
BRI	Basic Rate Interface
BSS	Base Station System
BSSMAP	BSS Management Application Part
CCAF	Call Control Access Function
CCF	Call Control Function
CCITT	Consultative Committee for International Telephony and Telegraphy¹
CEPT	Conference of European Post and Telecommunication
CM	Connection Management
CSPDN	Circuit Switched Public Data Network
DFP	Distributed Functional Plane
DP	Detection Point
DSL	Distributed Service Logic
DTAP	Direct Transfer Application Part
EIR	Equipment Identity Register
ETSI	European Telecommunication Standards Institute
FE	Functional Entities
FEA	Functional Entity Actions
GFP	Global Functional Plane
GSL	Global Service Logic
GSM	Global System for Mobile Communication
HLR	Home Location Register
IMSI	International Mobile Station Identifier
IN	Intelligent Network
INAP	Intelligent Network Application Protocol

¹ Now known as ITU-TSS (Telecommunication Standardization Sector)

INCM	Intelligent Network Conceptual Model
ISDN	Intergrated Services Digital Network
ISUP	ISDN User Part
LAPD	Link Access Protocol on the D-channel
MAP	Mobile Application Part
MM	Mobility Management
MSAT	Mobile Satellite System
MSC	Mobile Switching Centre
MTP	Message Transfer Part
NBTDMA	Narrow Band Time Divition Multiple Access
OSI	Open System Interconnection
PCN	Personal Communication Network
PCS	Personal Communication System or Service
PIC	Point In a Call
POI	Point Of Initiation
POR	Point Of Return
PRI	Primary Rate Interface
PSPDN	Packet Switched Public Data Network
PSTN	Public Switched Telecommunication Network
RDTD	Restricted Differential Time Delay
SCCP	Signalling Connection Control Part
SF	Service Feature
SCEF	Service Creation Environment Function
SCF	Service Control Function
SCP	Service Control Point
SDF	Service Data Function
SDP	Service Data Point
SMAF	Service Management Access Function
SMF	Service Management Function
SIB	Service Independent Building blocks
SS7	Signalling System No. 7

SSF	Service Switching Function
SSP	Service Switching Point
STM	Synchronous Transfer Mode
TCAP	Transaction Capability Application Part
TCP	Trigger Check Point
TIA	Telecommunication Industry Association
TMN	Telecommunication Management Network
TMSI	Temporary Mobile Station Identifier
TSSD	Time Slot Sequence Integrity
UPT	Universal Personal Telecommunication
VLR	Visitor Location Register
VPN	Virtual Private Network

Chapter 1

Introduction

1.1 Introduction

The beginning of this century brought about an increased interest in the field of telecommunication with the invention of the telephone set. In recent times there has been another leap forward in the field of communication and this is due to the evolution of personal communication system. With the demand for flexible and more convenient telecommunication services on the rise, technologies and concepts are evolving that will provides such services as mobile telephone service, cordless home telephone service, telefax, fascimile, videotex and paging services. With these services, a subscriber can be reached anywhere at anytime via a very small inexpensive handheld terminal. The user will also have sufficient control on the choice of services. Such services have come to be known as Personal Communication Services (PCS) [1-5].

The increase in demand for PCS has mainly occurred in the business sector. However, with the promise of personal convenience and freedom of mobility through wireless, lightweight, low power, portable handsets at relatively low cost, and current social trends of customizing services to an individual's life style, the PCS is becoming more and more attractive to the general public. Today, in North America alone, there are about 37 million users of mobile radio services [6]. By 1995 this number is expected to double and there are no signs of this rate of growth slackening through the later part of the decade. By the year 2000 we can expect to see a market for mobile unit of over 100 million. The trend is also increasing in Europe [7] and Japan [8]. Recent market studies indicate that one out of every four individuals with income of more than twenty-five thousand US dollars (\$25,000) would like to own a portable telephone set and would like to subscribe to a PCS as soon as service becomes ubiquitous in their normal travel area [6]. Ubiquity of service means the availability of service throughout the normal travel area. In order to provide this capability efficiently, it is imperative, therefore, to have an enormous infrastructure to provide the services to satisfy this demand and in addition to ensure a quality of service comparable that currently available in the wired networks.

Despite the increased interest in PCS, the direction in which PCS network will develop is not fully understood. There are different interpretations and perceived uses of the PCS concept and terminologies in existence today. Nonetheless, the technology to provide PCS is evolving based on presently available technologies. In today's highly mobile society, such technologies as cordless telephone, paging and cellular system are emerging to provide the mobile communication needs. These systems are based on cellular technologies, that is, providing services within areas divided into cells and as the user moves from one cell to the other, the network tracks this movement and follows the user with his service. But each of the aforementioned systems are designed to satisfy only a limited number of portable communication needs and none provides the ubiquitous, universal personal communication services. For instance, current digital cellular systems such as NB-TDMA (for North America) and GSM (for Europe) were designed to operate with vehicle mounted equipments and therefore have inherent design features that render them unsuitable for low power portable handheld usage [1].

Furthermore, vehicles are normally used outdoors whereas true PCS should be available anywhere including within office buildings, shopping malls, sidewalks, etc. Cordless telephone handsets are designed to be wireless extension (access network) of the fixed wired network. They operate at lower power levels and hence are battery efficient, they are generally restricted to their own base unit and therefore have limited coverage area, at most 100–200m around the base unit. In addition, limited channel availability and lack of coordination of channel usage can lead to an unacceptable level of interference and call drops during usage in densely populated buildings. These factors limit their utility in a personal communication environment although most of the complexities needed to provide PCS services, such as channel assignment logic, elaborate signalling mechanism, signal channel quality are present. Radio paging, though inexpensive, provide an incomplete communication in one direction and the user must find a way to complete the communication, usually by going to a near by telephone. Its use as a means to provide PCS is very limited. Certain advanced telecommunication service features provided in stored program control switches, such as automatic call forwarding and call transfer provide some form of service portability however it is still restricted to a wired connection and requires interaction of a user or the system.

The marriage between these technologies will therefore lead to the development of

new technologies to better meet the requirements of personal communication users. PCS can therefore be achieved by the ability to integrate these technologies so that the true needs of the user is realized, that is, ubiquity of service in any environment.

1.2 Literature Review

PCS is undoubtedly an exciting and challenging development in communication in recent times [3–5, 9, 10]. Several technologies and standards are evolving to provide PCS and these technologies come with names like Personal Communication Networks (PCN) , Digital Mobile Radio (DMR), Digital European Cordless Telecommunication (DECT), second and third generation advanced Cordless Telephone systems (CT2, CT3) , Wireless Integrated Networks (WIN) etc. [3, 6, 11–14]. Some applications are expected to penetrate the business market soon. The use of these different names reflects the different concepts that designers and engineers hold on these technologies.

The ultimate goal in PCS is to provide instant communication from any place in the world to any other place without any time or user location limitations at an affordable cost. In other words, the communication should be to person rather than to a location as it exists now. To achieve such ultimate aim, two major challenges need to be addressed. Firstly, the need for a survey to determine what the communication needs of people are and how flexible and convenient should such a communication service be and at what rate subscribers are willing to pay for such a service. Secondly, the need for an agreement on the part of designers and engineers on which technology would best deliver such instant communication service with all the convenience and flexibility. Joint effort is therefore needed to bring PCS to provide the ultimate in telecommunication. In general a better technology means high cost of implementation and a better market for the service is influenced by low equipment and service provision cost. Providing better quality of service therefore calls for addressing these challenges.

1.2.1 PCS Needs

The question of what the user expects from the PCS has been considered by many authors [11, 15–17]. In the surveys conducted, it was found that users want a telecommunication service that is very reliable, flexible, easy to use and inexpensive.

Also, users will prefer a telecommunication services that will provide them instant communication with anyone and anywhere. [17].

Users want to have control over the service features they are subscribing[16]. They want a service that will provide them with the ability to screen calls so that they can know who is calling. Users also expect to have a choice of equipment and service providers and demand that there should be no problems with frequency allocation. Interference and inconvenience from other users should be minimum. These expectations place a number of challenges to the designers and raise a number of questions that need to be answered. For instance, the problem of frequency allocation need to be seriously considered in a places where most of the frequency bands have already been allocated for other purposes. The designer must also look at the capabilities of present technologies and see how best to improve upon these capabilities within the limit of cost constraints to provide the type of services potential users are looking for.

1.2.2 Spectrum Allocation

Since users are demanding services that will allow them to communicate whiles on the move, wireless signalling may be the key to instant communication from any person to any other person, thus there must be a high level of radio components in the technology chosen. The allocation of radio channels for signalling and user access can limit the capacity of the system. Increase in the demand for service as well as the need to provide ubiquitous coverage requires a large radio frequency spectrum. Ubiquitous coverage means the availability of radio channels throughout the normal travel area. Already there is overcrowding on the radio frequency spectrum caused by use of large portion of the radio spectrum for military communication, marine navigation, space exploration , air traffic control, AM and FM radio and television [4]. There is, therefore, the need to determine, the range of available frequencies best suited for PCS. Most often the desired frequencies bands have already been allocated to other systems and services and holder of those frequencies are not willing to give them up for economic as well as market reasons. Therefore, other frequency bands need to be explored. Unless other frequency bands becomes available, the present allocation of spectrum to mobile services is grossly inadequate. The available frequency, therefore, must be used to provide the highest possible utilization. Extensive field trials and studies need to be done with

existing technologies prior to allocating radio frequency spectrums for PCS [5], this way it will prevent the unguarded allocation of frequencies in the spectrum and will determine the appropriate frequency range in the spectrum to allocate for better PCS performance.

In order to provide accessibility in heavily built buildings and tunnels it is desirable to allocate frequencies in the range that can penetrate such buildings. This will help reduce the number of cells needed at such places and associated reduction in the cost to the end user. Frequency allocation must also be compatible with international PCS transmission standards. This will allow easy global access to PCS. Several techniques have been studied that makes efficient use of the scarce spectrum. The most notable ones are multiplexing techniques such as TDMA [18, 19] and spread spectrum techniques such as CDMA [20]. Frequency reuse techniques in macro- and micro-cellular technology is essential to meet the traffic needs given a limited spectrum [21].

1.2.3 Interconnection with other Networks

Although the wireless access network is a key segment of the personal communication network, the fixed network still forms an important part in the provision of an affordable universal personal communication service. Studies has shown that cost effective and rapid service delivery can be realized by using fixed network part of the public switched telephone network (PSTN) as a transport vehicle in PCS [14]. This also assures continuous service since it provides both wired and wireless access. Continuity of service can also be guaranteed by the interworking of personal communication service with other networks. The interconnection must be transparently compatible in order to support PCS services. Presently, most cellular calls either terminates at fixed telephone, the same may be true for PCS. Compatible signalling will be required for such advanced services as call screening, selected call forwarding, call identification etc. As fixed networks evolve towards ISDN, the same services should be extended to PCS subscriber. To address these problems, the appropriate standard bodies have been working to unify the radio access interface for both public network access and other network access [2, 22]. However strict adherence to these standards by implementing bodies is necessary to ensure universal coverage. Results of PCS experiments undertaken by individual companies must also be made available to all interested parties and be used to address

such PCS issues as network interfaces and compatibility, network interoperability, inter-system handoffs, validation and registration and performance specification. For fixed PSTN to be used as part of a PCS network, there is the need to modify the existing telecommunication standards governing switching, common channel signalling, billing etc. in them to meet the requirements of PCS.

1.2.4 Marketplace Issues

The concept of developing a person centered, mobility independent communication service for the mass market has made PCS an exciting and challenging development in communications industry [23]. Recent market studies has shown that PCS has received far greater acceptance in the marketplace than any other telecommunication product or service that has been tested in the past decade [6]. This acceptance is not only from a narrow targeted market but from all socio-economic classes, (i.e., from the businessman to the man on the street). Buyers are ready to subscribe as soon as ubiquity of service is available in their normal travel area. This varies according to geographic area and requires an enormous infrastructural investment. However there is a great price sensitivity among buyers with a large percentage likely to buy at a reasonably low price. The engineer and the marketer must work closely to provide the service at affordable price. Broad international standards for PCS products will produce large economies of scales in manufacturing, which in turn will have a direct impact on the cost of handsets [4]. Participation in the development of international standards is therefore essential as this will give manufacturers access to global demand for PCS products. This will also help reduce the cost of PCS providers and end users buy increasing competition in the global market.

1.2.5 Intelligent Networks

PCS users desire the ability to have some control over the services to which they are subscribing. They want to be able to customize the services to suit their requirements. This requires a system with a high degree of flexibility both in signalling and service capabilities. This makes it necessary to implement PCS with an intelligent network infrastructure [24]. The concept of IN defined by the CCITT, will allow service providers to increase the flexibility and speed of service through service creation and deployment

capabilities [25]. It will provide the service creation environment to allow the user or network provider to freely define newly requested or created services.

The IN concept lies in the abstraction of network capabilities into Service Independent Building blocks (SIB). SIBs are standard reusable network wide capabilities and they are used by the service logic for the definition of services. They enable application developers to rapidly develop and modify IN-based services by providing them with the flexibility to easily reuse the existing capabilities of the network. Several references on PCS [10, 26–29] discuss architectures with high flexibility of service. The use of IN infrastructure in PCS is presented in [10, 30–32]. In [10] a PCS network architecture based on the interconnection of multiple networks and distributed service control architecture are described. The paper argued that IN is an ideal architecture for implementing the PCS service control system but fell short of providing performance results to back this argument. In [29], it is shown that services provided in GSM can be more easily developed using the Intelligent Network Conceptual Model (INCM) approach. The paper also identified the set of SIBs for some GSM applications such as call routing. The work done in this thesis has identified the SIBs for all the basic GSM application to support PCS. In addition we have looked at the performance evaluation of the Intelligent Network by defining some key performance metrics and developing a simulation model using the Block Oriented Network Simulator (BONeS™).

1.3 Thesis Objectives

This thesis has three main objectives :

- 1. To demonstrate that the requirements of a Personal Communication Service can be met by using current set of IN capabilities identified in the CCITT Q.1200 Series Recommendations.**
- 2. To propose an architecture based on existing technology capable of providing personal communication services using the concept of independent functional modules as provided in the IN capability standard**
- 3. To evaluate the performance of the proposed architecture, by modelling the service independent network functional units and the real-time execution of service logic of PCS service requests.**

Implementation of IN concepts in available networks to provide personal communication services has three practical benefits:

1. **Service customizing and rapid service delivery:** With IN capabilities, the time to create and deliver service will be reduced since designers are likely to make use of generic building blocks to create services. Also an infrastructure already exists to enable downloading of the logic programs of the new services into the appropriate nodes in the network.
2. **Less expensive:** The use of existing networks such as the PSTN to transport most of the traffic within the network will greatly reduce the cost of providing service as compared to customizing a completely new system to handle the PCS traffic.
3. **Easy to Use:** Since users are already familiar with the use of systems like cordless telephone, cellular phone etc., building a PCS infrastructure based on these existing technologies will cause the user to adapt quickly to PCS use since the learning time will be shorter. The increased interest then may help PCS to grow.

1.4 Thesis Scope

The objectives of this thesis are achieved by breaking the problem into different steps shown below:

1. A view of PCS evolving from currently available technology is presented. The factors leading to the choice of this view is also presented. This will give us a baseline on which to develop the IN concept. The PCS service concept and the service control architecture are also presented.
2. We describe how the PCS applications can be easily developed using the IN conceptual model approach. In addition, we describe a call model which incorporates ISDN call control and IN feature interaction control.
3. We analyze methods for generating traffic in a PCS environment and determine the SS7 bytes generated within the network.
4. The performance of the model is assessed by simulation using BONEs software package on a SPARC workstation. The performance bottlenecks are determined by investigating various scenarios of providing PCS service.
5. Finally, we present some performance results obtained from the computer simulation,

interpret the outcome and extract factors that influence system performance and suggest recommendations based on performance results for system planning and configuration.

1.5 Thesis Organization

The thesis is divided into six chapters including the current chapter. Chapter 2 starts with a discussion on PCS. The objectives and requirements of PCS are presented. The PTN service concepts are addressed as well as the telecommunication services to support PCS. The chapter also presents a system configuration based on a presently existing technology and provides a detailed description of a signalling architecture. Chapter 3 presents the concept of the Intelligent Network and its capabilities. It also describes the various parts that can be incorporated into our proposed architecture. This will enable us to have a basis for comparison of our simulation results. Chapter 4 covers the details of the traffic generation analysis used in the simulation of the system under investigation. It outlines the various assumptions considered in the simulation and describes the methodology used to estimate the traffic rates and SS7 bytes. The functions of actual modules such as the traffic generation modules and the service switching modules are constructed. These modules as well as the simulation package used are described in Chapter 5. The chapter also provides the simulation results and presents them into appropriate sets of curves which form the basis of discussion. The usefulness and practical application of the results are also discussed.

Chapter 6 summarizes the thesis work and suggests some areas for further research. A detailed description of the BONEs package and the Signalling System Number 7 protocol are provided in the appendices.

Chapter 2

Personal Communication System

2.1 Introduction

PCS is a service concept that allows a person to communicate instantly from any place in the world to any other person, using a single personal telephone number (PTN) by means of a personal radio communicator. In this chapter we discuss the objectives of PCS and the services that can be provided. We define an architecture which evolves from the present day fixed and mobile networks towards the goal of worldwide communication mobility for people wherever they may be at home, in the office or outdoors. The functionality of the various network entities are indicated and a signalling protocol required to support service interoperability is proposed. The concept of intelligent network has been around for a while. In the next chapter, we outline the IN requirements for implementing PCS and describe an architecture for PCS network based on the IN concept presented in CCITT Intelligent Network Compatibility Set 1 along with some implementation aspects of the architecture.

2.2 Objectives for PCS

The primary aim of PCS is to make telecommunication service available to subscribers at all times without mobility induced restrictions and at an affordable cost. PCS therefore aims to achieve the following objectives:

Universal Service Coverage

Telecommunication services should be made available to users regardless of their location. These services must compare to the ones provided by the PSTN in quality, integrity and security and must be available over a wide range of user densities and geographic coverage areas.

Capacity Enlargement

There should be a modular structure for PCS; to start from as small and simple configuration as possible and grow as needed both in size and complexity with a capability to provide service to unlimited number of subscribers.

Equipment Portability

A variety of terminals must be accommodated. Network should provide access via both wired and wireless terminals. The wireless terminals include terminals for voice and data communication ranging from small pocket size terminals to those mounted in vehicles.

Number Portability

A unique user identification number must be provided irrespective of the network through which the service is provided or the physical equipment used. This must be a dialable number and the network must be capable of mapping this number onto a physical equipment. This number is called the Universal Personal Telecommunication Number (UPTN)

Service Portability

Service should be provided by more than one network. Mobile network services should be extended to the fixed networks and vice versa. Mobile users should be connected to other mobile or fixed users through other mobile or fixed (PSTN/ISDN) networks or other telecommunication networks as appropriate. This may require the establishment of signalling interface standards in terms of the OSI model.

User Flexibility

Each user should be able to request a particular service and initiate and receive calls as desired. Users must also be able to customize their service to suit their life style.

Economizing

PCS should make efficient and economical use of radio spectrum. A flexible access technique and an efficient management system for traffic associated with the various services are therefore desirable. The cost of providing the service must also be affordable.

2.3 PCS Requirements

PCS may be either an adjunct to or an integral part of the PSTN/ISDN. The services offered in the PSTN/ISDN and other public networks should as far as possible be available in PCS bearing in mind the differences in the characteristics of the fixed networks and mobile radio environment. PCS may also offer additional services, taking into account the special nature of mobile communication. PCS should be designed to allow international operation and automatic roaming of mobile subscribers and stations. The degree to which the roaming facility between networks (or countries) shall be automatic has not yet been decided [33]. There will also be a need for PCS to have the capability to restrict roaming and access to and from unauthorized mobile subscribers or stations. PCS should be designed in such a way that the location of a roaming user should not be disclosed to the calling user without the agreement of the called party. Conversely, the location of a calling user, when roaming should not be disclosed to the called party without the called party's agreement. The services and facilities offered in PCS may not always be the same over all the system. The PCS should be designed so that under favorable circumstances services requiring high information rates can be provided. For example, if the mobile terminal is stationary, or moving slowly, and is close to a base station which is not heavily loaded otherwise, it may be possible to provide service up to the primary ISDN rate of 1536 (23B+D) or 1920 (29B+D) kbit/s. Under less favorable circumstances (e.g. mobile terminals at greater distances from the base station or when the base station is carrying a heavy load of traffic), the service may be restricted to those using lower rates. It should be possible for terminal stations in PCS to be used in the maritime and aeronautical environment, to the extent permitted by national or international regulatory authorities. PCS should be capable of providing service to a variety of mobile terminal ranging from those which are small

enough to be easily carried on the person to those which are mounted on a vehicle [34]. It should be possible for PCS to be used as a temporary or permanent substitute to fixed networks where fixed network facilities are limited or not available or more generally, where reasons of convenience or economics make it desirable. The system should then be capable of adaptation to these conditions (i.e. higher traffic per user, no movement between cells or even within a cells).

2.4 PCS Service Concept

The PCS service concept is based on a mobility independent service. This is the ability to access and use telecommunication service without any mobility induced restriction, by providing telecommunication facilities which are available everywhere and at all times. This concept is one which is desired by users served by the mobile network as well as the fixed network. Thus it may be considered to be a part of future fixed telecommunication systems as well as mobile communication system. The degree of mobility is likely to be different between the networks. In a mobile network, mobility involves the ability of the user to be in continuous motion (including the stationary condition) whilst accessing and using telecommunication services. This requires the telecommunication services to be available through a spatial volume and ideally at all times. In the fixed network, mobility is conferred by flexibility of access to telecommunication service provision, which is available at discrete locations, in such a way that the user identifies with and may configure, any one of these fixed terminals to meet his requirements. These requirements may then be relocated from terminal to terminal without restriction or extent.

PCS will provide continuous mobility within a coverage area and discrete mobility will be provided by the fixed network to support PCS. PCS can also benefit from this flexibility of access concept. These "higher layers" of service (mobility, flexibility of access) are offered by PCS in addition to the conventional telecommunication services. Services which previously could be considered as specific to separate mobile (radio) systems, such as dispatch services, could now be included in the supplementary telecommunication services of PCS.

Figure 1 Evolution towards PCS

2.5 The System Architecture

As noted earlier, depending on the implementer, PCS may comprise one integrated system or several separable systems. It could be an overlay of existing cellular mobile radio network, an extension of the existing PSTN or a completely separate network offering wireless network access and services. In this section, we present one view of PCS which involves a wide range of communication capabilities and not just microcellular based, low power portable radio access. We believe that the basic structure

Figure 2 PCS Network Architecture

for the PCS can be provided by a system architecture which evolves from the present-day fixed and mobile networks toward the goal of worldwide communications mobility for people wherever they may be, at home or away, with services interworkable with those offered by the fixed networks such as PSTN/ISDN. Figure 1 illustrates this view. To provide services portability across different environment, the interworking of the multiple wired and wireless network must be done using the IN approach.

Figure 2 present a hierarchical PCS architecture which can provide some advanced services by introducing IN capabilities within the architecture. It consists of three layers: the intelligent layer, the transport layer and the access layer.

The intelligent layer include equipments and functions responsible for the storage and retrieval of personal information of all network users. These informations can be made available to the call control functions in the transport layer. It also contains functions responsible for the provision of customer-defined services.

The transport layer include equipments and functions related to the transmission and delivery of information across the network. This include the call control and switching functions. The functions of this layer are directly under the control of the intelligent layer.

The access layer include both the wired and wireless access system. It include equipments and function to provide network access to all users irrespective of the users' location and to provide the means by which users can be reached for service. The UPT service provides a flexible means of accessing PCS services.

Key elements in the network are: the MS (Mobile Station), PS (Personal Station), Cell Site (CS) for PSs, BSS (Base Station System), MSC (Mobile Switching Centre), VLR (Visitor Location Register) and (Home Location Register) for the mobile side. There are other elements that can be included such as EIR (Equipment Identity Register) and AC (Authentication Check) however we include these functionalities as part of the HLR. For the fixed side, key elements are: Cordless Telephone (CT), Plain Ordinary Telephone System (POTS), Local Switch (LS), Toll Switch (TS), Public Branch eXchange (PBX) and a DataBase (DB) site.

The MS and PS are the physical equipment used by the user to access the services of the network. The MS is usually vehicle mounted and is characterized by high rate of motion. The PS, which is usually a pocket-size portable, can be used in various

ways. When used within buildings, the rate of motion is limited to walking speed. PS can be used in a moving vehicle with the MS acting as a tandem link to the network. PS can also be used outdoors for rate of motion up to that of slow moving vehicle. Other access equipment can also be used, for instance MET (Mobile Earth Terminal) to provide access to rural and remote areas via satellite and must be designed to operate at supersonic aircraft speeds.

The BSS terminate the radio links to mobile stations and is composed of BST (Base Station Transceivers) and BSC (Base Station Controllers). The distribution of functions within the BSS is not considered in this thesis and the term BS (Base Station) is used to denote the complete BSS configuration;

The MSC sets up calls to mobile users via the BSS and maintain connections to other users and the PSTN.

The HLR contains the permanent subscriber parameters and features for a group of users in a given network. It contains a pointer to the VLR to assist in routing incoming calls.

The VLR is a dynamic database whose entries move with users and which contains information for all users within one or more location areas. The VLR communicates with the HLR to obtain user parameters and to provide updates regarding the status of user location. The VLR also performs authentication and allocate temporary phone numbers to users.

2.5.1 Radio Interface

The radio interfaces are also shown in Figure 2. The personal station (PS) interface R2 is used in two separate cell plans, one for mainly outdoor and another for mainly indoor coverage. The two cell plans may use different sets of frequencies within the same band. A separate interface R1 and separate frequencies may be envisaged for the mobile station (MS) as distinct from the personal station. However, a common radio interface ($R1=R2$) may be possible, as well as a common frequency band for the different cell plans. The radio interface should be designed to allow different applications to use the same interface where this can be shown to be technically and economically feasible. If the same radio interface cannot be used for all applications then the individual interface should have maximum commonality to allow interworking with the minimum extra

complexity. With the low transmit powers considered for personal station, coexistence with older services (e.g. private wireless PABX) is a possible consideration. It is generally assumed that the location of the personal station changes slowly with respect to the personal base station.

A common worldwide radio interface may be a desirable but not essential goal to achieve mobility, although it may be desirable from the point of view of cost reduction and wider availability of equipment.

2.5.2 Signalling Access Protocol

The support of service interoperability will require the provision of network functions and the definition of signalling protocols for use both over the radio system signalling channels and within the fixed network. The PCS should provide universal personal telecommunication service through a universal service access mechanism such as UPTN. The PCS should perform service origination and termination regardless of user's location. Given a UPTN number, any user terminal can be used as a means for access; consequently, the terminal may vary as access is provided through either mobile or fixed means.

A user-network interface standard needs to be defined so that user terminals will be transparent to transmission technology. CCITT has developed a three stage method for the development of protocols (see CCITT Recommendation L130, Blue Book), used effectively in the development of the ISDN, which may be useful for the design of PCS. The GSM standard of ETSI defines the service requirements and interfaces of a new high capacity 900 MHz digital cellular network operating in member countries in Europe [35]. ETSI has reviewed and modified the GSM recommendations to provide Personal Communication Services in the 1800 MHz band. The standard uses the same basic signalling procedure as GSM and is known as Digital Cellular System DCS1800 [11, 36]. TIA's IS-41 and IS-54 [37, 38] specify the protocol standard for the North America digital cellular networks. They are still under revision.

One of the main goals of GSM is to support widespread roaming. Since widespread roaming is a key factor in PCS, we have adopted the GSM as the signalling protocol to provide PCS service and would be used subsequently.

Figure 3 Signalling Protocol Structure

2.5.2.1 The GSM Protocol Architecture

The GSM is defined in terms of functional entities and interfaces as shown in Figure 2. Two interfaces are defined. They are the radio interface (R1) and the BS to MSC interface (A reference point). SS7 is specified as the signalling protocol connecting different networks. The radio interface is always a physical interface and will support an ISDN-like access via traffic and control channels. The traffic channels are analogous to ISDN B-channel and the control channels correspond to the D-channel. Figure 3 illustrates the layered protocol structure of the GSM. Layering allows the communication procedures to be split into a hierarchy of signalling protocol functional layers. Each layer offers a specific communication services to the upper layer at distinct service access points. Three signalling layers are identified

Signalling Layer 1

This is the lowest layer of the radio interface. It represents the functions necessary to

transfer bit streams on the physical radio link. It offers a number of functional channels to the next layer. The physical layer includes the following characteristics:

- TDMA structure with 8 time slots per radio carrier.
- 124 radio carriers in the 935 — 960, 890 — 915MHz band with 200kHz channel spacing. *Further studies are required on radio propagation at different frequency band to determine which ones are best suitable for PCS application. The DCS1800 considers propagation at 1800MHz.*
- The modulation scheme is Gaussian Minimum Shift Keying (GMSK). *Other schemes have to be investigated for the PCS to determine the one with better spectrum utilization and ease of detection.*

The functional channels use the TDMA structure according to their specific requirements. The advantage of the TDMA is that it allows the mobile stations to measure its radio environment during the idle time between the time slots assigned to a dedicated connection.

The various functional channels are defined in Table 1 and their relation to each other is illustrated in Figure 4. From the figure, two classes of functional channels are defined. The traffic channels and the control channel. The traffic channels are used solely for the transfer of user information. Two information rates are possible. The full rate TCH may carry full rate speech or 9600 bit/s data services. The half rate TCH is intended for the transport of half rate coded speech and data services not exceeding 4800 bit/s.

The multiple access TDMA structure is shown in Figure 5. Transmission on the radio channel occur at the rate of 270.833 kbit/s or 3.692 μ s/bit duration. A single time slot consisting of 156.25 bits will therefore have a duration of 0.557ms and a frame will have a duration of 4.615ms. Figure 5 also shows the different types of bursts which can occur in a time slot. Except the Access burst, all the bursts utilize 148 bits and the rest are used as guard bits. The Access burst is relatively short and has a guard bit of 68.25. This is to prevent any problem arising from time frame misalignment which cannot be tolerated during random access. The synchronization and training sequences are used by the mobile to synchronize itself to the base station. The TDMA frames are organized into multiframes consisting of 26 frames for traffic channels. The traffic channels are organized such that a number of time slots carrying traffic information is followed by a time slot carrying control information. It has been decided [39] that for half-rate traffic,

AGCH	Access Grant Channel: A unidirectional downlink common control channel used to assign a dedicated channel to the MS
BCCH	Broadcast Control Channel: A unidirectional downlink common control channel used broadcast information about system configuration parameters.
FACCH	Fast Associated Control Channel: A bidirectional dedicated control channel used for signalling by higher layers.
PCH	Paging Channel: A unidirectional downlink common control channel used to page mobile stations
RACH	Random Access Channel: A unidirectional uplink common control channel used by mobile stations to indicate a request for a dedicated channel to the network.
SACCH	Slow Associated Control Channel: A duplex signalling channel always associated with FACCH and SDCCH. It is used for power control and transmission of periodic measurements for hand-over.
SDCCH	Stand alone Dedicated Control Channel: A duplex channel used for signalling by higher layers, for instance, during call setup or location update.
TCH	Traffic Channel: Used to transmit speech or user data. Depending on the service type, a full rate traffic channel TCH/F or a half rate traffic channel TCH/H can be hosted by the same physical channel

Table 1 Functional Channels

there is one ACCH time slot for 12 TCH time slot. For full-rate traffic, there is one ACCH time slot for 24 TCH time slot as Figure 6 illustrates.

Figure 4 Channel Structure

Signalling Layer 2

The main purpose of layer 2 is to provide reliable dedicated signalling link connections over the radio channel between the mobile station and the base station. A separate protocol entity is allocated to each logical control channel. The signalling layer 2 protocol, LAPDm, is based on the principles of the ISDN LAPD protocol [40]. However, it is modified according to the specific cellular constraints, e.g. high frame error rate and possible changes of radio channels during a signalling transaction. For instance on the dedicated control channels, both acknowledged and unacknowledged schemes are used whereas on the common control channels only the unacknowledged scheme is implemented. Other functions provided in this layer are:

- signalling layer 2 connections activation and deactivation
- mapping service data units provided from signalling layer 3 into signalling layer 2 protocol units for transmission.

Figure 5 TDMA Frame and Types of Burst, a) Normal burst, b) Synchronization burst, c) Frequency correction burst and d) Access burst

- multiplexing and de-multiplexing of several signalling layer 2 connections on a dedicated control channels.
- delimiting protocol data units for transmission.
- error detection, recovery and notification.
- identification and parameter exchange with peer signalling layer 2 parties.
- adaptation of speed by sender and receiver

On the 'A' interface, SS7 signalling protocol is used [41]. MTP provides the link

Figure 6 a) Multiframe structure for traffic channels, b) Full rate traffic channel, c) Half rate traffic channel.

layer functions and connection multiplexing is achieved by the use of SCCP in the connection oriented mode. A description of SS7 is presented in Appendix A.

Signalling Layer 3

The functions of the signalling layer 3 can be divided into 3 groups and are arranged in a hierarchy of sublayers as shown in Figure 3.

The Connection Management (CM) sublayer include the Call Control (CC) protocol, the Supplementary Service (SS) protocol and the Short Message Service (SMS) protocol. The CC protocol control the end-to-end connection establishment of telecommunication services. The SS protocol performs similar functions as the CC but for supplementary services. The SMS protocol provides the means of transferring short information messages between users and the network. Short message service is presented in [42]

The Mobility Management (MM) sublayer provides the necessary control procedures to enable unrestricted mobility of users. Typically, it provides control for roaming, registration and security. Specific procedures performed under roaming and registration are:

- **Location updating:** this procedure allows the MS to indicate to the network, which location area terminating calls to the MS user be routed.
- **IMSI attach/detach:** this procedure is performed at switch-on/switch-off time to indicate to the network that the MS is active/inactive thereby controlling the unnecessary use of paging messages.
- **Periodic updating:** this procedure is used to periodically confirm the availability of an MS to the network by constantly updating the location databases. It is also used to recover from location registration failures.

Specific procedures performed under security are:

- **Authentication:** this procedure is used to check against unauthorized users of the network. It involves an exchange of information with respect to user identity which is checked by an authentication algorithm in the network.
- **User data confidentiality:** this procedure is used to provide confidentiality of user and signalling information by ciphering data and signalling bits transmitted on the radio path. A ciphering key, which is set during authentication is used and the a ciphering mode setting procedure is used to synchronize both sides of the radio interface prior to ciphering.
- **Subscriber identity confidentiality:** this procedure is used to prevent a particular user's location to be noticed by trespassers monitoring the radio channel. It involves sending a TMSI to the MS in ciphered mode which is used in paging messages instead of the IMSI.

The Radio Resource management (RR) sublayer provides all the necessary control procedures for establishing, maintaining, modifying and releasing the common and dedicated channels. Specific functions performed on the common control channels are:

- random access to the network at the beginning of transaction;
- allocation of dedicated control channels;
- system information broadcasting;

- paging of specific mobile station.

Specific functions performed on the dedicated control channels are:

- assignment and release of specific combination of dedicated functional channels;
- initiation of ciphered operation;
- periodic measurement of the radio link performance;
- handover of a call or a transaction in progress, when a mobile station enters a new cell.

The RR sublayer is terminated at the base station since it only performs functions relating to the radio link. The RR messages are interpreted and processed at the base station and mapped into the Base Station System Management Application Part (BSSMAP). The MM and CM sublayers are both terminated at the MSC. Their messages are directly forwarded without interpretation at the base station via the Direct Transfer Application Part (DTAP). At the MSC, interworking functions between the CM, MM, BSSMAP entities and the corresponding SS7 entities [41] (refer to Appendix A), i.e. ISDN-UP, MAP, SCCP and TCAP are performed.

2.5.2.2 Signalling Procedures

The basic service procedures and supplementary service procedures to which this signalling protocol structure is applied is described in this section. The key signalling procedures to support wireless PCS service are: Location Updates procedures, Call establishment procedures and Handover procedures. A structured approach is used for the specification of these procedures and include the following:

- RR connection establishment
- MM connection establishment
- authentication
- ciphering
- ISDN signalling procedures
- MM connection release
- RR connection release

Procedures for PCS services in the fixed network involves only the ISDN signalling procedures.

Call establishment

These procedures enable a user to connect to another user for the purpose of sharing telecommunication services. The set of messages exchanges in this procedures are described in [35].

Mobile Origination:

These procedures allow calls to be set up from mobile users to either the fixed network or other mobile users. Figure 7 and 8 shows the sequence of information flow for successful call establishment and release at the radio interface and within the network for mobile originated calls. A mobile wishing to initiate a call sends a setup request.

1. The CM sublayer in the mobile station generates and sends a connection establishment request to the MM sublayer. After allocating a transaction identifier, the MM sublayer initiates a connection establishment request to the RR sublayer. The RR sublayer in the mobile station communicate with the RR sublayer in the base station to obtain a dedicated control channel for the transaction. This is done by sending a channel assignment request message on the RACH and setting the appropriate parameters for the dedicated channel. Upon RR connection establishment, a connection indication is sent to the MM sublayer in the network and a connection confirmation is sent to the MM sublayer at the mobile to mark the end of the MM and RR connection establishment phase.
2. Upon the establishment of MM and RR connection, the network initiates authentication by sending authentication request on the assigned dedicated control channel to the MM sublayer of the mobile. This request contains a random number with an authentication key which the mobile station use to compute a result. It sends the result through the authentication response message to the network. The network verifies the result and if correct, initiates cipher mode setting procedures. During authentication, cipher mode setting parameters are also exchanged. The ciphering procedure involves synchronizing both sides of the radio interface and assigning TMSI for use on the radio channel instead of IMSI.
3. After authentication and ciphering, call establishment will then be initiated using procedures similar to ISDN user-to-network signalling procedures. A SETUP

message is sent from the CM layer in the mobile to the network. This message contains the address information, requested network bearer capability and compatibility information. The network accepts the call establishment by returning a CALL PROCEEDING message. After initiating user alerting procedures at the called side, the network sends an ALERT message to the calling mobile and a ring tone is generated on the traffic channel. When the call is accepted at the terminating end, a CONNECT message is sent to the mobile station which responds with a CONNECT ACK message and then enters the active state. Transfer of information can begin at this time.

4. The procedure for disconnecting a call is much easier than setting up a call. It starts when a DISCONNECT message is sent by one party's terminal to the network. The network returns a RELEASE message to the terminal that initiates the disconnect and initiates a procedure to have DISCONNECT message sent to the other terminal. The initiating mobile completes the exchange of messages with a RELEASE COMPLETE message. The traffic channel is released followed by procedures to release connections in the MM and RR sublayer.

The signalling procedures within the network is shown in Figure 8. The SS7 signalling protocol is used. Upon receipt of a connection establishment request, the MSC sends an ACCESS REQUEST message to the VLR for the location area occupied by the mobile that initiated the call. The VLR determines the additional procedures that need to be invoked before the call can be completed. Initiation of authentication and ciphering is performed by the VLR. It sends an authentication message to the mobile via the originating MSC. Upon successful authentication, it initiates the ciphering procedure and assigns a new TMSI. Next, the MSC requests the user's personal service profile from the VLR. The user profile contains information such as special service or features which the mobile user has subscribed. This information is needed to complete the call. If the called party is mobile, the routing information is requested from the HLR. The call is then setup using standard ISDN-UP messages: IAM, ACM, and ANS. When the call is cleared, the SS7 network connection is released using the ISDN-UP messages REL and RLC.

Figure 8 Mobile origination (Network to Network Signalling)

Mobile Termination:

This procedures allow calls to be setup to mobiles from either the fixed network or other mobiles. Figure 9 and 10 shows the sequence of successful incoming call establishment. The process begins at the originating switch. The originating MSC requests routing information from the HLR. After receiving this information, a connection is setup to the destination MSC using IAM message. The destination MSC requests the user's profile from the VLR. The MSC then sends a paging request to the mobile station. If a mobile station responds to the paging request, then the procedures for MM and RR connection establishment, authentication and ciphering are initiated. The rest of the procedures are the same as for mobile originated calls. If the call had originated from a PSTN then specific bearer capabilities need be negotiated for the mobile station to be compatible with the services required.

Location Update

A key function of cellular networks is to route incoming calls automatically to a distinct mobile station by means of a UPTN which does not indicate the current location of the mobile station. Location update procedures are used to achieve this function. Location updates are used by the mobile station to inform the network of their location at any particular time and to indicate to the network their state whether attached or detached. The location information is stored in a home database (HLR) and the home database points to the appropriate visitor database (VLR) which pages called mobile stations in its location area. Location update procedures are necessary when: the mobile enters a new location area; a location update timer expires; or a mobile station is switched on or switch off.

The procedures for attach/detach is described in [35]. When the mobile station is activated, it selects the optimum BCCH according to defined quality selection criteria. It initiates a location update procedure if the broadcast location area identification is different from the one stored in the mobile before it was deactivated. Upon successful completion of the procedure, the mobile station enters the idle state and is ready for normal operation.

Figure 11 shows the location update procedures when a mobile moves to a new location area or when the location update timer expires. By constantly monitoring the

Figure 9 Mobile terminating (User to Network Signalling)

Figure 10 Mobile terminating (Network to Network Signalling)

broadcast channels, a mobile station can detect when it has moved into a different location area by checking the location area identification. When this happens it sends a location update message to the MSC which forwards it to the new VLR. The message contains the TMSI that is currently being used by the mobile. This identifies the VLR where the mobile came from. The new VLR communicates with the old VLR to request IMSI and authentication parameters. Authentication and ciphering procedures are then initiated by the new VLR. If successful, the HLR is informed of the new location of the mobile. The HLR then directs the old VLR to delete the entry of the mobile from its database. Depending on the implementation, location updates can be performed within the same VLR, when this happens the HLR need not be informed of the change in location.

Handover

Handover procedure enables continuity of service as mobile stations move through the wireless network. The quality of the radio channel is the main factor in determining the initiation of a handover. There are currently two criteria in the determination of handover [43]. The radio channel quality can be measured in terms of the relative distance of the mobile station from the current base station. This is determined by comparison of propagation delay times. If this becomes greater than a threshold which has been predefined then the mobile or the network might initiate handover procedures to select radio frequency channels for base stations closest to the mobile. The channel quality can also be determined by relative field strength measurements. This criterion selects the base station which receives the strongest signal from the mobile station. Handover procedures are not performed as long as the measured field strength of the current channel exceeds a prescribed threshold. There are several types of handovers:

- within the same base station: this usually occur when the noise level in the present radio channel becomes unacceptable in which case another radio channel is assigned by the same base station. The mobile remains in the cell even after the handover procedure.
- between different base station in the same switching area: this is the most common form of handover. It occurs when a mobile moves into another cell controlled by a different base station. The MSC coordinates the handover procedure in this case.

Figure 11 Location update

- between different switches: this occurs when the mobile moves from its current cell area to a cell area managed by a base station which is controlled by a different switch. Since this type of handover involves the exchange of control signals between switches, it bears some effect on the SS7 network connecting the various switches.

Figure 12 shows the handover procedures between different switches. When the current switch (MSC1) receives a hand over request, it sends a request to perform handover message to the receiving MSC (MSC2) with all the necessary parameters needed for the allocation of radio channel. MSC2 requests a handover number from its VLR and replies MSC1 with the identity of the new radio channel number and a roaming number.

A connection is then established between MSC1 and MSC2 using ISDN procedures. At the end of the handover procedure, a report is sent to the VLR to which the received MSC is associated.

Figure 12 Handover

As mentioned earlier, there are a number of benefits gained by introducing IN concepts in the PCS. In the next chapter we presents an overview of the IN concept and describes the set of SIBs necessary to provide the basic PCS applications.

Chapter 3

Intelligent Network Architecture for PCS

3.1 Introduction

In Chapter 2, we described a number of features available to PCS subscribers. An important requirement in PCS is that features available to PSTN subscribers should also be available to mobile subscribers. We also described the components and characteristics of the network that provides these features. The introduction of the Intelligent Network concept provides great advances in the provision of these service features to the user while allowing the service provider to do so with increased ease and flexibility. However, this comes with some penalties that must be identified and quantified so that their effect can be accounted for through proper planning strategies. Several authors have addressed the technical issues of integrating the IN concept into the existing network architectures [44-46].

The CCITT recommendation [25] gives an introduction to IN by providing the objectives and overall description of IN and by describing its main characteristics and overall capabilities. It defines the service aspects, network aspects and the functional relationships that forms the basis of the CS-1 capabilities. CS-1 is the first standardized stage of IN as an architectural concept for the creation and provision of telecommunication services. In this chapter, we describe an architecture for PCS network based on the IN concept presented in Appendix B along with some implementation aspects of the architecture. We examine those aspects of PCS from which IN functionality can be derived and specifically describe subscriber mobility which requires identification, location registration, and call management.

3.2 Intelligent PCS Network Configuration

The components and possible architecture of PCS network was presented in Section 2.5. The architecture was based on the view that PCS will evolve from presently existing technology with the DCS1800 (a modified version of GSM for PCS application) signalling protocol. However, as we stated earlier, the IN architecture concept will

facilitate the introduction and modification of advanced services and allow enhanced call control feature to PCS subscribers. Figure 13 shows a possible implementation of an Intelligent PCS architecture. Key elements in the PCS network corresponds to FEs of the IN conceptual model. These elements (Figure 4) are mapped onto the physical architecture of the IN.

The Service Control Point (SCP) provides the functionality of the Home Location Register (HLR), the Authentication Centre (AC), and the Equipment Identity Register (EIR). The AC is used to verify the authentication of a requested service and the EIR is used to maintain records associated with the identity of a mobile terminal. This is used to determine whether a terminal is an authorized equipment for service. The SCP also provides the functionality of the Network Management Centre (NMC). This helps to coordinate and monitor activities within the network, both fixed and mobile part. There are a number of different ways in which the Service Switching Points (SSP) can be implemented. In this thesis, all the functionality that constitute an exchange in a network is implemented as SSP. and so in modelling the mobile network, we made the SSP represent all the functions performed by the BSS, MSC and VLR. For the fixed network, the SSP represents the PBX, LS, TS and the fixed network database. The gateway switch (GW) is also considered an SSP linking the mobile network to the fixed network. The exchange of information between the SCP and SSP takes place via SS7.

3.2.1 IN Requirements for PCS

The fundamental concept of PCS is the identification of a user in the network via a personal telecommunication number (UPTN). This together with terminal mobility is required to deliver true personal mobility. The UPTN separates the user's logical address from the physical address of terminals used to access the network. The network must therefore be able to associate the user's UPTN with his current physical address so that the user can have the ability to receive calls addressed to his UPTN. The set of IN requirements for PCS has also been addressed in [31]. They were classified according to calling situations.

Using IN concept to provide PCS means that PCS subscribers will have access to IN services. IN services need to be defined, therefore, for PCS subscriber's call origination access, call termination access, call transfer access, subscriber service call

Figure 13 PCS Intelligent Network Architecture

access and personal mobility. Not all PCS services may require IN access. Subscribers having access to IN network will need to have their subscription, along with the relevant data, defined in the SCP. It is practical therefore to provide access classes to subscribers, allowing access to the relevant IN services. Certain services may also be made generally available to all subscribers even though the services may be provided by intelligent network. The following list describes some of the PCS features that can be provided in the DFP. We show the information flows that are needed for each feature.

1. **Access, Identification and Authorization:** This feature describes how a PCS user gets access to the PCS service by dialing a special service code and identifying himself in a prompt and collect sequence by his UPTN and authorization code. The IFs needed to provide this feature are:
Initial DP; Request Status Report; Connect to Resource; Status Report; Request Report BCSM; Prompt and Collect User Info; Specialized Resource Report; Query; Query Results.
2. **Procedure Identification:** This feature shows how a PCS user can identify the required procedure for successful provision of PCS service, e.g. procedure for location registration, call setup, etc. This involves a query intended to check whether the PCS user is allowed to use the requested procedure. The IFs are:
Prompt and Collect User Info; Request Status Report; Connect to Resource; Specialized Resource Report; Query; Query Results.
3. **Network Access Registration:** This feature registers the location of a PCS user, against a specific terminal whose physical address is known to the network. The required IFs are:
Analyze Info; Query; Prompt and Collect User Info; Update Data.
4. **PCS Call Setup:** This feature manages the call setup procedure for the PCS network. The required IFs are:
Prompt and Collect User Info (called number); Query (the SCP's database is accessed for user authorization and for the location of the called PCS user's home database, a second query to the home database returns the current location of the called PCS user); Furnish Charging Info (instructions to prepare PCS charging record); Connect (to destination number); Release Call (if something goes wrong); Disconnect Forward (at the end of call setup); Update Data (charging info stored in user's home PCS

database).

5. **Call Direction:** This feature will route calls according to parameters such as calling number and time of day previously set by the PCS user. The required IFs are:
Analyze Info; Query; Query Result; Route Call.
6. **Profile Status Interrogation:** This feature will provide the ability of the PCS user to query the settings and status of his PCS user profiles. For example, the number of times a feature is used. The required IFs are:
. Analyze Info; Query; Query Result; Prompt and Collect User Info.
7. **Profile Modification:** This feature will provide the ability of the PCS user to modify the settings of his PCS profiles. For example, change the data associated with the call direction procedures for certain UPTNs. The required IFs are:
Analyze Info; Query; Query Result; Prompt and Collect User Info.

In the discussion that follows, we show how these IFs interact in the IN network to support some PCS services. Figures 14 shows the information flow diagrams for the Access, Identification and Authentication feature.

3.2.2 PCS Call Processing

As depicted in Figures 69 and 70 in appendix B, the basic call processing for an IN shows the call states and interaction points where the SSP and the SCP can exchange information and service control commands to provide customized call treatment. Likewise, PCS call processing can also interact with the SCP in the same way. A possible call model for PCS is illustrated in Figures 15 and 16. Figure 15 describes the main phases of call origination by the PCS user, and Figure 16 describes a call termination on the PCS user. The call processing sequence do not differ from the IN call processing model with the exception of authentication, channel assignment, paging of mobile and handoff.

Figures 17 and 18 show the Information Flow diagram for PCS call origination and PCS call termination. The Access, Identification and Authentication feature is used to provide authentication and the PCS Call Setup feature is used to manage the call setup procedure.

**Figure 14 Access, Identification and Authentication
Procedure Information Flow Diagram**

Figure 15 Originating PCS Call Model

Figure 16 Terminating PCS Call Model

Figure 18 PCS Call Termination Information Flow Diagram

3.2.3 Location Registration

Personal mobility refers to the fact that the subscriber is reachable using a location independent number as opposed to terminal equipment. This is achieved through location registration and updating. Location registration in PCS can be provided using the Access, Identification and Authentication feature, the Procedure Identification feature and the Network Access Registration feature. The procedure can be performed manually or automatically. In the manual case, it is the responsibility of the user to inform the network of his location. This can be performed by dialling an identification code at a terminal. The network databases are updated by associating the present location of the user with the location of the terminal. All incoming calls will be directed to this terminal until the network is updated again. This type of registration can be quite tedious and so the use of smart cards has been suggested [6]. Smart cards can be designed so that by inserting them into terminals they can perform the registration automatically. But a more flexible way will be to have a personal radio communicator that periodically initiates the registration procedure to the network and identifies the nearest terminal location to update the network database without the user's interaction. This is the automatic location registration. Needless to say that this may increase the traffic in the network and as such need careful study.

Figure 19 shows the information flow diagram in relation to basic call processing model for location registration. In the manual PCS registration, a caller requests connection to a mobility management function residing in the SCP by dialling a code. The SSP receives the request and triggers the Analyze_Info TCP. This initiates procedures for access identification and authentication at the SCP. After the usual authentication procedures have been checked, the Procedure Identification stage is initiated. This procedure is used to determine if the outgoing call is a registration request, in which case a query to the SCP for further instruction is initiated. The SCP uses the Analyze Called Number FEA to invoke the mobility management program and responds with instructions for the IP to prompt the user for a UPTN and a password (PIN) using the Present Information FEA followed by Collect Information FEA. The IP passes the collected information to the SCP. The SCP again uses the Analyze Information FEAs to verify the UPTN and PIN format and uses the Data Access FEA to query the database to verify their validity. Upon validation, the SCP then prompts for new location. An

area code and a subscriber number (terminal number) are used to provide the current location using the Analyze Calling Number FEA. The SCP updates the user's location in a mobility database and returns instruction to the SSP to terminate the call.

3.2.4 Introducing Advanced Telephone Services

Some advanced telephone services are already being provided in the fixed network. These services such as Freephone Service can be extended to PCS subscribers via the Intelligent network. The process of introducing new services, such as 1-800 service in North America, into the Intelligent Network begins with the specification of the service features of the new service and the associated data. This fall into the Service Plane. A global service logic is then designed for each service using available SIBs. If necessary a new SIBs is constructed to meet specific needs not supported in the present set of SIBs. A method is formulated to map the functions of the GSL to functions performed at various network elements. These are recorded in the Distributed Service logics. The DSLs are combined with service specific data and tested by the SMF before being downloaded at the participating network nodes. The trigger tables at the SSPs need to be modified to accommodate this new service. For example, in Freephone service a subscriber wants all calls to his company to be free of charge to the callers and that the same telephone number should be used in the whole country. Assuming the company has only one UPTN to each office for fixed number of lines and during office hours all calls go to the nearest office. After office hours calls are to be directed to the central office up to nine o'clock in the evening. At night and over week ends an announcement shall be given. There must be a queue to the central office. The service subscriber can be provided with statistics about incoming calls to the different location offices so he can determine the right amount of agents in the different answering places. The service logic for this service can be constructed by using the Call Direction service feature with pre-specified parameters and the Billing Negotiation and Charging service features. Figure 20 shows the global service logic that can be constructed for this service. The application program is stored in a database located in the SCP.

The following list identifies the SIBs used to develop this logic. Some modifications to the CS-1 have been made and additional ones are shown in *italics*.

Distribution: This SIB distributes calls to logical ends of the SIB based on user

Figure 19 PCS Location Registration Information Flow Diagram

Figure 20 Freephone Service Logic

specified parameters. The Day In Week, Time of Day Origin of Call are examples of this SIB but with different parameters.

Queue: This SIB provides all the processing needed to provide queuing for a call. It will pass the call if resources are available, queue the call if resources are not available, dequeue the call when resources becomes available or direct the call elsewhere if queue is full.

Log Call Information: This SIB logs detailed information for each call into a file. However a slight modification is needed here. In CS-1, the collected information may be used only by management services, this should also be available to service subscribers by keeping a copy in their service profile so they can have access to it. The Count is an example of this SIB.

Call Processing & Billing: This SIB is added to provide the processing functions

needed to complete a PCS call. It basically is a combination of the following CS-1 SIBs: Algorithm, Translate, Verify, Charge.

3.2.5 Implementation

A trigger table in the SSP enables the detection of a call that needs special routing service. The Analyzed Information TCP recognizes such a detection. A query to the database will be triggered. The query, which is transmitted to the SCP via SS7 using TCAP Protocol, include the dialled subscriber service number. The SCP reads the Freephone application program from the database and begins execution with the translation of the dialled number to a real destination number according to previously established service parameters like time of day/week/month, geographical area where call originated etc. The results of this analysis in addition to service logic instructions are sent to the SSP via SS7. The SSP then completes the call setup, routing the call through the network to the appropriate destination. The Route Select Failure TCP is used to recover from failed PCS number translation. Figure 21 details the information flow between the participating network elements i.e. SCP, SSP and the IP.

Using the concept of IN to implement and deploy PCS services as has been described will provide benefits both to the user as well as the network operator. Creation of new services will no longer take a long time to be deployed. This, to the service provider, will mean the ability to capture a large market share for the new services resulting in an increase in revenue. To the service subscriber, it will mean the freedom to customize their services to suit their private life styles. Despite the anticipated advantages, the provision of PCS using intelligent network concept may generate new problems and it is imperative to develop ways and means to quantify to what extent these problems might have on the performance of the network and to address ways to mitigate these effect. The use of generic functional units to create new services results in a large number of information flow between network elements. Processing of information elements at the various nodes also increases. This might result in an overall degradation of the system performance in terms of longer service response and some services being rejected. In the next two chapter, a model is built using the concept presented so far to investigate this problem. The performance of the system is measured in terms the number of processes performed at the various network elements as well as the load on the network due to the

Figure 21 Freephone Information Flow

information flow. We also look at the delay in the network when providing advanced call features.

Chapter 4

Performance Analysis

The performance analysis of the proposed system is incomplete without considering the effect of traffic rates on the protocol processing characteristics. It is therefore necessary, for the purpose of analysis, to identify and assess limitations in the resources at the disposal of the users. The traffic rates are directly related to the number of user triggered activities occurring in the PCS network. The busy-hour-call-attempt (BHCA) is a measure used to indicate traffic in a telecommunication network. In the absence of any available data describing PCS network traffic, it is necessary to develop a method of forming these estimates. In this chapter, we present a method to estimate the BHCA rates for a personal communication system providing services for users in both cellular and microcellular environment. We also provide a method to quantify signalling load when SS7 and Intelligent network concepts are used in a PCS environment.

We use these methods to investigate the performance of frequency channel assignments between micro cell and macro cell and the overall throughput in our proposed network. The performance is measured in terms the probability of blocking. We investigate suitable channel allocation algorithms with the view of minimizing the overall probability of blocking by determining the optimum values of various parameters and variables used by computer simulation. The service area is made up of a grid of macrocells with microcells deployed where high traffic density is expected. The macrocell provides a universal coverage throughout the switching area and may overlap areas covered by the microcell. Two classes of user terminals are identified. The personal station (PS) or handheld portable terminals and the mobile station (MS) or vehicle mounted terminals. The PS operates in the microcell and the MS operates in the macrocell. These two classes are characterized by different mobility statistics as shown in Table 3

4.1 Traffic Consideration

PCS in large cities where different categories of traffic may be in high demand requires both MS and PS system integration. Traffic volume estimates may be obtained

$\text{Orig}_{\text{cell}}/\text{Term}_{\text{cell}}$	The Number of call origination/termination generated per hour in a cellular area
$\text{Orig}_{\mu\text{cell}}/\text{Term}_{\mu\text{cell}}$	The Number of call origination/termination generated per hour in a microcellular area
P_{term}	Probability that a person is equipped with terminal
P_{on}	Probability that a terminal is powered on
P_{cell}	Probability that a call originates from a cellular area
$P_{\mu\text{cell}}$	Probability that a call originates from a microcellular area
H	Call hold time (in hours)
ρ	User density: The number of potential users per km^2 whether they are equipped with terminals or not.
A	The area covered by the mobile switching centre (in km^2)
erl	The number of erlangs generated per terminal
n_{cell}	The total number of cells in an area
n_{crossing}	The number of people, with or without terminals and with or without call in progress that leave the cell per hour
A_{cell}	Area covered by a cell

Table 2 Definition of Traffic Parameters

according to the nature of the cities. Paging and Land mobile satellite have not yet been estimated but are considered to be proportionally insignificant [33].

4.1.1 Rate of Transactions

To estimate the SS7 signalling related traffic overhead generated by PCS traffic, we determine the load associated with the main transactions within the network:

call origination and termination, location updates and inter-MSC handovers. There are other transactions such as those used to provide network administration and management service i.e. billing, network statistics gathering for system configuration and performance. However, we concentrate on the basic voice services although in the future the use of data is expected to grow. Detailed message flow for each transaction has been given in chapter 3

In order to determine the total traffic offered to the network, we need to know how often each transactions occur per hour. We determine the rate of origination, termination, handover and location update in terms of the location area served by an MSC.

The number of calls originations/terminations generated per hour in a cellular environment covering an area A can be expressed as [47]:

$$Orig_{cell}(A)/Term_{cell}(A) = A \times \rho \times P_{term} \times P_{on} \times erl \times \frac{1}{H} \quad (4.1)$$

Where H is expressed in hours. All the parameters are defined in Table 2

Assuming that an intra-MSC handover is made every time a terminal with a call in progress crosses a cell boundary internal to the switch and an inter-MSC handover is made every time a call in progress leaves the MSC area, then, the total number of handover requests generated in an area A (both intra and inter) is

$$Handovers(A) = n_{cell(A)} \times n_{crossing(A)} \times P_{term} \times P_{on} \times erl \quad (4.2)$$

The inter MSC handover is given by:

$$InterHandovers(A) = n_{crossing(A_{cell})} \times P_{term} \times P_{on} \times erl \quad (4.3)$$

This results in the intra-MSC handover being:

$$IntraHandovers(A) = Handover(A) - InterHandover(A) \quad (4.4)$$

The number of location updates is computed by noting the number of location update areas in the switching area and the fact that only terminals that are powered on and do not have a call in progress generate location update requests. For terminals with calls in progress, location updates are assumed to be performed at the end of the call. The

total number of location updates in a switching area generated per hour is given by equation (4.5)

$$LocUpd(A) = n_{loc}(A) \times n_{crossing}(A_{loc}) \times P_{term} \times (P_{on} - erl) \quad (4.5)$$

where n_{loc} is the number of location areas in the switch coverage area, and is given by

$$n_{loc}(A) = \frac{n_{cell}(A)}{n_{cell}(A_{loc})} \quad (4.6)$$

In order to calculate equations (4.2), (4.5) and (4.6), a mobility model is needed to model the rate of flow of traffic across cell boundaries. This can be used to compute the number of cells, location area and switching area crossing. We use a model given in [47]. The model assumes that people are uniformly distributed in an area A and that the direction of travel of each user relative to the border is uniformly distributed in the interval $[0, 2\pi]$. We defined v to be the average speed of a person in km/hr, L the length of the perimeter of the area A , then by conservation of flow, F , the traffic flow in and out of area A , which is also the average number of people entering and leaving the area A per hour is given by

$$F = \frac{\rho v L}{\pi} \quad (4.7)$$

An expression for L as a function of A can be derived depending on the geometry of the cell sites

4.1.2 Traffic Model

As noted earlier, two types of traffic exists in the network, traffic arising from the micro cell and traffic arising from the macro cell. We refer to calls arriving from a macro cell environment as Type A and calls arriving from a microcell environment as Type B. In order to estimate the total traffic offered to the network, we need to know the frequency of origination, termination, handover and location updates per hour. To do this we use the parameters shown in Table 3 to model the PCS traffic in a highly populated urban area. These traffic parameters were obtained from [33]. The macro cell traffic parameters were chosen to reflect larger cell sizes and smaller probability

Parameters	Macro cell	Micro cell
Radio Coverage (%)	90	90
Cell area (km ²)	1.0	0.016
P_b (%)	2	1
ρ (people/km ²)	100,000	100,000
P_{term}	0.1	0.8
P_{on}	0.5	1.0
H (sec)	120	120
v (km/hr)	25	5
erl (erlang/terminal)	0.1	0.04
VLR area	MSC area	MSC area

Table 3 Cellular and microcellular traffic parameters

that the terminal is powered on. Whereas the micro cell parameters represents smaller cell sizes and a high terminal usage probability. We assume square shaped cell sites. Equations (4.1 — 4.5) can now be combined with the traffic parameters given in Table 3, to compute the rate of originations, terminations, handovers, and location updates per hour. Table 4 displays the number of attempts generated for the two traffic types in a cell area during a busy hour period. We observe that the traffic generated per cell

	Macro cell	Micro cell
Call Origination/Termination	15000	1536
Handover(intra-switch)	1590	320
Location Updating	6400	82

Table 4 Signalling traffic generated per busy-hour per cell area

area for type B is low as compared to that generated for type A. This is because the comparison is made on a per cell basis with the microcell area being much smaller than the macrocell area. With the chosen cell layout and parameters, a macrocell area is comparable to 62.5 microcell area. Table 5 shows the traffic rate when compared on a per switch basis. Each switch serves an area comprising 4 base station sites with each base station serving an area comparable to a macrocell area. It can be seen that traffic arriving from the micro cell environment forms a greater proportion of the total traffic at the switch. This information is to generate PCS traffic in our simulation.

4.1.3 Signalling Network load

The transport of signalling messages in a PCS network is performed on the SS7 network, there is therefore, the need to determine the signalling network load. GSM specifications [35, 48], defines the SS7 MAP protocol and CCITT Q.1218 recommendations [25] defines the SS7 INAP protocol. Using these specifications, estimates of message lengths for the MAP and INAP messages described in Section 3.2 can be calculated. A description of the SS7 network is provided in Appendix A. Tables 6—9 gives the estimates of the message lengths. Each table shows the message name, the Transaction Sub-layer (TSL) part, and Component Sub-layer (CSL) primitives, and the Service Information Field (SIF) length. Description of the parameters can be

	Macro cell	Micro cell
Call Origination/Termination	60000	384000
Handover	6360	80000
Location Update	25600	20480

Table 5 Signalling traffic generated per hour per switch area.

found in [35]. The number of bytes of SS7 messages generated per transaction in a PCS network that uses the Pan-European standard GSM is given in [47]. Though SS7 message lengths and parameters are variable, for this study only mandatory parameters having maximum lengths are assumed. The parameters in each message can be found in [25]. Field sizes were estimated as follows:

- Transaction Sub-layer (TSL):
 - BEGIN = 10 bytes
 - CONTINUE = 16 bytes
 - END = 10 bytes
- Component Sub-layer (CSL):
 - INVOKE = 10 bytes
 - RETURN RESULTS = 12 bytes
 - RETURN ERROR = 5 bytes
 - RETURN REJECT = 5 bytes
- Signalling Connection Control Part (SCCP):
 - = 10 bytes
- Message Transfer Part (MTP):
 - Layer 2 = 7 bytes
 - Layer 3 = 4 bytes

We assume that all invoke operations are successful and so only the RETURN RESULT primitive is returned. The ISDN messages are generated in the ISDN-UP and so do not use the TCAP protocol. The SCCP and the MTP field sizes were not shown in the table however they were included in the total message length in the last column.

Table 10 displays the number of SS7 bytes generated per transaction between network elements for our PCS network (refer to section 3.2). This is obtained from Tables 6 — 9 by adding the elements in the last columns. In column 1 of the table, all possible transactions between the relevant networks entities are listed. The rest of the columns show the number of bytes for each type of transaction: Call originations ; Call terminations on an MS or PS; inter-MSC handovers; and Location Updates. The first row shows the average number of bytes exchanged between two SSPs. The second row indicates the number of bytes exchanged between the SCP and the SSP. The SCP which houses the HLR contains information needed for call processing as well as user location. This is responsible for the large number of bytes in this row. The third row shows the number of bytes exchanged between the SSP and the SDP. Though we have assumed that the VLR is collocated with the SSP, and as a result, the exchange of messages is internal to the SSP, we have provided this information to indicate the signalling bytes exchange that would result if they were not collocated. The fourth row shows the number of bytes that is transferred between the SCP and the SDP which contains the VLR. The last row indicates the number of bytes transferred between the SCP and the IP. It can be seen from the table that most traffic occur between the SSP-SCP link. This could cause an overload in the network if not controlled. We will use these traffic flow estimates in the next chapter to show how congestion in the network can be controlled.

4.2 Performance Analysis

In the previous section we developed estimates for the PCS traffic generation rates and also estimates for SS7 message bytes for the key activities in the network. In this section, we develop the analysis used in our simulation. We first examine the spectrum requirements for services provided in both the micro cell and the macro cell. This will give us an idea of how much bandwidth is required. Having decided on the total bandwidth needed, we next analyze the system performance in terms of the probability of attempted calls being blocked. We also discuss upper and lower bound on the blocking

Message Name	TSL	CSL	SIF	Total
BEGIN[INV(Authorize Orig. Attempt)]	10	10	17	58
CONT[INV(Connect to Resource)]	16	10	8	55
CONT[INV(Collect Authen. Info)]	16	10	29	76
CONT[RES(Authen. Info Collected)]	16	12	23	72
CONT[INV(Activate Cipher Mode)]	16	10	10	57
CONT[INV(Collect MSI Info)]	16	10	15	62
CONT[RES(MSI Info Ack)]	16	12	12	61
END[RES(Orig Attempt Authorized)]	10	12	17	60
BEGIN[INV(Send Routing Info)]	10	10	16	57
END[RES(Routing Info Ack)]	10	12	32	75
IAM				54
ACM				12
ANS				10
REL				14
RLC				10

Table 6 Mobile Origination Message Length (bytes)

Message Name	TSL	CSL	SIF	Total
IAM				54
BEGIN[INV(Authorize. Term. Attempt)]	10	10	16	57
CONT[INV(Contact to Resource)]	16	10	8	55
CONT[INV(Page MS)]	16	10	21	68
CONT[RES(Paging Response)]	16	12	5	54
CONT[INV(Collect Authen. Info.)]	16	10	29	76
CONT[RES(Authen. Info Collected)]	16	12	23	72
CONT[INV(Activate Cipher Mode)]	16	10	10	57
END[RES(Term.Attempt Authorized.)]	16	12	10	59
ACM				12
ANS				10
REL				14
RLC				10

Table 7 Mobile Termination Message Length (bytes)

Message Name	TSL	CSL	SIF	Total
BEGIN[INV(Query Send IMSI)]	10	10	15	56
END[RES(Query Resp. IMSI)]	10	12	50	93
BEGIN[INV(Update Data Req.)]	10	10	26	67
END[RES(Update Data Resp.)]	10	12	24	65
BEGIN[INV(Loc. Cancel Req.)]	10	10	14	55
END[RES(Loc. Cancel Resp.)]	10	12	6	40
BEGIN[INV(Loc. Reg. Request)]	10	10	27	68
CONT[INV(Collect Authen. Info)]	16	10	29	76
CONT[RES(Authen. Info Collected)]	16	12	23	72
END[RES(Loc Reg Accepted)]	10	10	17	58

Table 8 Location Update Message Length (bytes)

Message Name	TSL	CSL	SIF	Total
BEGIN[INV(Authorize HO Attempt)]	10	10	57	88
END[RES(HO Attempt Authorized)]	10	12	31	74
BEGIN[INV(Req. HO Info.)]	10	10	6	47
END[RES(HO Info. Resp.)]	10	12	12	55
BEGIN[INV(HO Report Notification)]	10	10	8	49
BEGIN[INV(Query HO No. Info)]	10	10	12	53
BEGIN[INV(Resp HO No. Info)]	10	12	36	79
IAM				54
ACM				12
ANS				10
REL				14
RLC				10

Table 9 Handover Message Length (bytes)

	Origination	Termination	Handover	Loc. Update
SSP - SSP	100	100	100	–
SSP - SCP	485	376	246	275
SSP - SDP	–	389	151	397
SCP - SDP	–	–	280	361
SDP - SDP	–	–	–	149
SCP - IP	148	122	–	148

Table 10 SS7 Signalling network bytes per Transaction between network elements

probability and show how a channel allocation strategy can be used to optimize this probability.

4.2.1 Spectrum Estimation

The main goal in the design of PCS is to provide adequate capacity to serve intended users while using the resources efficiently. To that effect, the propagation characteristics, the traffic generated in the network for various services offered, the available techniques (channel coding such as block codes and convolutional codes; multiple access techniques such as TDMA, FDMA or CDMA) and the time scale for meeting the users' needs should be taken into account in the consideration on spectrum usage. Other consideration on cost, size, weight and power consumption of the MS and PS should also be made.

The service requirements for PCS was discussed in Chapter 2. The demand for these services is very high in large cities and include different categories of traffic, each with different spectrum requirements. No estimates exists for the traffic generated by the paging system but is considered to be insignificant [33]. The traffic for land mobile satellite has also not been considered. We therefore discuss spectrum requirements for

only voice traffic. Non-voice traffic, such as telefax and interactive data services will in the future constitute a significant portion of total traffic however, they can be handled using queuing procedures which improve spectrum utilization.

A general equation for obtaining the total bandwidth (MHz) required for voice services is given in [33], assuming a relatively uniform traffic load over a reuse cluster of cells.

$$\text{bandwidth} = b \times n \times B(a, E, P_b) \times 10^{-3} \quad (4.8)$$

where $B(\bullet)$ is the Erlang B formula, E is the traffic load in Erlang/km², a is the cell area in km², n is the number of cells per cluster, P_b is the blocking probability, and b is the equivalent bandwidth per duplex channel in kHz.

It has been estimated [33] that on a busy street, there will be 400 vehicles per km of street length. If 50% of the vehicles are equipped with a mobile station, with each generate 0.1 Erlang, the traffic density originating from vehicle mounted mobile stations will be 20E/km of street length or 250E/km² for a street block length of 200m. If we consider the same amount traffic originating from mobile stations carried by pedestrians, then the combined traffic for mobile stations would be around 500E/km² in the highly dense city areas. For personal stations, assuming 3000 pedestrians per km of street length, with 80% equipped with PS, and each PS generating 0.04E, the peak traffic is estimated at 1500E/km².

Estimation of b , bandwidth for duplex channel, will depend on the choice the system. For instance, a duplex bandwidth of 25kHz per voice channels can support a GSM system with 7kbit/s speech codec with high quality speech in a macro cellular environment with a cluster size of 3 cells with each cell having $3 \times 120^\circ$ sectors. Lower bandwidth may be employed in systems that have good quality low bit rate codecs. Linear predictive coding (LPC) techniques combined with adaptive quantization and adaptive prediction is an attractive possibility.

Due to the severe limitations imposed on cost and power consumption for personal stations, a higher duplex channel bandwidth is required in a micro cellular environment. It is assumed that a duplex channel bandwidth of 50kHz will be needed to provide high quality speech. Figures 22 and 23 are plots of the Erlang blocking probability for different number of channels assigned to a cell. In Figure 22, the traffic is set to

500 E/km² and a is set to 1.0 km² to represent a macro cell environment and in Figure 23 the traffic is set to 1500 E/km² and a is set to 0.016 km² to represent a micro cell environment. It can be seen that about 17 duplex channels, each of 50kHz bandwidth, need to be assigned to a micro cell to achieve a 1% blocking probability and about 230 duplex channels, each with 25kHz bandwidth, are required in a macro cell to achieve a 2% blocking probability. This information is useful in determining the total bandwidth required. In the next chapter we show the results obtained on the number of channels needed in a cell to meet the demand as the cell area is increased.

4.2.2 Blocking Probability Calculation

This section discusses the steps involved in the analysis of the blocking probability as the traffic in the network varies according to the environment (macro cell or micro cell). We have stated earlier that there are two classes of traffic generated in the network: by mobile station (MS) and by the personal station (PS). We have also estimated the rate of transactions for these classes. Due to the different characteristics of these traffic as seen in Table 3, the effect on the performance of the network due to the traffic mix will vary. We study this variation by looking at the blocking probability as the traffic mix changes. In the discussion that follows, we determine upper and lower bounds to the probability of blocking and in the next chapter, results of our simulation to determine these bounds are presented.

Assume that call requests are generated in the BSS area according to a Poisson distribution with rate λ calls/min, and that the arrival of a call of a certain type (PS or MS) is independent of all other events. Assume also that each call last a random amount of time which is exponentially distributed with mean $\frac{1}{\mu}$ minutes, and the occurrence of this is independent of the type of call, and assuming further that all blocked calls are cleared from the system, then the Erlang blocking formula $B(n,a)$ given below applies.

$$P_b = B(n, a) = \frac{\frac{a^n}{n!}}{\sum_{i=0}^n \frac{a^i}{i!}} \quad (4.9)$$

where

Figure 22 Blocking probability for different number of channels, traffic=500E/km², A=0.94 km², b = 25kHz

Figure 23 Blocking probability for different number of channels, traffic=1500E/km², A=0.016km², b= 50kHz

P_b = probability of a call being blocked
 a = the load measured in Erlang ($a = \frac{\lambda}{\mu}$)
 n = the number of available channel

We assume also that a fraction α , ($0 < \alpha < 1$) of these call attempts originates from the PS and the remaining $(1-\alpha)$ call attempts originates from the MS.

The rate of arrival of calls from a PS can be considered as a Poisson process with arrival rate :

$$\lambda_{PS} = \alpha \lambda \quad (4.10)$$

and similarly the arrival of calls from an MS form a Poisson process with arrival rate:

$$\lambda_{MS} = (1 - \alpha) \lambda \quad (4.11)$$

In order to be able to evaluate the performance we need to obtain the performance bounds and then compare the simulated system performance with this bounds. We obtain the upper bound by determining the conditional probabilities of a call arrival being PS or MS. If we let

$$P_b(\alpha) = \alpha P(b/PS) + (1 - \alpha) P(b/MS) \quad (4.12)$$

where $P(b/PS)$ = the probability that a call originates from a personal station and

$P(b/MS)$ = the probability that a call originates from a mobile station,

then we can obtain the upper bounds of $P_b(\alpha)$ by finding the upper bounds on $P(b/PS)$ and $P(b/MS)$. We obtain bounds for each of these probabilities on the assumption that one type of calls are given a higher priority and consume a lot more resources leaving less resources available for the other. In particular, for a PS, calls are assigned channels after all MS calls have been assigned. The availability of a lower number of channels for the PS calls will cause them to be blocked. The channels allocated to PS calls will therefore depend on the total number of MS calls in progress. Thus we get

$$P(b/PS) \leq B(n_{PS}(\alpha), \lambda_{PS}) \quad (4.13)$$

where $n_{PS}(\alpha)$ = the total channels available to PS calls. This is a function of the number of MS calls being serviced which in turn depends on α .
 λ_{PS} is the load normalized over the service time
 Similarly by giving higher priority to the PS calls we get

$$P(b/MS) \leq B(n_{MS}(\alpha), \lambda_{MS}) \quad (4.14)$$

Equations 4.12, 4.13, 4.14 implies that

$$P_b^u(\alpha) = \alpha B(n_{PS}(\alpha), \lambda_{PS}) + (1 - \alpha) B(n_{MS}(\alpha), \lambda_{MS}) \quad (4.15)$$

This is clearly an upper bound to the probability of blocking $P_b(\alpha)$ because in fact the calls from PS arrive to a larger and faster pool of servers than the assumption. Thus, the actual probability of blocking will always be lower than the upper bound i.e.

$$P_b(\alpha) \leq P_b^u(\alpha) \quad (4.16)$$

Similarly we obtain the lower bound by determining the conditional probabilities of a call arriving from PS or MS. (Equation 4.12). The lower bound of $P_b(\alpha)$ is obtained by finding the lower bound on $P(b/PS)$ and $P(b/MS)$. We obtain bounds for each of these probabilities assuming that the other type of calls do not consume any resources. In particular then for PS call arrival, we assume that the MS calls do not take any channels and the entire set of channels are available to the PS calls. Thus we get

$$P(b/PS) \geq B(n_T, \lambda_{PS}) \quad (4.17)$$

where n_T = the total channels available

Similarly by ignoring the resources used by the MS calls we get

$$P(b/MS) \geq B(n_T, \lambda_{MS}) \quad (4.18)$$

n_T depends on the environment as well as on the cell arrangement i.e. cluster size.

Equations 4.12, 4.17, 4.18 implies that

$$P_b^l(\alpha) = \alpha B(n_T, \lambda_{PS}) + (1 - \alpha) B(n_T, \lambda_{MS}) \quad (4.19)$$

In the next chapter we show the effect on the overall blocking probability as α increases from 0 to 1. We implement a control strategy which aims at optimizing the admission of a call of a particular type into the network in order to minimize the overall blocking probability and hence improve the performance.

Chapter 5

Simulations and results

In the previous chapter, we developed a method for the estimation of the traffic rate in a personal communication system. A method for determining the signalling load in bytes generated in the network was also discussed (Section 4.1.3). We use the results obtained in that analysis as input to our simulation. The system model presented in Chapter 3 is used.

5.1 BONEs

The simulation was performed using BONEs 2.0TM (Block Oriented Network Simulator version 2.0)TM (refer to Appendix C for a description of BONEs and its capabilities). BONEsTM is a software package that provides an interactive graphical environment for simulation based analysis and design of a broad range of communication networks and distributed processing systems at any level of detail. In this environment, our network model is specified graphically and hierarchically using building blocks from the BONEs library. This enabled us to construct modules representing the independent functional blocks (see Chapter 3) and simulate the IN concept of providing service using independent building blocks. Another advantage of BONEs is that it enables a model component written as a C program to be incorporated into the BONEs modelling environment. This allowed us generate our own traffic for the model. In working with bones, our major concern was the processing delay at each module. A detailed simulation of the network requires constructing modules to represent every level of detail. On the other hand having modules to represent every level of detail will make the simulation model too complicated to extract any useful information since most of the delay in the model will be due to processing at the modules rather than in the network. In order to prevent this from happening, not every activity that occur in providing PCS service was simulated in detail.

Some of the capabilities we did not consider were: procedures involved in authentication and ciphering, and transmission error handling. We modelled the processing of requests at the various network entities using multiple servers with

processor sharing scheduling discipline. By assigning more processing time to higher priority requests, we are able to measure the system performance for different call scenarios.

5.2 Simulation Model

5.2.1 Data Structures

Three BONEs data structures were created for the model (Refer to Appendix C.3). The Radio Link DS is used to model information flow across the radio link (Figure 24). The fields of this data structure contains Temporary Station Identifier (TSI) which is assigned to every terminal that initiates a call. The Channel Number field is designed to enable the BSS allocate channels to a call. The Calling and Called PTN provides the necessary information needed for PTN service provision. The Service Type field indicate the type of service being requested. The DTAP DS is shown in Figure 25. Its function is similar to the Direct Transfer Application Part in the GSM standard. The fields of this data structure contains information that allows the transport of signalling information between the BSS and the MSC. The third data structure shown in Figure 26 is used to model information flow between the network entities using the SS7 signalling protocol. The information carried in this data structure is similar to that specified in the INAP [25].

5.2.2 The System Module

The BONEs block diagram representing the top level PCS system consists of 6 block diagram as shown in Figure 27. The SSP module generate and process service requests at each switching area. The SCP module performs procedures and functions needed to complete a transaction such as providing the location information of a PCS subscriber to the SSP. It also maintains a database into which is fed information relating to the status of the system. We assume that 5% of total PCS traffic originates from the PSTN. The PSTN module generates this traffic load and also acts as a sink for outgoing signalling data structures to the PSTN. The Router module is a simple network which connects the SSPs and the SCP, it models a router with a bidirectional SS7 links to each service point.

Name	Type	Subrange	Default Value
Time Created	REAL	[0,+infinity)	
TMSI	INTEGER	[0,1000)	0
Ch_Num	INTEGER	[-1,+infinity)	0
Calling PTN	INTEGER	[0,+infinity)	0
Called PTN	INTEGER	[0,+infinity)	0
Service Type	Service Set		Basic Call Proc.
Status	Status Set		OK
Length	INTEGER	[0,+infinity)	0
Flag	INTEGER	[0,1]	0

Figure 24 Radio Link Data Structure

Name	Type	Subrange	Default Value
O_MSC	INTEGER	[0,2]	0
D_MSC	INTEGER	[0,2]	0
O_BSS	INTEGER	[0,10]	0
D_BSS	INTEGER	[0,10]	1
Msg Length	INTEGER	[0,+infinity)	0
Msg Type	MAP Msg		Setup Req
Status	Status Set		OK
Link Num	INTEGER	[0,+infinity)	0
Setup Flag	INTEGER	[0,1]	0

Figure 25 DTAP Data Structure

2 OF 2

PM-1 3 1/2 x 4 PHOTOGRAPHIC MICROCOPY TARGET
NBS 1010a ANSI/ISO #2 EQUIVALENT

Name	Type	Subrange	Default Value
OPC	INTEGER	[0,10]	
DPC	INTEGER	[0,10]	0
SLS	INTEGER	[0,+infinity)	0
SIF	INAP		Auth.Orig.Attempt
Status	Status Set		OK
Length	INTEGER	[0,+infinity)	0

Figure 26 SS7 Link Data Structure

5.2.3 The SSP Module

Each SSP consist of a Service Application module, four BSS modules numbered 1 to 4, an MSC module and a few other modules which model the link between these blocks as shown in Figure 28. The Service Application module creates service requests for both Mobile Station (MS) and Personal Station (PS). It also generates the arrival of both the Handover and the Location Update requests. The rate of arrival for each type is determined from Table 5. The length in bytes for each service request is chosen from Tables 6—9. The Service Application module also collects data for all transactions in the BSS switching area through probes inserted at various points. It collects information on successful calls as well as blocked calls. Figure 29 displays the expanded module. The Wireless Link module, between the Service Application module and the BSS modules, models a full duplex radio channel links. The GSM's TDMA protocol with 8 slots per frame is specified as the protocol for this link. In this module the link is modelled as a multiple server (8 servers) with FIFO queueing discipline. This choice provides a reasonable trade-off between accuracy and simulation performance. The module between the BSSs and the MSC, the Wired Network, also models a full duplex link. The GSM's "A" interface protocol is specified for this link. The "A" interface is based on the use of 1 or more 2Mbit/s (31*64kbits/s) digital transmission system. In this module we model the links as FIFO servers with server rate of 64kbits/s. The module also contains

Figure 27 PCS System Module

a router module with bidirectional links.

5.2.3.1 BSS Module

The BSS module is expanded in Figure 30. It performs operations similar to an actual BSS. Most of the radio channel management functions are performed at the BSS. The radio channel management messages are interpreted and processed at the BSS. The BSS also terminates the radio links to the mobile station and personal stations. In Figure 31 the Radio Resource Manager module performs the procedures for the allocation and release of radio channels as shown in the expanded form in Figures 32 and 33. These modules contains a number of vector memories that are used to monitor channel usage. When a service request is received, these memories are accessed and if a channel is

Figure 28 SSP Module

Service Application [30-Sep-1992 16:12:09]

Figure 29 Service Application Module

available, then it is assigned and the state of that particular channel is changed to busy.

A channel may be released if :

- the call successfully completes or
- a Handover takes place or
- if for some reason the call is dropped

When a channel is released, the state of the channel is changed to idle and it becomes available again for other users.

Messages that are not interpreted at the BSS are forwarded to the MSC by the DTAP. The Build & Store DTAP-DS module creates the DTAP data structure for requests originating at the base station area and the Set & Store DTAP-DS module forwards response and acknowledgment information regarding requests from other stations. Part of the function of the mobile stations are implemented in the BSS module. This is to reduce the complexity of the model and yet model to some level of detail. The Call Proc. Manager module and Handover & Loc. Upd Manager module are used to activate certain TCPs which in turn causes the processing of events at other areas. Activation of TCPs are modelled using event timers in the BONEs library. Thus, a Handover failure at another module due to a lack of available channels at that module can be detected and the appropriate resources released.

5.2.3.2 MSC Module

The MSC provides the interface functions between the radio system and the fixed networks. It includes all functions relating to IN call and connection processing in the SSP. It also coordinates activities within a cell. Figure 34 shows the expansion of the MSC module. The MSC Service Processor module schedules the activities occurring within the MSC. It is implemented as a processor sharing server with parameter that must be set to indicate the priorities given to each request and responses associated to it. The Basic Call Manager module provides basic call and connection control to establish communication paths for users. It detects basic call and connection control events that requires the invocation of IN service logic execution and finally, it manages resources required to support the basic call and connection control. Figure 35 is an expansion of the Basic Call Processing module. The IN Switching module, Figure 36, interacts with the SCF in the course of providing IN service features to the user. It detects triggers or IN call processing events in the BCM and initiates a query to the SCF for

Figure 30 BSS Module

Figure 31 Radio Resource Manager Module

further processing. When a response is received from the SCF, it notifies the Basic Call Processing module to continue with call processing.

5.2.4 The SCP Module

The SCP module provides the service logic execution environment on which a service logic program runs to provide IN service. It also performs the functions of the HLR. It contains memory vectors acting as databases which holds the subscription records of every mobile station. It also contains vector memories that acts as temporary storage of the actual location of the mobile station. To simplify the operation, most activities that occur in this module are restricted to either retrieving information or updating the records as shown in Figure 37. The SDP which performs the functions of the VLR is incorporated in each MSC and contains data necessary for call handling. It is

Figure 32 Radio Channel Allocation Module

Figure 33 Radio Channel Release Module

modelled as a module containing just a vector memory to store and retrieve information relating to the service profile of a user (typically the PTN number).

5.3 System Performance

In this section we present the results of several simulation runs performed to check the validity of our proposed network model. The necessary parameters and equations have already been discussed in the previous chapter. All the simulations performed on the network is focused towards assessing the IN capabilities of the network and as

Figure 34 MSC Module

such in some cases the results may not match those obtained for existing networks. However, our main objective is to introduce intelligent network capabilities into the personal communication network and to investigate whether with these capabilities, new and advanced telecommunication services can be offered with much flexibility.

The first simulation performed on our network measures the performance in terms of the blocking probability within a base station area (Figures 40—44). Two types of base station areas were compared, the macrocell to support vehicle-mounted and hand held mobile terminals (MS) and the microcell to support portable personal stations (PS). Using the capabilities offered by the intelligent network, a control strategy is adopted with the view to minimize the blocking probability within the base station area and to

Figure 35 Basic Call Processing Module

improving the performance and resource utilization. The second simulation measures the flexibility that the proposed network can offer to a user requesting service of a particular type (Figures 45—47). First, the utilization of the channels available to a base station area are determined under all kinds of service, then again by implementing a strategy made possible by the IN capabilities, the performance of the network in its ability to deliver the right services to its users is determined. The third simulation measures the delay performance of the network in terms of the average service delay

Figure 36 IN Switching Module

(Figures 48—49). The services are classified into basic and advanced depending on the amount of processing required at the SCP, which depends upon the number of service independent building blocks required. An approach is then investigated that seeks to reduce delay in offering these services while maintaining the highest level of service capability. It is inevitable that the introduction of personal communication services with IN capabilities will cause additional signalling load in the network over conventional PCS systems based on GSM. The fourth set of simulation is performed to assess the impact of this additional signalling load on the overall system performance (Tables 11—13 and Figures 50—55). A complete distribution of SS7 message bytes between the various network entities are presented. We also show the signalling link dimensioning required to accommodate such a load.

Figure 37 SCP Module

5.3.1 Blocking Probability

In Section 4.2.1 we discussed a method for estimating the spectrum of the PCS network. Using the parameters shown in Table 14, we determine the number of channels per cell required in order to maintain a certain level of performance. Figures 38 and 39 shows a plot of the required number of duplex channels as a function of the cell area for both microcell and macrocell calls. In general, the number of channels in the cell increases with cell size. Also, as the number of cells per cluster increases, the number of channels also increases. The probability of blocking in the microcell was maintained at 1% and that in the macrocell at 2%. The number of channels based on the Erlang B formula is also plotted. It can be seen from the result that for the microcell, a cell area

Figure 38 Number of duplex channels required per cell as a function of the cell area for microcell environment ($P_b \approx 1\%$)

Figure 39 Number of duplex channels required per cell as a function of the cell area for macrocell environment ($P_b \approx 2\%$)

of about 0.016km^2 requires about 30 equivalent 25kHz duplex channels ($15 * 50\text{kHz}$ duplex channels) in order support voice traffic. For the macrocell, an area of 1km^2 the channels requirement is are about 230 equivalent 25kHz duplex channels. This means a total of 0.75MHz bandwidth is required per cell for microcell and 5.75MHz bandwidth is required for the macrocell. Using a cluster size of 4 cells with four sectors per cell (90° sectorized antenna) for the micro cell and a cluster size of 4 cells (omnidirectional antenna) for the macro cell, the total bandwidth needed in a micro cell is 12MHz and for the macro cell it is 23MHz giving a total of 35MHz for the PCS environment. Based on the above results we have adopted a total bandwidth of 35MHz for the rest of the simulation.

In our simulation, by varying the microcell call arrival rate α , we determine the effect on the overall blocking performance. In Figure 40 the upper and lower bound probability of blocking ($P^u(\alpha)$ and $P^l(\alpha)$) as a function of the fraction of micro cell requests α (refer to Section 4.2.2 for a detail discussion) are plotted. The blocking probability increases from 0.06 to 0.8 in the extreme case when all calls are originating from the micro cell. As seen from the result, there is a large gap in the intermediate region between the upper and lower bounds. In this simulation, resources are assigned randomly and no control strategy was employed. Five simulations were performed using different seeds and the results are based on the average of these. Based on the results in Figure 40, it can be seen that the simulated blocking probability has about the same margin from both the lower and the upper bounds. For values of α approaching 1.0 the performance is unacceptable. Even for a medial fraction ($\alpha = 0.5$) the blocking probability is quite high ($P_B = 0.5$). The simulation was performed again this time adopting a control strategy that improves the overall blocking probability. The control strategy is implemented as follows:

For a given value of the fraction of microcell calls α , a fraction β of the total network resources available within the cell are reserved for microcell calls. The resources here imply all entities that need to be assigned in order for the call to be successful and this include traffic channels, base station processors MSC processors rates etc. Thus if a microcellular call request arrives, the number of available resources in use is determined. If the number is less than a threshold whose value depends on β then the call is processed. If on the other hand the number of resources already in use exceeds this threshold, the call

is blocked. The reason behind this strategy is simple. Rather than randomly assigning resources to the two types of calls as observed in the previous simulation, this strategy allows a minimum number of resources to be available for each type of call while at the same time allowing more resources to one type of service with higher traffic load provided the other type has a lower traffic load. The threshold level varied depending on the value of β . Figure 41 shows the effect on the blocking probability when the threshold level varies directly as β for different values of α , ($\alpha=0.3$ and $\alpha=0.5$). It can be seen that reserving too much resources for microcell calls ($\beta=1$) results in the blocking of macrocell calls that could be served without any significant adverse effect and reserving too few resources ($\beta=0$) would prevent substantial number of microcell calls from being processed. This is reflected in the high blocking probabilities recorded at both extremes of the curve. From the figure the value of β that minimizes the blocking probability can be obtained. Observe that P_{Bmin} , i.e. the minimum point on the curves, increases with α , this is an indication that services provided to microcell users need more resources than that provided to macrocell users as shown in Figure 42

Figure 43 compares the results from the random assignment (no control) strategy with the control strategy described above. The lower bound is added for comparison. We observe that the control strategy has an advantage in considerably lowering the blocking probability. This improvement is significant in the region of low to moderate α but becomes insignificant at high α because of much higher traffic load from microcell.

5.3.2 Handover Blocking Probability

The next simulation investigates how the increase of the fraction of microcell calls, α , affects the rejection of inter MSC Handover, i.e. handover between switches. As described earlier, several kinds of handovers can occur in a PCS. We have chosen this type of handover because it is performed over the SS7 network and as a result increases the load on the network. A handover request could be rejected due to a lack of available channel resources in the incoming cell area. The request rate for handover depends on the user mobility, more specifically the average speed of users. Table 5 (in chapter 4) displays the handover requests generated per busy hour period in a switch area. By varying the fraction of microcell traffic allowed in the network, we can study the effect that this has on the rate of rejection of handover requests. We expect a general increase

Figure 40 Effect of α on the Blocking Probability for the proposed network

Figure 41 The influence of β on the Blocking probability (for $\alpha = 0.3$ and $\alpha = 0.5$)

Figure 42 Optimum β for various α

Figure 43 Effect of α on the Blocking Probability ($\beta = \text{optimum}$)

since microcell traffic are characterized by high user densities. The handover rate is provided as a percentage of the network traffic. In Figure 44 the rejection of handover is shown as a function of α . It can be seen that there is a significant reduction in handover rejection when assignment strategies are based on user speed (i.e., slow moving users are assigned to a microcell and fast moving users are assigned to a macrocell) are implemented. This is due to the fact that with the use of the control strategy, network resources are allocated in a way that more channels are made available which can be used during handover procedure.

5.3.3 Link Utilization

The transmission link on the 'A' interface (i.e. the interface between the BSS and MSC) is based on the use of 1 or more 2048kbps digital transmission link. From this a number of equivalent channels can be obtained. We use one such link to connect the BSS and MSC, and divide the digital channel into three types as follows:

Type I: 64kbps traffic channel

Type II: 32kbps traffic channel

Type III: 16kbps traffic channel

From this we can have 32 Type I channels, 64 Type II channels, 128 Type III channels or several combinations there of. Each of the channels can be used as signalling or traffic channels. We present some simulation results to show the utilization of these traffic channels between the MSC and the BS. At the end of every channel allocation and release, the network database is updated with the type and number of channels in use as well as available for allocation. The load on the network is normalized to the total BHCA for all transaction in a mobile switch area.

In the first simulation, traffic channels are allocated in a first come first serve basis. Depending on the type of service, any one type of channel is allocated. For instance, if a 24kbps data rate is requested, the network will allocate a 32kbps channel, since a 16 kbps channel is insufficient. This type of allocation though better than allocating a full 64kbps channel, is still wasteful as shown in Figure 45. From the figure, we observe that the utilization of different types of channels is very similar, however with increasing traffic, more channels are used and there is not enough channels remaining to handle additional traffic in case of link failure.

Figure 44 Handover rejection probability as a function of α

Figure 45 Utilization of traffic channel, without IN capabilities

Figure 46 Utilization of traffic channel, with IN capabilities

Figure 47 Processor Utilization for various network components

In the second simulation, before a traffic channel is allocated, the type of service being sought is identified by the network. The capacity of the channel required for such a service is determined. Processor rates at the various network points are also determined for the service. Resources are allocated such that maximum efficiency or minimum wastage is achieved. For example, if a 100 kbps channel is demanded, the network will determine the combination of Type I, II, and II channels needed to provide the needed capacity and also ensure that there are enough channels of each type available; instead of allocating two Type I channels, the network will allocate one channel each from all the three types. Figure 46 shows the channel utilization when the choice is made based on an efficient channel usage. It can be seen that even at high traffic load (equal to 1.0) there are sufficient channels available for each type to handle additional traffic in case of link failure. This improvement comes at a cost of increasing the signalling bytes in the network. The additional SS7 signalling bytes introduced into the network to improve system performance is further investigated later.

The processor utilization of the various network components are depicted in Figure 47 for all services. At nominal load equal to 1.0, the SCP (HLR processors), SSP (MSC processor), and BSS processor are 68%, 62% and 23% busy respectively. This is a useful information to help network planners and administrators to determine how much processing time to allocate for specific functions.

5.3.4 System Response Time

One of the very important performance criterion in signalling networks is delay performance. There are no performance objectives given in the standards concerning the end-to-end signalling delays. Formulas are provided in [41] for signalling link queueing delays based on M/G/1 queueing models. However, no recommendations are given in CCITT standards on what delay criteria should be used in teletraffic engineering and signalling dimensioning. Some objectives have been provided for cross-office delays for networks that use SS7 signalling [49]. We present results from our investigation of the effect that an increase in load has on the delay in the network. Two type of workload was considered. Type A workload involves procedures that provide only the basic call processing features (i.e. call establishment, maintenance and tear-down) and type B requires some advanced call features in addition to the basic call processing

features, such advanced features are delivered by the intelligent network services for example freephone service. The call model which includes these features was discussed in Chapter 3. Figure 48 shows the variation of the mean response time with respect to the nominal load for the two workloads. A confidence intervals of 95% has been used. The response time is expressed in seconds and includes processing time as well as waiting time at the SSP, SCP. The curve possesses very important information. Firstly, the nature of the curve suggests that there is still much to be done as far as the response time objectives of the proposed network are concerned. Secondly, the delays identical for both types of workload until about $\text{load}=0.4$.

To improve on the delay in the network, a scheme is implemented in the proposed network that selectively allocates more resources and processing time to certain advanced call features at the expense of others based on the level of delay in the network. We capitalize on the extensive database structure and the IN capabilities to implement this scheme. The system collects data periodically regarding the state of the average delay in the system for advanced features. If the delay for a particular service feature exceeds a certain specified threshold, the network then decides to allocate resources for this service feature according to a priority level provided by the user during call setup. This could mean more resources at a higher cost for some service and less resources at a lower cost for others. The basic call feature is always provided. This scheme will provide an improvement on the overall blocking probability because calls which otherwise would have been dropped due to delay will now have higher priority and as such experience low delay. The scheme will also generate revenues because the user has the option of requesting this advanced prompt service at a higher cost than the basic service.

Figure 49 shows the results of adopting this scheme. The improvement gained is seen at the point where the curves begin to rise. This occurs at $\text{load}=0.6$. This is a tremendous improvement over Figure 48 since it means that the system can handle more requests without and significant increase in delay. It is however, noted that the minimum delay is still quite high. This may be attributed to the delay caused by processing of the additional signalling traffic. Despite this, the information provided by these curves could help network planners to find out which key components within the network contribute immensely to expected delays and consequently take the necessary action.

Fig. 48 Response Time vs Load

Figure 49 Improved Response Time vs Load

5.3.5 Signalling load and Message distribution

The signalling load estimation was discussed in chapter 4. In this section we present simulation results assessing the effect of additional signalling load on system performance due to introduction of IN capabilities.

Service	Basic	Advanced
Call processing	485	712
Handover	246	388
Delay Improvement	–	98
Channel Utilization	–	86

Table 11 Additional SS7 bytes generated to provide enhancement on System performance

In designing systems with improved performance, we must take note of the additional signalling traffic introduced into the network. Table 11 compares the SS7 bytes per request between the SSP and the SCP for basic services (without IN features) and advanced services (with IN features). It can be seen that 60% more SS7 bytes are generated to provide advanced intelligent services over the basic call services. This is quite high and can lead to system overload and blocking at high load if some kind of control strategy is not used. With the scheme discussed in the previous section aimed towards improving the system response time, it was shown that selectively discouraging certain service features can bring about an improvement as seen in Figure 49. This is because at low traffic the additional SS7 bytes generated do not contribute any significant delay. But as the load in the network increases, this improvement becomes less and

less because the number of additional SS7 bytes generated increases and their delay contribution no longer remains insignificant.

We next present traffic load and message distribution for the various network entities. For this particular simulation we use one BSS and assume that it is an aggregate of many BSS required to support the call traffic. This simplifies our determination of the signalling message load at each network entity. We also assume that the VLR and the MSC are separately located. This is to provide the necessary rates for SS7 link dimensioning. Table 12 shows the various SS7 link messages and their average sizes in the various network entities. From this results, the control links between the network entities can be dimensioned. For examples, the number of 64kbps links between the SSP and SCP can be obtained as follows [50]

$$No. of links = \frac{(422msg/sec \times 60bytes/mgs \times 8bits/byte)}{(64kbps \times 0.5util)} = 6.33 \quad (5.1)$$

This means seven 64kbps links are needed for signalling messages to flow from the SSP to the SCP when the utilization of the links is not more than 50%.

Table 13 shows the number of links and their utilization. In determining the number of links needed we considered the utilization to be less than 0.5 so that each pair would be capable of carrying additional traffic load should a link fail. Since not all network entities have relatively balanced traffic load, the utilization of the links can be significantly different for the uplink and downlink. This is seen in the SCP to SDP (HLR to VLR) link, which indicates that when traffic imbalances are ignored, the utilization on one side of the signalling link far exceeds the maximum allowed even though the utilization on the other side is within acceptable limit. This imbalance is due to the fact that the HLR communicates very frequently with the VLR during periods of location update. The information provided in Table 12, though useful, does not truly represent the real story in the network and so are not reliable when used to configure the network links. The real picture is depicted in the distribution of message traffic between the network entities as presented in Figures 50—55. For instance from Table 12, the average message (worst case) size for the links between the BSS and MSC is 48 bytes, however as seen from Figure 53, the actual message size ranges from 10 bytes to 100 bytes. Clearly, a better way to configure a link is to include the range of message sizes

Network Entities	Ave. Msg. Size (bytes)	Msg load (msg/sec)
BS to MSC	48	1240
MSC to BS	53	1335
SSP to SCP	60	422
SCP to SSP	70	546
SSP to SDP	61	741
SDP to SSP	67	964
SCP to SDP	65	522
SDP to SCP	67	65
SSP to SSP	34	992
SSP to PSTN	42	496

Table 12 Signalling link message load between network entities

and their associated distribution. Figure 55 shows the total signalling load distribution as seen at the MSC. This provides an understanding of the traffic load on the MSC and hence may enable network planners to allocate resources based on this load distribution.

Network Entities	Utilization (%)	No. Links
BS to MSC	41.33	18
MSC to BS	49.13	18
SSP to SCP	31.17	10
SCP to SSP	47.78	10
SSP to SDP	33.23	17
SDP to SSP	47.49	17
SCP to SDP	84.82	5
SDP to SCP	10.89	5
SSP to SSP	46.82	9
SSP to PSTN	43.40	6

Table 13 SS7 links Utilization between network entities

Figure 50 Message Size Distribution between BSS and MSC

Figure 51 Message Size Distribution between SSP and SCP

Figure 52 Message Size Distribution between SCP and SDP

Figure 53 Message Size Distribution between SSP and SDP

Figure 54 Message Size Distribution between SSP and PSTN

Figure 55 Total Message Size Distribution at the SSP

Chapter 6

Conclusion

6.1 Conclusion

We have shown that a moderate set of capabilities within the SSP, SCP and IP, which are the intelligent network components responsible for real time service execution, can support the basic PCS features. The set of capabilities identified in the CCITT Capability Set 1 are found to be sufficient. However, with some modifications on the existing capabilities a wide range of advanced services can be extended to PCS users. In some cases it might even be necessary to create new capabilities specific to PCS needs.

Our simulation results show some limitations of using IN to providing PCS service. As described in previous sections some IN-based services results in significant increases in the number of SCP-SSP messages. This causes increased processing in the network elements. This can lead to significant delay in response as perceived by the user. Several approaches were considered to improve the system performance. We have shown that when network resources are allocated based on the type of PCS traffic (i.e. microcell or macrocell), an improvement in system performance results. We have also shown that by selectively discouraging certain types of services, the additional increase in signalling load due to introduction of IN capabilities can be controlled. Our results were based on the speed of the SPARCstation processor. However, with processor speed in switching systems becoming more and more faster, this will compensate for the increased processing required for IN. In the interim, efficient switching designs incorporating distributed and multiprocessing architectures are essential to the success of the SSP in the IN concept.

Another strategy for improving system performance is moving some of the control logic to the SSP. Because most services currently being considered in IN involves simply fetching records and instructions from the SCP, significant increases in performance level can be obtained by moving frequently accessed records into the SSP. This can be achieved as follows. By monitoring the trigger tables at the SSPs for most frequent entries, the service logics for such entries can be downloaded at the respective SSPs. Triggers from such entries would result in the records being accessed at the SSP instead

of the SCP. This would significantly reduce the number of SCP-SSP messages and improve the delay response, but would require a much larger data store in the SSP.

6.2 Future Work

While the personal communication needs of today will be met by the merging of present technologies, this will not be adequate to meet the requirement for both the quantity and range of services required in future PCS. There is need to develop a new generation of personal communication network, building on the convergence of these technologies. The development of the CT-2+ as the first personal communicator is a step towards the evolution of true personal communications allowing us to meet the ideal customer requirement for a small, low cost, 2-way voice communicator which can be used anywhere by the end of the century. The Digital European Cordless Telephone (DECT) working group is also considering the development of the next generation of digital cordless phone — the CT-3. The functionality of the CT-3 could be specified flexibly to enable the product to operate on both the second generation cordless frequencies (likely to be around 1.6GHz) and the digital cellular frequencies (900MHz), thus creating the early evolution of the personal communicator.

However, challenges still remain to be addressed. As an extension to this thesis, future work should investigate the traffic performance of networks with extensive overlaid cell configurations since this is the environment in which PCS will ultimately be. There is also the need to study the effect of other services such as data and video on the network performance. Theoretical analysis using queuing network models could also be performed with traffic rates from microcells and macrocells being represented as different classes of calls.

Given that PCS is a strong candidate to be the next major user of IN, future work must also focus on developing a new set of call model TCPs and FEAs capabilities to implement the mobility aspects of PCS and to attest that IN is economically feasible. Once a proven baseline architecture is established and operational experience is gained, then a wider variety of IN capabilities can be included. Simulations should also be performed to study the effect of mobility pattern on the performance.

Appendix A

Signalling System No. 7

A.1 Introduction

SS7 is a common channel signalling system defined by CCITT [41]. It is a key element in supporting a large number of applications in telecommunication networks ranging from call control in the interconnection of the nodes of an ISDN to Intelligent Networks and Mobile Telecommunication Services. Unlike traditional signalling systems that uses in-band signalling techniques, SS7 is a packet switching system that uses separate channels dedicated for signalling between network elements. This provides a vehicle for reliable transfer of circuit related information between network elements for call setup and release and reduces the connection setup time significantly. The existence of signalling path separate from user information path facilitates call control prior to and during conversation phase adding more flexibility to the network capability. SS7 also provides an opportunity to distribute the network database resources.

The signalling network consist of signalling points and signalling links connecting the signalling points together as shown in Figure 56. A Signalling Point (SP) is similar to a node in a network. It contains message discriminating and distribution functions. If it has a transfer capability, then it is referred to as a Signalling Transfer Point (STP).

A.2 The Protocol Architecture

SS7 is based on a layered protocol architecture similar to the Open System Interconnection (OSI) reference Model. It consists of four functional levels as shown in Figure 57. The levels corresponding to the lower three layers of the OSI model, is referred to as the SS7 Network Service Part. This consist of the MTP, which provides a simple sequenced connectionless service, and the Signalling Connection Control Part (SCCP). All other functional block are contain in the fourth level which is referred to as the SS7 User parts. These include the ISDN-UP, Telephone User Part (TUP) and Transaction Capability Application Part (TCAP).

Figure 56 Signalling Network Structure

A.3 SS7 Network service Part

A.3.1 Message Transfer Part

The purpose of the MTP is to provide a reliable transfer and delivery of signalling information across the signalling network and to have the ability to react and take necessary actions in response to system and network failure to ensure that reliable transfer is maintained. The three levels in the MTP are described below:

Signalling Data Link Function

The signalling data link function corresponds to the physical layer of the OSI model. It is responsible for the physical, electrical and functional characteristics of the signalling path. It consists of a full-duplex digital channels exclusively dedicated to the use of signalling. The recommended bit rate for the ANSI standard is 56kb/s, and for the CCITT international standard it is 64kb/s. The minimum bit rate allowed for telephone call control applications is 4.8kb/s. Recommendations for higher bit rates e.g. 1.544 Mb/s is possible but require further study.

Figure 57 SS7 Protocol Architecture

Signalling Link Function

The signalling link function corresponds to the data link layer of the OSI model. Together with the signalling data link, this function provides a signalling link for the reliable transfer of signalling messages between two directly connected signalling points. The signalling messages are transferred over the signalling link in variable length messages called Signal Units (SU). Depending on the functionality of the signal unit, three types of SUs can be identified. These are: Message Signal Unit (MSU), Link Status Signal Unit (LSSU) and Fill-In Signal Unit (FISU) as shown in Figure 58. All of the signal units contain the following:

(a) Format of MSU

(b) Format of LSSU

(c) Format of FISU

BIB: Backward Indicator Bit

BSN: Backward Sequence Number

CK: Check Bit

F: Flag

FIB: Forward Indicator Bit

FSN: Forward Sequence Number

LI: Length Indicator

SIF: Signalling Information Field

SIO: Service Information Octet

SF: Status Field

Figure 58 Signal Unit Format

Flag Bits (F): A unique 8 bit pattern (01111110) at the start and end of the signal unit. This is used to mark the beginning and ending of the variable bit signalling information.

Check bit (CK): Two octets of error detection bits are appended to the end of the bit stream for the detection of transmission errors. To prevent the reception of corrupted signals, a retransmission method is employed. Two method of retransmission are defined: the basic method and the preventive cyclic retransmission.

Forward Sequence Number (FSN): This 7 bit field represents the sequence number of an MSU. Each successive MSU increments the last assigned FSN by a value of 1 using modulo 128. The LSSU and FISU carries the FSN associated with the last MSU transmitted.

Backward Sequence Number (BSN): This 7 bit field represents the sequence number of an MSU being acknowledged. A positive acknowledgment is sent whenever an MSU is received correctly. The receiving point assigns the FSN value of the received MSU to the BSN of the SU used to acknowledge this reception.

Forward Indicator Bit (FIB): This 1 bit field is used to indicate retransmission for a transmitted signal unit. The retransmission are indicated by inverting the FIB value upon beginning of retransmission. This inversion is maintained in subsequent signal units until another negative acknowledgment is received.

Backward Indicator Bit (BIB): This 1 bit field is used to indicate a negative acknowledgment for a transmitted signal unit. The negative acknowledgment are indicated by inverting the BIB value and maintaining it in subsequent signal units until another negative acknowledgment is received.

Length Indicator (LI): This 6 bit field is used to determine the three types of SUs. A value of zero represents FISU. A value of One or Two represents LSSU. Any value greater than two represents MSU. The value of the LI represents the number of bytes between the LI and the CK. When this number is larger than the maximum number presented by LI, the LI is set to 63.

In addition to the above, the MSU contains the Signalling Information Field (SIF) and Service Information Octet (SIO). The SIO is one octet and the SIF consists of an integer number of octets between 2 and 272. The LSSU also carries a Status Field (SF). This is a one or two byte field that indicates the status of the link. Six different status indications have been defined including: Out of Alignment; Normal; Emergency; Out of Service; Processor Outage; and Busy status.

Signalling Network Function

The signalling network functions corresponds to the lower half of the OSI network layer. They provide the functions and procedures for the transfer of messages between signalling points. The two classes of signalling network functions are the signalling

Figure 59 Signalling Network Functions

message handling and the signalling network management. Figure 59 illustrates these classes.

Signalling Message Handling — Signalling message handling functions consists of message routing, message discrimination and message distribution. These functions are necessary for the transfer of messages between two users of MTP over a signalling link. The part of the message that performs these functions are the Routing Label and the SIO field. The routing label is placed at the beginning of the SIF and consists of the

Figure 60 Routing Label

Destination Point Code (DPC), the Origination Point Code (OPC) and the Signalling Link Selection (SLS) fields as shown in Figure 60. The SLS is used to perform load sharing among multiple signalling links. In the CCITT standards, the DPC and OPC are 14 bits each while the SLS field is 4 bits long. In the ANSI standard, the DPC and OPC are 24 bits long representing an 8-bit network indicator, 8-bit network cluster number and 8-bit network cluster member number. The SLS field is 5-bits long with 3 spare bits. The SIO field includes a Service Indicator (SI) field and a Sub-Service Field (SSF), each 4 bits long. The SI is used for message distribution or to support the signalling network management at the destination point. The SSF is used to distinguish between ANSI and CCITT routing label types.

Signalling Network Management — The signalling network management functions provide reconfiguration of the signalling network in the case of signalling link or signalling point failures. They also control traffic in the case of congestion or blocking. The signalling network management consist of three functions as shown in Figure 59. The signalling traffic management function is used to divert signalling traffic, without causing message loss or duplication from unavailable signalling links or routes, to one or more alternative signalling links or routes, or to reduce traffic in the case of congestion. The signalling route management function is used to distribute information about the signalling network status in order to block or unblock signalling routes. A number of procedures to implement the different situations that can occur have been defined by CCITT and are documented in [41, 49, 51]. The signalling link management function is used to restore failed links, activate new signalling links and deactivate aligned signalling links.

A.3.2 Signalling Connection Control Part

The SCCP enhances the services provided by the MTP to include connection-oriented services, connectionless control, routing and management. The addressing capability of the MTP is also enhanced by the SCCP using DPC and Sub-System Numbers (SSN). Messages with global titles can be addressed. These messages do not explicitly contain information usable for routing at the MTP. A translation capability is provided in the SCCP to translate the global title to a DPC + SSN that can be used for routing at the MTP. This translation function can be performed at the originating point or at an STP. The following classes of services are provided by the SCCP:

- Class 0: Basic connectionless class
- Class 1: Sequenced (MTP) connectionless class
- Class 2: Basic connection-oriented class
- Class 4: Flow control connection-oriented class

A.4 SS7 User Part

This section describes the major SS7 user parts that use the transport services provided by the MTP and the SCCP. They are the ISDN-UP, TCAP, and other parts

such as the Mobile Application Part (MAP), Intelligent Network Application Protocol (INAP) that uses the services provided by the TCAP.

A.4.1 ISDN User Part

The ISDN-UP protocol provide call related services in support of ISDN. It supports the basic bearer services and supplementary services. The ISDN-UP can access the MTP directly or via the SCCP to transfer signalling information between user parts (Figure 57). ISDN-UP messages consist of a routing label identifying the origin of the message, a Circuit Identification Code (CIC) for referencing a particular circuit, a message type code that uniquely defines the function and format of each ISDN-UP message, and a set of parameters. Figure 61 illustrates the format of the ISDN-UP message. The mandatory fixed part refers to a set of predefined fixed length parameters for a particular message type. The mandatory variable part refers to the set of predefined variable length parameters with pointers for a particular message type. A parameter length indicator holds the length of each parameter. The optional part consists of a set of parameters, fixed length or variable length, that may or may not be present in any particular message type. A large set of message types have been defined which performs specific functions. Table 14 provides a list of the messages types used in call setup and release. Only fixed and variable parameters have been specified.

A.4.2 Transaction Capability Application Part

TCAP provides the set of features in a connectionless environment that can be used by an application at one signalling point to invoke execution of a procedure at another signalling point and exchange the results of such invocation. TCAP uses the services provided by the SCCP. The distributed applications that uses TCAP may reside in exchanges and in database sites, examples are freephone service, credit card and calling card validation and any other services that require a query/response type of interaction. Figure 62 illustrates the structure of the application service layer. The TCAP is divided into two sublayers, the component sublayer and the transaction sublayer.

Transaction sublayer

The Transaction Sub-Layer (TSL) is responsible for the management of dialogue between two Transaction Capability (TC) users. Two kinds of dialogue can take place

Routing Label
Circuit ID Code
Message Type Code
Mandatory Fixed Parameters
Mandatory Variable Parameters
Optional Parameters

Figure 61 ISDN-UP Message Format

between peer transaction sublayers: the unstructured and the structured dialogue service. In the unstructured dialogue service, the TSL provides the means for a TC-user to send to its remote peer, one or more components that do not require any responses. In the structured dialogue service, the TC-user sends a TC-BEGIN primitive containing a unique dialogue ID to the component sublayer which maps it into a TR-BEGIN primitive with a unique transaction ID and forwards it to the TSL. The TSL manages all transactions by grouping each transactions with the same transaction ID into BEGIN messages and transmits them to the remote TSL. Other primitives are CONTINUE, END and ABORT.

Message Type		Description	Parameters	Length (bytes)
Initial Address Message	IAM	Initiates Call setup by reserving out going trunk. Carries information about routing and call handling	Message type, Nature of connection indicators, Calling party's category, Transmission medium requirements, Called party number	10-17
Address Complete Message	ACM	Indicates the complete receipt of the called party address at the final destination exchange	Message type, Backward call indicator	3
Answer Message	ANM	Indicates that the call has been answered	Message type	1
Release Message	REL	Indicates that the circuit is being released	Message type, Cause indicators	4
Release Complete	RLC	Response to REL and indicated ready for seizure	Message type	1

Table 14 ISDN-UP Messages

Component sublayer

The Component Sub-Layer (CSL) consist of either an invocation to perform a remote operation or a reply to an invocation. It allows several dialogues to be run concurrently between two TC-users. A unique ID is associated with every invocation of an operation. This allows several invocation of the same remote operation to be active simultaneously. The value of the ID is returned in any reply to that operation. CSL provides for four classes of remote operations:

- Class 1: both success and failure in performing the remote operation are reported;

- Class2: only failures is reported;
- Class 3: only success is reported;
- Class 4: neither failure nor success is reported.

The reply to an operation can contain one of the following: RETURN RESULT (last), RETURN ERROR or REJECT depending on whether a result, an error or a notification of syntax error in performing the operation was encountered.

Figure 62 Application Layer Structure

A.4.3 Other Application Part

There are a number of other SS7 user parts such as the Telephone User Part (TUP) and Data User Part (DUP). The TUP is specified to provide the signalling functions to support control of telephone calls on national and/or international calls. DUP provides the signalling functions for control of data transfer.

One of the main features of Intelligent Networks is the use of the SS7 as the transport

vehicle for intelligent network signalling. The Intelligent Network Application Protocol (INAP) specified in Q.1218 [25] is a remote operation protocol and so is a TC-User. The recommendation defines the interface relations between remotely located network entities (SSF, SCF, SRF, SDF) using TCAP.

Mobile telephony signalling is also another important area where the use of SS7 is growing. The distributed application that uses the TCAP to provide the signalling function to support telephone call from mobile terminals is referred to as the Mobile Application Part (MAP) and is defined in CCITT Rec. Q.1051 [48].

Appendix B

The Intelligent Network Concept

B.1 Introduction

This section provides an overview of the Intelligent Network concept offered in CCITT's Drafted recommendation for Intelligent Network Compatibility Set 1 (CS-1) as specified in the Q.1200 series [25]. The recommendation gives an introduction to IN by providing the objectives and overall description of IN and by describing its main characteristics and overall capabilities. It defines the service aspects, network aspects and the functional relationships that forms the basis of the CS-1 capabilities. CS-1 is the first standardized stage of IN as an architectural concept for the creation and provision of telecommunication services.

B.2 IN Concept

Definition

The term Intelligent Network (IN) is used to describe an architectural concept which is intended to be applicable to all telecommunication networks. Its aim is to make the introduction of new services such as UPT, VPN, Freephone etc. much more easier by offering more flexibility and new capabilities.

The Intelligent network therefore is an architectural concept for the creation and provisioning of telecommunication services which is characterized by:

- a. extensive use of information processing techniques
- b. efficient use of network resources
- c. modularization of network functions
- d. integrated service creation and implementation by means of standard reusable network functions
- e. flexible allocation of network functions among physical entities
- f. portability of network functions to physical entities

- g. standardized communications between network functions via service independent interfaces
- h. service subscriber control of subscriber specific service attributes
- i. standardized management of service logic

The interest in IN is motivated by the increasing demand of telecommunication service providers to rapidly, cost effectively and differentially satisfy their existing and future market needs for services. Also these service providers seek to improve system quality and reduce network operating and management cost. In addition, there is a greater degree of intelligence and greater freedom in the allocation of intelligence in current telecommunication networks technology, for instance, advances in digital transmissions and switching, common channel signalling, distributed data processing, database management and expert systems has led to improved mobility being derived from pocket size electronic components, this allows for greater degree of distributed functionality within and between service provider networks.

B.3 Functional Requirements and Network Capabilities

IN Functional requirements arise as a result of the need to provide network capabilities for both:

1. customer needs (service requirements) and
2. network operator needs (network requirement);

A service user is an entity external to the network that uses its services. A service is that which is offered by an administration to its customers in order to satisfy a telecommunication requirement. Part of the service used by the customer may be provided and/or managed by other customers of the network. In order to provide a network with the capability to deliver IN services to the user and service provider, it is important to determine those requirements pertaining to the customer that provides him with enough flexibility to customize his service

B.3.1 Service Requirements

The service requirements emphasizes what the customer expects the network to be

capable of doing in order to deliver IN services. A number of such requirements have been stated below:

1. it should be possible to access services by the usual user — network interface e.g. POTS, ISDN;
2. it should be possible to access services that span multiple networks;
3. it should be possible to invoke a service on a call by call basis, or for a period of time, in which case the service may be deactivated at the end of the period;
4. it should be possible to perform some access control to a services;
5. it should be easy to define and introduce a service;
6. it should be possible to support services involving calls between two or more parties;
7. it should be possible to record service usage in the network e.g. for the purpose service supervision, tests, performance and charging;
8. it should be possible to provide services that imply the use of functions in several networks;
9. it should be possible to control the interactions between different invocations of the same service.

B.3.2 Network Requirements

The network requirements view IN services provision from the network operator's point. It emphasizes what the operator expects the network to be capable of doing in order to provide IN services to customers. The following reflects the overall network requirements of IN. Specific detailed requirements can be found in [25]

1. it should be possible to migrate from existing network bases to target network bases in a practical, flexible and cost-effective manner;
2. it should be possible to reduce redundances among network functions in physical entities;
3. it should be possible to allow flexible allocation of network functions to physical entities;
4. there is a need for a communication protocol that allow flexibility in the allocation of functions;
5. it should be possible to provide new services from network functions in a cost and time efficient manner;

Figure 63 Network Capabilities

6. it should be possible to guarantee the integrity of the network when a new services is being introduced;
7. it should be possible to manage the network elements and network resources such that quality of services and network performance can be guaranteed.

The following areas of network capabilities has been identifies to meet the service and network requirements: service creation, service management, network management, service processing and network interworking. Figure 63 illustrates the network capabilities and their relation to the service and network requirements.

B.3.3 Service Creation

This is an activity whereby supplementary services are brought into being through service specification, service development and service verification.

Service specification is the first step in the service creation process and includes activities such as refinement of detailed service description requirements, functional analysis, generation and verification of a service specification and definition of a high level structured design. Service deployment is the step which transforms a high level structured design into a detailed structured software design and subsequently develops the necessary software components, data definitions, etc. required to realize that design. Service verification is the stage in the service creation process where the developed

service software, including supporting documentation, is rigorously tested to validate that the resulting service application completely satisfies the specification. The environment that allows the easy creation of new services in a network-type and configuration independent way by means of service independent building blocks (SIBs) is the Service Creation Environment (SCE).

B.3.4 Service Management

This involves the network capabilities that are necessary to support the proper operation of services and to provide service management services to the customers. Service management activities during deployment stage include limited deployment of services for testing purposes followed by downloading of service logic, management logic etc. into appropriate network elements if testing were successful, and the activation of these logics. During service provisioning phase, service management involves such activities as the creation of customer's service profile in the service logic and the activation of the service for the customer. Service activation is realized by initialization and activation of trigger conditions within the service logic. Service management activities after service provisioning has taken place, i.e. during utilization stage include:

1. service activation, deactivation, service maintenance and service customizing. These activities make services usable by specific customers to meet their specific needs. The network supports these activities by providing the customer with the ability to modify parameters which control their services and also by verifying data entered by the customer to ensure that, for example, the customer has permission to change the indicated parameters and also to check the validity of the modified parameters;
2. charging which is mainly to collect data on service usage and to generate report for billing either on demand or automatically. The network may provide flat rate charging or usage sensitive charging which may include not only the duration of the call but also the network resources required to provide such a service;
3. service monitoring which provides the capability to collect and accumulate statistics on a given service with a view to determine the quality of service. Service monitoring can either be automatic or manual. Typical manual operations are queries regarding status information and network configuration.

B.3.5 Network Management

This refers to the network capabilities that are necessary to support the proper operation of the intelligent network. They are organized into the following management capabilities:

1. Performance management
 - performance monitoring
 - traffic management and network management
 - quality of service observation
2. Fault (maintenance) management
 - alarm surveillance
 - failure localization
 - testing
3. Configuration management
 - provisioning
 - status and control
 - initialization
4. Account management
5. Security management

B.3.6 Service Processing

This includes the network capabilities necessary for the provision of basic and supplementary services by an IN-structured network. From the user's point of view, the provision of basic and supplementary services using IN concept is transparent to the customer, i.e. the customer is not aware as to how the service is provided either using IN or not therefore no service processing requirement can be identified that has specific reference to the IN. Regardless of this, the IN should be capable of supporting a broad range of basic and supplementary services. The requirements for service processing can be identified for the following capabilities:

1. Service Capabilities: these are necessary to support a broad range of basic and supplementary services. IN concepts can be applied for the support of supplementary services for the following basic services:

Bearer services:

- circuit-mode unrestricted
- circuit-mode speech
- circuit-mode audio
- packet switched data services
- circuit switched data services
- other

Teleservices:

- Telephony
- Telefax
- Videotex

Broadband interactive services:

- Conversational services
- Messaging services
- Retrieval services

Broadband distribution services:

- Distribution service without user individual presentation control
- Distribution services with user individual presentation control

Mobility services²:

- Location
- Handover
- Personal Service Profile
- Paging

² These services are needed to support PCS and so are considered basic

2. Access Capabilities: these are necessary to interface with the network and to access the services. In order to allow freedom of access to services, it is necessary that there is a sufficient degree of independence between the service capabilities on one hand and the access capability on the other hand. The following access capabilities are foreseen for IN:

Fixed Network:

- PSTN access
- ISDN access
- PSPDN access
- CSPDN access

Private network access

Mobile network

- Cellular access
- Microcellular access
- Paging access
- MSAT access

Broadband networks

- Asynchronous Transfer Mode (ATM) access
- Synchronous Transfer Mode (STM) access

B.3.7 Other Capabilities

These may include network interworking capabilities. The need for such capabilities results from the fact that customers may want to access services which span multiple networks. These networks may have different access types (Cellular, PSTN etc.) and may have different level of IN structuring (full, partial or no IN structuring). Regardless of the access type and the level of IN structuring, services should be provided to the customer in a consistent way. Network interworking requirements exists at different levels and in each level, some network interworking related gateway functions need to be defined. The gateway functions for the service processing, service management and service creation stages has been identified in [25].

B.4 IN Service Processing Model

The main requirements for IN service processing arose from the inability of “non-IN” networks to rapidly create and deploy new supplementary services. To overcome this inability, the IN aims for:

- rapid service implementation by means of reusable network functions;
- modularization of network functions;
- standardized communication between network functions via service independent interfaces.

The IN service processing model is designed to achieve the aims listed above. The three main elements of this model as shown in Figure 64 are: the basic call processing, the “hooks” that allow the basic call processes to interact with IN service logic, and the IN service logic that can be “programmed” to implement new supplementary services. The basic call process should be available all over the network and is designed to support services that do not require special features, with optimum performance. In order to achieve flexibility in service processing, the basic call processing needs to be modularized into service-independent sub-processes such that these can be executed autonomously (without interference from the outside during execution).

Figure 64 IN service processing model

B.5 IN Architectural Concept

The functions addressed in [25] are grouped into the following categories

- call control related functions
- service control related functions
- management related functions

Figure 65 gives an overview of the functional components in the IN.

Figure 65 IN Functional Components

B.5.1 Call Control Related Functions

The following are definitions of the capabilities required for the IN functional components relating to call control:

- **Call Control Agent Function (CCAF)**

The CCAF provides access for users. It is the interface between user and network call control function. It interacts with the user to establish, maintain, modify and release, a call or instance of service. It accesses the service-providing capabilities of the call control function (CCF), using service requests (e.g. setup, transfer, hold, etc.) for the establishment, manipulate and release of a call or instance of service. It receives indication relating to the call or service from the CCF and relays them to the user as required.

- **Call Control Function (CCF)**

The CCF provides call/connection processing and control in the network. It establishes, manipulates and releases call/connection control. It provides the capability to associate and relate CCAF functional entities that are involved in a particular call and/or connection instance (that may be due to SSF request). It manages the relationship between CCAF functional entities involved in a call (e.g. supervises the overall perspective of the call and/or connection instance). It provides trigger mechanism to access IN functionality (e.g. passes events to the SSF).

B.5.2 Service Control Related Functions

The following are definitions of the capabilities required for the IN functional components relating to service control:

- **Service Switching Function (SSF)**

The SSF in association with the CCF, provides the set of functions requested for interacting between the CCF and the SCF (service control function). It extends the logic of the CCF to include recognition of service control triggers and to interact with the SCF. It manages signalling between the CCF and the SCF. It modifies call/connection processing function (in the CCF) as required to process request for IN provided service usage under the control of the SCF.

- **Service Control Function (SCF)**

The SCF contains the logic and processing capability required to handle IN provided

service attempts. It commands call control functions in the processing of IN provided and/or custom service requests. The SCF may interact with other functional entities to access additional logic or to obtain information (service or user data) required to process a call/service logic instance. It interfaces and interact with SSF/CCF, SRF, SDF FEs. It directs the CCF via the SSF in the realization of IN supported services.

- **Special Resource Function (SRF)**

The SRF provides the specialized resources required for the execution of IN provided service (e.g. digit receivers, announcements, conference bridges, etc.). It interfaces and interacts with SCF and SSF (and with the CCF). It may contain logic and processing capability to receive/send and convert information. It may contain functionality similar to the CCF to manage bearer connections to the specialized resources.

- **Service Data Function (SDF)**

The SDF contains customer and network data for real-time access by the SCF in the execution of an IN provided service. It handles access to service related data and network data and provides consistency checks on that data. It also provides a logical data view to the SCF thereby removing any dependency relative to the real data implementation in the SDF.

B.5.3 Management Related Functions

The following are definitions of the capabilities required for the IN functional components relating to management of service in the network:

- **Service Management Function (SMF)**

The SMF involves service management control, service provision control and service deployment control. It updates the SCF with new data or programs and collects statistics from the SCF. It also enables service subscribers to control their own parameters by specifying for example, the day and time when an 800 number should be routed to a specific office.

- **Service Management Agent Function (SMAF)**

This provides an interface to the SMF.

- **Service Creation Environment Function (SCEF)**

The SCEF allows an intelligent network service to be defined, developed tested and

Figure 66 IN Physical Components

input to the SMF. It produces service logic and service data templates as its output.

Each functional element are mapped into physical components as shown in Figure 66. Communication between the various network entities is via an SS7 network.

B.6 IN Conceptual Model

This provides a framework within which various models and concepts used in the standardization of IN are identified. This makes it possible to define clearly the purpose, value and limitation of any IN concepts and its relationship to other such concepts.

INCM consists of four planes, they are the Service, Global Functional, Distributed Functional, and Physical planes, Figure 67. Each plane represents a different abstract view of the capabilities provided in an IN controlled network.

B.6.1 Service Plane

This plane describe IN supported services by means of generic blocks called Service Features. A service is a stand-alone commercial offering, characterized by one or more core service features, and can be optionally enhanced by other service features. A service feature is a specific aspect of a service that can also be used in conjunction with other service or service features as part of a commercial offering. It is either a core part of a service or an optional part offered as an enhancement to a service. The service plane represents an exclusive service oriented view. This view contains no information whatsoever regarding the implementation of the service in the network (for instance, an IN type of implementation is invisible). All that is perceived is the network's service related behavior as seen for example, by a service user. A set of targeted services and service features have been identified for CS-1. Tables 15 and 16 displays some of these services. These SFs can be used as building blocks in the specification and design of new and more complex services. SFs are comprised of one or more SIBs.

B.6.2 Functional Plane

Global Functional Plane

The global functional plane models network functionality from a global or network wide point of view. As such, the IN structured network is said to be viewed as a single entity in the GFP. In this plane, services and service features are redefined in terms of the broad network functions required to support them. These functions are neither service nor service feature specific and are referred to as service independent building blocks (SIBs). The GFP contains the Basic Call Processing (BCP), SIBs and Global Service Logic (GSL). The BCP which is an SIB, identifies the normal call processing from which IN services are launched. This includes the POI and POR which provides the interface from the BCP to GSL. The GSL describes how SIBs are linked together to describe service features and also describes the interaction between the BCP and the

Figure 67 IN Conceptual Model

ABD	Abbreviated Dialling	CS	Call Screening
ACC	Account Card Calling	CON	Conference Calling
AAB	Automatic Alternate Billing	SCF	Selective Call Forwarding
CF	Call Forwarding	SPL	Split Charging
CRD	Call Rerouting Distribution	VPN	Virtual Private Network
CCC	Credit Card Calling	UPT	Universal Personal Telephone
CD	Call Distribution	VOT	Televoting
FPH	Freephone	UDR	User Defined Routing

Table 15 CS-1 Services

SIB chains. Table 17 shows the set of SIBs defined in CS-1. In Figure 68, normal or non-IN supported services are processed within the BCP. When an IN supported service is to be invoked, its GSL is launched at POI by a triggering mechanism from the BCP. Knowledge of the connection pattern, decision options and data required by the SIBs are available in the GSL. At the end of the chain of SIBs, the point of return to the BCP is indicated by the POR. For a given service, at least one POI is required. However, depending on the logic required to support the service, multiple PORs may be defined.

Distributed Functional Plane

The DFP models a distributed view of an IN structured network. This plane contains the functional entities which provides service capabilities in IN CS-1. A Functional Entity (FE) may perform a variety of Functional Entity Actions (FEAs). A FE is a unique group of functions in a single location and a subset of the total set of functions required to provide a service. The following functional entities have been defined in

ABD	Abbreviated Dialling	CW	Call Waiting
ACB	Authomatic Call Back	TDR	Time Dependent Routing
AUTC	Authentication Check	LOG	Call Logging
AUTZ	Authorisation Code	CS	Call Screening
CD	Call Distribution	TRA	Call Transfer
CF	Call Forwarding	CUG	Closed User Group
CHA	Call Hold with Anouncement	PN	Personal Numbering
GAP	Call Gapping	SPLC	Split Charging

Table 16 CS-1 Services Features

CS-1, i.e. CCAF, CCF, SSF, SCF, SDF, SRF, SMF, SMAF, SCEF. The relationships between the above functions are maintained via a set of defined interfaces for the service control relationship in CS-1. The DFP also contain the Distributed Service Logic (DSL) programs which holds a set of FEA to be executed and defines the link between them.

B.6.3 Physical Plane

The physical plane models the physical aspects of IN-structured network. It also identifies the different physical entities and interfaces between these entities. Functional entities in the DFP are mapped unto physical entities in this plane. A physical architecture can be formed by a group of physical entities. The following PEs are defined to support IN.

- **SSP** : This contains the CCF and SSF functionality. If it is a local exchange then it may contain the CCAF. It may also optionally contain an SCF, and/or SRF and/or

Figure 68 GFP Modelling

SDF in which case it becomes a Switching Node (SN)

- **SCP** : This contains an SCF and may optionally contain an SDF. It contains the SLPs that are used to provide IN services. The SCP can be connected to the SSP and optionally to IPs through the SS7 network.
- **SDP** : It contains the SDF functionality and provides the storage facilities for keeping data used by the SLPs to provide individualized services. It can be accessed directly by SCP and/or SMP through the SS7 network.
- **IP** : This contains the SRF functionality and provides special resources for the customization of services, and supports flexible information interactions between a user and the network.
- **SMP** : This contains the SMF, and optionally, SMAF and SCEF functionality. It performs service management control, service provision control and service deployment control.

Figure 66 shows the various PEs and how they are related.

B.6.4 SIBs and FEAs

SIBs are standard reusable network wide capabilities residing in the global functional

ALGORITHM	BASIC CALL PROCESSING
CHARGE	COMPARE
DISTRIBUTION	LIMIT
LOG CALL INFORMATION	QUEUE
SCREEN	SERVICE DATA MANAGEMENT
STATUS NOTIFICATION	TRANSLATE
USER INTERACTION	VERIFY

Table 17 CS-1 Service Independent Building-blocks

plane used to create service [25]. Each has a unified and stable interface with one or more inputs and one or more outputs. A service feature is provided by a combination of one or more SIBs linked together in a particular order specified in the service logic of the GFP, i.e. GSL. FEAs are actions performed at a given location (FE) and comprise specific set of basic functions that cannot be further decomposed. SIBs are provided by a combination of one or more FEAs in the DFP following a particular order specified in the DSL. FEAs involve the transfer of information flows between the different FEs. These information flows are messages exchanges between the different FEs in the DFP. Table 18 shows the set of information flows between the different FEs for CS-1 [25]. Information flows between physically separated FEs are implemented in the physical plane through a standardized OSI-based application layer protocol, the Intelligent Network Application Protocol (INAP). The rest of the layers is based on CCITT's Signalling System Number 7 (SS7) protocol [49, 51].

B.6.5 CS-1 Call model

The Call Model is a set of Points in Call (PICs), transitions and Trigger Check Points (TCPs) or Detection Points (DPs) used to illustrate the different stages involved in a call, from off-hook (the user picks up the phone) to on-hook (the user hangs up).

<u>SCF - SSF</u>	
Apply Charging	Connect to Resource
Analyze Information	Status Report
Collect Information	Initial DP
Call Information Request	Select Route
Furnish Charging Info	Request Report BCSM
Request Status Report	Release Call
Disconnect Forward	Connect
<u>SCF - SRF</u>	
Play Announcement	Specialized Resource Report
Prompt and Collect User Information	Cancel Announcement
<u>SCF - SDF</u>	
Query	Query Result
Update Data	Update Confirmed

Table 18 CS-1 Information Flows

There are two separate sets of call processing logic in switching systems, one for the originating side of the call and one for the terminating side of the call. Figures 69 and 70 illustrates the IN basic call model for both the originating and the terminating side of a call. Each side of the call model includes a number of PICs and DPs. PICs are active operations performed by the switch, such as collecting information or selecting a

(O) Origination Attempt	(O,T) Answer
(O) Collected Information	(O,T) No Answer
(O) Analyzed information	(O,T) Busy
(O) Route Select Failure	(O,T) Abandon
(T) Termination Attempt	(O,T) Disconnect
	(O,T) Mid-Call

Table 19 Trigger Check Points for BCPM

route. TCPs refers to the completion of such operation. They are events in the basic call processing model which are made visible to the IN service logic. If a TCP is encountered by the switch during call processing, two things happens depending upon the information at hand. If the TCP is armed, then call processing is suspended and the SSP initiates a query to the SCP service logic for further instructions. This allows the IN service feature to supplement the call processing. Resumption of the call processing model at specific PICs is dictated by the SCP. If on the other hand, the TCP is unarmed, then call processing will continue without interruption and the SSP will proceed to the next PIC. A detailed description of the CS-1 Call Model can be found in the CCITT Q.1214 Series Recommendation [25]. A number of TCPs have been specified for CS-1 call processing model and are displayed in Table 19.

Figure 69 Originating Call Model

Figure 70 Terminating Call Model

Appendix C

BONeS

C.1 Introduction

The Block Oriented Network Simulator (BONeS) provides an interactive graphical environment for simulation-based analysis and design of a broad range of communication networks [52]. In the BONeS environment, the network model is specified in terms of network topology, traffics, data structures, and protocols functions. The model is constructed graphically using the BONeS library [53]. The network model is translated into a C program and executed using an event-driven Monte-Carlo simulation. Statistical measures of network performance are computed and displayed graphically. The key components in the BONeS software are: block diagram editor, data structure editor, simulation manager, post processor, each of these components are described briefly in the following sections.

C.2 Block Diagram Editor

The Block Diagram Editor (BDE) is used to create, edit, document and store block diagrams. The network model is constructed graphically using the BDE. This involves adding functional blocks on screen and connecting them to form protocol functions. These functions are combined to form processes. A group of process can be combined to form a processor which are combined to form a node. Networks and systems can be formed by combining a group of nodes. The network is constructed by specifying the structure of the entities flowing in the network (data structures), the rule governing the flow (protocol functions), the topology and the traffic. Figure 71 shows a sample of modules from three levels of a BONeS FDDI model [52].

C.3 Data Structure Editor

The Data Structure Editor (DSE) is used to create, edit, document and store Data Structures (DS). The DS necessary to specify a network model is created using the DSE.

Figure 71 A BONeS FDDI Model

The DS can have an arbitrary number of fields. The fields can range from very simple ones (e.g. integer) to more complicated ones (e.g. a set type or another DS). The DS are defined prior to constructing data flow block diagrams and the design typically focus on simulation needs. Figure 72 shows a DSE window and the hierarchical organization of data structures in the BONEs framework. A DS at a particular node in the hierarchy tree inherits all the fields of its parent. This arrangement permits a processing block to accept a number of different DS.

C.4 Simulation Manager

Once a model has been completely defined, BONEs performs error consistency checks, translate the model into a C program, and stores the graphical representations and corresponding code in the BONEs library. The Simulation Manager (SM) supervises the execution of the simulation. Before simulation is performed, the SM can be used to specify parameter values such as traffic rates, transmission rates, processor rates, buffer sizes etc. Probes are attached at various network points where data is to be collected during simulation. The execution of the simulation can be specified on the local SPARCStation or a different workstation on the network. BONEs then executes an event driven simulation of the network model and stores the simulated data in the library.

C.5 Post Processor

The Post Processor (PP) analyzes and displays the simulated results. Statistical measures, such as averages and confidence intervals, or performance measures, such as throughput and delay, can be computed and displayed. The library managers group together multiple simulations of the same model so these results can be plotted as a function of the simulation parameters.

C.6 BONEs Library

BONEs has a set of lower level models called primitives as well as a number of higher level models that were built using the primitives. These system supplied libraries include models for traffic generation, protocol functions queues, timers, and execution

DSE Window

Inherited Fields

Dimensioned Basic Statistics			
File	Edit	View	Select
Add	Help		
Name	Type	Subrange	Default Value
Interval Mean	REAL	(-Infinity, +Infinity)	0.0
Interval Variance	REAL	(-Infinity, +Infinity)	0.0
Minimum	REAL	(-Infinity, +Infinity)	..
Maximum	REAL	(-Infinity, +Infinity)	..
Num Samples	INTEGER	[0, +Infinity)	0
Dimension	INTEGER	(-Infinity, +Infinity)	-1

DSE (Composite)

Hierarchical Organization

Figure 72 Data Structure Editor

control. New models can be constructed using the building blocks from the various libraries and the BDE and added to the BONEs model library. Primitive models can also be written in C programs and incorporated into the BONEs library.

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