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sis of the Wireless Communication Systems and Traffic Modeling

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Cemal Kavalcıoğlu: Analysis of the Wireless Communication Systems and Traffic Modeling

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I commenced higher education with Near East University in 1995 and after my graduation; I was employed as an assistant at same University for Programmable Logic Controllers and Power Electronics Laboratories at the Engineering Faculty in 2000. While I was employed with NEU, I took up M.Sc. Degree in "Analysis of the Wireless Communication Systems and Traffic Modeling". Also I wish to thank all engineering staff, friends and colleagues, for their valuable contributes, towards my studies.

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Dedicated to My Father and My Mother

CNIP	Calling Number Identification Presentation
CNIR	Calling Number Identification Restriction
CRC	Cyclic Redundancy Check
CCIR	International Consultative Committee on Radio
DCC	Digital Color Code
DCCH	Dedicated Control Channel
DECT	Digital European Cordless Telecommunications
DMH	Data Message Handler
DQPSK	Differential Quadrature Phase Shift Keying
EIA	Electronic Industry Association
EIR	Equipment Identity Register
EIRP	Effective Isotropic Radiated Power
ESN	Electronic Serial Number
E-TDMA	Extended Time-Division Multiple Access
ETSI	European Telecommunications Standards Institute
FACCH	Fast Associated Control Channel
FC	Fast Channel
FCC	Federal Communications Commission
FCCH	Frequency Correction Channel
FDD	Frequency-Division Duplex
FDMA	Frequency-Division Multiple Access
FE	Functional Element
FM	Frequency Modulation
FSK	Frequency Shift Keying
FPLMTS	Future Public Land Mobile Telecommunications System
GHz	gigahertz
GMSK	Gaussian Minimum Shift Keying
GSM	Global System for Mobile communications
HAAT	Height Above Average Terrain
HLR	Home Location Register
IAM	Initial Address Message
ID	Identification
IEEE	Institute of Electrical and Electronic Engineers
IMSI	International Mobile Station Identification

Improved Mobile Telephone Service

Interim Standard

Integrated-Service Digital Network

Interworking Function

Japanese Digital Cellular

kilohertz

Location Area Identity

Local Area Network

Linear Predictive Coding

Mobile Application Part

Multiple Carriers

Million chips per second

Megahertz

Multilevel Precedence and Preemption

Mobile Station

Mobile Switching Center

Mobile Termination

S Narrowband-Advanced Mobile Phone Service

Network Switching Subsystem

P Operation, Administration, Maintenance, and Provisioning

Operation Maintenance Center

Operation and Maintenance Subsystem

Operations Systems

Operational Subsystem

Personal Access Communications System

Packet Assembler/Disassembler

Private Branch Exchange

Power Control Channel

Paging Channel

Pulse Code Modulation

Power Control Pulse

Personal Communications Services

Personal Communications Switching Center

Personal Identification Network

LMN	Public Land Mobile Network
IC	Personal Mobility Controller
ID	Personal Mobility Data store
	Pseudonoise
S	Personal Station
SC	PCS Switching Center
SPDN	Public Switched Packet Data Network
STN	Public Switching Telephone Network
AM	Quadrature Amplitude Modulation
PSK	Quadrature Phase Shift Keying
RACF	Radio Access Control Function
RACH	Random Access Channel
RASC	Radio Access System Controller
RCF	Radio Control Function
RELP	Residual Excited Linear Prediction
RES	Radio Equipment System
RF	radio frequency
RP	Radio Port
RPE-LTP	Regular Pulse Excited-Long-Term Predictive
RPI	Radio Port Intermediary
RPT	Radio Personal Terminal
RS	Radio System
SACCH	Slow Associated Control Channel
SAT	Supervisory Audio Tone
SC	Slow Channel
SCH	Synchronization Channel
SDCCH	Stand-alone Dedicated Contol Channel
S/I	Signal-to-Interference ratio
SIM	Subscriber Identity Module
SMR	Specialized Mobile Radio
SRF	Specialized Resource Function
SSD	Shared Secret Data
SSF	Service Switching Function
TACS	Total Access Communications System

TCH	Traffic Channel
TCH/F	Traffic Channel/Full rate
TCH/H	Traffic Channel/Half rate
TDMA	Time-Division Multiple Access
TDD	Time-Division Duplex
TE	Terminal Equipment
TIA	Telecommunications Industry Association
TMC	Terminal Mobility Controller
TMD	Terminal Mobility Data store
TMSI	Temporary Mobile Station Identification
UMTS	Universal Mobile Telecommunications System
UPCH	User Packet Channel
UPT	Universal Personal Telecommunications
USC	User Specific Channel
VLR	Visitor Location Register

ABSTRACT

Wireless Communication was first developed to enable communication of ships on the sea. Wireless Communications technology has evolved along a logical path, from simple first-generation analog products designed for business use to second-generation digital wireless telecommunications systems for residential and business environments. The International Telecommunications Union (ITU) started work a decade ago to create a "framework" for new systems in which true global coverage could be achieved. This scheme is now known as the International Mobile Telecommunications 2000 (IMT-2000). According to the International Telecommunication Union (ITU) International Mobile Telecommunications 2000 initiative ("IMT-2000") third generation mobile ("3G") system services are scheduled to be initiated around the year 2000, subject to market considerations.

This thesis presents "Analysis of the Wireless Communication systems and Traffic Modeling".

Personal Communications Systems (PCS) is a name given to wireless systems that are starting to operate in the 1800 MHz frequency band. Initially the concept was that these systems would be very different than cellular; better, cheaper, simpler.

Global System for Mobile (GSM) is one of the most advanced digital cellular communication systems in the world today. GSM is an integrated system that can support voice communication (mobile telephony), data and Short Message Services.

Personal Digital Cellular (PDC) is the standard digital cellular system in Japan and was built as a high capacity system to replace the Japanese analog cellular system (J-TACS). One of the main differences between this and other digital systems is the use of a frequencies in the 1500 MHz band thus this system requires special radio hardware. PDC was also originally named as a JDC (Japan Digital Cellular).

When designing of any Wireless System as a case study the planning and engineering of a cellular radio system, including engineering philosophy, engineering considerations, quality of service criteria, and types of analyses must be taken into consideration at each design.

TABLE OF CONTENTS

HISTORICAL INVENTIONS AND EVENTS IN

COMMUNICATION SYSTEMS&WIRELESS COMMUNICATION

- **1799** Invention of Telegraph by Samuel Morse.
- **1844** First active telegraph lines between Washington and Baltimore.
- **1858** The first transatlantic cable between US and Europe.
- **1870's** Invention of telephone by Alexander Graham.

Introduction of wireless communication.

- Wireless communication was first developed to enable communication of ships on the sea.
- **1906** Human voice was transferred succesfully over radio for the first time.
- **1915** The invention of mobile radios.
- **1921** In the Detroit police department the first Vehicular mobile radio was used(One way communication system).
- **1930s** First half duplex mobile communication systems were introduced in U.S.
- **1935** Invention of FM (Frequency Modulation).
- **1969** Nordic countries made an attempt to standardise the Telecommunication aspects of that countries.
- **1973** That group (NMT group) specifies a feature allowing mobile telephones to be located within different networks. This is the beginning of the roaming concept.
- **1979** The installation and testing of first cellular systems were authorised by FCC.
- **1981** The installation of first cellular systems in the world which was using an analog system called NMT (North Mobile Telephony).

Introduction of GSM

- **1981** A group of specialists was formed to determine a series of standards for Mobile communications by Conference of European Posts and Telecommunications (CEPT). This group was called Groupe Speciale Mobile.
- The primitive aims of this comitee was as follows:
  - * Spectrum efficiency
  - * International roaming
  - * Low mobile and base stations costs
  - * Good subjective voice quality
  - * Compatibility with other systems such as ISDN (Integrated Services Digital Network)
  - * Ability to support new services
- **1989** The responsibility of GSM was passed from CEPT to ETSI(**European Telecommunications Standards Institute**)

INTRODUCTION

Over the last decade, deployment of wireless communications in North America and Europe has been phenomenal, the first mobile telephone service was introduced in the United States in 1946 by AT&T in St. Louis. It was used to interconnect mobile users (usually in automobiles) to the public telephone land-line network, thus allowing telephone calls between fixed stations and mobile user.

For the North American PCS and Cellular market, five digital air interfaces and one analog air interface supporting a common MAP (Mobile Application Part) have been defined by the standards bodies TR-46 and T1P1.

The first cellular radio system in Europe was installed in Scandinavia in 1981 and it served initially only a few thousand subscribers.

The aims of this M.Sc.Thesis are:

- To give explanations about overview of Wireless Technologies.
- To give general informations about European, North American Cellular and Japanese Cellular also PCS (PCS 1900) in their own systems.
- To give informations about Air interface unique capabilities.
- To give explanations about System design and application of Traffic Modeling.

This M.Sc.Thesis consists of the introduction, five chapters, and conclusion.

Chapter-1 presents the historical background of Wireless Communications starting from 1946 and examines the evolution of Wireless technologies in the United States and Europe. The brief and important explanations about 3G Mobile Communication and Wireless systems are also given in this chapter.

Chapter-2 presents the analog and digital systems used in the United States, typical call flows for origination, termination, handoff and so on is provided. This chapter also addresses PCS 1900 derived from the GSM for PCS applications in the North America.

Chapter-3 presents how the analog cellular and each of five digital air interfaces (the U_m-Interface in the reference model) support the functions of basic encoding of the digital data being sent. The various air interfaces that are standarized for PCS in the United States is also discussed with the operation of these air interfaces and the services they support.

Chapter-4 presents an overview of the Global System for Mobile Communications (GSM) system including architecture, channel and frame structure, speech processing and typical call flow scenarios as described in the European Telecommunication Standard Institute's (ETSI's) Recommendations. This chapter will be concluded with a brief description of Japanese Digital Cellular (JDC) system.

Chapter-5 presents the planning and engineering of a radio system. the process of growing a wireless system by considering a growth scenario with a frequency reuse factor of 7 is illustrated, also traffic models of a wireless serving area for both Cellular and PCS systems operating in a large Metropolitan area is presented.

1. AN OVERVIEW OF WIRELESS TECHNOLOGIES

OVERVIEW

the last decade, deployment of wireless communications in North America and ope has been phenomenal. Wireless communications technology has evolved along gical path, from simple first-generation analog products designed for business use to ond-generation digital wireless telecommunications systems for residential and siness environments.

As the industry plans and implements the second-generation digital networks in mid-1990s, a vision of a next-generation wireless information network is emerging. mplete Personal Communications Services (PCS) will enable all users to onomically transfer any form of information between any desired locations. The new twork will be built on and interface with the separate first- and second-generation rdless and cellular services and will also encompass other means of wireline and ireless access such as Local Area Networks(LANs) and Specialized Mobile Radio SMR). For at least part of the 1990s, we will see some systems that are cellular at new equencies and some with new services. By the end of the decade, true third-generation systems offering high bandwidth multimedia applications may emerge [1].

1.2 HISTORICAL BACKGROUND

The progression of the U.S cellular market has been heavily influenced by previous generations. The North America market can be classified in three areas: Pioneer, pre-cellular, and cellular. The pioneer area (1921 to 1947) began in Detroit where one-way broadcasts were made to mobile police cars. In the 1930's two-way radios were in at least 194 municipal police radio systems and 58 state police stations. These early mobile radio systems used amplitude modulation (AM) and their capacity quickly became saturated as the number of users grew. This spawned the first spectrum crisis for the Federal Communication Commission (FCC). World War II accelerated improvements in mobile communications and built the foundation for the commercial mobile communications market.

The pre-cellular area (1946 to 1968) experienced the first rapid growth which would be characteristic of the wireless market. The first mobile telephone service was introduced in the United States in 1946 by AT&T in St. Louis [2]. It was used to interconnect mobile users (usually in automobiles) to the Public Telephone Land-Line Network, thus allowing telephone calls between fixed stations and mobile users. A single powerful transmitter radieted frequency modulated (FM) waves in a coverage area of up to 50 miles. In 1949, the FCC officially recognized mobile radio as a new class of service and as demand grew, it soon became evident that the handful of channels that the systems provided would not be enough.Within a year, mobile telephone service was offered in more than 25 American cities. These mobile telephone systems were based on Frequency Modulation (FM) transmission. Most of these systems used a single powerful transmitter to provide coverage of up to 50 miles or more from the base. The FM mobile telephone channels used 120 kilohertz (kHz) of spectrum to transmit a voice with an effective bandwidth of only about 3 kHz. While these systems were technically advanced for their day, modern improvements in transmitter stability, receiver noise figure, and receiver bandwidth have shown how inefficient these systems were.

Demand for mobile telephone service grew quickly and stayed ahead of the available capacity in many of the large urban cities. Offered traffic was more than the effective traffic capacity of the system. Loading factors of 50, 100 or more subscribers per radio channel were common. Service quality was terrible; Blocking probabilities were as high as 65% or more. The usefulness of the mobile telephone decreased as users found that blocking often prevented them from getting a circuit during the peak periods. Users and the telephone companies, alike, realized that a handful of channels would not be enough for a true mobile telephone service to develop. Large blocks of spectrum would be needed to satisfy the demand in the urban areas.

In 1950, the FCC split the 120 kHz channel into two equal 60 kHz channels double capacity. Initially FM receiver technology was unable to utilize this increased capacity, but by 1960 advances enabled the spectrum efficiency of these systems to increase four fold. In the mid-1960s the Bell System introduced the Improved Mobile Telephone Service (IMTS) with enhanced features, including automatic trunking, direct dialing, and full-duplex service. Improvements in transmitter and receiver design enabled a reduction in the FM channel bandwidth to 25-30 kHz. IMTS showed the fundamental viability of narrowband FM channel, automatic interconnection, trunking, and other key modern features.

The result was a growth from 86,000 users in 1948, to approximately 650,000 users by 1958, and 1.4 million users by 1962. The cellular area took root with the

...ation of concepts that appeared in system proposals from Bell Labs. In these ...sals the broadcast model, where one or a synchronized few high-powered ...ers cover the area of interest, was repleced by a distribute models, where the ... area is partitioned into multiple cells each serviced by its own lower power ...ter. In the late 1960s and the early 1970s work began on the first cellular ...e systems. The term "cellular" refers to dividing the service area into many ... regions (cells) each served by a low-power transmitter with moderate antenna ... Frequencies are not reused in adjacent cells to avoid interference.

It should be recognized that the first-generation analog cellular radio was not so ... a new technology as it was a new idea for organizing existing IMTS technology ... a large scale. While the voice communications used the same analog FM that had ... used since the end of World War II, two major technological improvements made ... cellular concept a reality. In the early 1970s the microprocessor was invented; While ... complex control algorithms could have been implemented in wired logic, the ...croprocessor made the concepts easier to implement. It also allowed more complex ...rol algorithms to be implemented. The second improvement was in the use of a ...gital control link between the mobile telephone and the base station (or cell site as it ...s later called). In IMTS, the base station transmitted an idle tone to inform a mobile ...ephone that the channel was idle. The mobile telephone in turn transmitted dial pulses ...ing tones and transmitted its own identification (ID) as a four digit number. This use ...f limited information between base station and mobile telephone severely limited the ...rvices available to users.

In the cellular service, channels were assigned to transmit data between the base ...ation and the mobile telephone and from the mobile telephone to the base station using ...e digital data transmission. Now the mobile telephone could function more like a ...ireline phone, and true telephone service could be offered to people in automobiles ...d walking on city streets.

In 1971, Bell Labs submitted a proposal to the FCC for a new analog cellular FM telecommunications system. What evolved from this proposal was the Advanced Mobile Phone Service (AMPS) cellular standard.

In the late 1980s interest emerged in a digital cellular system, where both the voice and the control were digital. The use of digital technology for reproduction of music with compact disks popularized the quality of digital audio. The idea of eliminating noise and providing clean speech to the limits of each serving area were attractive to engineers and lay people alike. By 1991 digital cellular service began to emerge to reduce the cost of wireless communications and improve the call-handling capacity of an analog cellular system [3]. In 1993, a digital system was placed in service in some parts of the United States.

1.3 STANDARDS

In the United States, a decision by the Federal Communications Commission (FCC) to require one nationwide standard, and thus support roaming between any systems in the country, resulted in the rapid deployment of cellular systems. In Europe, several nations built their own systems that were incompatible with systems in other European nations. Similarly, Japan built its own cellular system. Thus, throughout the world, there are at least five different, incompatible, first-generation cellular standards. Each of these standards depends on frequency modulation of analog signals for speech transmission and in-band signaling to send control information between a Mobile Station (MS) and the rest of the network during a call.

Since there was already one nationwide standard in the United States, with roaming, the push for second-generation systems was not as strong in the United States as it was in Europe and Japan. Europe and Japan developed their second-generation digital mobile communications system by using new dedicated frequency bands, whereas the North American standards specify band sharing with the already successful analog cellular systems.

Current digital cellular systems use one of three standards:

1. Western Europe-Global System for Mobile Communications(GSM)
2. North American Electronic Industry Association (EIA) Standards IS-54
3. Japanese Digital Cellular (JDC).

The North American standards permit coexistence with the first-generation standard, the Advanced Mobile Phone System (AMPS), and add a digital voice transmission capability for new digital equipment. In one implementation, channels in a certain geographic area may be assigned to either a digital or analog signal, whereas in another area the two signals may share channels. Thus, the North American Standard IS-54 enhances, rather than replaces, the analog cellular technology [4].

Because of this history, North America and Europe appear to be moving in opposite directions. Europe is migrating from a set of incompatible analog systems to a single digital system. North America is deploying various dual-mode technologies that share a common analog standard, but with two radically different digital technologies. There is also a narrowband version of the analog standard (N-AMPS) and of the digital Time-Division Multiple Access (TDMA) standard Extended Time-Division Multiple Access (E-TDMA). Further compounding the problem is that the FCC has not mandated a single nationwide standard for PCS as they did for analog cellular and high definition television. They are letting the marketplace decide. U.S. PCS may have six or more incompatible digital standards in widespread use.

European research for digital cellular telephony began in the United Kingdom and Sweden in the early 1980s. In 1985, Conference Europeenne des Postes et Telecommunications (CEPT) started standardization of second-generation cellular telephones. Digital cellular standards were first published in the United Kingdom in 1987. These national standards specified parameters associated with operating frequency, transmitter power and spectrum, and interworking with the Public Switching Telephone Network (PSTN), but left the issue of radio interfaces open to manufacturers. There were no common air interface specifications. Debate continued over the relative merits of different technical solutions, with Sweden pushing for a Time-Division Multiple Access/Time-Division Duplex (TDMA/TDD)solution in contrast to the United Kingdom's Frequency-Division Multiple Access/Time-Division Duplex(FDMA/TDD). In January 1988, CEPT decided on the new European standards based upon a Time-Division Multiple Access/Time-Division Duplex/Multiple Carrier (TDMA/TDD/MC) approach operating just below 2 GHz frequency, subsequently known as the Digital European Cordless Telecommunications (DECT) standards. DECT was intended to support data as well as voice communications.In 1988, with the formation of the European Telecommunications Standards Institute (ETSI), responsibility for DECT standardization was moved to the ETSI Radio Equipment & System (RES) 3 subcommittee. In August 1992, DECT became a European Telecommunications Standard, ETSI 300-175. DECT has a guaranteed pan-European frequency allocation supported by the ETSI members and enforced by European Commission Directive 91/287. Figures 1.1 and 1.2 show the evolution of wireless technologies in the United States and Europe.

The success of first-generation analog mobile communications systems and the enormous investment in developing second-generation digital technologies have testified to the world that PCS is in great demand. Wireless communications continue to perience rapid growth, and new applications and approaches have spawned at an precedented rate. The number of subscribers for cellular mobile telephones, cellular rdless phones, and radio pagers have increased manifold. Market studies continue to ow large mass markets for new types of PCS, perhaps exceeding 50 to 100 million bscribers in the United States alone.

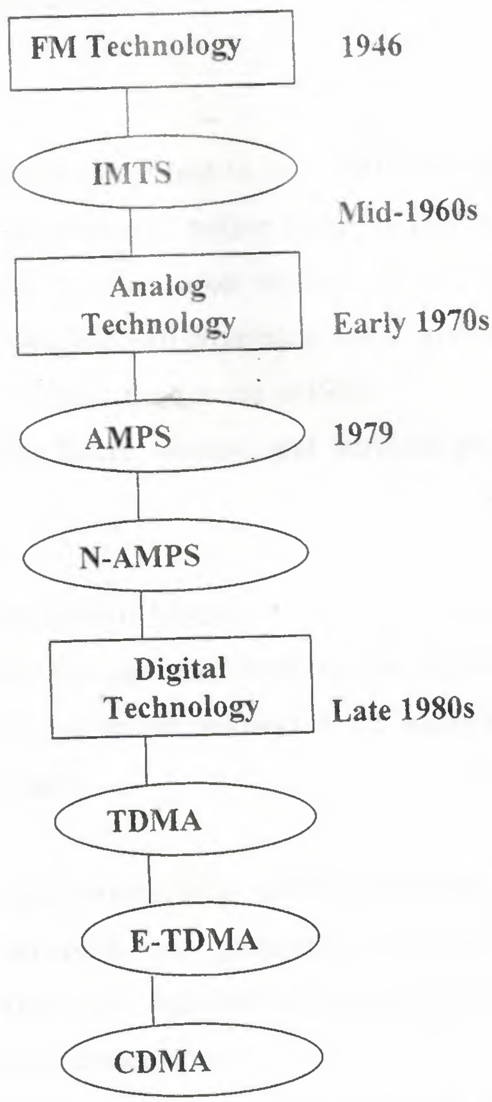

Figure 1.1 Evolution of Wireless Technologies in the United States

Within the International Consultative Committee on Radio (CCIR), many international studies on the subject of future mobile services, radio interface design, and network architectures of PCS have been conducted under the banner of Future Public Land Mobile Telecommunications System (FPLMTS). The International Consultative Committee on Telephone and Telegraph (CCITT) is developing standards for wireline personal communications known as Universal Personal Telecommunication (UPT) communications. Also, ETSI has created a special mobile group to prepare standards for a Universal Mobile Telecommunication System (UMTS). In th U.S. standards arena, several bodies are engaged in developing standards for PCS, e.g., subcommittees T1E1, T1M1, T1S1, and T1P1 of committee T1 of the Alliance for Telecommunications Industry Solutions (ATIS), committee TR46 of the Telecommunications Industry Association (TIA), and committee 802 of IEEE.

1.4 VISION OF PCS

PCS is likely to explode worldwide in the mid to late 1990s with global revenues for services and handsets ranging from about 2 billion Dolar in 1996 to about 12 billion Dolar by the end of the decade. As the market expands, it will draw in a growing number of low-end residential users and will drive basic voice services up from half the PCS market in the mid-1990s to about three-fourths in 1999.

PCS will use both existing and future wireline and wireless networks. Three key elements of PCS will be:

1. *An easy-to-use, high-functionality handset.*
2. A single, personal number that can reach the subscriber anywhere.
3. An individualized feature profile that follows the user and provides a customized set of services at any location.

S will not be a single, all-encompassing wireless solution, but it will be a bination of standards, networks, and products that meet a range of user uirements at a reasonable price with a high level of support. PCS will fill gaps left by er modes of wireless and wireline telephony.

In the long run, microcellular and macrocellular services are likely to merge. s will allow seamless handoff and roaming between network types. Major data-nted players like IBM, Apple, and DEC may also enter the market to integrate their products with voice products.

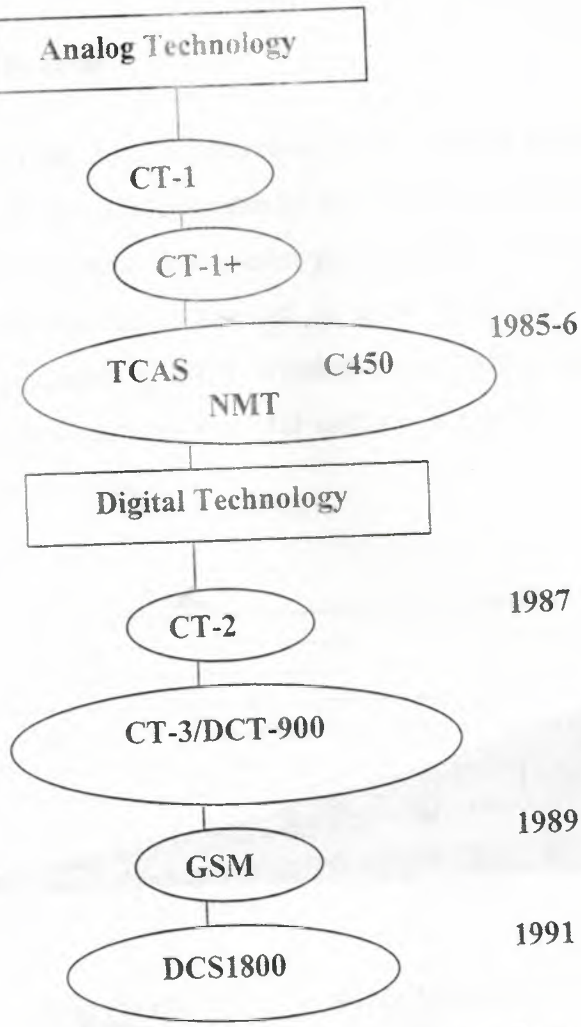

Figure 1.2 Evolution of Wireless Technologies in Europe

Vendors and carriers will increasingly market their offerings through mass merchants, electronics stores, and general merchandising outlets. Different price levels will be established for different end-user segments with business users along with emergency service, health care, and public protection personnel targeted for feature-rich portable voice and/or data units easily integrated with other types of equipment.

1.5.Third Generation Mobile Communication Systems

1.5.1.Introduction to 3G Systems

The phenomenal growth in the mobile communications industry has been one of the success stories of the last decade. Growth rates of up to 165% per annum in developing countries, and almost 5 million new mobile users per month have meant that the global market for mobile communications is forecast to grow from today's figure of 200 million to around 2.4 billion users by 2015. Wireless access will overtake fixed access to global telecommunications early in the 21st century. Figure 1.3 demonstrates the expected growth into the next century.

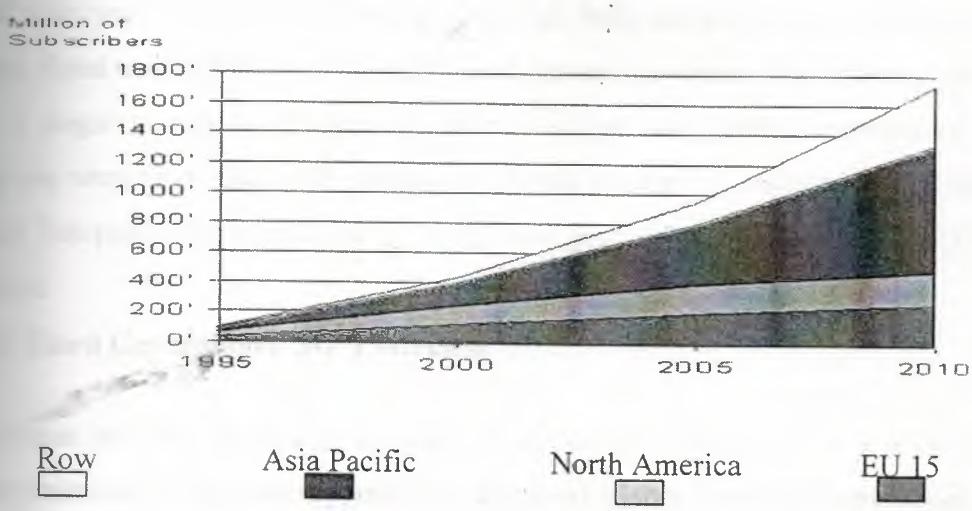

Figure 1.3 World Mobile Subscribers (Reproduced from UMTS- The next generation of mobile radio, IEE Review, March 1999, pp. 59-63.) [6]

It is clear that mobile communications is set to play an integral part in future society. Furthermore, with the trend towards globalization of the world's economy it is desirable that communications should be able to take place globally providing 'communications anywhere-anytime.' This is not possible with today's mobile industry. The communications scene of today is an international bazaar, with a wide variety of equipment on offer to provide for difference types of networks. These include first generation analogue cellular, second generation digital cellular, analogue and digital cordless, satellite networks for land, sea and air mobiles, as well as fixed networks. Despite the fact that some of the more recent systems work on similar principles most of them are incompatible with each other. Another drawback of existing mobile systems is that they are not capable of supporting the high bandwidth applications, which characterize the kind of systems users, will demand as we move into the next century. Today's cellular phones generally operate at a maximum speed of just 9600 bits/sec - which is too slow to support applications like high quality audio and video, real time video conferencing or high-speed Internet connection. It is clear that a new generation of wireless services are required which can address the above problems. This new generation of wireless services is described as third generation mobile systems (TGMS), also referred to as 3G systems. The International Telecommunications Union (ITU) started work a decade ago to create a 'framework' for new systems in which true global coverage could be achieved. This scheme is now known as the International Mobile Telecommunications 2000 (IMT-2000). It's purpose is to provide wireless access to the global telecommunication infrastructure through both satellite and terrestrial systems, serving fixed and mobile users in public and private networks. The ultimate goal is to have a single system for residential, office, cellular and satellite environments. The European version of the third generation mobile systems is known as the Universal Mobile Telephone System (UMTS). UMTS will play a core role within the IMT-2000 structure.

1.5.2 Third Generation ("3G") Wireless

3G systems will provide access, by means of one or more radio links, to a wide range of telecommunication services supported by the fixed telecommunication networks and to other services that are specific to mobile users. A range of mobile terminal types will be encompassed, linking to terrestrial and/or satellite-based networks, and the terminals may be designed for mobile or fixed use. Key features of 3G systems are a high degree of commonality of design worldwide, compatibility of services, use of small pocket terminals with worldwide roaming capability, Internet and other multimedia applications, and a wide range of services and terminals. According to the International Telecommunication Union (ITU) International Mobile Telecommunications 2000 initiative ("IMT-2000") third generation mobile ("3G") system services are scheduled to be initiated around the year 2000, subject to market considerations. The following Table describes some of the key service attributes and capabilities expected of 3G systems [5].

Table 1.1 3G System Capabilities

Capability to support circuit and packet data at high bit rates: • 144 kilobits/second or higher in high mobility (vehicular) traffic • 384 kilobits/second for pedestrian traffic • 2 Megabits/second or higher for indoor traffic
Interoperability and roaming
Common billing/user profiles: • Sharing of usage/rate information between service providers • Standardized call detail recording • Standardized user profiles
Capability to determine geographic position of mobiles and report it to both the network and the mobile terminal
Support of multimedia services/capabilities: • Fixed and variable rate bit traffic • Bandwidth on demand • Asymmetric data rates in the forward and reverse links • Multimedia mail store and forward • Broadband access up to 2 Megabits/second

On October 13. 2000, the President executed a memorandum that articulated the need to select radio frequency spectrum to satisfy the United States' future needs for mobile voice. high-speed data, and Internet-accessible wireless capability. The Presidential Memorandum established for the Executive Agencies guiding principles to be used in selecting spectrum that could be made available for 3G wireless systems. and strongly encouraged independent federal agencies to follow the same principles in any actions they take related to the development of 3G systems. Noting the joint spectrum management responsibilities of the Executive Branch and the Commission, the Presidential Memorandum directed the Secretary of Commerce to work cooperatively with the FCC to develop a plan to select spectrum for third generation wireless systems by October 20, 2000; and to issue by November 15, 2000 an interim report on the current spectrum uses and potential for reallocation or sharing of the bands identified at the 2000 World Radio communication Conference that could be used for 3G systems. These actions were taken to enable the Commission to identify spectrum for 3G systems by July 2001 and auction licenses by September 30, 2002. In accordance with the Presidential Memorandum, the Department of Commerce released a "Plan to Select Spectrum for Third Generation (3G) Wireless Systems in the United States" (Study Plan) on October 20, 2000. The Study Plan noted that although various frequency bands have been identified for possible 3G use, the Commission and the National Telecommunications and Information Administration (NTIA) needed to undertake studies of the 2500-2690 MHz and the 1755-1850 MHz frequency bands in order to provide a full understanding of all the spectrum options available. The Study Plan called for the Commission to complete an Interim Report on the 2500-2690 MHz band and for NTIA to complete an Interim Report on the 1755-1850 MHz band by November 15, 2000. In March 2001, the Commission issued a Final Report on the 2500-2690 MHZ band and NTIA issued a Final Report on the 1755-1850 MHz band. The NTIA Final Report also addressed the 1710-1755 MHz Federal Government band. Comments were received on these reports in April 2001. In July 2001, FCC Chairman Powell and Commerce Secretary Evans exchanged letters, in which they agreed to postpone the July 2001 deadline for the Commission to identify spectrum for 3G systems. Secretary Evans informed Chairman Powell that he has directed the Acting Administrator of NTIA to work with the FCC to develop a new plan for the selection of 3G spectrums, to be executed as quickly as possible. In September 2001, the Commission added a mobile allocation to the 2500-2690 MHz band to provide additional near-term and long-term flexibility for use of this spectrum, thereby making this band potentially available for advanced mobile and fixed terrestrial wireless services, including 3G and future generations of wireless systems. However, because the 2500-2690 MHz band is extensively used by incumbent Instructional Television Fixed Service and Multichannel Multipoint Distribution Services licensees, and in order to preserve the viability of the incumbent services, the Commission did not relocate the existing licensees or otherwise modify their licenses.

1.5 Key features of third generation systems

- **Flexibility:** 3G mobile systems should be able to incorporate a variety of systems, terrestrial and satellite based, to give true global 'seamless' access (see figure1.4). The provision of a small pocket worldwide terminal is also an important aim within the IMT-2000 and UMTS structure.

- **High speed data:** 3G mobile systems will support a variety of high-speed broadband services including audio, video, multimedia, internet, data and speech services in a variety of environments. To achieve this 3G systems will have to provide adequate performance for each user in each of the operating environments, shown in figure 1.4

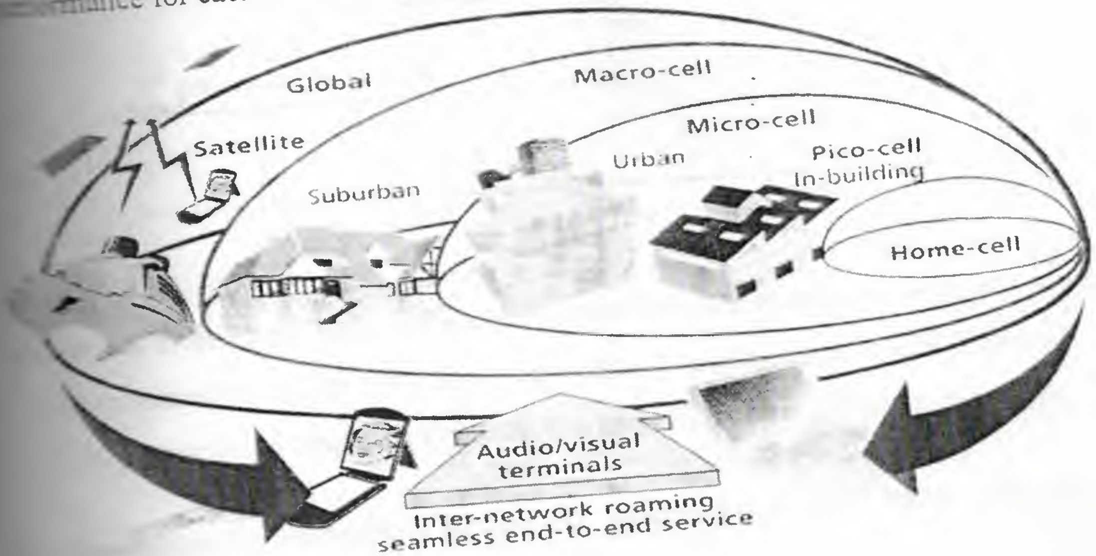

Figure 1.4 Illustration demonstrating how future mobile systems will provide a worldwide seamless end-to-end service for the consumer (Reproduced from Rupert Goodwins, 'UMTS proposes new technologies to improve Wireless Communications', PC Magazine, May 1999, pg50.) [7].

Future mobile systems architecture will be divided into a set of different coverage regions. Some of these regions will be privately operated, while others may be publicly operated. Pico-cells will typically cover building areas, with micro-cells covering dense urban areas. Within the UMTS model, rates of up to 2 Mbit/s are expected in these environments. Macro cells will provide wide-area coverage giving data rates of around 384 kbit/s. Finally, satellite segments will give terminal data rates starting from 144 kbits/s.

- **Compatibility:** 3G services have to be compatible with existing networks. The enormous investments made in developing the world's 2nd generation cellular networks over the last decade means there has to be a transitional phase for the implementation of 3G generation systems. Figure 1.5 demonstrates the expected transitional periods required for the switchover to 3G systems.

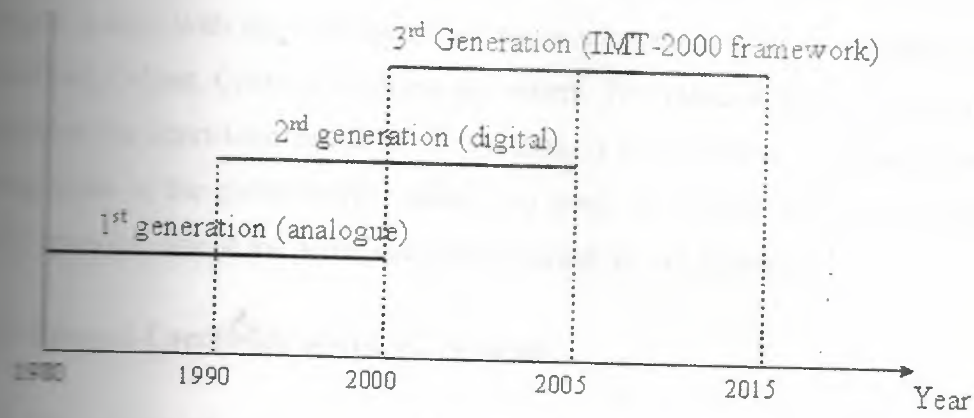

Figure 1.5 Transitional Periods Required for Different Generations of Mobile Systems (Reproduced from The Evolution of Personal Communications', IEEE Personal Communications Magazine (Volume 1, Number 2), Second Quarter 1994.) [8].

- **Affordable:** For commercial success 3G systems will have to be as affordable as today's mobile systems.

- **Spectrum Efficiency:** Efficient use of the radio spectrum consistent with providing service at an acceptable cost will have to be accomplished. IMT-2000 based systems have been assigned the frequency bands 1885 - 2025 MHz and 2110-2200 MHz.

- The different broadband service aspects and service environments for IMT-2000 systems calls for a new network architecture. The IMT-2000 based systems will be based upon common standardized flexible platforms to meet the demands of fixed and mobile markets around the world.

1.5.4 Influence of market forces upon third generation systems

The successful implementation of the previous described features by no means guarantees the success of third generation systems. Ultimately, the market forces will determine if the new IMT-2000 based systems are a success. It is therefore important to preserve a competitive environment for manufacturers in order to create incentive and stimulate innovation. One of the first countries to issue a license for the new 3G (UMTS) systems will be the United Kingdom, in the summer of 1999. Interest in these licenses is high with bids expected from major telecommunications companies such as Vodafone, Cellnet, Cable & Wireless and others. The value of these licenses may well determine the short-term future of 3G systems. A high auction price would provide a strong boost to the global market, whereas a weak or unsuccessful auction would cast doubt upon the size of the perceived global market for 3G systems.

1.5.5 General Conclusion about 3G Systems

With the advent of the global mobile boom and the globalization of the world's economy the long term prospects for the development of third generation mobile communication systems is secure. As we approach the new millennium 3G systems offer a way to provide not only global communications, but also global broadband services such as multimedia applications and high-speed Internet access. The success of third generation mobile systems will depend upon market forces. Third generation systems will not only have to deliver the promised services but also at a price attractive to the consumer. The success of the UMTS licenses to be awarded these days by the United Kingdom will provide an early indication as to the success of third generation mobile communication systems.

1.6. SUMMARY

This chapter presented the historical background of Wireless Communications starting from 1946 and examines the evolution of Wireless technologies in the United States and Europe. The brief and important explanations about 3G Mobile Communication and Wireless systems were also given in this chapter.

2. NORTH AMERICAN CELLULAR AND PCS SYSTEMS

2.1 OVERVIEW

The North American cellular systems have evolved to two digital standards; IS-136 (enhanced IS-54) using TDMA and IS-95 using CDMA. The cellular versions of the standards support both analog AMPS and the digital protocols. Thus, there are three types of phones in general use for cellular in North America: An analog-only AMPS phone, a dual-mode AMPS and TDMA phone, and a dual-mode AMPS and CDMA phone.

When work was started on PCS, each of the two groups that supported TDMA and CDMA made modifications to their standards to remove the analog capabilities and support the new PCS frequencies. The standards also support a dual frequency phone that can function on cellular or PCS frequencies.

Other groups did not want to use the existing CDMA and TDMA protocols and set about to define a new set of protocols and services for PCS. Oki, a Japanese company with an extensive market share throughout the world, has set about defining a wideband CDMA protocol that will be used in the United States and Asia [9].

Motorola, Bellcore, and others have worked extensively in defining a protocol for use by wireline companies that want to provide switching functions while letting radio companies provide the radio portion of the cell. This protocol is called PACS and uses TDMA. Its primary use is for low-mobility applications in residential wireless. Omnipoint has defined a combined CDMA/TDMA protocol, PCS2000, for use in some systems.

While PCS was initially thought of as a new service, clearly different companies have different interests in what PCS is and how they will provide services. Existing cellular companies are looking at PCS as a means to provide wide coverage areas and fill in gaps in their current service offerings. For these companies, PCS is an extension of cellular to the 1900-MHz band using identical standards for both bands. Other companies see PCS as the opportunity to offer new services that will compete with cellular by offering lower costs, additional services, and better quality of existing services. These companies do not want the existing protocols but want new ones or extensions of the old ones to support the new services.

This chapter discusses the original analog cellular air interface and five digital air interfaces and compares the characteristics of them. All of these air interfaces share a common heritage from two sources. The original analog cellular system was defined in [illegible] by the EIA. All of the digital air interfaces inherit their characteristics from the analog protocol. In addition, all of the air interfaces discussed in this chapter have services based on the IS-41 intersystem communications protocol. This is the protocol supported in North America for communications between wireless systems. IS-41 and IS-104 (for supplementary services) define the functionality for a Mobile Application Part (MAP) for U.S. based PCS. Since the GSM system supports a different MAP.

First this chapter examines the TR-46 and T1P1 reference models and compares them. We discuss both the elements of the models and the interfaces. Then we examine the basic services from the T1P1 service description and the supplementary services from IS-104. Call flows for basic call processing are discussed with operations traced from the personal station to the radio system to the PCS switch and other network elements. Since each of the six air interfaces accomplish the same purpose with different protocols, we examine each interface and describe the similarities and differences of the protocols.

Work on PCS has progressed in five different standards groups:

T1P1: This committee is under the ATIS and is responsible for PCS services requirements.

TR-46: This committee is under the TIA and is responsible for PCS services and protocols.

T1S1: This is an ATIS body responsible for the signaling protocols and is undertaking the role of the upper-layer signaling protocols between various elements of the PCS system.

T1M1: This ia an ATIS body responsible for operations, administration, maintenance, and provisioning (OAM&P) and is undertaking the role of the OAM&P services and protocols for PCS.

The Joint Technical Committee of T1P1 and TR-46: This committee is result of a cooperative effort between T1P1 and TR-46 and has the responsibilities of developing the air interface requirements for PCS.

Clearly with five standards bodies examining the problem, there is overlap between some of the bodies.

2.2 PCS REFERENCE MODELS

Key to the North American systems is the use of a common reference model. Both TR-46 and T1P1 have a reference model, but each model can be converted into the other one. All of the North American systems follow the reference model. The names of each of the network elements are similar, and some of the functionality is partitioned differently between the models. The main difference between the two reference models is how mobility is managed. Mobility is the capability for users to place and receive calls in systems other than their home system. In the T1P1 reference model, the user data and the terminal data are seoarate; Thus users can communicate with the network via different radio personal terminals. In the TR-46 reference model, only terminal mobility is supported. A user can place or receive calls at only one terminal(the one the network has identified as owned by the user). The T1P1 functionality is migrating toward independent terminal and user mobility, but all aspects of it are not currently supported. Although these models are for PCS, they apply equally well to cellular systems.

2.2.1 TR-46 Reference Model

The main elements of the TR-46 reference model (see Figure 2.1) are:

The Personal Station (PS): Terminates the radio path on the user side and enables the user to gain access to services from the network. The PS can be a stand-alone device or can have other devices (e.g., personal computers, fax machines) connected to it.

The Radio System (RS): Often called the base station, terminates the radio path and connects to the personal communications switching center. The RS is often segmented into the base transceiver system, and the base station controller:

a) The Base Transceiver System (BTS): Consists of one or more transceivers placed at a single location and terminates the radio path on the network side. The BTS may be colocated with a base station controller or may be independently located.

b)The Base Station Controller (BSC) : The control and management system one or more BTSs. The BSC exchanges messages with both the BTS and the personal communications switching center. Some signaling messages may pass through BSC transparently.

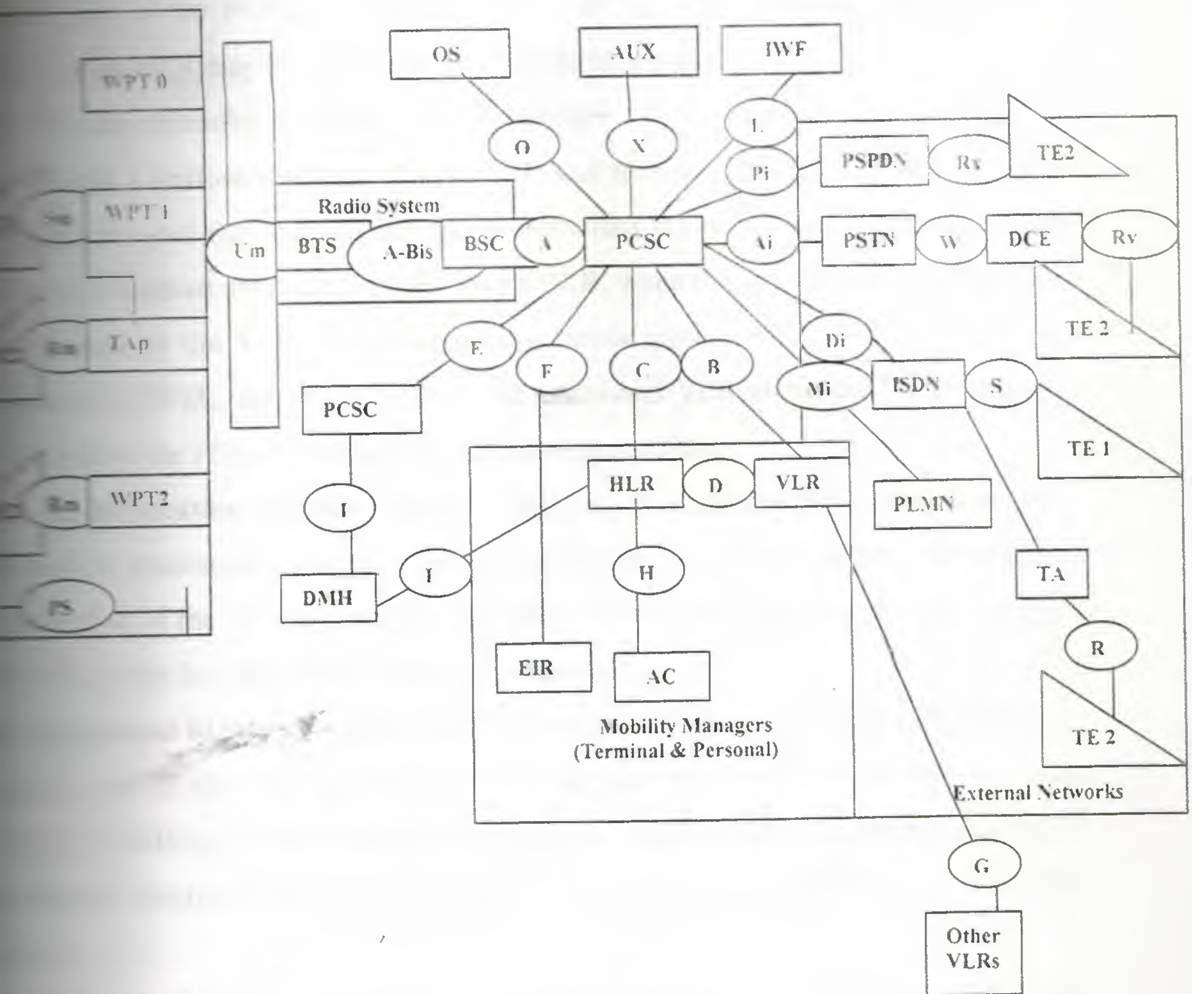

Figure 2.1 TR-46 Reference Model

The Personal Communications Switching Center (PCSC): An automatic system that interfaces the user traffic from the wireless network to the wireline network or other wireless networks.

ocation Register (HLR): The functional unit used for management of ibers by maintaining all subscriber information (e.g.. Electronic Serial). Directory Number (DN), International Mobile Subscriber Identity rofiles, current location). The HLR may be collocated with a PCSC, an of the PCSC, independent of the PCSC. One HLR can serve multiple HLR may be distributed over multiple locations.

Handler (DMH): Used for billing.

ocation Register (VLR): Is linked to one or more PCSCs. The VLR is unit that dynamically stores subscriber information (e.g., ESN, DN, user tion) obtained from the user's HLR, when the subscriber is located in the y the VLR. When a roaming mobile station enters a new service area CSC, the PCSC informs the associated VLR about the PS by querying he PS goes through a registration procedure.

ication Center (AC): Manages the authentication or encryption sociated with an individual subscriber. As of this writing, the details of f the AC have not been defined. The AC may be located within an HLR y be located independently of both.

t Identity Register (EIR): Provides information about the PS for record f this writing, the details of the operation of the EIR have not been R may be located within a PCSC or may be located independently of it.

stem (OS): Is responsible for overall management of the wireless

Function (IWF):Enables the PCSC to communicate with other

rnal networks are other communications networks; The Public Switched work (PSTN), the Integrated Services Digital Network (ISDN), the bile Network (PLMN), and the Public Switched Packet Data Network

The following interfaces are defined between the various elements of the system:

RS to PCSC (A-Interface): The interface between the RS and the PCSC supports signaling and traffic (both voice and data). A-Interface protocols have been defined using SS7, ISDN BRI/PRI, and frame relay.

The BTS to BSC Interface (A-bis): If the RS is segmented into a BTS and BSC, this internal interface is defined.

The PCSC to PSTN Interface (Ai): This interface is defined as an analog interface using either Dual Tone Multifrequency (DTMF) signaling or Multifrequency signaling.

PCSC to VLR (B-Interface): This interface is defined in the TIA IS-41 protocol specification.

PCSC to HLR (C-Interface): This interface is defined in the TIA IS-41 protocol specification.

HLR to VLR (D-Interface): This interface is the signaling interface between an HLR and a VLR and is based on SS7. It is currently defined in the TIA IS-41 protocol specification.

PCSC to ISDN (D_i-Interface): This is the digital interface to the public telephone network and is a T1 interface (24 channels of 64 kbs) and uses Q.931 signaling.

PCSC to PCSC (E-Interface): This interface is the traffic and signaling interface between wireless networks. It is currently defined in the TIA IS-41 protocol specification.

PCSC to EIR (F-Interface): Since the EIR is not yet defined, the protocol for this interface is not defined.

VLR to VLR (G-Interface): When communication is needed between VLRs, this interface is used. It is defined by TIA IS-41.

HLR to AC (H-Interface): The protocol for this interface is not defined.

DMH to PCSC (I-Interface)

PCSC to the IWF (L-Interface): This interface is defined by the IWF.

PCSC to PLMN (M_i Interface): This interface is to another wireless network.

PCSC to OS (O-Interface): This is the interface to the OS. It is currently being defined in ATSI standard body T1M1.

PCSC to PSPDN (P_i-Interface): This interface is defined by the packet network that is connected to the PCSC.

erminal Adapter (TA) to Terminal Equipment (TE) (R-Interface): These erfaces will be specific for each type of terminal that will be connected to a PS.

DN to TE (S-Interface): This interface is outside the scope of PCS and is defined thin the ISDN system.

S to PS (U_m-Interface): This is the air interface. Later sections in this chapter will scuss this interface in detail.

PSTN to Data Communication Equipment (DCE) (W-Interface): This interface is utside the scope of PCS and is defined within the PSTN system.

PCSC to Auxiliary (AUX). (X-Interface): This interface depends on the auxiliary equipment connected to the PCSC.

2.2.2 T1P1 PCS Reference Architecture

The T1P1 architecture (Figure 2.2) is similar to the TR-46 model but has some differences. The following elements are defined in the model [10]:

Radio Personal Terminal (RPT): Is identical to the PS of the TR-46 model.

Radio Port (RP): Is identical to the BTS of the TR-46 model.

Radio Port Intermediary (RPI): Provides an interface between one or more RPs and the radioport controller. The RPI allocates radio channels and may control handoffs. It is dependent on the air interface.

Radio Port Controller (RPC): Identical to the BSC of the TR-46 model.

Radio Access System Controller (RASC): Performs the radio-specific switching functions of call delivery and origination, handoff control, registration and authentication, and radio access management (control of signaling channels).

PCS Switching Center (PSC): Similar to the PCSC of the TR-46 reference model. Some of the functions of the PCSC in the TR-46 model are distributed into other elements.

Terminal Mobility Controller (TMC): Provides the control logic for terminal authentication, location management, alerting, and routing to RPTs. This function is supported by the VLR and the PCSC in the TR-46 reference model.

Terminal Mobility Data Store (TMD): Maintains the associated terminal data and is similar to the VLR in the TR-46 reference model.

Personal Mobility Controller (PMC): Provides the control logic for user authentication, location management, alerting, and routing to users. This function is supported by the HLR and the PCSC in the TR-46 reference model.

Personal Mobility Data Store (PMD): Maintains the associated user data and is similar to the HLR in the TR-46 reference model.

OAM&P Systems: Identical to the OS in the TR-46 reference model.

Auxiliary Services: A variety of services such as voice mail and paging that may be provided by the PSC.

Interworking Function (IWF): Identical to the IWF of the TR-46 reference model.

External Networks: Those networks (i.e., wired and wireless) that are not part of the described wireless network.

The following interfaces are described between the elements in the system:

RP to RPT (a-Interface): This interface is identical to the U_m-Interface of the TR-46 model.

PSC to RPC (c-Interface): This is similar to the A-Interface of the TR-46 model. If an RASC is used, then the c_{rpc}-,c_{rasc}-,and c_r-interfaces are defined.

TMC to other elements (d-Interface): The d-Interface is between the TMC and the RASC and between the PMC and the PSC. The d_d-Interface is between the TMC and the TMD and between the TMC and the PMC.

RPI to RPC (f-Interface): This interface is between the RP and the RPI; It may or may not be internal to a radio system.

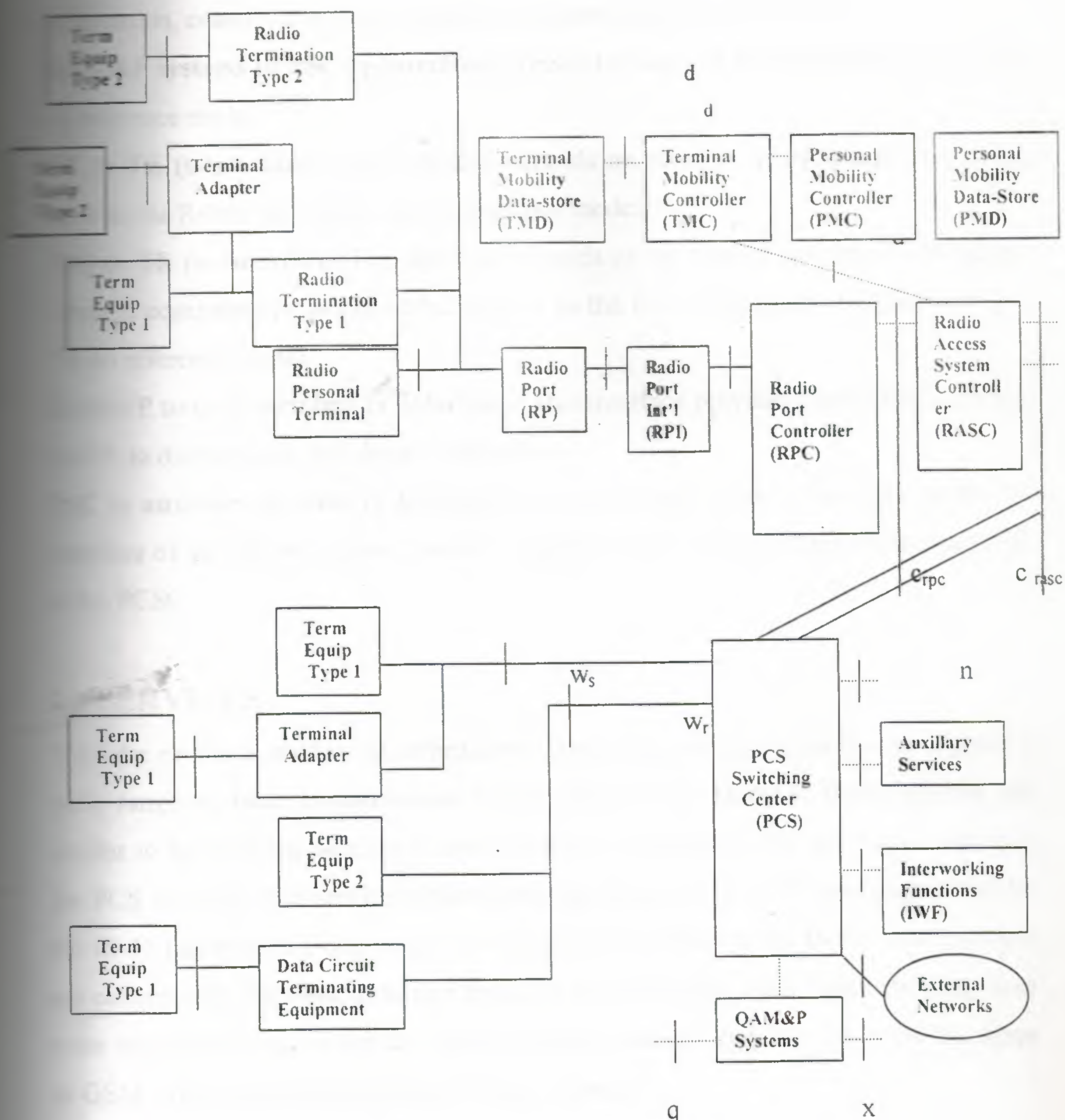

Fig 2.2 T1P1 Reference Model

PSC to interworking functions(l-Interface): This is the same as the L-Interface in the TR-46 reference model. As in the TR-46 model, this interface is defined by the IWF.

external networks (m-Interface):** This is the same as the TR-46 reference o external networks except that TR-46 segments this interface into type of (A_i, D_i, M_i and P_i).

other PSC (n-Interface): This interface is to other PSCs.

RP (p-Interface): This interface carries baseband bearer and control tion, contained in the air interface, between the RPI and the RP.

P systems to PSC (q-Interface): This is the same as the O-Interface in the TR-ence model.

TE (t-Interface): This interface depends on the type of equipment and is the the R-Interface of the TR-46 reference model.

TE (w-Interface): This interface depends on the type of equipment and allows l equipment to be connected directly to the PSC. There is no equivalent in the reference model.

P to craft terminal (x-Interface): This interface provides capabilities to access to display, edit, add/delete information.

auxiliary services (y-Interface): This interface, which is the same as the X-e of the TR-46 reference model, depends on the auxiliary equipment connected CSC.

RVICES

ne reference models described above, there are enough capabilities to support a ange of telecommunications services over PCS. Many of these services are to those of the wireline network; Some are specific to the untethered approach S provides. The services defined here are based on an MAP that is supported by 41 Intersystem Communications Protocol [11]. Most of the IS-41 based services nceptually the same as those offered by GSM based systems. Since these services riginally designed for the North American market, they may not work the same M. which was designed for the European market.

Basic Services

tandards body T1P1 is in the process of defining basic call functions and mentary services for PCS. The T1P1 Stage 2 Service description [12] defines 15 services (information flows) that can be grouped as follows:

- Registration and deregistration functions to support the process where a PS informs a PCS system of its desire to receive service and its approximate location:
    1. Automatic registration
    2. Terminal authentication and privacy (using private key cryptography)
    3. Terminal authentication and privacy (using public key cryptography)
    4. User authentication and validation
    5. Automatic personal registration
    6. Automatic personal deregistration
    7. Personal registration
    8. Personal deregistration
- The Registration and deregistration process requires that a PS identify itself to the PCS network and requires that the PCS network communicate with the home PCS network to obtain security and service *profile information.*

Roaming: This is the process where a PS registers and receives service in a PCS system other than its home system. Call establishment, call continuation, and call clearing procedures:

1. Call origination
2. Call delivery (call termination)
3. Call clearing
4. Emergency (E911) calls
5. Handoff

2.3.2 Supplementary Services

Supplementary services are defined in the IS-104 Personal Communications Service Descriptions for 1800 MHz (PN-3168) [13]. The IS-41 C specification defines those services that can be made available to users as they roam. Obviously, these services would also be available to users in their home system. Additional services may be available in a specific home PCS or cellular system, but users would not necessarily em available in other systems since no common set of procedures and protocols en defined to support other services. These services are:

atic Recall: Allows a wireless calling a busy number to be notified when the arty is free and have the PCS system re-call the number.

atic Reverse Charging (ARC): Allows a wireless subscriber to be charged for e special ARC number. This service is similar to wireline 800 service in North a.

old and Retreve: Allows a wireless subscriber to interrupt a call and return to l

orwarding-Default: Represents the ability to redirect a call to a PS handset in ituations: "Unconditional," "Busy," and "No Answer." The PS call forwarding F) features build upon the PS call terminating capability. Under all of these es, calls may be forwarded by the network to another PS or to a DN associated a wireline interface. There are no additional information flow for PS call nating.

Forwarding-Busy: Permits a called PCS subscriber to have the system send ning calls addressed to the called personal communications subscriber's personal er to another personal, terminal, or directory number when the PCS subscriber is ged in a call.With "personal call forwarding-busy" activated, a call incoming to the subscriber will be automatically forwarded to the forward-to number whenever the subscriber is already engaged in a prior call.

Forwarding-No Answer: Permits a called PCS subscriber to have the system send ncoming calls addressed to the called PCS subscriber's personal number to another onal, terminal, or directory number when the PCS subscriber fails to answer or sn't respond to paging. With "personal call forwarding-no answer" activated, a call ming to the PCS subscriber will be automatically forwarded to the designated vard-to number whenever the PCS subscriber does not answer within a specified od after transmission of the alert indication.

ll Forwarding-Unconditional: Permits a PCS user to send incoming calls addressed the PCS subscriber's personal number to another personal, terminal, or directory mber (forward-to number). The ability of the served PCS subscriber to originate calls unaffected. If this service is activated, calls are forwarded independent of the state of PS (busy, idle, and so on).

Call Transfer: Permits a PCS user to transfer a call to another number on or off the PCS switch. When a call is transferred, the PCS personal terminal is then available for other calls.

Call Waiting: Provides notification to a PCS subscriber of an incoming call while the user's personal station is in the busy state. Subsequently, the user can either answer or ignore the incoming call. With call waiting activated, the PCS user who is already engaged in conversation on a prior call will receive a notification signal of a new incoming call attempt. This may be repeated a short time later if the PCS user takes no action. The calling party will hear an audible ringing signal either until they abort the call attempt or until the PCS user acknowledges the waiting call. The PCS user may indicate acceptance of the waiting call by : 1)Placing the existing call on hold or 2) Releasing the existing call.

Calling Number Identification Presentation (CNIP): Is a supplementary service offered to a called party. It provides to the called part the number identification of the calling party. If the calling party has subscribed to Calling Number Identification Restriction (CNIR), the calling number will not be presented.

Calling Number Identification Restriction (CNIR): Is a supplementary service offered to a calling party that restricts presentation of that party's calling number identification to the called party. When the CNIR service is applicable and activated, the originating network provides the destination network with a notification that the calling number identification is not allowed to be presented to the called party. CNIR may be offered with several options. Subscription options applied are: Not subscribed (inactive for all calls); Permanently restricted (active for all calls); Temporary restricted (specified by user per call), default_ restricted; Temporary allowed (specified by user per call), default_ allowed.

Conference Calling: Is similar to three-way calling except more than three parties are involved in the call.

Do Not Disturb: Allows a wireless subscriber to direct that all incoming calls stop at the PCS switch and not page the PS.

Flexible Alerting: Allows a call to a directory number to be branched into multiple attempts to alert several subscribers. The subscribers may have wireless or wireline terminations.

Message Waiting Notification: Is the service where a message is sent to the PS to inform the user that there are messages stored in the network that the user can access.

Call Transfer: Permits a PCS user to transfer a call to another number on or off the PCS switch. When a call is transferred, the PCS personal terminal is then available for other calls.

Call Waiting: Provides notification to a PCS subscriber of an incoming call while the user's personal station is in the busy state. Subsequently, the user can either answer or ignore the incoming call. With call waiting activated, the PCS user who is already engaged in conversation on a prior call will receive a notification signal of a new incoming call attempt. This may be repeated a short time later if the PCS user takes no action. The calling party will hear an audible ringing signal either until they abort the call attempt or until the PCS user acknowledges the waiting call. The PCS user may indicate acceptance of the waiting call by : 1)Placing the existing call on hold or 2) Releasing the existing call.

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Message Waiting Notification: Is the service where a message is sent to the PS to inform the user that there are messages stored in the network that the user can access.

ss Hunting: Is the service where call delivery is presented to a series of umbers. If the first number is not available, the system will try the second e down a list. The terminating numbers can be mobile or nonmobile where in the world.

Precedence and Preemption (MLPP): Permits a group of wireless to have access to wireless service where higher-priority calls will be ead of lower-priority calls and may preempt (i.e.. force the termination of) calls. Only calls within the same group will override each other.

Call Acceptance: Is the service where calls to the wireless subscriber are and the calling party is asked to correctly enter a password before the PS is

Language: Is the capability for a user to hear all network announcements in ed language.

ccess and Channel Assignment: Allows the PCS service provider to abilities to a subscriber that allows priority access to radio resources. This mits emergency services personnel (e.g., police, fire, and rescue squads) ess to the system Multiple levels of access may be defined.

eature Call: Permits a wireless subscriber to call a special directory number vireless or wireline phone) and, after correctly entering account code n and a Personal Identification Number (PIN), change the operation of one or res of the service. For example, the selective call list can be modified by this

call Acceptance: Is the service where a wireless subscriber can form a list of ctory numbers that will result in the PS being paged. All other directory vill be blocked.

er PIN Access: Is the ability to block access to the PS until the correct PIN is to the PS.

er PIN Intercept: Is the ability of a wireless subscriber to bar outgoing calls e correct PIN is entered. This feature can be implemented in the network or in

Vay Calling: Permits a PCS user authorized for three-way calling to add a third an established two-way call regardless of which party originated the established add a third party, the PCS user sends a request for three-way calling service to ce provider, which puts the first party on hold. The PCS user then proceeds to establish a call to the third party. A request by the controlling user for disconnection of the third party (i.e., the last added party) will release that party and will cause the three-way connection to be disconnected and return the call to its original two-way state.If either of the noncontrolling parties to an established three-way call disconnects, the remaining two parties are connected as a normal two-way call. If the controlling PCS user disconnects, all connections are released.

Voice Message Retrieval: Is service where the user can retrieve voice message stored in the network. These messages are typically left by parties calling the user while the user was busy, did not answer, or was not registered with a system.

Voice Privacy: Is the service where the user's voice traffic over the radio link is encrypted to prevent casual eaves dropping. In PCS in the United States, this is a required feature and is not optional.

Short Message Service: Permits alpha and alphanumeric short messages to be sent to or from a PS.

2.4 OPERATION OF A PCS SYSTEM

This section describes the operation of a PCS system. We trace call flows from a PS to an RS to the PCS switch to other network elements. The flows are based on the TR-46 reference model and an A-Interface based on the ISDN. The ISDN model assumes that there are ISDN terminals associated with the switch, one for each directory number on the switch. Since PCS allows PSs to be associated with any RS, there is not a one-for-one correspondence between PSs and ISDN terminals. Thus, with the ISDN model, each PS registered at an RS is assigned a temporary directory number (also called virtual terminal number or interface directory number) that the RS and PCS switch use to refer to the PS in the ISDN signaling messages, while that PS is registered at the RS. The ISDN A-Interface defines a PCS Application Protocol (PCSAP) that uses ISDN signaling. With basic ISDN and PCSAP, the PCSC can support terminal mobility. The RS and switch can interact in either of two methods. In method one, the RS is equivalent to a Private Branch Exchange (PBX) and the switch is an ISDN switch that does minimum call control. In method two, the RS is a virtual ISDN terminal with all call control in the ISDN switch.

2.4.1 General Operation of a Personal Station

All of the air interfaces share a common approach to the operation of the mobile telephone, called a Personal Station (PS) for the remainder of this chapter. This section will discuss the common characteristics of the operation and later sections will discuss how each particular air interface meets those characteristics.

Each of the air interfaces defines control channels that are used for data communications between the PS and the PCS system and traffic channels that are used for user-to-user communications (voice or data). When a PS is first powered up, it will find and decode data on a control channel before any further processing can be done. The operation of the PS can be segmented into following tasks (see Figure 2.3):

Power-up: Whenever power is applied to the PS, it will initialize itself and then enter the Scan Control Channels task.

Scan Control Channels and Decode Data: Under certain conditions (i.e., the PS first powers up, loses communications with the PCS system, is directed to by an order or moves into a new system) the PS will analyze the transmissions from the RS. It does this by scanning the control channels in the system, choosing the best one (usually based on signal strength), and determining the system characteristics.

Idle: When the PS is in the Idle task, the PS is waiting for further action to occur. It needs to process data on the control channel and monitor for its own ID to be paged or for other global orders to be received.

Registration/DeRegistration: The registration process informs the PCS system of the location of the PS. Thus, the PS can be paged in the correct system or region of the system without having to resort to paging in multiple areas. This reduces the paging load in large system. Registration also results in the visited PCS system obtaining security and services data from the home PCS system's database and populating the data into the visited system's database. These databases are known as the Home Location Register (HLR) and Visited Location Register (VLR), respectively.

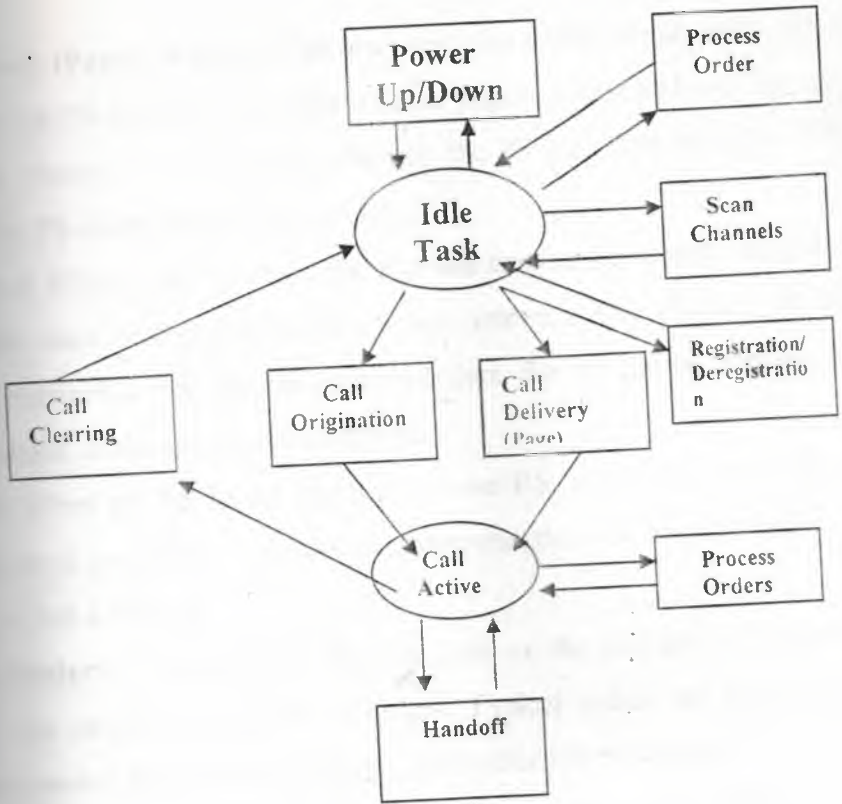


Figure 2.3 General Operation of a PS

When the PS first powers up, when a registration timer expires, when it enters a new system, or when directed to by a specific or general order on the control channel, the PS will register with the system. The registration consists of sending information to the PCS system to properly identify and authenticate itself.

When the PS is powered down (user pushes the on/off button to "off"), some PSs (but not all, depending on the air interface protocols) will send a deregistration message to the PCS system. The PS powers down only after the request is confirmed (or a time-out occurs). Those PSs not supporting this feature will power down without deregistering.

Call Origination: When the user wishes to place a call, dialed digits are collected by the PS and sent to the PCS network in an origination message. The PS then waits for an order to go to a traffic channel and enters the call active task.

ry (Page): Whenever the user receives a call, the network will send a page the PS. The PS will respond to the page message and wait for an order to go channel. On the traffic channel, the PS will alert the user. When the user e PS enters the call active task.

e: When a call is active, the user can communicate with the network over the voice or data information is sent over a traffic channel. In the call active etwork or the PS can process orders, the PS can be handed off to another nnel, or the call can be terminated.

When the PS is out of range of one RS, either the PS or the network can et of procedures to establish communications on another traffic channel. This lled a handoff.

)rders: During either the idle task or the call active task, the PS or the an initiate or respond to orders. Typical orders are three-way calling, call ference, short message service, and caller identification.

ring: When either party in the call wants to end the call, the call is cleared. PS initiates the action or the network initiates the action. The PS then enters earing task and ends the call.

he five digital air interfaces and the analog cellular air interface supports the ctions of the PS in different ways but accomplishes the same goal of supplying the user.

sic Services

PS can originate or receive a call, it will register with the PCS system. An n is made for emergency (911) calls. During the registration process, the PS is Temporary Mobile Subscriber Identity (TMSI) that is used for all subsequent cessing. For the call flows in the remainder of this chapter (except for ncy Calls) the Personal Station is assumed to be registered on the PCS system.

gination:

ation is the service where the PS user calls another telephone on the telephpne network. It is a cooperative effort among the PCS switch, the the RS.

tailed call flow steps are (see Figure 2.4 for the call flow diagram):

1. The PS processes an Origination Request from the user and sends it to the RS.
2. The RS sends a PCSAP Qualification Request to the VLR.
3. The VLR returns a Qualification Request Response to the RS.
4. The RS then processes an ISDN Setup message and sends it to the PCSC.
5. The PCSC sends an SS7 (ISDN User Part) Initial Address Message (IAM)to terminating switch.(wireline or wireless).
6. At the same time the PCSC returns an SS7 Call Proceeding message to the RS.
7. The RS assigns a traffic channel to the PS.
8. The PS tunes to the traffic channel and confirms the traffic channel assignment.
9. The terminating switch checks the status of the called telephone and returns an SS7 Address Complete Message (ACM) to the PCSC.
10. The PCSC returns an ISDN alert message to the RS.
11. The PCSC provides audible ringing to the user.
12. The terminating user answers.
13. The terminating switch sends SS7 ANSWER Message to the PCSC.
14. The PCSC sends an ISDN CONNect message to the RS.
15. The PCSC removes audible ringing and makes the network connection.
16. The RS returns an ISDN CONNect ACKnowledge message.
17. The two parties establish their communications.

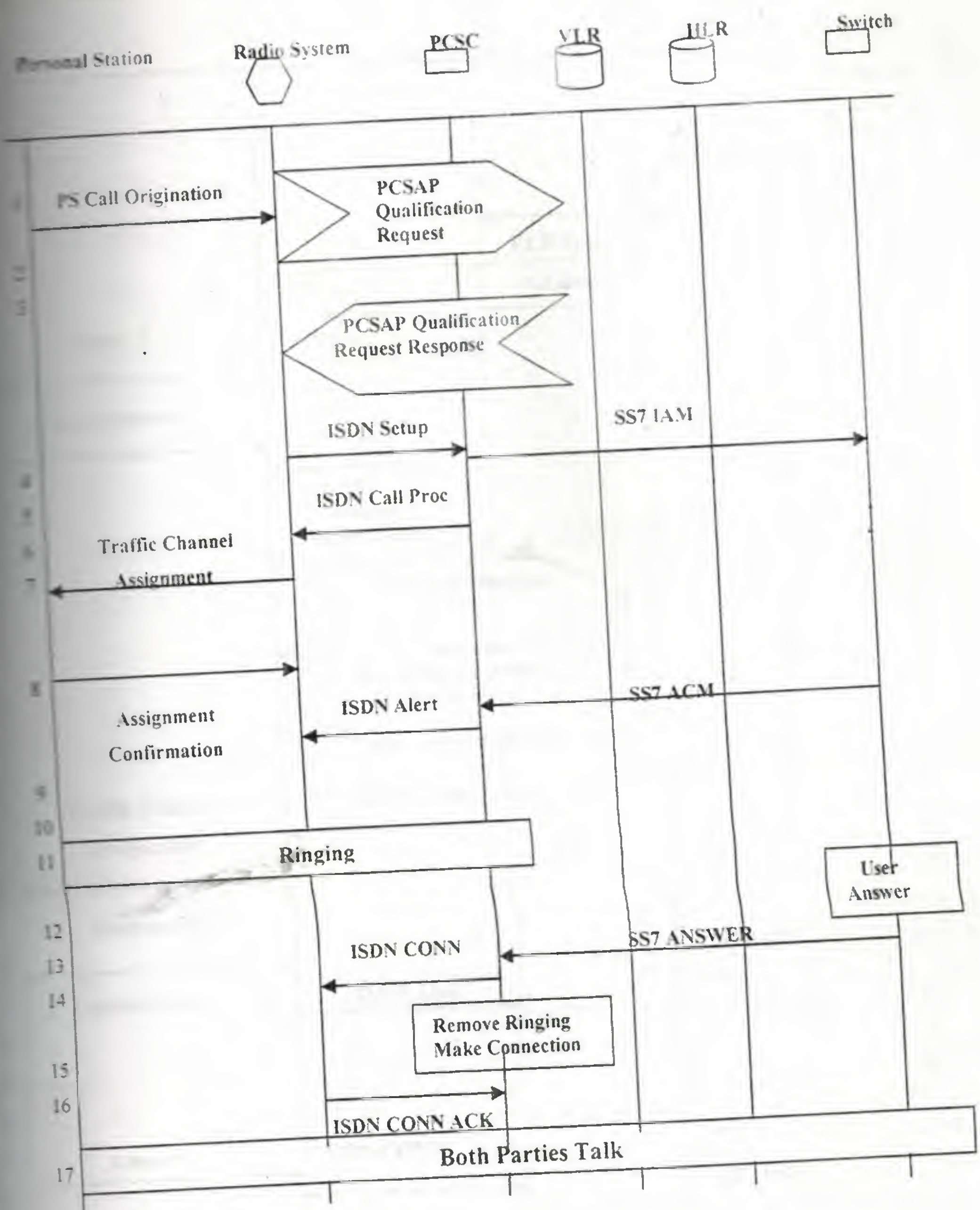


Figure 2.4 Personal Station Call Origination

b) Call Termination:

Call termination is the service where a PS user receives call from other telephones in the worldwide telephone network. The following discussion is for calls terminating to a PS registered at its home PCSC.

Call termination is a cooperative effort among the PCS switch, the VLR, and the RS. The detailed call flow steps are (see Figure 2.5 for the call flow diagram):

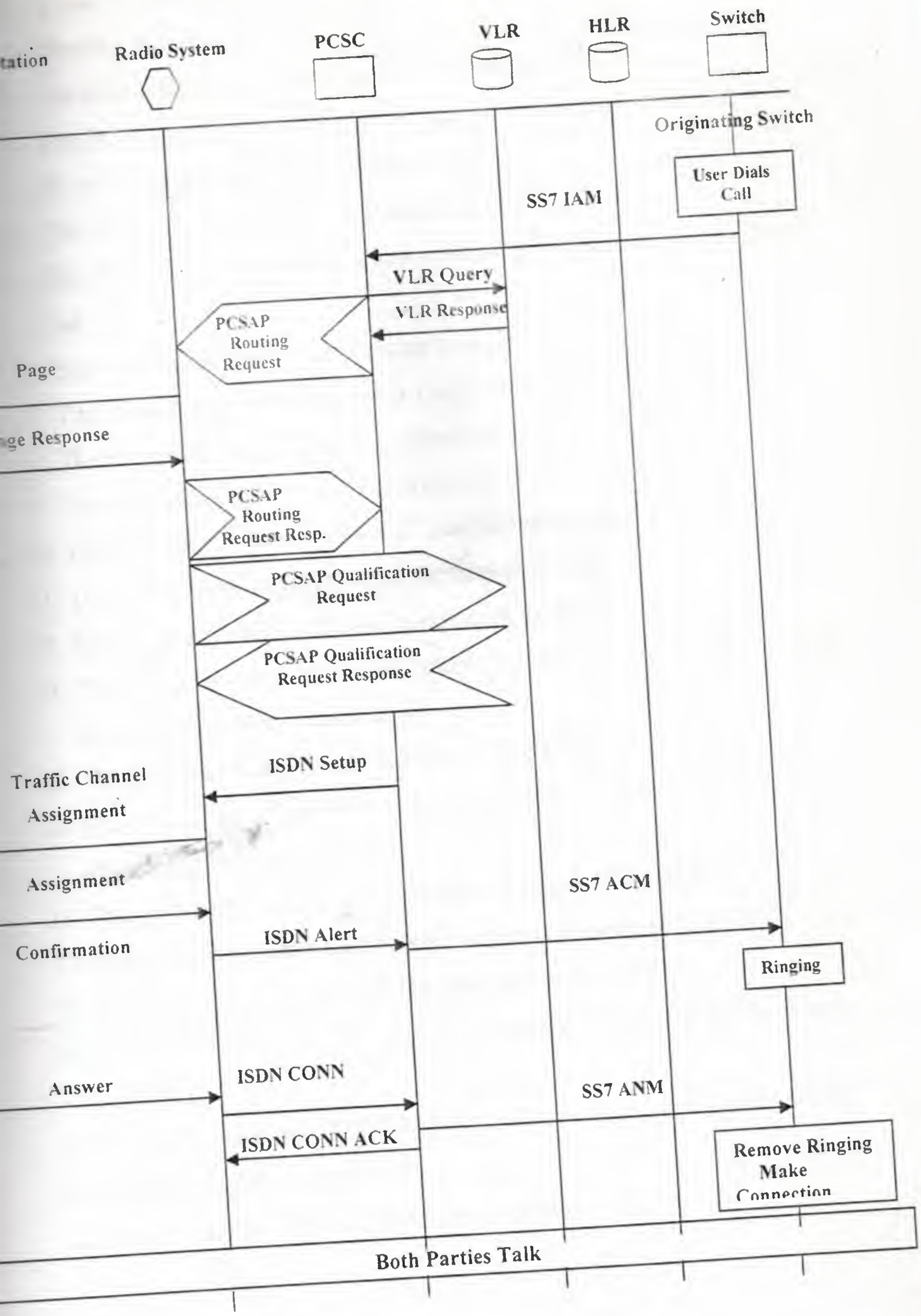


Figure 2.5 Call Termination to a PS

ser in the worldwide phone network (wired or wireless) dials the
tory number of the PS.

originating switch sends an SS7 IAM to the PCSC.

PCSC queries the VLR for the list of RSs (one or more) where the
will be paged, and for the TMSI of the PS.

VLR returns with the TMSI and list of RSs.

PCSC sends an PCSAP Routing Request message to all RSs on the

h RS broadcasts a page message on appropriate control channels.

e PS responds to the page with a page response message at one RS.

e RS sends a PCSAP routing request response to the PCSC.

e RS sends a PCSAP Qualification Directive message to the VLR.

e VLR responds with a PCSAP qualification request response

e PCSC sends an ISDN setup message to the RS.

e RS sends a traffic channel assignment to the PS.

e PS tunes to the traffic channel and sends a traffic channel
signment confirmation message.

e RS sends an ISDN alert message to the PCSC.

e PCSC sends an SS7 Address Complete Message (ACM) to the
iginating switch.

he originating switch applies audible ringing to the network.

he user answers and the PS sends an Answer message to the RS.

he RS sends an ISDN CONNect message to the PCSC.

he PCSC sends an SS7 Answer Message (ANM) to the originating
witch.

The PCSC send an ISDN CONNect ACKnowledge message to the RS.

Audible ringing is removed.

The two parties establish their communications.

learing:

her party in a conversation wishes to end a call, then the call clearing function d. The exact call flows depend on which side ends the call first. Call clearing is ative effort among the PCS switch, the VLR, and the RS. The detailed call flow a PS-initiated call clearing are (see Figure 2.6 for the call flow diagram):

he PS user hangs up

he PS sends a release message to the RS.

he RS sends an ISDN DISConnect message to the PCSC

he PCSC sends an SS7 RELease message to the other switch.

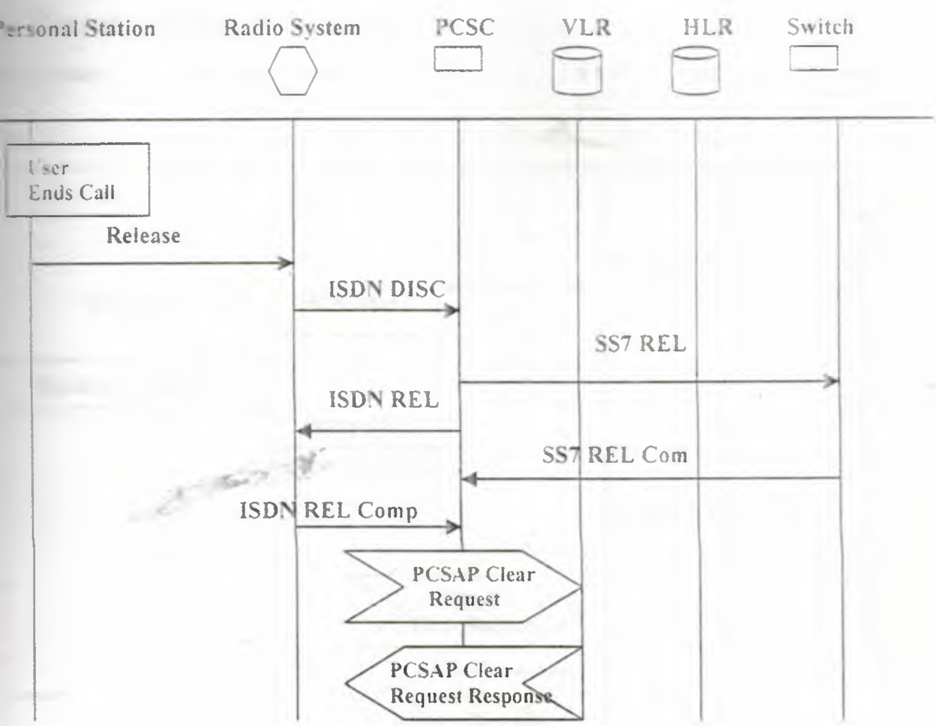


Figure 2.6 Call Clearing- PS Initiated

he PCSC sends an ISDN RELease message to the RS.

he other switch sends an SS7 RELease complete message to the PCSC

he RS sends an ISDN RELease complete message to the PCSC

he RS sends a PCSAP Clear Request message to the VLR.

he VLR closes the call records and sends a PCSAP Clear Request response nessage to the RS.

The detailed call flow steps for a far-end-initiated call clearing are (see re 2.7 for the call flow diagram):

The far-end user hangs up.

The other switch sends an SS7 RELease message to the PCSC

The PCSC sends an ISDN DISConnect message to the RS.

The RS sends a Release message to the PS.

The PS confirms the message and disconnects from the traffic channel.

The RS sends an ISDN RELease message to the PCSC.

The PCSC sends an ISDN RELease Complete message to the RS.

The PCSC sends an SS7 RELease Complete message to the other switch.

The RS sends a PCSAP Clear Request message to the VLR.

). The VLR closes the call records and sends a PCSAP Clear Request Response message to the RS.

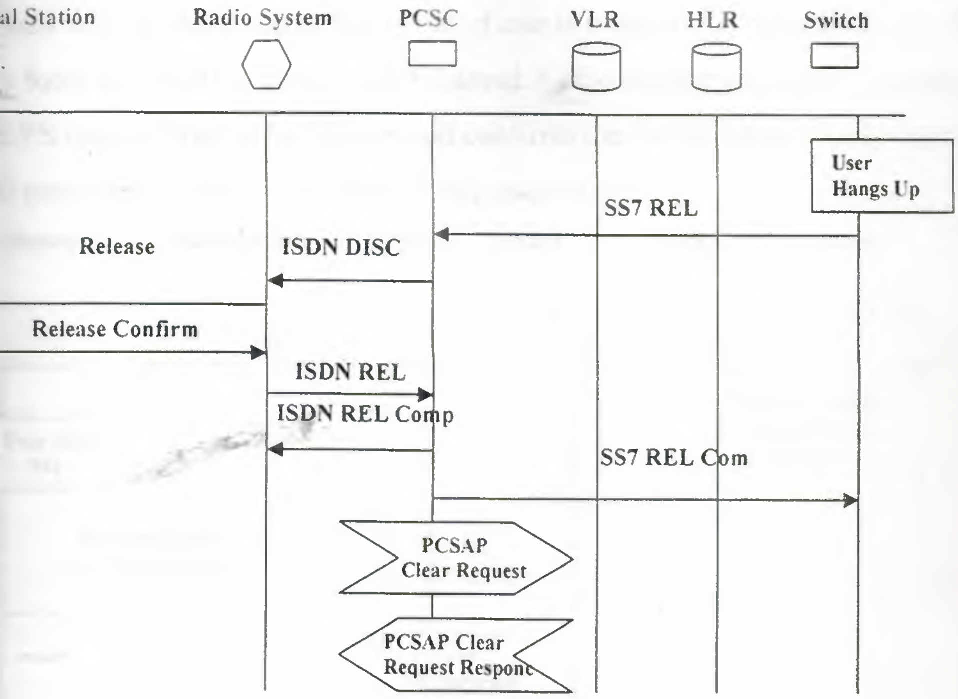


Figure 2.7 Call Clearing-Far-End Initiated

gency Calls:

cy calling is a service that enables a user of a PS to reach an emergency service through a simple procedure of dialing 911 or pushing an emergency button on Emergency calling is offered to unregistered and/or unsubscribed PSs (at the oviders option).The goal is to process the call independent of any failures that ur. Thus, authentication failures are ignored during call processing for y calls.For emergency calling to operate with the following procedures, a y call indication will be set in the origination message from the PS. If that bit

...then the call will be processed with normal handling. The following are the ... processing an emergency call (see Figure 2.8 for the call flow diagram):

1. The user dials 911-SEND or pushes an emergency button on the PS.
2. The PS recognizes the unique number 911 or the emergency button depression and forms an origination message with the emergency calling indication set.
3. The RS sends a PCSAP Emergency Request Message to the VLR.
4. The VLR sends a PCSAP Emergency Request Response Message to the RS.
5. The RS forms an ISDN Setup Message to PCSC.
6. The PCSC sends an SS7 IAM to the Personal Safety Access Point (PSAP).
7. The PCSC sends an ISDN Call Proceeding message to the RS.
8. The RS sends a traffic channel assignment to the PS. If no channel is available, the call will go to the top of the queue, if one is maintained. Alternately, the RS may force a handoff to free a traffic channel. Calls in progress are not dropped.
9. The PS tunes to the traffic channel and confirms the traffic channel assignment.
10. Call processing proceeds at step 9 of call origination.

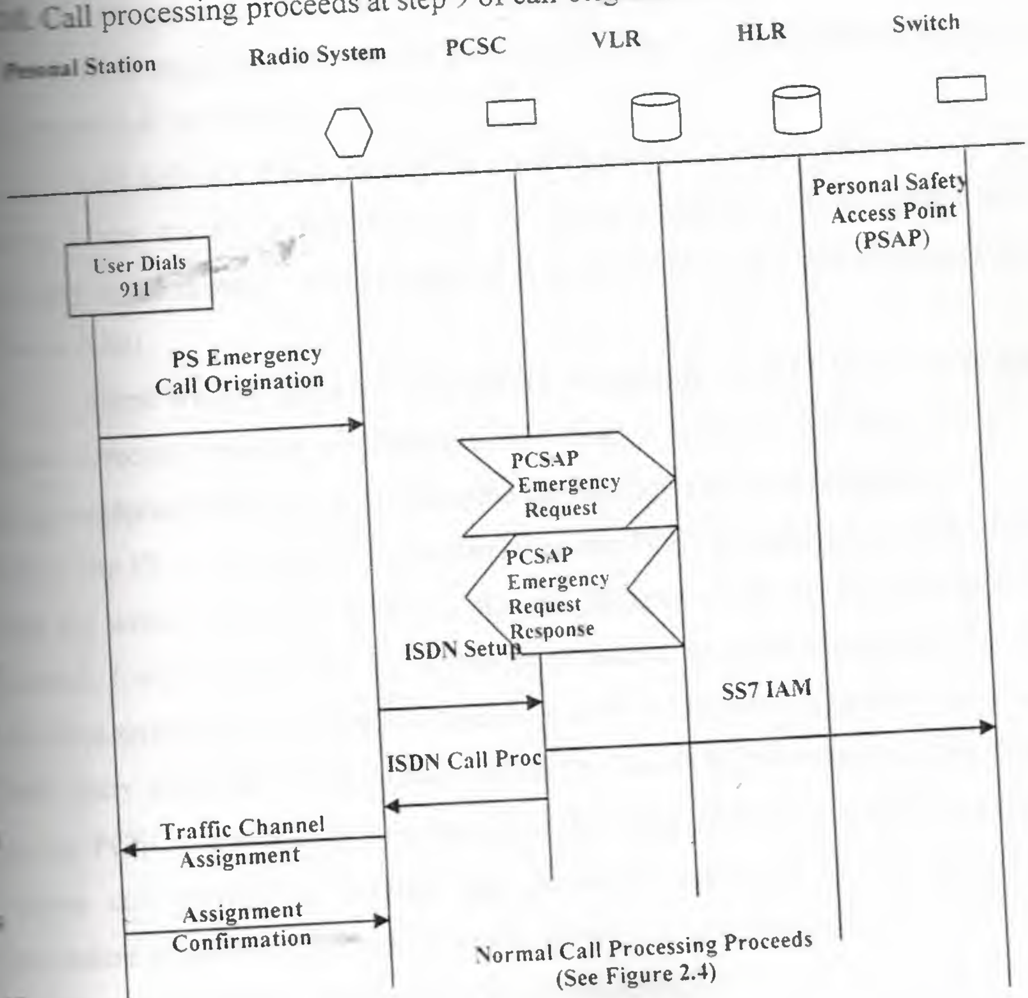


Figure 2.8 PS Emergency Calling

ng:

s the ability to deliver services to PSs outside of their home area. When a PS g. registration, call origination and call delivery will take extra steps. data will be retrieved from the VLR, and the data is not available, then the send a message to the appropriate HLR to retrieve the data. The data consists \ to MIN conversion. service profiles, Shared Secret Data (SSD) for ication. and other data needed to process calls. The most logical time to retrieve ta is when the PS registers with the system.

Once the data on a roaming PS is stored in the VLR, then call processing for any ting services (basic or supplementary) is identical to that of home PSs. However, may be times when the PS originates a call before registration has been plished or when the VLR data is not available. At those times, an extra step will ed for the VLR to retrieve the data from the HLR. Thus any originating service o optional steps where the VLR sends a message (using IS-41 signaling over SS7 e HLR requesting data on the roaming PS. The HLR will return a message with per call information.

Call delivery is not possible to an unregistered PS since the network does not where the PS is located. Once the PS is registered with a system, then call y to the roaming PS is possible. This section will discuss call delivery to roaming detail.

There are two cases of call delivery to roaming PS: The PS has a geographic- directory number (indistinguishable from a wireline number); The PS has a graphic number. We will describe the call flows for both operations.

he PS has a geographic number, then the PCSC is assigned a block of numbers within the local numbering plan for the area of the world where the PCSC is Call routing to the PS is then done according to the procedures for that of a e telephone. If a PS associated with a PCSC is not in its home area, the PCSC ery the HLR for the location of the PS. The PCSC then invokes call forwarding CSC where the PS is located, and the connection is made to the second PCSC call terminating services are deliveried according to the procedures.This re is inefficient because it results in two sets of network connections_originating o home PCSC, and home PCSC to visited PCSC.

to a roaming PS is a cooperative effort between the home and
.R and HLR, and the RS.

call flow steps for call delivery to a roaming PS with a geographic
(see Figure 2.9 for the call flow diagram):

ne worldwide phone network (wired or wireless) dials the directory
the PS.

ating switch sends an SS7 IAM to the home PCSC.

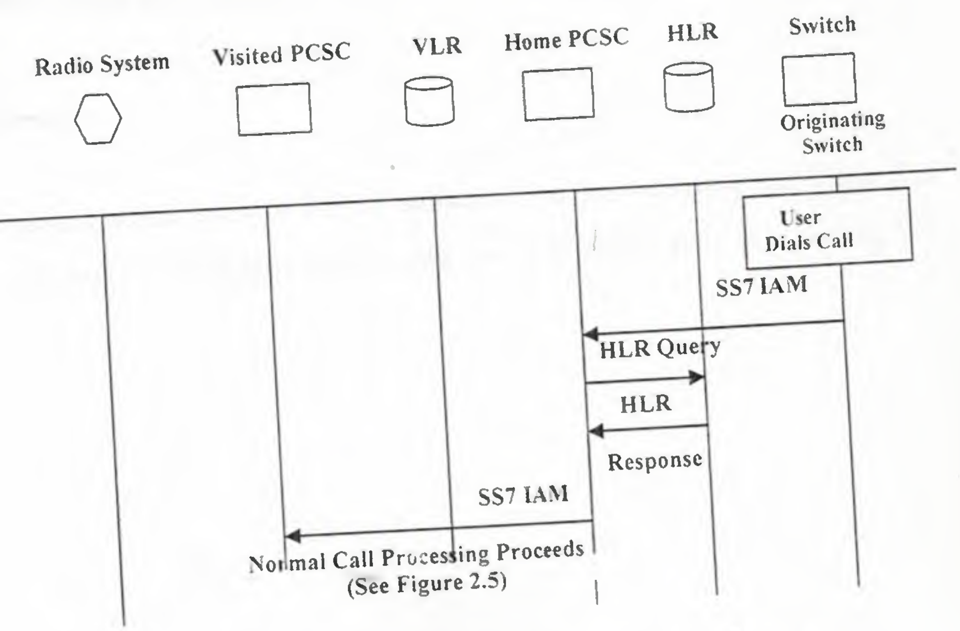
PCSC queries the HLR for the location of the PS.

returns the location of the visited system.

C invokes call forwarding to the PCSC in the visited system and the
g (home) PCSC switch sends an SS7 IAM to the visited PCSC.

essing proceeds at step 3 of the terminating call flow.

s a nongeographic number, then calls can be directed from an
n directly to the visited switch. Call delivery to a nongeographic
that the originating switch recognize the number as a nongeographic
pecial call processing for routing. This special processing is known as
gent Network (AIN) processing. If the originating switch does not
n it will route the call to a switch that supports AIN. With AIN support,
switch will recognize the nongeographic number and send an SS7
ILR with a request for the location of the PS.



re 2.9 Call Termination to Roaming PS with Geographic Number

ll return a temporary directory number (on the visited PCSC) that can be
to the PS in the visited system. Calls then proceed according to normal
call flows. Call delivery to a roaming PS with a nongeographic number is.
cooperative effort between the visited PCSC, the VLR and HLR, and the
ailed call flow steps for call delivery to a roaming PS with a nongeographic
mber are (see Figure 2.10 for the call flow diagram):

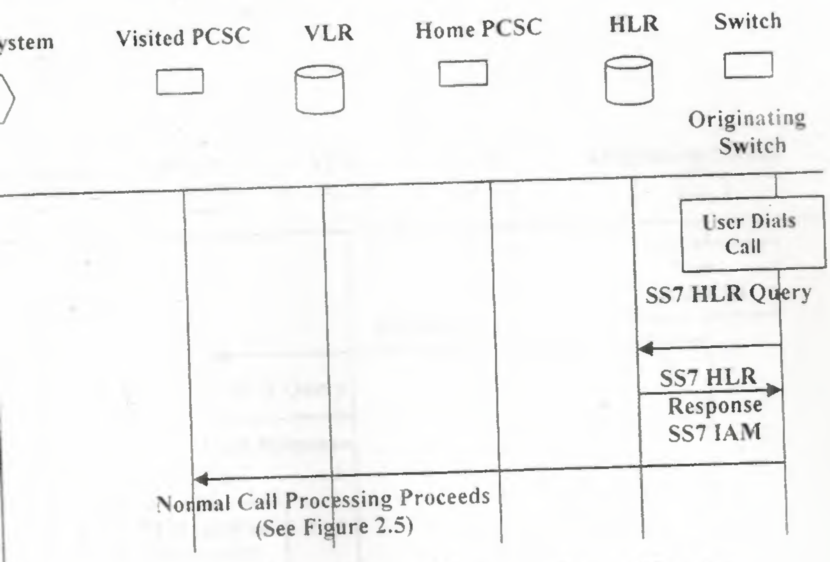
ser in the worldwide phone network (wired or wireless) dials the directory
nber of the PS.
originating switch recognizes the number as a nongeographic number and
ds an SS7 query message to the HLR at the home PCSC.
e HLR returns the location of the visited system with a directory number to
for further call processing.
e originating switch sends an SS7 IAM to the visited PCSC.
ll processing proceeds at step 3 of the terminating call flow .

plementary Services:

ports several supplementary services. However, only the call flows for a few
e common ones are described herein.

aiting:

ing provides notification to a PCS subscriber of an incoming call while the
is in the busy state. Subsequently, the user can either answer or ignore the
call.



Termination to Roaming PS with Nongeographic Number

ered, the user can switch between the calls until one or more
either distant party hangs up, then the call reverts to a normal
If the PS user hangs up, then both calls are cleared according to
nctions. The detailed call flow steps for call waiting delivery are
call flow diagram):

l.

switch sends an SS7 IAM to the PCSC.

ries the VLR.

ns with a location of the PS that is within the serving system. If it
call will be forwarded to the serving PCSC.

termines that the PS is busy and subscribers to call waiting and
call waiting tone.

es the "flash" button (may be "send" on some PSs) to answer the
dication, and the PS sends a flash message to the RS.

an ISDN Hold message to the PCSC.

s the first call on hold and connects the second call.

ds a Hold Acknowledge to the RS.

0. The user presses the flash button (may be SEND on some PSs) to talk to caller 1, and the PS sends a flash message to the RS.
1. The RS sends an ISDN Hold message to the PCSC.
12. The PCSC puts the second call on hold and connects the first call.
13. The PCSC sends a Hold Acknowledge to the RS.
14. The user wants to drop the current call (either 1 or 2) and pushes the "drop" (or "END") key, and the PS sends a Drop message to the RS.
15. The RS sends an ISDN Drop message to the PCSC.
16. The PCSC drops the current call and connects the other call (the one currently on hold).
17. The PCSC sends an ISDN Drop Acknowledge message to the RS.

Call Forwarding:

all Forwarding represents the ability to redirect a call to a PS handset in three uations: Unconditional, Busy, and No Answer.The call forwarding features build oon the PS cal terminating capability. Callforwarding is separate from call delivery to roaming PS; Thus, if a roaming PS has call forwarding invoked, two stages of call orwarding may occur. The detailed call flow steps for call forwarding unconditional are see Figure 2.12 for the call flow diagram):

1. A user in the worldwide phone network (wired or wireless) dials the directory number of the PS.
2. The originating switch sends an SS7 IAM to the PCSC.
3. The PCSC determines that the PS has invoked call forwarding for all calls.
4. The PCSC sends an SS7 IAM message to the destination switch.
5. Call processing proceeds according to normal procedure for a wireline or wireless user.

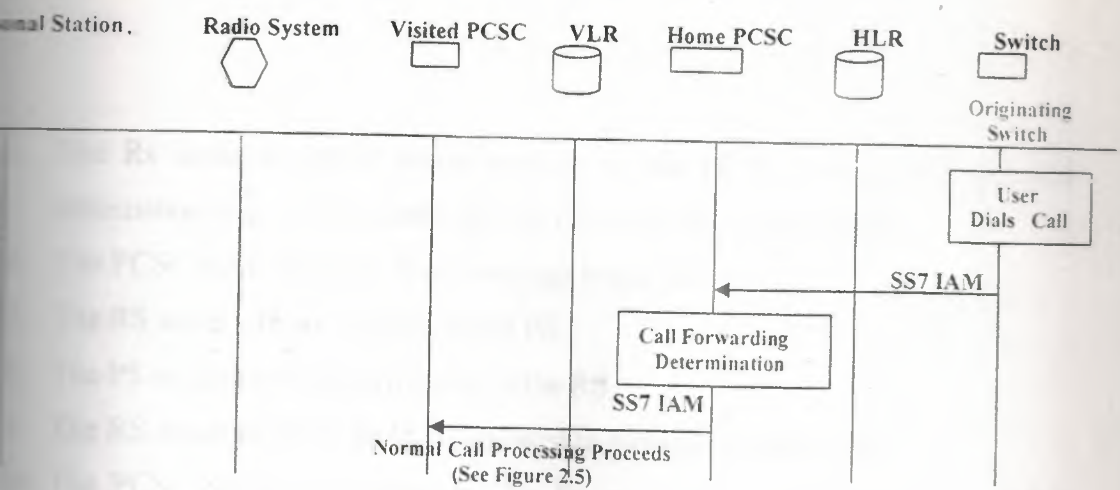


Figure 2.12 Call Forwarding to a PS

...n the user has invoked call forwarding busy or no answer, additional steps between ...d 3 above are needed to page the PS or to determine that it is busy.

...hree-Way Calling:

...e-way calling is the service where a PS user can connect to two parties ...ltaneously and all three parties are part of the call. If the PS subscribers to three-...calling, then the PS can add a third party independent of who originated the call. ...n a PS user wants to place a three-way call, the user puts the first party on hold, ...second party, and then requests a three-way connection. When either of the two ... parties disconnects, the call reverts to a normal two-way call. If the PS user ...nnects, then all connections are released. The detailed call flow steps for three-way ...g establishment are (see Figure 2.13 for the call flow diagram):

1. The PS user signals a flash indication to the RS.
2. The RS signals to the PCSC a ISDN Information message with a feature activator of three-way calling.
3. The PCSC allocates resources for the three-way call.
4. The PCSC sends an ISDN Information message to the RS with a feature identifier of three-way calling and active status.

5. The Rs sends an ISDN Setup message to the PCSC with appropriate call information (e.g., calling party ID, call reference, B channel used).
6. The PCSC sends an ISDN Hold message to the RS.
7. The RS sends a Hold message to the PS.
8. The PS sends a hold acknowledge to the RS.
9. The RS sends an ISDN Hold Acknowledge message to the PCSC.
10. The PCSC begins call processing for the new call by sending an SS7 Initial Address Message (IAM) to the terminating switch.
11. The PCSC sends a ISDN CALL PROCeeding message to the RS.
12. The terminating switch returns an SS7 Address Complete Message (ACM).
13. The PCSC sends a ISDN ALERT message to the RS.
14. The network provides audible ringing.
15. The terminating party answers.
16. The terminating switch sends an SS7 Answer Message to the PCSC.
17. The PCSC sends an ISDN CONNect message to the RS.
18. The RS sends an ISDN CONNect ACKnowledge message to the PCSC.
19. The two users begin talking.
20. The PS user sends a flash message to the RS and recognizes the flash as a request to join the two calls.
21. The RS sends a ISDN RETrieve message to the PCSC with the call record for the held call.
22. The PCSC forms a three-way call from the two call records.
23. The PCSC sends an ISDN Retrieve ACKnowledge message to the RS.
24. The PCSC sends an ISDN RELease message to the RS to release the unneeded second call reference.
25. The RS sends an ISDN RELease COMPlete message to the PCSC.

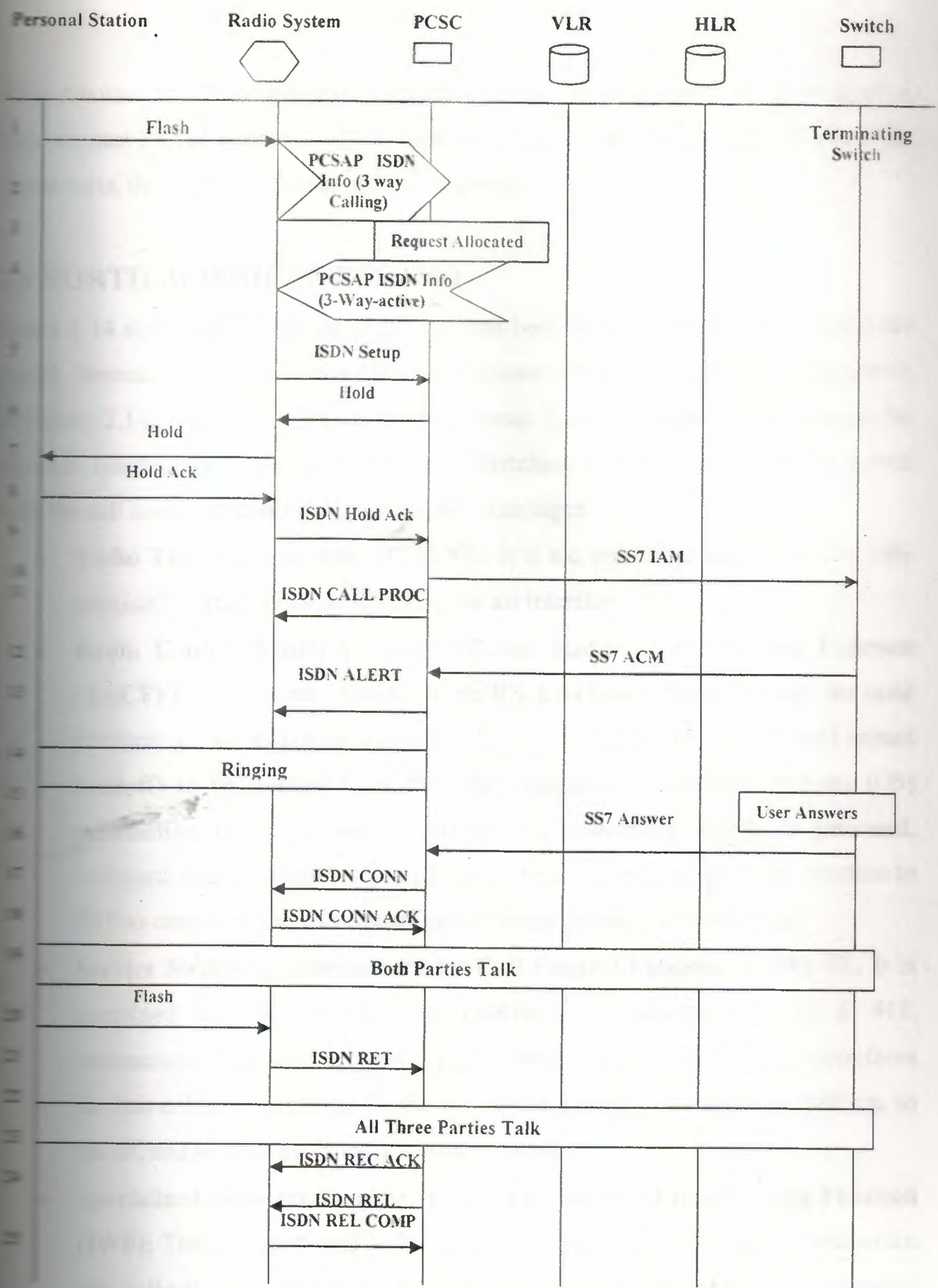


Figure 2.13 PS Three-Way Calling

ser desires to release the connection to the last added party, the user signals a st to the network, which then releases the last added party. If the User , the network releases both other parties.

TH AMERICAN PCS 1900

shows the functional model that has been derived from the T1P1 reference eral physical scenarios can be developed using the functional entities shown 14. Figure 2.15 shows the Functional Entity (FE) grouping in which the erface between the RS and the Switching System Platform (SSP) carries control and mobility management messages.

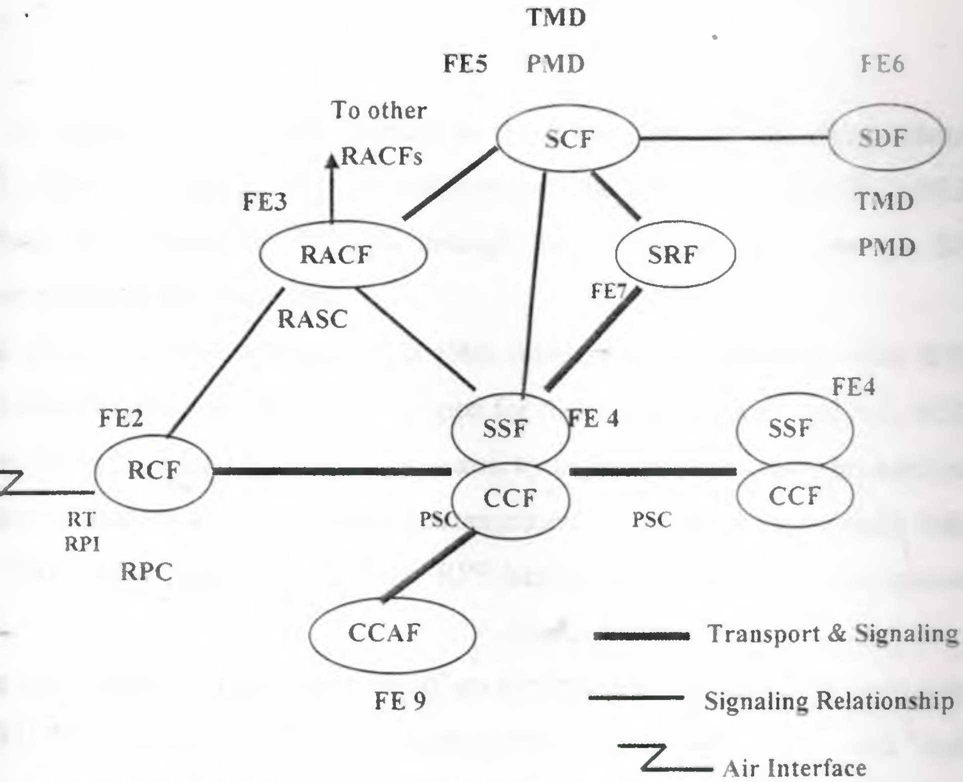
io Terminal Function (RTF) FE: It is the subscriber unit (SU). The only ical interface is to the RS using the air interface.

o Control Function (RCF) FE and Radio Access Control Function CF) FE: These are included in the RS. Combining these FEs onto the same orm allows air-interface-specific functions (such as those that would impact off) to be isolated from the other interfaces. Operations Systems (OS) mation, including performance data and accounting records, is generated, cted, and formatted on this platform. There is only one physical interface to o carry both the call control and mobility management signaling.

ce Switching Function (SSF)/ Call Control Function (CCF) FE: It is ined in SSP and provides interfaces to operator services, E 911, ational calls, and network repair/ maintenance centers. Physical interfaces is collection include: To the RS, to the mobility management platform, to , and to other SSPs and external networks.

alized Resource Function (SRF) FE and data InterWorking Function): They are contained in the Internal Peripheral (IP). Physical interfaces for ollection include one to the SSP and another to the mobility management rm.

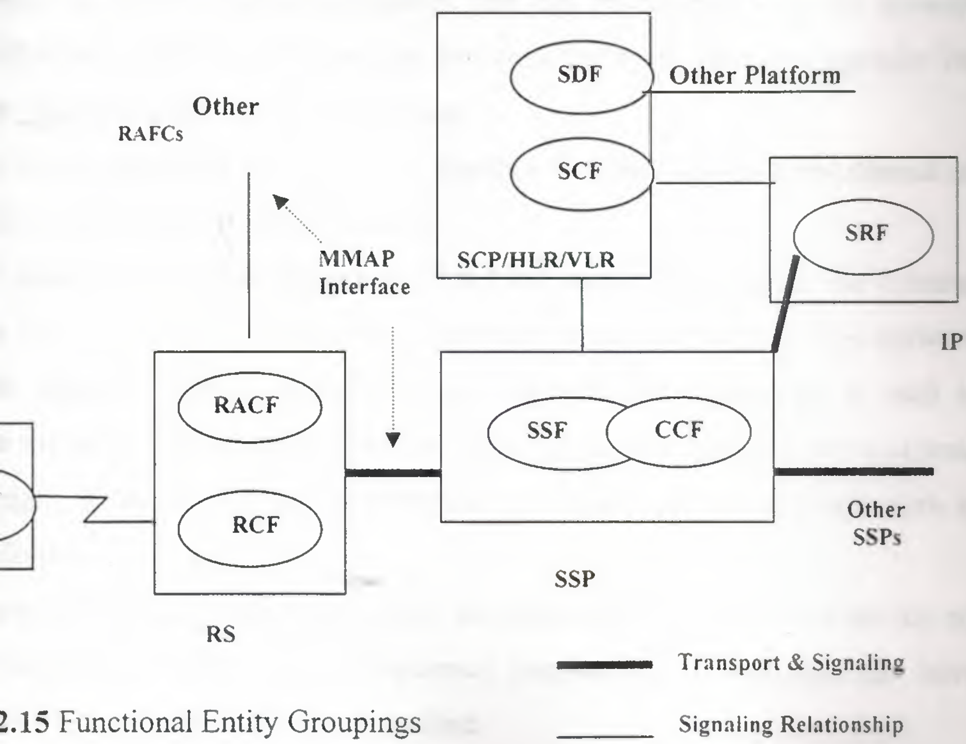
the SSF/CCF FE and CCF FE represent interswitch and inter-network ity collections and physical interfaces.



ontrol Agent Function.
ntrol Function.
l Element.
al Mobility Data.
Access Control Function.
Control Function.
ort Controller.
ort Interface.

RPT: Radio Port Terminal.
RT: Radio Terminal.
SCF: Service Control Function.
SDF: Service Data Function.
SRF: Specialized Resource Function.
SSF: Service Switching Function.
TMD: Terminal Mobility Data.

gure 2.14 Functional Model Derived from T1P1 Reference Model



2.15 Functional Entity Groupings

nown in Figure 2.15, the only interface to the RS is from the Switching Control (SSP). There is no direct physical path between the RS and the SCP/HLR/VLR. perations to or from the RS pass through the SSP, whether or not the SSP umes or produces the operation.

The proposed North American PCS 1900 standard is an adaptation of the ETSI 1800 standard that was initially developed for the digital cellular band, i.e., GSM t consists of 200-kHz radio channels shared by eight time slots, one per terminal. tandard supports a frequency duplex arrangement for forward and reverse links. PCS 1900 system supports a fixed-rate RPE based on a speech coder that operates kbs.

The North American types of handoff are network initiated and Mobile-Assisted off (MAHO). Incase of the network-initiated handoff, both "hard" and "soft" ff are supported. The PCS 1900 standard defines support for MAHO and a form work-initiated handoff that applies only to "hard" handoff. For PCS 1900 systems ction as an integral part of the North American PCS environment, handoff needs e supported between PCS 1900 and North American systems.

900 supports voice privacy through the encryption capabilities. The encryption privacy) is an air interface capability that can be controlled by the network or rather than a service that may not be controlled by the network operator but lso be offered as a service to the end user.

The GSM encryption is only an air interface function and does not depend on SM MAP function or the home network.

The authentication algorithm in the PCS 1900 uses IMSI as input. The terminal ses a "key" that is the same "key" known by the home network. The network tes a signature that is specific for an end user. This signature is used to icate the end user through the duration of the service. This authentication scheme strength in the authentication algorithm. However, there is no mechanism to ze clones.

To satisfy the PCS needs requirements for accessibility and seamless service, air e transparency must exist. Transparency implies that an end user can have o service regardless of the access method.

the initial phase of PCS, multiple air interface may exist, and therefore "dual mode" r "dual spectrum" terminals may be used. The aim is to attain some level of teroperability with the existing North American networks. If interoperability does not xist between PCS 1900 air interface and analog AMPS 800-MHz air interface, the biquity of service is precluded. The PCS 1900 air interface may access the network at provides GSM services. The AMPS analog air interface may have access to the IS-1 services. Access to two different types of networks and services may preclude biquity of service from an end-user prospective.

.6. SUMMARY

his chapter presented the analog and digital systems used in the United States, we rovided typical call flows for origination, termination, handoff and so on. This chapter so addressed PCS 1900 derived from the GSM for PCS applications in the North merica.

3. AIR INTERFACE UNIQUE CAPABILITIES

OVERVIEW

s section will discuss how the analog cellular and each of five digital air interfaces U_m-Interface in the reference model) support the functions of basic encoding of the tal data being sent (i.e., the framing of the channel), speech encoding method, the of the frequency allocation for PCS, the output power of the PS and how it is trolled, the modulation method, and support for hand-offs.

Framing of Digital Signal

log cellular uses a digital control channel with analog voice channels. During a e call, digital data is sent via blanking the audio and sending the data (blank and st) during the blanked audio period. Data transmission is handled via either voice d modems or Cellular Digital Packet Data (CDPD). For the digital air interfaces, the ital format of each interface will multiplex voice and data traffic, control rmation, and other data as needed. Each of the interfaces uses a different approach this multiplexing. After the signal is multiplexed, it will be encoded and then dulated before transmission. The receiver will undo each step. This section discusses v each of the air interfaces support the sending and receiving of the data.

.1 Framing of Analog Cellular

e analog cellular system uses a digital format at 10 kbs sent on a separate control nnel. The basic frame is 463 bits long. Figure 3.14 shows the framing on the forward trol channel (RS to PS). Synchronization of the data receiver is established via a 10- 1010101010 (dotting) pattern and 11-bit 11100010010 framing pattern.

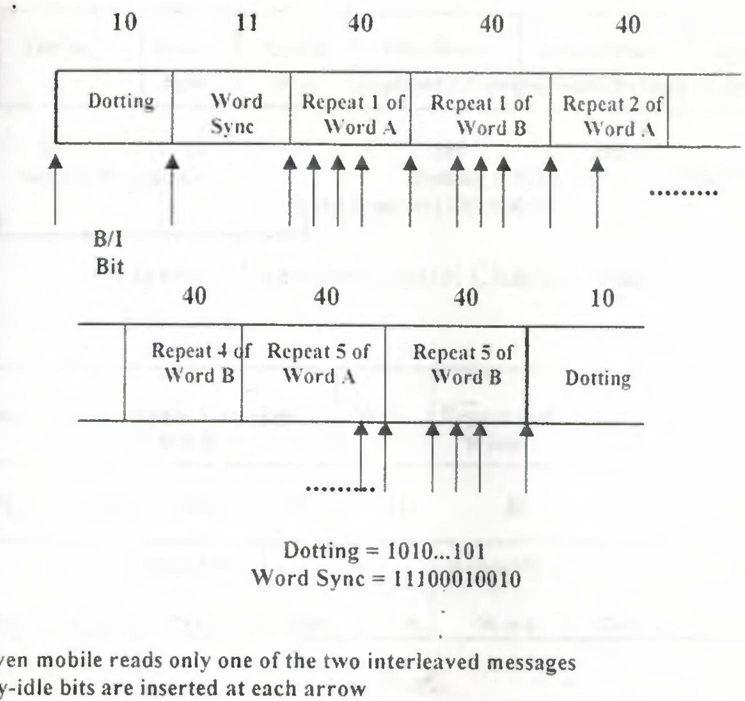


Figure 3.1 Forward Control Channel Framing

ssages are then sent on an A or B frame and repeated five times. Each message uses 40,28,5: 28 data bits, 12 parity check bits for a total of 40 bits and capability of ection of five errors) BCH code. The combination of the BCH code and the five etitions of the message provide the error detection and correction in the data eiver. After the data is encoded , it undergoes a further encoding, called Manchester coding, using two bits per baud.

Every 10 bits in the frame, a busy/idle bit is sent to inform PSs of the status sy or idle) of the reverse control channel. A PS can transmit on the reverse control nnel, only when the channel is idle, as indicated by the busy/idle bits sending an idle us.

ures 3.2, 3.3, and 3.4 show the frame information of the reverse control channel and forward and reverse blank and burst channel. Different channels use a different nber of repeats of the message and a different length of the dotting pattern. On the erse channel a (48, 36, 5) BCH code is used.

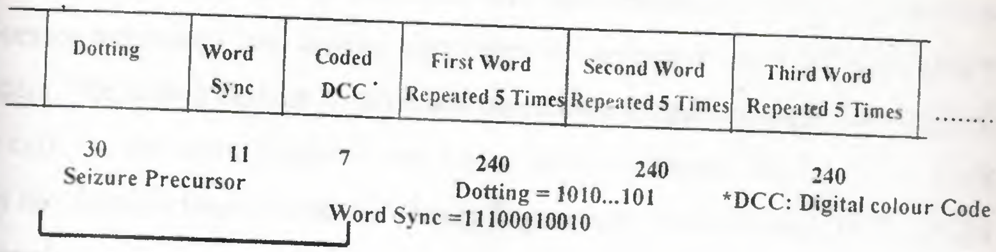


Figure 3.2 Reverse Control Channel Framing

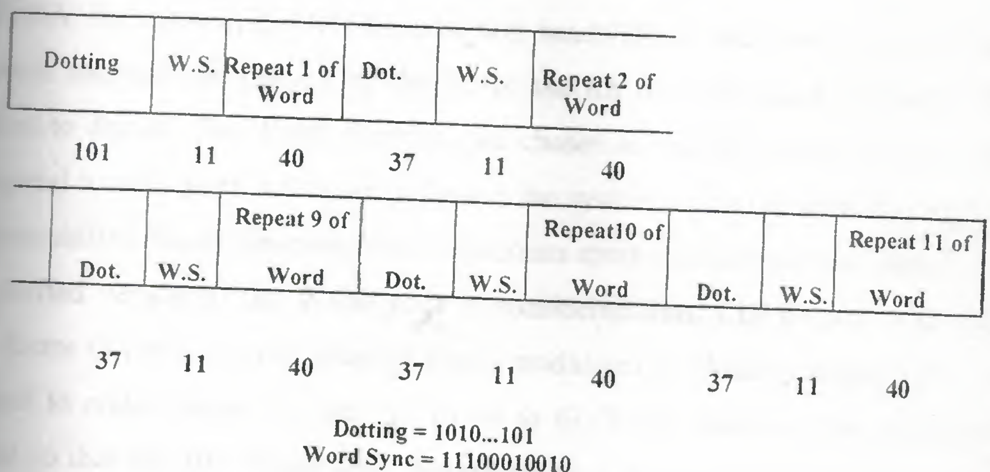


Figure 3.3 Forward Voice Channel Framing

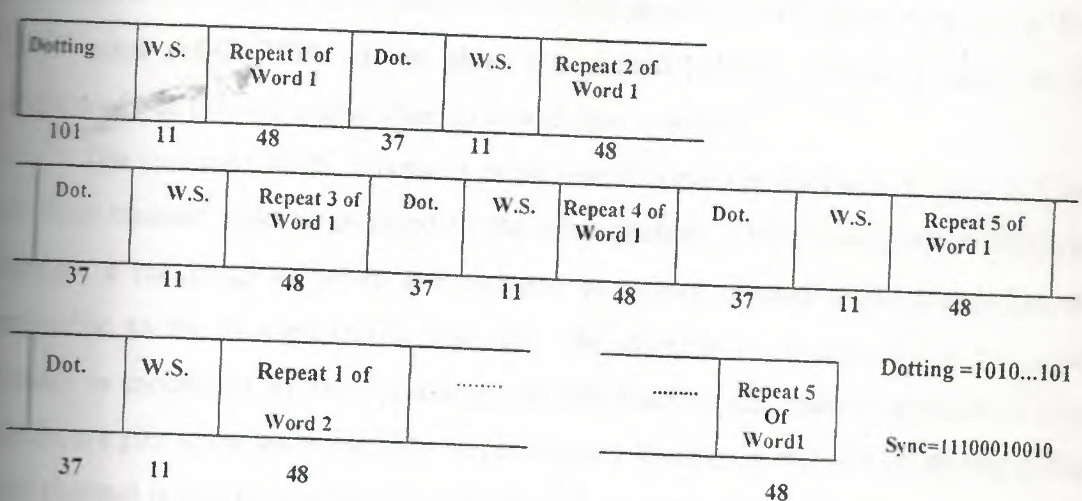


Figure 3.4 Reverse Voice Channel Framing

The analog control channel was designed in the 1970s when most data receivers were implemented in hardware, and therefore it uses inefficient coding systems. The overall throughout of the various channels (forward and reverse) is 300 to 600 bits per second even though the signaling is done at 20 kilobaud per second.

The digital cellular and PCS protocols have taken advantage of improvements in microprocessor technology and coding algorithms to implement more efficient coders and decoders. The analog cellular systems are based on a frequency reuse factor (N) of ... Nearby cells on the same frequency are coded with a different Digital Color Code (DCC) on the control channels and a different Supervisory Audio Tone (SAT) on the voice channels.

3.2.2 Framing of CDMA

In CDMA, the entire 1.25-MHz transmission bandwidth is occupied by every station. Different channels are chosen by the PS or the RS by selecting a particular Walsh function to decode. The Walsh functions are chosen so that the set of functions are all orthogonal to each other. All radio systems in the system are on the same frequency and use time-shifted Walsh functions.Walsh functions cross correlate to low values when a time shifted version of the Walsh code is considered thus, CDMA uses a frequency reuse factor (N) of 1. A code channel that is modulated by Walsh function "n" will be assigned to code channel number "n" (n =0 to 63).Walsh function time alignment is defined so that the first Walsh chip, designated by 0 in the column headings of Table (shown in Appendix), begins at an even second time mark referenced to radio system transmission time. The Walsh function spreading sequence will repeat with a period of 52.083 µsec (=64/1.2288 million chips per second [Mcps]), which is equal to the duration of one forward traffic channel modulation symbol.

The forward CDMA channel has the overall structure shown in Figures 3.5 and 3.6. One channel code is assigned to the pilot channel; One is assigned to the sync channel; a maximum of seven are assigned to paging channels; The remainder are assignable to the forward traffic channels. The information (voice or data) on each channel is modulated by the appropriate Walsh function and then is modulated by a quadrature pair of pseudonoise (PN) sequences at e fixed chip rate of 1.2288 Mcps. The pilot channel is always assigned to code channel number zero. When the sync channel is present, it will be assigned code channel number 32. If paging channels are present, they will be assigned to code channel numbers one through seven (inclusive) in sequence. The remaining code channels are available for assignment to the forward traffic channels. When a radio system supports multiple forward CDMA channels, then frequency division multiplex is used. Figure 3.7 shows a sample assignment of the code channels transmitted by a radio system.

example includes: The pilot channel (that is always required), 1 sync channel, 7 g channels (the maximum number allowed), and 55 traffic channels. Alternately, e a radio system does not need paging

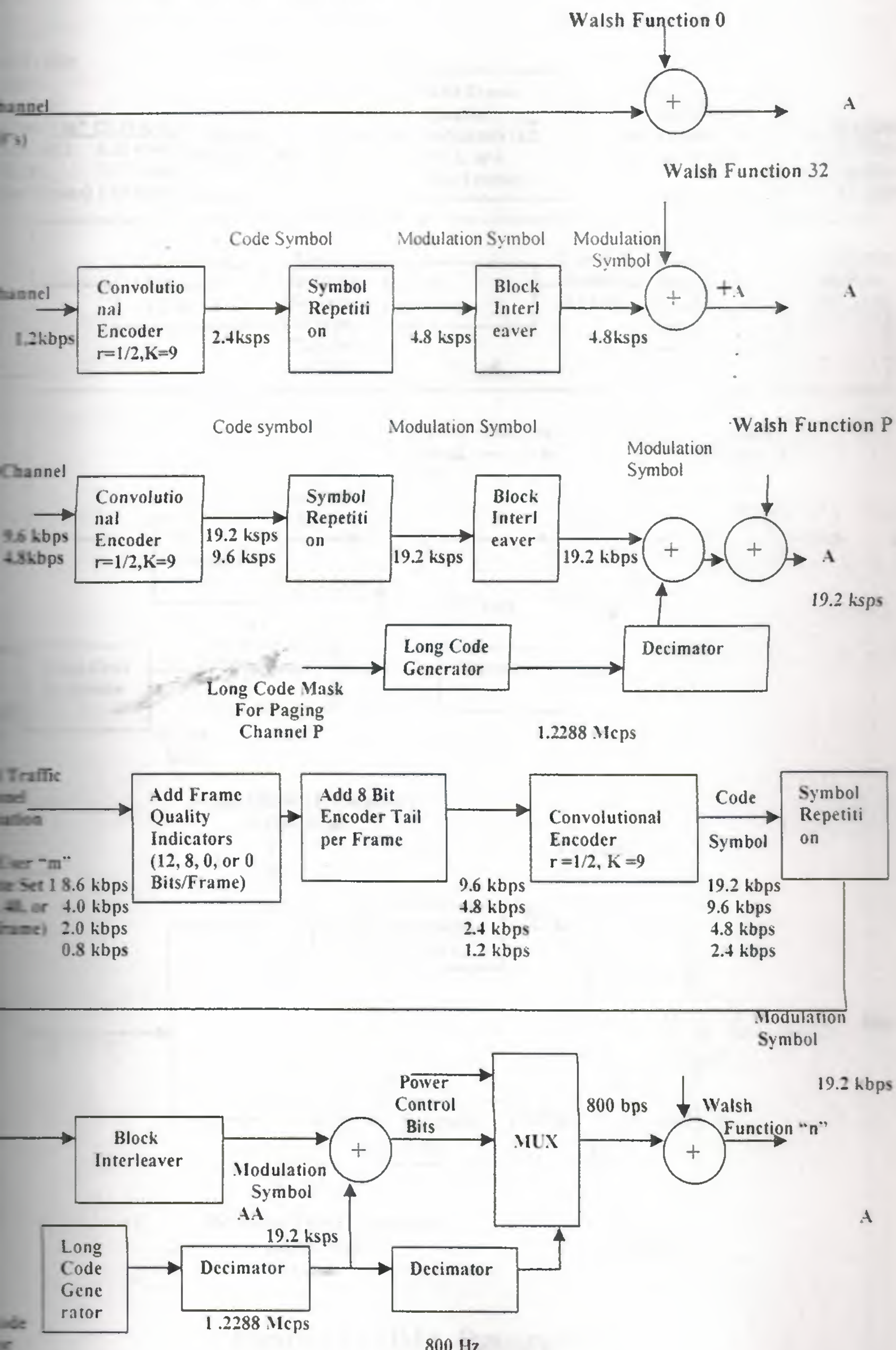


Figure 3.5 CDMA Framing

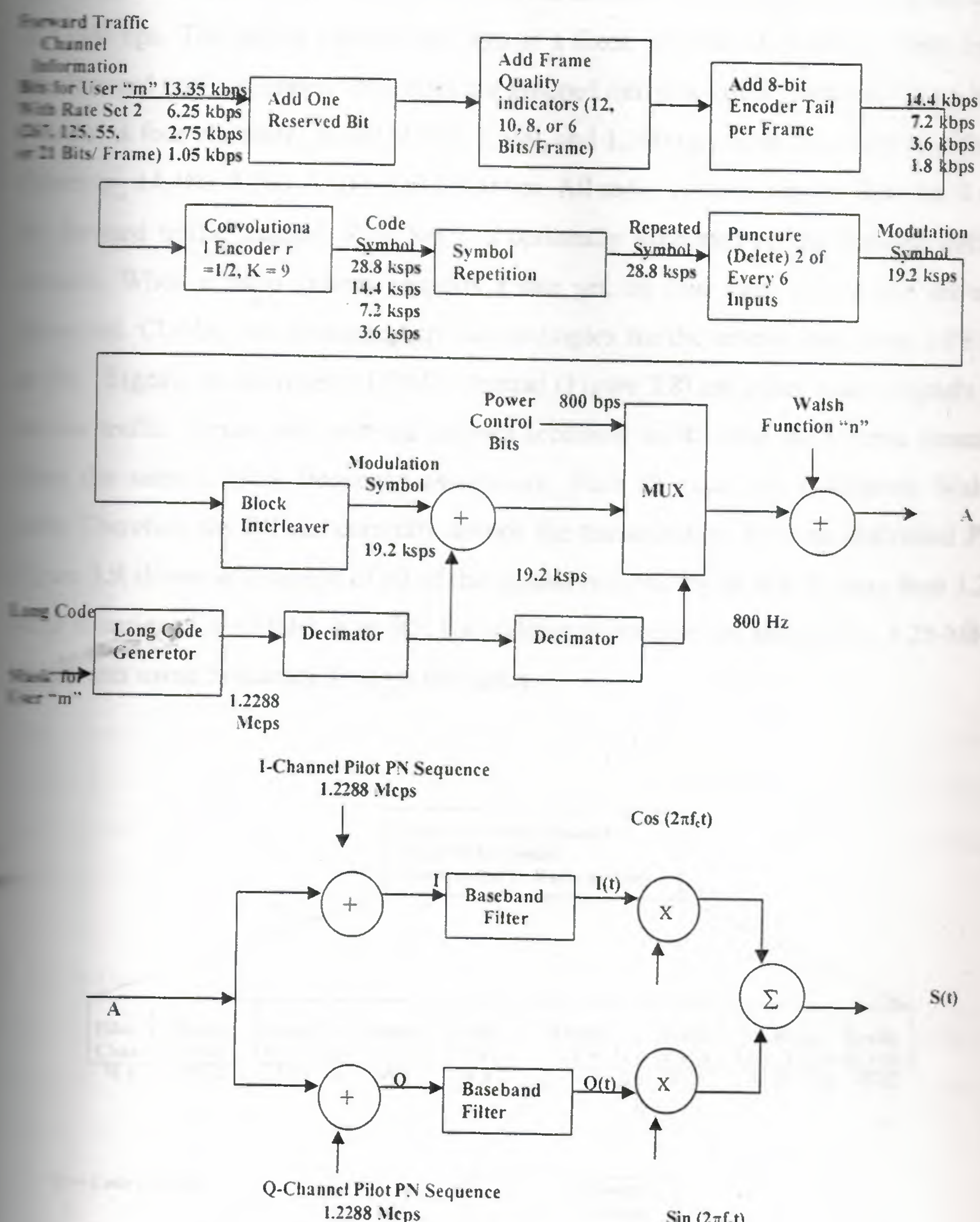


Figure 3.6 CDMA Framing (Part 2)

nc channels, the assignments could be one pilot channel, no paging channels, no
nannels, and 63 traffic channels. The sync channel always operates at a fixed rate
00 bps. The paging channel operates at a fixed data rate of 9,600 or 4,800 bps.
rward traffic channels' data rates are grouped into sets called "rate sets." Rate Set
tains four elements_ 9,600, 4.800, 2,400, and 1,200 bps. Rate Set 2 contains four
ents_ 14.400, 7,200, 3,600, and 1,800 bps. All radio systems support Rate Set 1 on
orward traffic channel. Rate Set 2 is optionally supported on the forward traffic
nel. When a radio system supports a rate set, all four rates of the rate set are
orted. CDMA uses frequency division multiplex for the reverse path from a PS to
S. Signals on the reverse CDMA channel (Figure 3.8) are either access signals or
rse traffic signals. All personal stations accessing an RS over the reverse channel
e the same CDMA frequency assignment. Each PS transmits a different Walsh
e; Therefore the RS can correctly decode the transmissions from an individual PS.
ure 3.9 shows an example of all of the signals received by an RS. If more than 1.25
Hz is assigned to CDMA at an RS, the additional channels are assigned in 1.25-MHz
crements using frequency division multiplex.

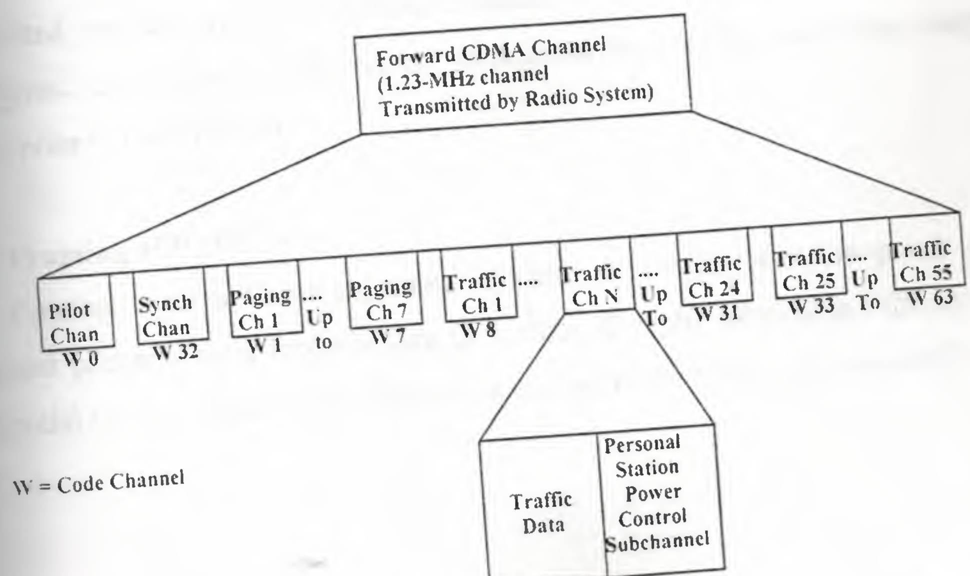


Figure 3.7 Example of a Forward CDMA Channel Transmitted by a RS

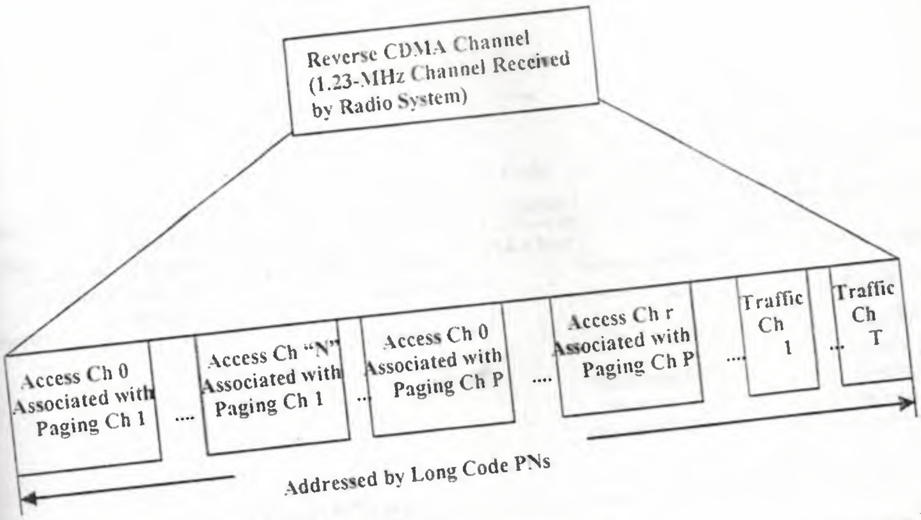


Figure 3.8 Example of Logical Reverse CDMA Channel Received at an RS

The reverse CDMA channel has the overall structure shown in Figures 3.10 and 3.11. Data transmitted on the reverse CDMA channel is grouped into 20-ms frames. All data transmitted on the reverse CDMA channel is convolutionally encoded, block interleaved, modulated by the 64-ary orthogonal modulation, and direct-sequence spread prior to transmission.

3.2.3 Framing of PACS

The Personal Access Communications System, or PACS, is a low-power system designed primarily as a low mobility (less than 30 mph) solution to PCS for use in residential wireless applications and other areas where the handset moves slowly.

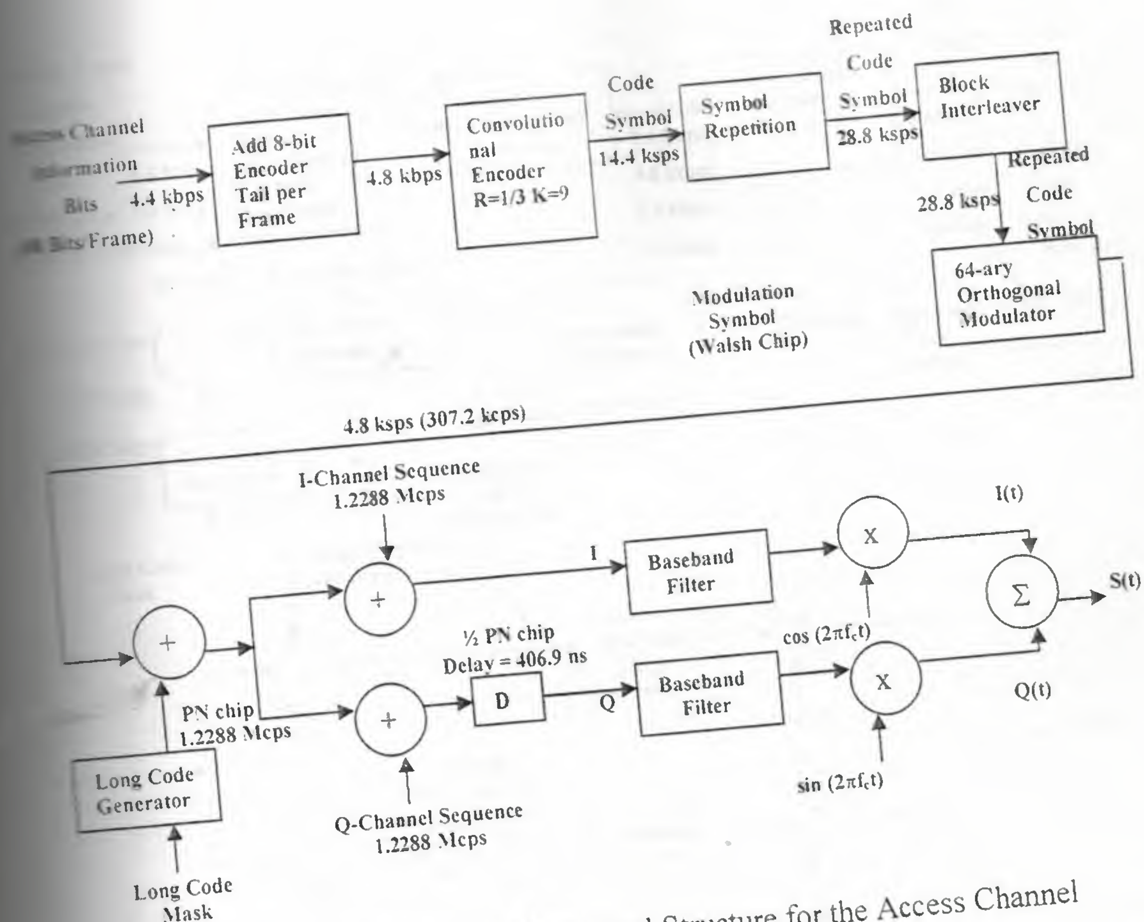


Figure 3.9 Reverse CDMA Channel Structure for the Access Channel

PACS uses Time Division Multiple Access (TDMA) with Frequency Division Duplex (FDD) on 300-kHz channels. Although the band plan shows channels on 100-kHz spacing, only every third channel can be used in a given region. The extra channels are identified to solve guard-channel problems at band edges. Once the channel at the band edge is established, all other channels increment in 300-kHz spacing. PACS uses a basic

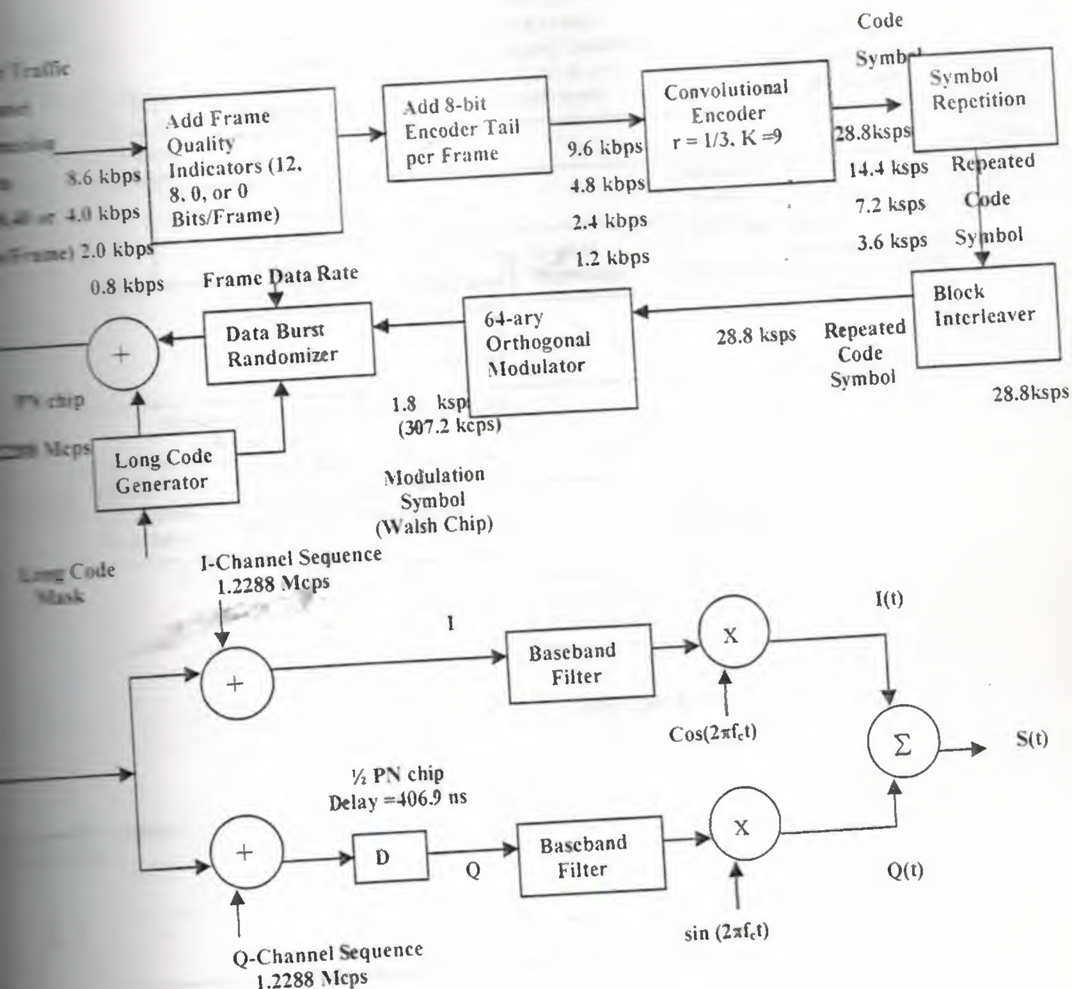


Figure 3.10 Reverse CDMA Channel for Traffic Channels with Rate Set 1

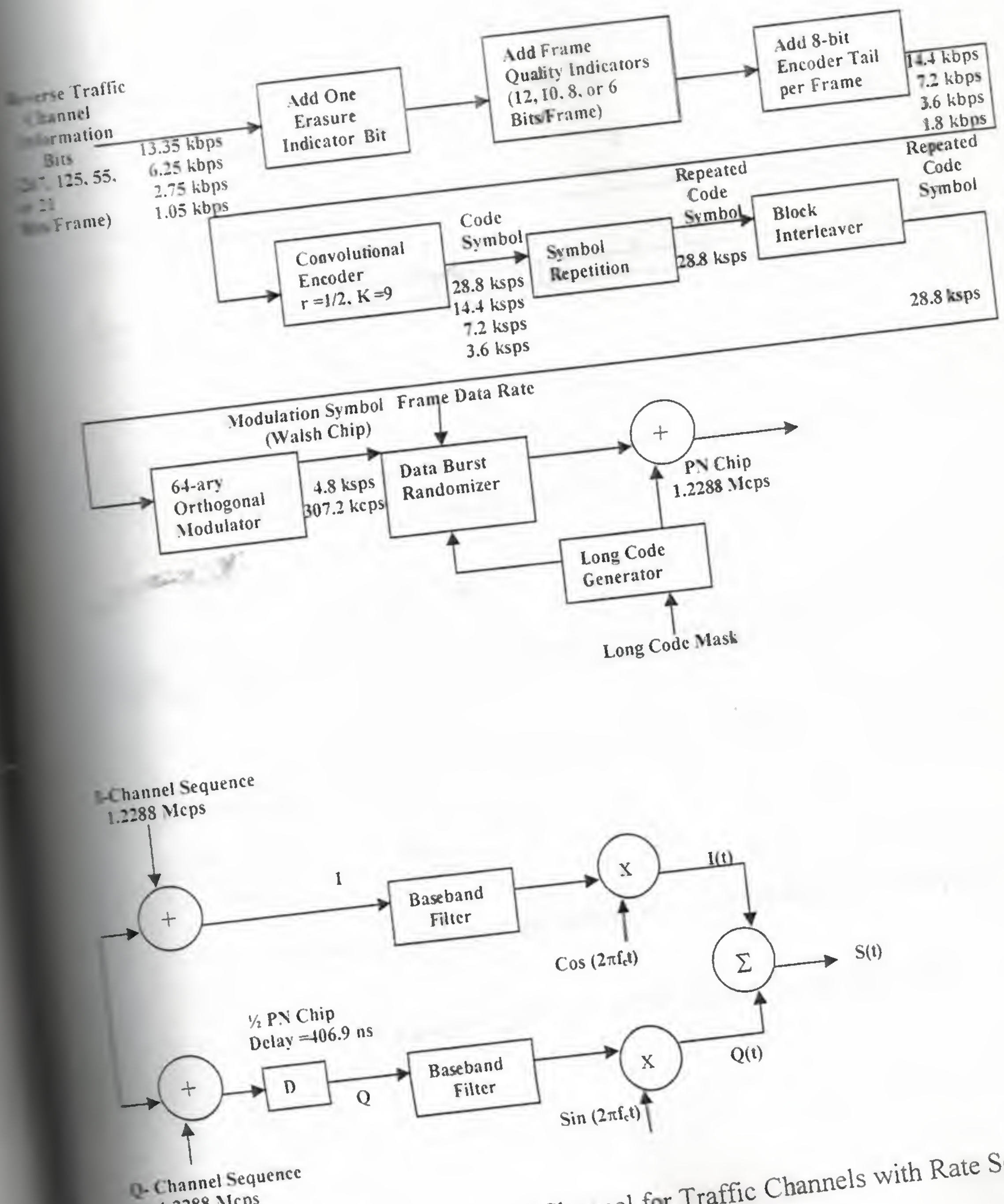
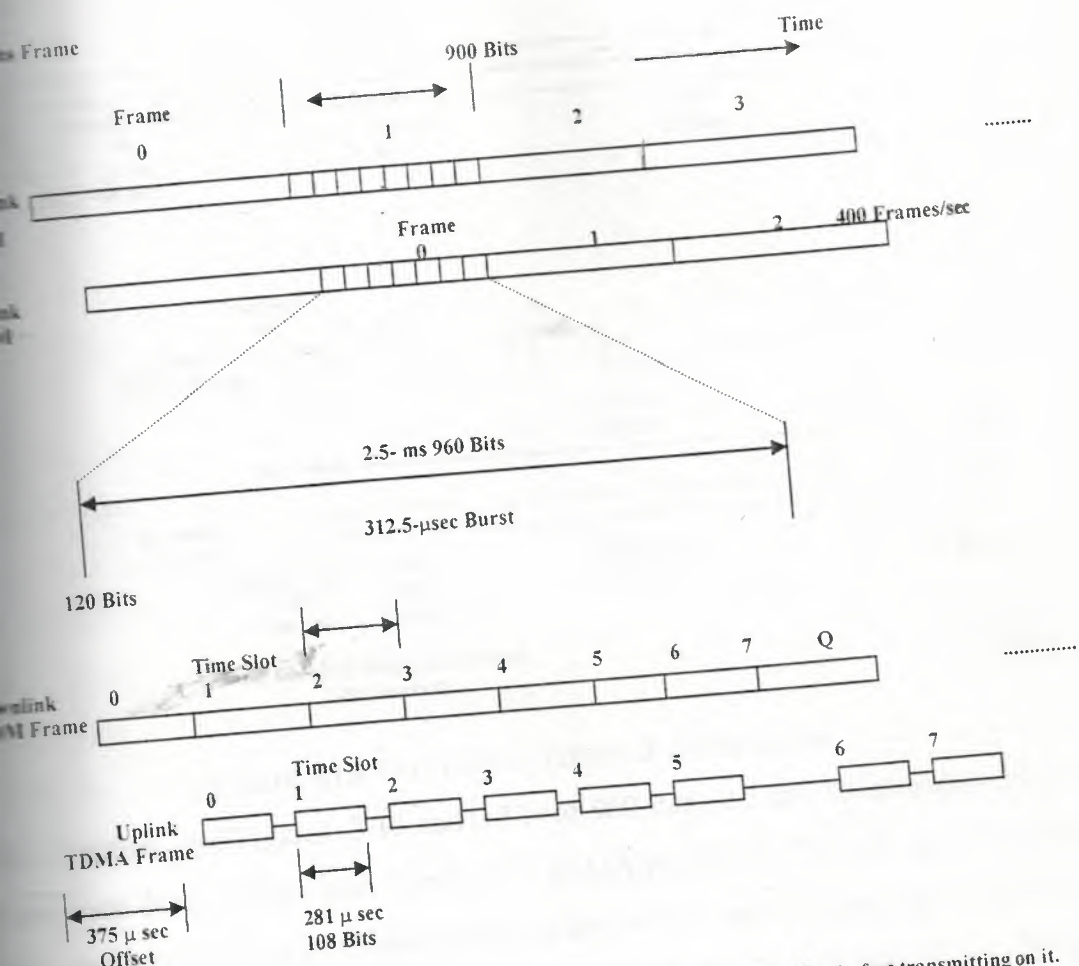


Figure 3.11 Reverse CDMA Channel for Traffic Channels with Rate Set 2



Note: Time slots will be "filled" only on the uplink if a subscriber unit (SU)is in fact transmitting on it.

Figure 3.12 PACS TDM/TDMA Frame Structure

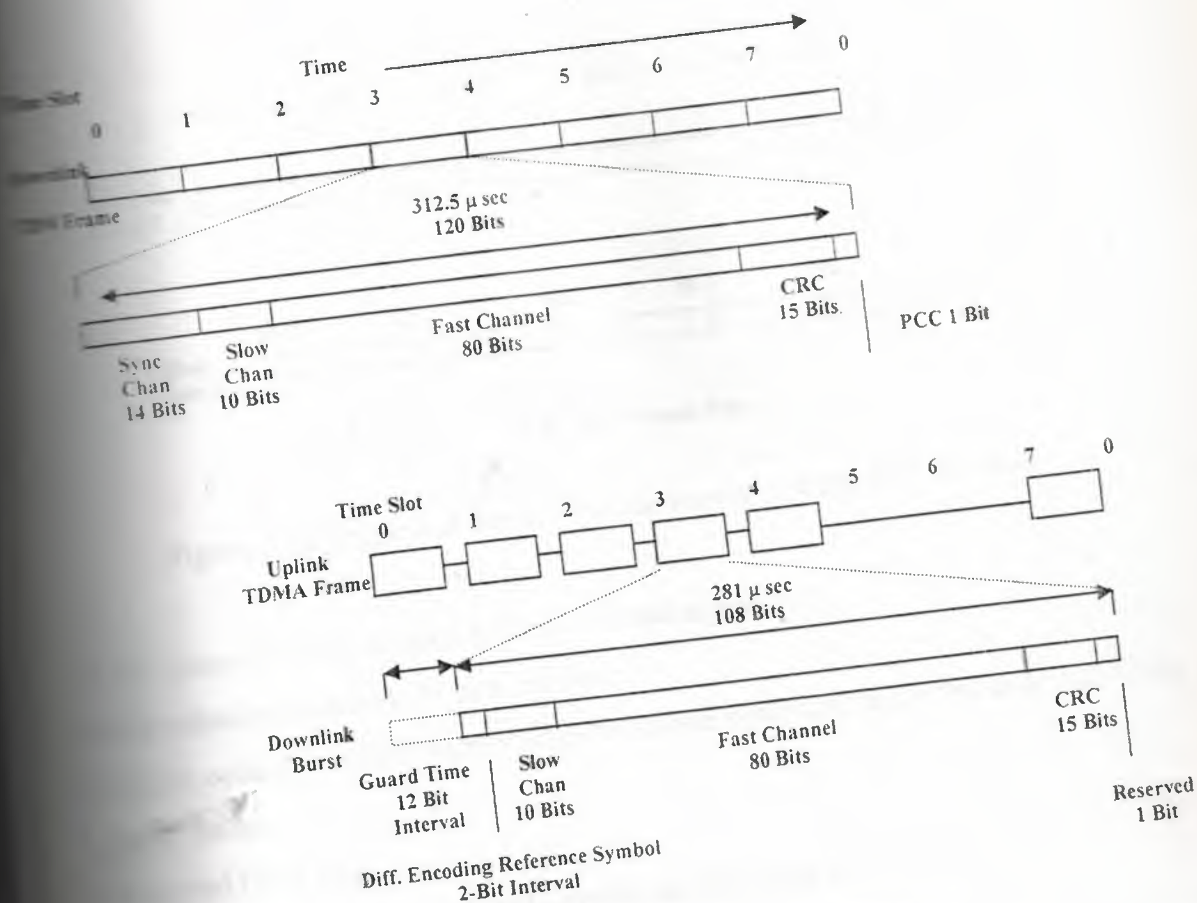


Figure 3.13 PACS TDM/TDMA Burst Structure

frame structure (see Figures. 3.12 and 3.13) of 960 bits (2.5 ms) to minimize speech transmission delay. Within each frame are 8 TDMA bursts of 312.5 μsec each. A super frame consists of a group of 8 frames and supports subrate multiplexing. The overall bit rate is 384 kbs that is transmitted at 2 bits per symbol . The PS transmits a 281.2-microsecond burst with a 31.3-microsecond guard time. To account for propagation delays, the PS delays its burst by 375 ± 5.2 microseconds with respect to the received burst from the RS (see Figure 3.14). The delay also allows the PS to operate without a duplexer.

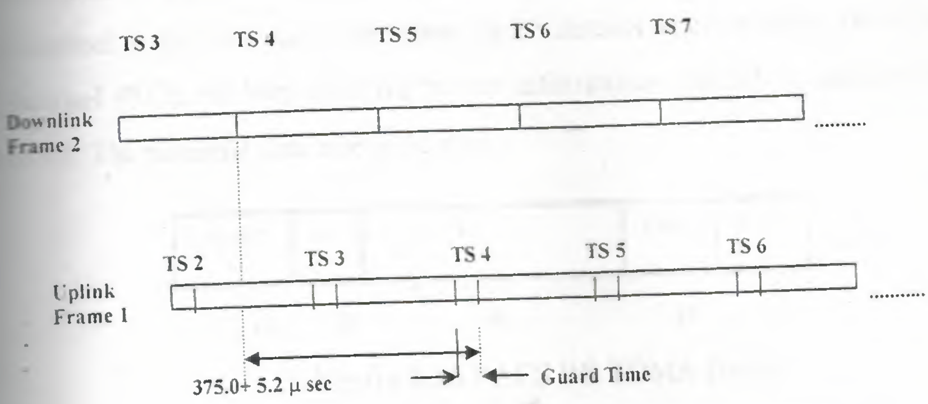


Figure 3.14 Frame and Burst Offset as measured at the PS's Antenna

e RS transmits bursts (Figure 3.15) of 120 bits as follows:

nchronization channel: 14 bits, used as:

Unique code (7 bits) that carries an initial frame synchronization that may vary from me to frame.

Reserved bit (1 bit)

Frame number (3 bits) numbered sequentially from 000 to 111.

Time slot number (3 bits) numbered sequentially from 000 to 111.

ow Channel (SC): 10 bits, used for signaling and control. The nominal rate is 4.0 bs. Bit 1 of this channel is the Word Error Indicator (WEI) that is set when the RS etects an error on the uplink.

ast Channel (FC): 80 bits, used for bearer information (speech or data) or signaling nd control. The nominal data rate is 32 kbs.

yclic Redundancy Code (CRC): 15 bits, used to check the SC an FC on each burst sing a (105, 90) code.

ower Control Channel (PCC): 1 bit, that controls the PS power on the uplink.

The PS transmits on its assigned time slot on the uplink and sends the following nformation in each burst (Figure 3.16).

Guard time: Equivalent to 12 bits, to avoid overlapping with adjacent transmission from other PSs. No signal is sent during the guard time.

Differential decoder start bit: 2 bits, with value undefined, that serve to derive a phase reference to decode subsequent bits.

Slow channel: 10 bits, used for signaling and control. The nominal rate is 4.0 kbs. Bit 1 of this channel is the WEI that is set when the PS detects an error on the downlink.

Fast channel (FC): 80 bits, used for bearer information (speech or data) or signaling and control. The nominal data rate is 32 kbs.

SYNC	SC	FC	CRC	PCC
14	10	80	15	1

Figure 3.15 PACS RS TDMA Burst

GT	DD Start	SC	FC	CRC	PCC
12	2	10	80	15	1

Figure 3.16 PACS RS TDMA Burst

Cyclic redundancy code: 15 bits, used to check the SC and FC on each burst using a (105, 90) code.

Reserved bit: 1 bit, will be set to "0." Each time slot of 2.5 ms has 80 bits of data; Thus, there are 400 frames per second and the nominal data rate of the system is 32 kbs. This is the normal data rate used for the ADPCM speech. The PACS system can transmit data at rates of 4 kbs, and 16 kbs by using subrate multiplexing. With subrate multiplexing, a PS transmits in fewer than 8 frames per superframe. This lower data rate can be used for data traffic or for alternative speech encoding methods. At initial deployment, PACS will support only ADPCM. The rate of a channel and its time slot assignment are identified by:

1) The letters "TS" followed by
2) The number of the time slot (0 to 7) and
3) The letters "a" to "p."

Table 3.1 summarizes the use of a traffic channel at different data rates. See Figure 3.17 for examples of channel usage at each data rate.

Table 3.1 Traffic Channel Designation for PACS

c Channel ta Rate	PS or RS Uses Frame Numbers	Time Slot Designation	Example, if RS or PS Uses Time Slot 4
2 kbs	0,1,2,3,4,5,6,7	TS # -a	TS 4 –a
6 kbs	0,2,4,6	TS # -b	TS 4 –b
6 kbs	1,3,5,7	TS # -c	TS 4 –c
3 kbs	0,4	TS # -d	TS 4 –d
3 kbs	1,5	TS # -e	TS 4 –e
3 kbs	2,6	TS # -f	TS 4 –f
3 kbs	3,7	TS # -g	TS 4 –g
4 kbs	0	TS # -h	TS 4 –h
4 kbs	1	TS # -i	TS 4 –i
4 kbs	2	TS # -j	TS 4 –j
4 kbs	3	TS # -k	TS 4 –k
4 kbs	4	TS # -l	TS 4 –l
4 kbs	5	TS # -m	TS 4 –m
4 kbs	6	TS # -n	TS 4 –n
4 kbs	7	TS # -o	TS 4 –o

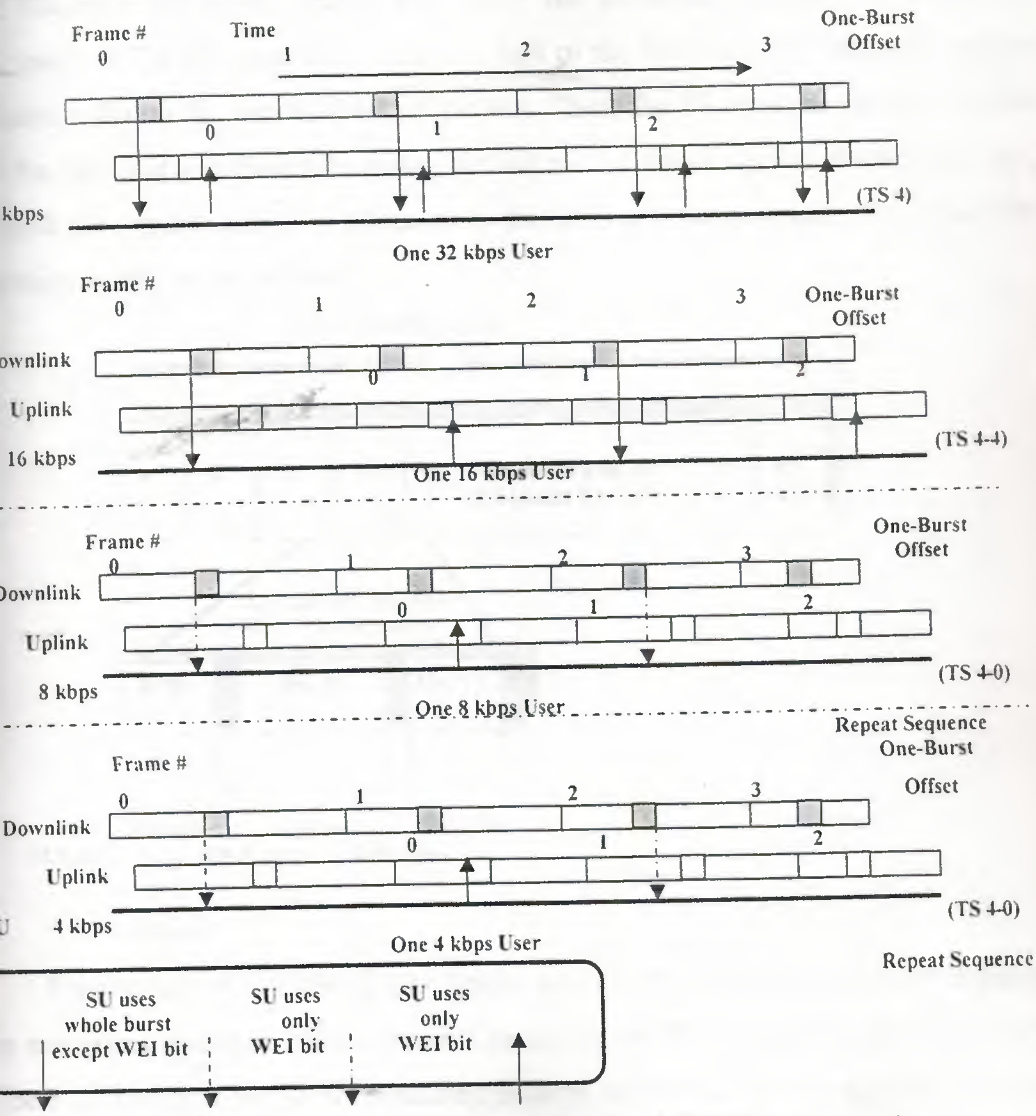


Figure 3.17 PACS Subrate Traffic Channel and WEI Bit Sequencing

aming of PCS 2000

nipoint, composite CDMA/TDMA, PCS 2000 system employs a combination A, FDMA, and CDMA for multiple user access to the PCS network. Within a ll, TDMA is used to separate users. FDMA is used to provide a greater area of e, or to provide a greater capacity for densely populated regions. In those areas, e cells or sectorized cells use FDMA. The Direct Sequence Spread Spectrum) method of CDMA is also used for each RF link to reduce cochannel rence between cells reusing the same RF carrier frequency. In PCS2000, the A frame and time slot (channel) structure is based on a 20-ms polling loop for user to the RF link (see Figure 3.18). Both the PS and the RS transmit on the same ency using Time Division Duplex (TDD). The 20-ms frame supports 32 full x channels within the frame. Each time slot (channel) supports an 8-kbs full x user[14].The RS transmits in the first half of the TDMA/TDD time slot, and the ansmits during the second half of the slot. Thus, the PS receives during the first of the time slot and transmits during the last half. A small portion of each time slot, gnated the "guard time," is allocated to the end of a transmission to account for agation delays in the system.

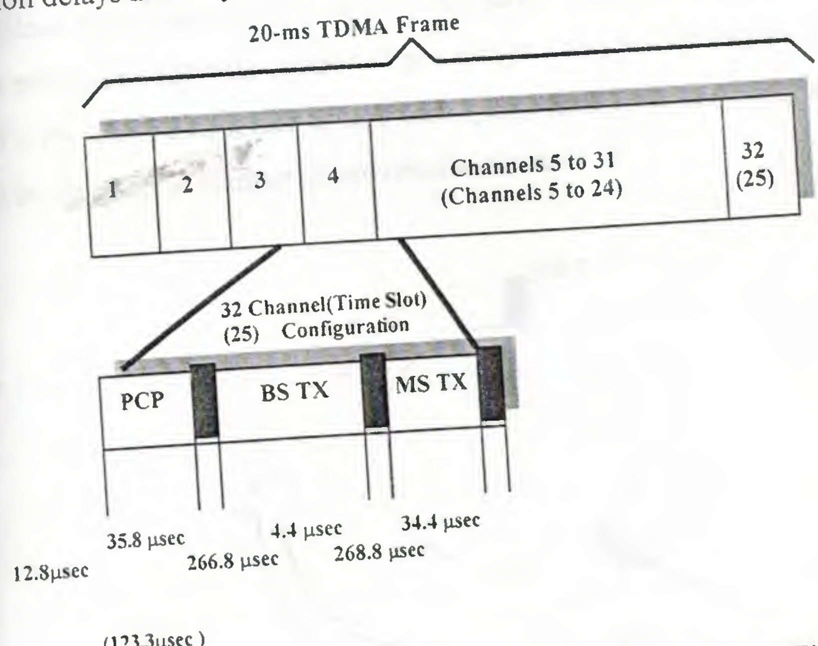


Figure 3.18 PCS-2000 TDMA Frame and TDMA Channel Time Slot Structure

Power control is managed by the Power Control Pulse (PCP) signal received from the PS. The PCP serves as a channel sounding pulse to determine link propagation loss and to serve as a measurement ok link quality for the PCS 2000 power control subsystem. It

o determine which of the multiple antennas to use for the spatial diversity
permits spatial diversity control to be updated during each TDMA time slot

arge cell system, where propagation delays are long, an RS may optionally channels (time slots). At the boundary between large and smaller cell 2- and 25- channel cells), the frequencies are assigned in nonoverlapping he 32-channel cells and the adjacent 25-channel cells.After sufficient etween the 32-channel/25-channel boundary, the channels can then be no restrictions. Each of the 32 full duplex channels per frame are provided s data capability (for voice or data) for use during the 20-ms frame period. es a TDMA burst data rate of 512 kbs in TDD mode. Guard timing and verhead bits required for link protocols and control functions make the total st rate of 781.25 kbs. An individual PS may negotiate for multiple or slots in the polling loop to have data rates higher or lower than 8 kbs. The may take place at any time via signaling traffic. The slots, if available, are the RS to a PS. Those PSs using higher rates than 8 kbs use additional me slots) per TDMA frame (see Figure 3.19). For example, a PS using two ime slots) operates at a 16-kbs data rate, versus 8 kbs for one channel (time maximum data rate supported per user is 256 kbs full duplex or 512 kbs half a PS wants a data rate of less than 8 kbs, time slots may be granted in frames y an integral number of intermediate frames

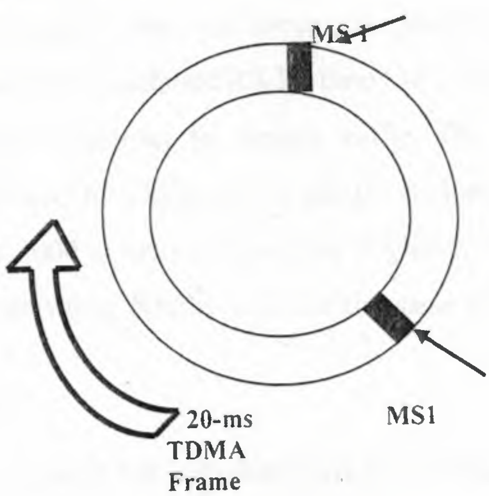
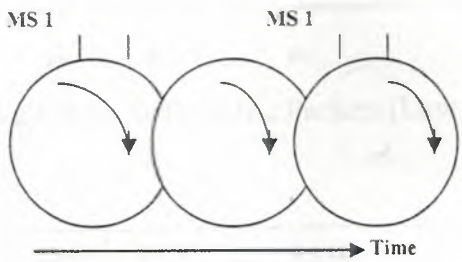


Figure 3.19 PCS 2000 Multiple TDMA Channel (Time Slots) per User

3.20). The maximum limit on the separation of slots allocated to a single PS ds for a data rate of 320 bits per second. To maximize system capacity, the ie times for all RSs within the same geographical area will be synchronized. mmon method for synchronization is to utilize a Global Positioning System ver at the RS controller to generate the primary-reference timing marker for frame timing. The synchronization of all RSs within a given area allows an er to temporarily turn off any TDMA time slot of a given cell that may be with a neighboring cell. It also facilities switching a PS to a different time ent time slot has interference from an adjacent cell using the same time slot. synchronize to a new RS within one channel (time slot) and is capable of ng with multiple RSs when those RSs are synchronized to a common digital



e 3.20 PCS-2000 Submultiple TDMA Channels (Time Slots) per User

S and the PS can transmit either of two packet types: Base poll packets and packets. Poll packets (Figure 3.21) are 200 bits in length and consist of 16 ader, 168 bits of data and 16 bits of a frame check word. For traffic packets s 3.22, 3.23, and 3.24), the system can operate in any of three modes: The ong packets, and PS sends short packets, RS sends short packets and PS packets, or RS and PS both send symmetric packets (symmetric bandwidth In all three cases, one combined RS/PS time slot is used and the D channel is gnaling and the B channel for bearer traffic. The system can optionally or checking and use the additional 16 bits for B-channel data. The frequency n for the PCS-2000 system is three (see Figure 3.25).Adjacent cells differ equency or code word; Nearby cells on the same frequency use a different id interference.

ing of TDMA

IA system, as supported at both 900 MHz for cellular and at 1,800 MHz for verall 30-kHz RF channel structure from analog cellular is maintained. Each l has six time slots and supports three PSs. In the future with half-rate

, an RF channel will be able to support up to six PSs. As in analog cellular, the are designated as control channels or traffic channels.

HDR	DATA	FCW
16	168	16

Figure 3.21 PCS-2000 Poll Packets

HDR	B-CH	D-CH	FCW
16	8	280	16

Figure 3.22 PCS-2000 Traffic Packets (High Bandwidth Signaling)

HDR	B-CH	D-CH	FCW
16	8	40	16

Figure 3.23 PCS-2000 Traffic Packets (Low Bandwidth Signaling)

HDR	B-CH	D-CH	FCW
16	8	160	16

Figure 3.24 PCS-2000 Traffic Packets (Symmetric Bandwidth Signaling)

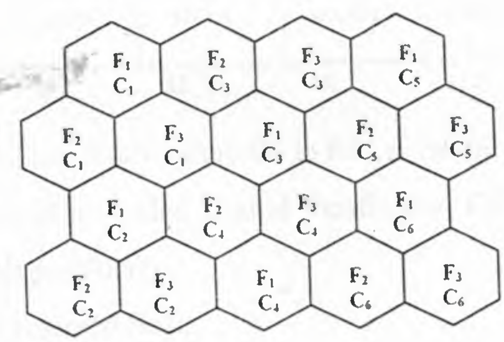


Figure 3.25 Frequency and Code Reuse for PCS-2000 N =3

e 3.26 shows the frame structure of the digital TDMA RF channel frame. The e length is 40 ms (i.e., 25 frames per second), with each frame being 1,944 bits symbols) long. The frame is segmented into six equally sized time slots (1-6) with symbols (or 324 bits). Each full-rate traffic channel utilizes two equally spaced time of the frame (i.e., 1 and 4, 2 and 5, or 3 and 6), whereas each half-rate channel uses time slot of the frame. Both control and traffic channels use a common frame nat with a slot format that is slightly different depending on whether the ion is on a control channel or a traffic channel. The slot format on the traffic s defined in Figures 3.27 and 3.28. The fields in the RS to PS slot are defined

YNC: 28 bits of synchronization

ACCH: 12 bits for the Slow Associated Control Channel

DATA: Two 130-bit fields of data; The data can be digitized voice or data

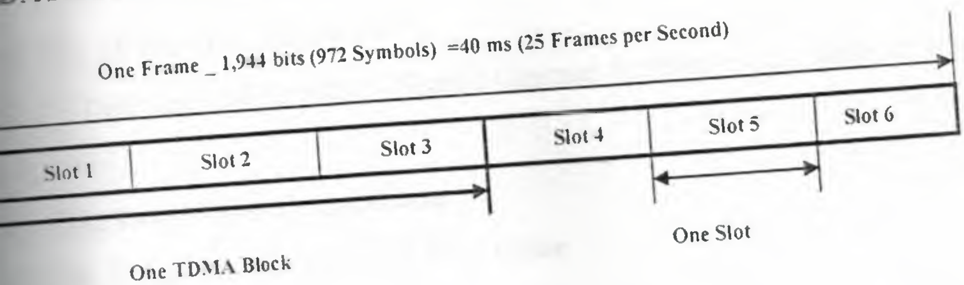


Figure 3.26 TDMA Frame Structure

SYNC	SACCH	DATA	CDVCC	DATA	RSVD =1	CDL
28	12	130	12	130	1	11

Figure 3.27 Time Slot Format: RS to PS on Traffic Channel

G	R	DATA	SYNC	DATA	SACCH	CDVCC	DATA
6	6	16	28	122	12	12	122

Figure 3.28 Time Slot Format: PS to RS on Traffic Channel

- **CDVCC:** 12 bits of a Coded Digital Verification Color Code (similar to the SAT tone for analog cellular).
- **RSVD:** 1 bit of a reserved field.
- **CDL:** 11 bits for a Coded Digital Control Channel Locator.

the reserve traffic channel (from PS to RS), the slot format (Figure 3.28) has the lowing field definitions:

- **G:** Equivalent of 6 bits guard time (the PS does not transmit during this time).
- **R:** Equivalent of 6 bits of ramp time (the PS is powering up).
- **SYNC:** 28 bits of synchronization.
- **DATA:** One field of 16 bits plus two fields of 122 bits; The data can be digitized voice or data.

CH: 12 bits for the slow associated control channel.

CC: 12 bits of a Coded Digital Verification Color Code (similar to the tone for analog cellular).

rd digital control channel (RS to PS), the fields in each slot are (see Figure

C: 28 bits of synchronization.

: Two fields of 12 bits of Shared Channel Feedback (i.e., status of reverse nel).

TA: Two 130-bit fields of data.

SP: 12 bits of Coded Super Frame Phase

VD: 2 bits of a reserved field

erse digital control channel (from PS to RS), the PS can transmit with a mat (Figure 3.30) or an abbreviated format (Figure 3.31). For these slots, the itions are:

SYNC	SCF	DATA	CFSP	DATA	SCF	RSVD
28	12	130	12	130	10	2

Figure 3.29 Time Slot Format: RS to PS on Digital Control Channel

G	R	PREAM	SYNC	DATA	SYNC+	DATA
6	6	16	28	122	24	122

Figure 3.30 Time Slot Format: PS to RS on Digital Control Channel

G	R	PREAM	SYNC	DATA	SYNC+	DATA	R	AG
6	6	16	28	122	24	78	6	38

e 3.31 Abbreviated Time Slot Format: PS to RS on Digital Control Channel

- **G:** Equivalent of 6 bits guard time (the Ps does not transmit during this time).
- **R:** Equivalent of 6 bits of ramp time (the PS is powering up).
- **PREAM:** 16 bits of a preamble.
- **SYNC:** 28 bits of synchronization.
- **DATA:** Two fields of 122 bits of data for a normal slot or one field of 122 bits and one field of 78 bits for an abbreviated slot; The data can be traffic data (e.g., voice) or control data.
- **SYNC+:** Additional synchronization.
- **R:** Equivalent of 6 bits of ramp time (the PS is powering down).
- **AG:** Guard time for abbreviated burst.

S will adjust its timing to account for delays in the system. The offset is equal to ne slot plus 45 symbols (i.e., 207 symbol periods or 414 bit periods). Time slot 1 ne N of the PS occurs 207 symbol periods (or 414 bit periods) before time slot 1 ne N of the RS. The delay can be further refined under command of the RS. When S is transmitting on a reverse control channel (access channel), guard times are d to prevent interference. The PS may be commanded to send abbreviated bursts handing off to other channels where the timing delays are not known.

Framing of Wideband CDMA

Wideband CDMA (W-CDMA) system is similar to the CDMA system of IS-95. DMA can support bandwidths of 5 MHz, 10 MHz, and 15 MHz. The basic ation of the system is the same independent of the bandwidth chosen. The overall k diagrams of the system are similar to the CDMA system for the operation of the em). The frequency reuse factor for W-CDMA is 1 as it is for CDMA. Tables 3.2 3.3 summarize the characteristics of the W-CDMA system.

Table 3.2 System Characteristics of Wideband CDMA at Various Bandwidths

Bandwidth (MHz)	Number of Channels	PN Rate (Mcps)	Code Rate (Mcps)	Data Rates Supported (kbs)	Channel Selection Method
5.0	256	4.096	4.096	2.0-64.0	Walsh 64x4*
10.0	512	8.192	8.192	2.0-64.0	Walsh 64x8*
15.0	768	12.288	12.288	2.0-64.0	Hadamard 48x4 and Hadamard 96x8**

- The 64 Walsh codes are repeated four times in normal and inverted patterns to construct a total of 256 different codes.
- The 64 Walsh codes are repeated eight times in normal and inverted patterns to construct a total 512 different codes.
- The codes are based on Hadamard 48 codes to generate a Hadamard 96 code which is then repeated in normal and inverted patterns to form up to 768 different codes.

Table 3.3 Channel Usage of W-CDMA at Various Bandwidths

Bandwidth (MHz)	Number of Channels	Pilot Channel Number	Sync Channel Number	Paging Channel Numbers
1.25	64	0	32	1-7 (no skipping)
5.0	256	0&64	32&/or 96	1-7 (no skipping)
10.0	512	0&128	64&/or 192	1-7 (no skipping)
15.0	768	0	384	1-7 (no skipping)

5-MHz Bandwidth System

In the 5-MHz bandwidth system, a fixed chip rate of 4.096 Mcps is used with 128 Walsh codes. A code channel that is modulated by Walsh function "n" will be assigned to code channel number "n" (n=0 to 64 for 64 kbs, n=0 to 127 for 32 kbs, and n=0 to 255 for 16 kbs). Tables in the appendix define the codes used for 64 kbs, 32 kbs, and 16 kbs. respectively. Walsh codes of length greater than 64 are generated by combinations of the basic Walsh 64 and inverted versions of the Walsh 64 code (see Tables in appendix). The Walsh code spreading sequence will repeat with a period of the duration e forward traffic channel code symbol. In a manner similar to the 1.25-MHz A system, the pilot channel is always assigned to code channel number 0 and/or hen the sync channel is present, it will be assigned code channel number 32 r 96. When paging channels are present, they will be assigned to code channel bers 1 through 7 (inclusive) in sequence. The remaining code channels are available ssignment to the forward traffic channels. When an Rs supports multiple forward MA channels, then frequency division multiplex is used. The maximum number of nels for 64 kbs is 64, for 32 kbs is 128, and for 16 kbs is 256. The same Walsh code ot be used simultaneously for two different data rates.

-MHz Bandwidth System

e 10-MHz system, a fixed chip rate of 8.192 Mcps is used with 256 Walsh codes. A de channel that is modulated by Walsh function "n" will be assigned to code channel mber "n" (n =0 to 127 for 64 kbs, n =0 to 255 for 32 kbs, and n =0 to 511 for 16 kbs). e of the time-orthogonal Walsh codes as defined in Tables in Appendix will be used r 64 kbs, 32 kbs, and 16 kbs, respectively. In a manner similar to the other CDMA ndwidths, the pilot channel is always assigned to code channel number 0 and/or 128, e sync channel is assigned to 32 and/or 96. And the paging channels are assigned to ode channel numbers 1 through 7 (inclusive) in sequence, with the remaining code channels available for assignment to the forward traffic channels. The maximum number of channels for 64 kbs is 128, for 32 kbs is 256, and for 16 kbs is 512. The same Walsh code cannot be used simultaneously for two different data rates.

15-MHz Bandwidth System

In the 15-MHz system, 768 Hadamard codes are used with a fixed chip rate of 12.288 Mcps. A code channel that is spread using Hadamard code "n" will be assigned to code index number "n" (n = 0 to 191 for 64 kbs, n = 0 to 383 for 32 kbs, and n = 0 to 767 for 16 kbs). One of the time-orthogonal Hadamard codes as defined in Tables in appendix will be used for 64 kbs, 32 kbs, and 16 kbs, respectively. The longer Hadamard codes are constructed like the longer Walsh codes. Similar to the other CDMA bandwidths, code index 0 will always be assigned to the pilot channel. When the sync channel is present, it will be assigned code index 384. When paging channels are present, they will be assigned to code indexes 1 through 7 (inclusive in sequence). The remaining code indexes are available for assignment to the forward traffic channels. There maximum number of channels for 64 kbs is 192 channels, for 32 kbs is 384, and for 16 kbs is 768.

other bandwidths, the same Hadamard code cannot be used simultaneously
ferent data rates.

ch Coding

ne network is based on sending voice using digital Pulse Code Modulation 54 kbs and sending data at rates of 64 kbs or higher multiples of 64 kbs. Many og facilities still exist, especially in residential areas, and use voice band t rates up to 28.8 kbs for data and analog electrical signals for voice. At the ice the analog voice and analog data is converted to digital signals using PCM nally, using modem pools for data. It would be nice if the identical systems used for PCS. Unfortunately the error rates on the PCS channels are many magnitude higher than those of the copper or fiber-optic cables. In addition, nefficient for use over scarce and expensive radio channels. Therefore, all of l PCS systems use some efficient method of voice coding and extensive error techniques to overcome the harsh nature of the radio channel. This section will various means used for each of the North American protocols to send signals radio channel.The simplest form of waveform-coding scheme is the linear which the speech signal is sampled, quantized, and encoded. This approach is sed for analog-to-digital conversion of a signal. The speech signal is band the frequency range of 200 to 3,300 Hz. To achieve telephone quality speech, er sample are required at a sampling rate of 8,000 samples per second. By using nic PCM, 8 bits per sample are sufficient. Each sample is quantized into one of ls. Two widely used variations of PCM for telephone quality speech (μ-law and CM) are based on a nonuniform quantization of the signal amplitude according garithmic scale rather than a linear scale. Such coders utilize the static ristics of amplitude nonstationary in speech to achieve good quality at a bit rate bs. This is the basis of PCM. High bit rates are not attractive for wireless .Better results are obtained with differential coders, in which a dynamic range of ssion can be applied, such as Adaptive Predictive Coding (APC) and Adaptive ntial Pulse Code Modulation (ADPCM). The reason for these coders is to a better signal-to-quantization noise performance over PCM.Differential coders e error signals as the difference between the input speech samples and the onding prediction estimates. The error signals are quantized and transmitted.

and APC differential coders are often used for intermediate bit rates-16 to 32

employs a short-term predictor that models the speech spectral envelope. achieves network-quality speech (Mean Opinion Score [MOS]of 4.1 or better) s. This is a low-complexity coder of reasonable robustness with channel bit es in the range of 10^{-3} to 10^{-2}. The ADPCM coder is well suited for wireless pplications. APC employs both short- and long- term prediction in a differential structure. APC out performs ADPCM at 16 kbs and offers communication-speech (MOS 3.5 to 4.0) at a bit rate as low as 10 kbs. Introduction of noise and post filtering in ADPCM and APC reduces the subjective loudness of ation noise. PCM, ADPCM, and APC operate in the time domain. No attempt is o understand or analyze the information that is being sent. Redunancy removal ques have been used successfully in the frequency domain. Frequency domain orm-coding algorithms decompose the input speech signal into sinusoidal onents with varying amplitudes and frequencies. Thus, the speech is modeled as a arying line spectrum. Frequency domain coders are systems of moderate lexity and operate well at a medium bit rate (16 kbs). When designed to operate in ange of 4.8 to 9.6 kbs, the complexity of the approach used to model the speech rum increases considerably. The other class of speech-coding techniques consists lgorithms called vocoders which attempt to describe the speech production hanism in terms of a few independent parameters serving as the information-bearing als. These parameters attempt to model the creation of the voice by the vocal tract, mpose the information, and send it to the receiver. The receiver attempts to modes lectronic vocal tract to produce the speech output. The model operates this way: oders consider that speech is produced from a source-filter arrangement. Voice ech is the result of exciting the filter with a periodic pulse train similar to the pulses erated by the vocal tract. Vocoders operate on the input signal, using an analysis cess based on a particular speech production model, and extract a set of source-filter ameters which are encoded and transmitted. At the receiver, they are decoded and d to control a speech synthesizer that corresponds to the model used in the analysis ocess. Provided that all the perceptually significant parameters are extracted, the nthesized signal, as perceived by the human ear, resembles the original speech signal. onspeech signals are often not modeled well, so this method works poorly for analog odems. Vocoders are medium-complexity systems and operate at low bit rates.

typically 2.4 kbs, with synthetic-quality speech. Their poor-quality speech is due to the oversimplified source model used to drive the filter and the assumption that the source and filter are lineraly independent. In the bit rates from about 5 kbs to 16 kbs, the best speech quality is obtained by using hybrid coders that use suitable combinations of waveform-coding techniques and vocoder techniques. A simple hybrid coding scheme for telephone-quality speech with a few integrated digital signal processors is the Residual Excited Linear Prediction (RELP) coding. This belongs to a class of coders known as analysis-synthesis coder based on Linear Predictive Coding (LPC).

RELP systems employ short-term (and in certain cases, long-term) linear prediction to formulate a difference signal (residual) in a feed-forward manner. RELP systems are capable of producing communications-quality speech at 8 kbs. These utilize either pitch-aligned high-frequency regeneration procedures or full-band pitch prediction in time domain to remove the pitch information from the residual signal prior to band limitation/decimation. At bit rates < 9.6 kbs, the quality of the recovered speech signal can be improved significantly by an Analysis by Synthesis (AbS) optimization procedure to define the excitation signal. In these systems both filter and the excitation are defined on a short-term basis using a closed-loop optimization process that minimizes a perceptually weighted error measure formed between the input and decoded speech signals. Table 3.5 summarizes the speech-coding method used by each of the North American protocols.

3.4 Frequency Allocation

The frequency allocation for cellular is 824-849 and 869-894 MHz. In this band, there are two carriers: A wireline carrier (B-band) and a nonwireline carrier (A-band). The frequencies used for each are defined in Table 3.5. The frequency allocation for PCS is 1850-1910 MHz and 1930-1990 MHz with a constant 80-MHz spacing between transmit and receive frequencies. The spectrum is segmented into six blocks. Table 3.6 summarizes the frequency bands of each block. Each of the air interfaces use the spectrum in different ways. TDMA segments the spectrum into channels, whereas CDMA uses a wider bandwidth than TDMA and uses different codes to select a channel.

Table 3.4 Speech Coding Used in PCS

	Coding Type	Coding Rate	Frame size	Frame/sec.
	None	analog	-	-
	QCELP (QUALCOMM CELP)	14.4 kbs	-	-
	ADPCM	32 kbs	80 bits	400
A	VSELP	8 kbs	320 bits	25
00	ADPCM PCM PCS HCA	32 kbs 64 kbs 8 kbs	640 bits 640 bits 640 bits	50 (4 slots used) 50 (2 slots used) 50 (1 slots used)
MA	ADPCM	32 kbs	-	-

Table 3.5 AMPS Frequency Spectrum (after Expansion)

ncy Band	MHz	PS Transmitter Band (MHz)	RS Transmitter Band (MHz)
A"	1.0	824.000-825.000	869.000-870.000
A	10.0	825.000-835.000	870.000-880.000
A'	1.5	845.000-846.500	890.000-891.500
B	10.0	835.000-845.000	880.000-890.000
B'	2.5	846.500-849.000	891.500-894.000

Table 3.6 Definition of Bands for PCS

Designator	PS Transmit Band (MHz)	RS Transmit Band (MHz)	Bandwidth (MHz)
A	1850-1865	1930-1945	15
D	1865-1870	1945-1950	5
B	1870-1885	1950-1965	15
E	1885-1890	1965-1970	5
F	1890-1895	1970-1975	5
C	1895-1910	1975-1990	15

nnel Spacing and Frequency Tolerance for Analog Cellular

ellular uses digital control channels and analog voice channels with the defined in Table 3.7. The frequency tolerance for an analog cellular PS is s per million (± 2.000 Hz at 800 MHz). The frequency tolerance for an analog S is ± 1.5 parts per million (± 1.200 Hz at 800 MHz).

Table 3.7 AMPS Channels with Expanded Spectrum

cy	Width (MHz)	Number of Channels	Channel Number	PS Transmitter Center Frequency (MHz)	RS Transmitter Center Frequency (MHz)
ed		1	(990)	(824.01)	(869.01)
	1.0	Voice: 991-1023 (33)	991-1023	824.04-825.00	869.04-870.00
	10.0	Voice: 001-312 (312) control :313-333 (21)	1-333	825.03-834.99	870.03-879.99
	1.5	Voice :667-716 (50)	667-716	845.01-846.48	890.01-891.48
	10.0	Control : 334-354 (21) voice : 355-666(312)	334-666	835.02-844.98	880.02-889.98
3	2.5	Voice : 717-799 (83)	717-799	846.51-848.97	891.51-893.97

3.4.2 Channel Spacing and Frequency Tolerance for CDMA

CDMA uses a bandwidth of 1.25 MHz and defines a set of channels on 50-kHz spacing. However, the recommended channels are on 1.25-MHz channel spacing.The basic same signal is encoded with a Walsh function. The Walsh function has the property that within the set of Walsh functions, each function is orthogonal to all others. Thus different signals are selected by choosing a different Walsh function to decode. After Walsh function encoding, the CDMA signal is modulated by a 1.23-MHz spreading signal called the long code. The channels for CDMA are described in Tables 3.8, 3.9, and 3.10 for PCS frequencies and Table 3.11 and 3.12 for cellular frequencies.The frequency tolerance stated as the PS transmit carrier frequency will be below the RS transmit frequency, as measured at the PS receiver, by 80 MHz $\pm$150 Hz for PCS frequencies and 45 MHz $\pm$300 Hz for cellular frequencies. The RS transmitter will maintain its frequency to within $\pm$5 parts per 100 million ($\pm$100 Hz at 2000 MHz).

3.4.3 Channel Spacing and Frequency Tolerance for PACS

PACS uses TDMA with FDD on 300-kHz channels. Although the band plan (Figure 3.14) shows channels on 100-kHz spacing, only every third channel can be used in a given region. The extra channels are identified to solve guard channel problems at band edges. Once the channel at the band edge is established, all other channels increment in 300-kHz spacing. The frequency stability of the PS and the RS will be within $\pm$1 ppm ($\pm$2,000 Hz at 2000 MHz) of their nominal values over the operating temperature ranges and voltage ranges. Short-term stability over a TDMA frame will not exceed $\pm$0.1 ppm.

...le3.8 Definition of Channel Numbers and Frequencies for CDMA at 1900 MHz

Band	Valid CDMA Frequency Assignments	CDMA Channel Number	PS Transmit Center Frequency(MHz)	RS Transmit Center Frequency (MHz)
A (...5 MHz)	Not Valid Valid Cond. Valid	0-24 25-275 276-299	1850.000-1851.200 1851.250-1863.750 1863.800-1864.950	1930.000-1931.200 1931.250-1933.750 1933.000-1934.950
D (...5 MHz)	Cond. Valid Valid Cond. Valid	300-324 325-375 376-399	1865.000-1866.200 1866.250-1883.750 1868.800-1869.950	1945.000-1946.200 1946.250-1943.750 1948.800-1949.950
B (...15 MHz)	Cond. Valid Valid Cond. Valid	400-424 425-675 676-699	1870.000-1871.200 1871.250-1883.750 1883.800-1884.950	1950.000-1951.200 1951.250-1963.750 1963.800-1964.950
E (5 MHz)	Cond. Valid Valid Cond. Valid	700-724 725-775 776-799	1885.000-1886.200 1886.250-1883.750 1880.800-1889.950	1965.000-1966.200 1966.250-1963.750 1968.800-1969.950
F (5 MHz)	Cond. Valid Valid Cond. Valid	800-824 825-875 876-899	1890.000-1891.200 1891.250-1893.750 1893.800-1894.950	1970.000-1971.200 1971.250-1973.750 1973.800-1974.950
C (15 MHz)	Cond. Valid Valid Not Valid	900-924 925-1175 1176-1199	1895.000-1896.200 1896.250-1908.750 1908.800-1909.950	1975.000-1976.200 1976.250-1988.750 1988.800-1989.950

Table3.9 CDMA Preferred Set of Frequency Assignments

Block Designator	Preferred set Channel Numbers
A	25, 50, 75, 100, 125, 150, 175, 200, 225, 250, 275
D	325, 350, 375
B	425, 450, 475, 500, 525, 550, 575, 600, 625, 650, 675
E	725, 750, 775
F	825, 850, 875
C	925, 950, 975, 1000, 1025, 1050, 1075, 1100, 1125, 1150, 1175

Table 3.10 Channel Number Designations for CDMA PCS at 1900

ransmitter	CDMA Channel Number	Center Frequency of CDMA Channel (MHz)
PS	1 < N < 1199	0.050 N + 1850.000
RS	1 < N < 1199	0.050 N + 1930.000

able 3.11 Definition of Valid Channel Numbers for CDMA at 800 MHz

ency Band	MHz	Valid Regions	Channel Number
A'	1	Not valid Valid	991-1012 1013-1023
A	10	Valid Not valid	1-311 312-333
A'	1.5	Not valid Valid Not valid	667-688 689-694 695-716
B	10	Not valid Valid Not valid	334-355 356-644 645-666
B'	2.5	Not valid Valid Not valid	717-738 739-777 778-799

ble 3.12 Definition of Preferred Channel Numbers and Frequencies for CDMA at 800 MHz

Band	Preffered CDMA Channel Number	PS transmit Center Frequency (MHz)	RS transmit Center Frequency (MHz)
A	283	833.490	878.490
A'	691	845.730	890.730
A''	-	-	-
B	384	836.520	881.520
B'	777	848.310	893.310

Channels for PACS are described in Table 3.13

13 Definition of Channel Numbers and Frequencies for PACS at 1900 MHz

Bandwidth(MHz)	Number Of Channels	Channel Number	PS Transmit Center Frequency (MHz)	RS Transmit Center Frequency (MHz)
15	149	1	1850.1	1930.1
		2	1850.2	1930.2
		3	1850.3	1930.3
		...	...	...
		147	1864.7	1944.7
		148	1864.8	1944.8
		149	1864.9	1944.9
5	50	150	1865.0	1945.1
		151	1865.1	1945.2
		152	1865.2	1945.3
		...	...	...
		197	1869.7	1949.7
		198	1869.8	1949.8
		199	1869.9	1949.9
15	150	200	1870.1	1950.1
		201	1870.2	1950.2
		202	1870.3	1950.3
		...	...	...
		347	1884.7	1964.7
		348	1884.8	1964.8
		349	1884.9	1964.9
5	50	350	1885.1	1965.1
		351	1885.2	1965.2
		352	1885.3	1965.3
		...	...	...
		397	1889.7	1969.7
		398	1889.8	1969.8
		399	1889.9	1969.9
5	50	400	1890.1	1970.1
		401	1890.2	1970.2
		402	1890.3	1970.3
		...	...	...
		447	1894.7	1974.7
		448	1894.8	1974.8
		449	1894.9	1974.9
15	50	450	1895.1	1975.1
		451	1895.2	1975.2
		452	1895.3	1975.3
		...	...	...
		597	1909.7	1989.7
		598	1909.8	1989.8
		599	1909.9	1989.9

Channel Spacing and frequency Tolerance for PCS 2000

PCS 2000 system uses 5-MHz channels separated by 5 MHz. Table 3.14 defines potential channels (on a 2.5-MHz spacing). Obviously, in any given region, only that are 5 MHz apart can be used.

RS, the radio frequency signals and the data clock will be generated from a common frequency reference. The RS frequency will be kept within ± 1 ppm (± 2,000 at 2000 MHz) of the nominal channel frequency when not synchronized to the data network and ± 10 ppm (± 20,000 Hz at 2000 MHz) of the nominal channel frequency when synchronized to the data network.

5 Channel Spacing and frequency tolerance for TDMA

TDMA, the channel spacing is 30 kHz. There are 1,999 TDMA channels in the PCS with the channels designated as 1 to 1999. Tables 3.15 and 3.16 define the channel numbers and frequencies for TDMA for the 1900-MHz PCS band. When TDMA uses cellular frequencies, the channels are the same as the analog cellular channels. The PS transmit carrier frequency will be 80 MHz ± 200 Hz below the RS transmit frequency. The RS transmitter will maintain its frequency to within ± 0.25 ppm (± 500 Hz at 2000 MHz).

Table 3.14 PCS-2000 Channel Plan

Band	Channel Number (Hex)	Frequency
A	0	1852.50
	1	1855.00
	2	1857.50
	3	1860.00
	4	1862.50
	5	1932.50
	6	1935.00
	7	1937.50
	8	1940.00
	9	1942.50
B	A	1872.50
	B	1875.00
	C	1877.50
	D	1880.00
	E	1882.50
	F	1952.50
	10	1955.00
	11	1957.50
	12	1960.00
	13	1962.50
C	14	1897.50
	15	1900.00
	16	1902.50
	17	1905.00
	18	1907.50
	19	1977.50
	1A	1980.00
	1B	1982.50
	1C	1985.00
	1D	1987.50
D	1E	1867.50
	1F	1897.50
E	20	1887.50
	21	1967.50
F	22	1897.50
	23	1972.50
Unlicensed	28	1920.625
	29	1921.875
	2A	1923.125
	2B	1924.375
	2C	1925.625
	2D	1926.875
	2E	1928.125
	2F	1929.375

able 3.15 Definition of Channel Numbers and Frequencies for TDMA at 1900 MHz

	Bandwidth (MHz)	Number of Channels	Boundary Channel Number	PS Transmit Center Frequency (MHz)	RS Transmit Center Frequency (MHz)
	15	499	1 499	1850.040 1864.980	1930.020 1944.960
d		1	500	1865.010	1944.990
	5	165	501 665	1865.040 1869.960	1945.020 1949.940
ed		1	666	1869.990	1949.970
ed		1	667	1870.020	1950.000
	15	498	668 1165	1870.050 1884.960	1950.030 1964.940
sed		1	1166	1884.990	1964.970
sed		1	1167	1885.020	1965.000
	5	165	1168 1332	1885.050 1899.970	1965.030 1969.950
used		1	1334	1890.000	1970.980
used		1	1334	1890.030	1970.010
F	5	165	1335 1499	1890.060 1894.980	1970.040 1974.960
used		1	1500	1895.010	1975.990
C	15	499	1501 1999	1895.040 1909.980	1975.020 1989.960

Table3.16 Channel Number Designations for TDMA PCS at 1900

Transmitter	Channel Number	Center Frequency (MHz)
PS	1 < N < 1999	0.030 N + 1850.010
RS	1 < N < 1999	0.030 N + 1929.990

.6 Channel Spacing and Frequency Tolerance for Wideband CDMA

e Wideband CDMA system is similar to the CDMA system except than any one of ee wider bandwidths can be supported_ 5 MHz, 10 MHz, or 15 MHz. With different cations of bandwidths (5 or 15 MHz) in the United States, then the bands will be gulated as described in Tables 3.17 and 3.18. For a W-band CDMA transmitter, the S transmit carrier frequency will track within ± 200 Hz of a frequency value 80 MHz wer than the frequency of the corresponding RS transmit signal as measured at the PS

... The RS transmitter will maintain its frequency to within ±5 parts per 100 ... ±100 Hz at 2000 MHz).

...er Output Characteristics of the PS and the RS

... air interfaces use power control to minimize interference in the cellular ... TDMA uses discrete levels that are changed in response to local propagation ... and distance from the PS to the RS. CDMA systems require instantaneous ... control to maintain minimum interference levels at the RS. Each air interface ... various classes of PS transmitters that are permitted different power outputs. ... units typically have lower power outputs than units mounted in vehicles. The ... output of handheld units is kept lower to increase battery life and to reduce RF ... into the body of the user.The FCC has established standards for handheld ...ers to avoid radiation hazards. Two classes of users have been established_ one ... who are aware they are using transmitters and may have received training ... police, fire, rescue) and one for those who may not realize that they are using a ...itter (e.g., a cellular or PCS user). The radiation levels for the second class of ... are lower. All handheld PCS PSs will meet the second class of requirements. RS ... is typically higher than PS power and depends on the cell size. It too is limited by ... rules.

Table 3.17 Use of 5-MHz Spectrum with Wideband CDMA

0 MHz	**5 MHz Wideband CDMA**
5 MHz	

Table 3.18 Sharing of 1.25-MHz CDMA with Wideband CDMA

0 MHz	1.25 MHz Narrowband CDMA 1.25 MHz Narrowband CDMA 1.25 MHz Narrowband CDMA 1.25 MHz Narrowband CDMA
5 MHz	

Power Output characteristics of Analog Cellular

... classes of personal stations are defined for analog cellular with the fourth class ... for a dual-mode TDMA/analog cellular phone (see Table 3.19).

Table 3.19 Personal Station Nominal Power Levels

wer level (dBW)	Mobile Attenuation	PS Power Class			
(PL)	Code (MAC)	I	II	III	IV
0	000	6	2	-2	-2
1	001	2	2	-2	-2
2	010	-2	-2	-2	2
3	011	-6	-6	-6	-6
4	100	-10	-10	-10	-10
5	101	-14	-14	-14	-14
6	110	-18	-18	-18	-18
7	111	-22	-22	-22	-22
ual mode only					
8		-26 ± 3 dB			
9		-30 ± 6 dB			
10		-34 ± 9 dB			

control is done under control from the RS based on propagation conditions n the PS and the RS and based on cell size. All PSs are required to adjust power n command from an RS specifying power level 0 to 7.PSs in classes IV-VIII are d to change power to levels 0 to 10 by a physical layer control message from the

The power levels 0 to 7 are maintained within the range of +2 dB/-4 dB of its al level over the ambient temperature range of –30°C to + 60°C and over the voltage range of ± 10% from the nominal value. For power levels 8 through 10, wer emission is maintained within the range +2 dB/- 6 dB of the initial power nless a physical layer control message is received, over the same temperature and voltage conditions stated earlier. The RS EIRP depends on Height Above ge Terrain (HAAT) and power output.

Power Output Characteristics of CDMA

CDMA PS, each class of transmitter has an absolute maximum effective isotropic ed power (EIRP). For any class of personal station transmitter, it is 3 dBW (2.0

...le 3.20 defines the maximum output power and the minimum power output for ...ss of PS when it is commanded to its maximum power. The PS attempts to ... the power output based on received signal strength (open-loop control), and the ...ds power control messages to the PS and controls the power about once every ...cond (closed-loop control). The net effect is to control the power received at the ... within 1 dB for all PSs being received at that RS. The fine level of power control ...essary for proper operation of the CDMA system. The RS will not transmit more ...1,640 W of EIRP in any direction in a 1.25-MHz band for antenna HAAT less than ...eters. The RS antenna height may exceed 300 meters with a reduction in EIRP ...ding to current FCC rules. The transmitter output power of the RS in any 1.25-...z band of the RS's transmit band between 1930 and 1990 MHz and in any direction ...not exceed 100 W.

...3 Power Output characteristics of PACS

...ACS, the RS transmits continuously on a channel with a maximum power output of ... mW (+ 29 dBm). The PS transmits in TDMA bursts with an average power ...ermined by the power control process. The maximum average output power is 200 ...W (−23 dBm). There are no classes of PSs defined for PACS. The RS sends power ...trol messages to the PS that control the power in 1-dB steps (± 0.5 dB) over a range ...at least 30 dB. If a failure of RS control of power output occurs, the PS will transmit ...ts maximum power output.

Table 3.20 EIRP at Maximum Output Power for a CDMA PS

PS Classes	EIRP at Maximum Output Will Exceed	EIRP at Maximum Output Will Not Exceed
I	-2 dBW(0.63 W)	3 dBW(2.0 W)
II	-7 dBW(0.20 W)	0 dBW(1.0 W)
III	-12 dBW(63 mW)	-3 dBW(0.5 W)
IV	-17 dBW(20 mW)	-6 dBW(0.25 W)
V	-22 dBW(6.3 mW)	-9 dBW(0.13 W)

...5.4 Power Output Characteristics of PCS2000

...e PCS2000 system uses a maximum of 1 W EIRP for the PS and a maximum of ...,640 W EIRP for the RS in the licensed PCS band. Actual power output of the RS ...ansmitter is 2 W and an antenna with a gain of 29.15 dB (plus feed-line losses) is ...ermitted. Typical power output for the PS is 10 mW in each 8-kbs time slot. The peak f the RS is controllable over a 33-dB range in steps of 3 dB. The peak power of s controllable, by the RS, over a range of 33 dB in steps of 3 dB. The combined)0 system does not define power classes of PSs.

ower Output Characteristics of TDMA

ximum EIRP with respect to a half-wave dipole for any class of PS transmitter is (1.58 W). The nominal EIRP for each of transmitter is shown in Table 2.21. is controlled by the RS (via an order message) and accounts for the changes in ation between the RS and the PS. No instantaneous power control is used. The ver is not specified in the TDMA specification but will be coordinated with other systems in an area and depends on HAAT and cell size.

ower Output Characteristics of Wideband CDMA

-CDMA system uses power control similar to that of the CDMA system except ower power is used and fewer classes of transmitters are used. Table 3.22 arizes the PS power levels for all three bandwidths for the W-CDMA system.

Table 3.21 PS Power Levels for Wideband CDMA

ower Level	Mobile Attenuation Code	Nominal EIRP (dBW) for PS Power Classes						
(PL)	(DMAC)	II	III	IV	V	VI	VI	VIII
0	0000	2	-2	-2	-	-	-	-
1	0001	2	-2	-2	-	-	-	-
2	0010	-2	-2	2	-	-	-	-
3	0011	-6	-6	-6	-	-	-	-
4	0100	-10	-10	-10	-	-	-	-
5	0101	-14	-14	-14	-	-	-	-
6	0110	-18	-18	-18	-	-	-	-
7	0111	-22	-22	-22	-	-	-	-
8	1000	-22	-22	-26 ± 3dB	-	-	-	-
9	1001	-22	-22	-30 ± 6dB	-	-	-	-
10	1010	-22	-22	-34 ± 9dB	-	-	-	-

- Classes V,VI, VII, and VIII are reserved for future use

Table 3.22 EIRP at Maximum Output Power

PS Class	EIRP at Maximum Output Will Exceed
I	23 dBm (200 Mw)
II	13 dBm (20 mW)
III	3 dBm (2 Mw)

or W-CDMA will have no more than 1,000 W of EIRP

odulation Methods

3.23 summarizes the modulation methods used by each of the North American ols. We describe the information in more detail in the following sections.

nalog Cellular Modulation

cellular uses FM for voice band and analog data transmission with several voice sing stages:

- **Speech compressor:** A 2:1 compressor is used. For every 2-dB change in input, the change in output level is a nominal 1 dB. The compressor will have a nominal attack time of 3ms and a nominal recovery time of 13.5 ms as defined by the CCITT [15]. A calibration frequency of 1,000 Hz is used, and the level is set by the typical level from the PS microphone. This level will produce a nominal ± 2.9-kHz peak frequency deviation of the PS transmitted carrier.
- **Pre-empasis:** Since baseband noise in an FM RS increases with increasing (baseband) frequency, a pre-emphasis network is used.The pre-emphasis characteristics will have a nominal + 6 dB/octave response between 300 and 3,000 Hz.
- **Deviation limiter:** To avoid overmodulating and interfering with adjacent channels the maximum modulation will be limited to ± 12 kHz. The SAT can add to this deviation.
- **Post-deviation limiter filter:** The deviation limiter will be followed by a low-pass filter whose attenuation characteristics will exceed the characteristics in Table 3.24.

Table 3.23 Summary of Modulation Methods for Cellular and PCS

	Analog	CDMA	PACS	PCS2000	TDMA	W-CDMA
on	MFM-FSK	QPSK	π/4 DQPSK	QAM	π/4 DQPSK	QPSK
of n ion	2	4	8	32	8	4
ata	10 kbs	1.288 kbs	384 kbs	5 Mcps for CDMA 781.25 kbs (for TDMA frame)	48.6 kbs	4.096 Mcps 8.192 Mcps 12.288 Mcps
baud	1/2	-	2	-	2	-
dth	30 kHz	1.25 MHz	288 kHz	5 MHz	30 kHz	5 MHz 10 MHz 15 MHz
/Hz	0.333	-	0.75	-	0.62	-

eceiver will have de-emphasis and an expander to match the transmitter. When and data is sent for signaling, either on the control channel or the voice channel, ideband data streams will be further encoded (Figure 2.45) so that each non- -to-zero binary one is transformed to a zero-to-one transition, and each non- -to-zero binary zero is transformed to a one-to-zero transition. The filtered band data stream will then be used to modulate the transmitter carrier using direct y FSK. A one (i.e., high state) into the modulator will correspond to a nominal frequency deviation 8 kHz above the carrier frequency, and a zero into the lator will correspond to a nominal peak frequency deviation 8 kHz below the er frequency. When the SAT is sent, it will have a deviation of 1/3 radian (± 2 . The signaling tone used to signal a disconnect or a flash will have a tone of 10 ± 1 Hz and a deviation of ± 8 kHz.

Table 3.24 Post-Deviation Limiter characteristics

Frequency Band	Attenuation relative to 1,000 Hz for PS	Attenuation relative to 1,000 Hz for RS
3,000-5,900 Hz	40 log (f/3,000) dB	40 log (f/3,000) dB
5,900-6,100 Hz	35 dB	40 log (f/3,000) dB
6,100-15,000 Hz	40 log (f/3,000) dB	40 log (f/3,000) dB
Above 15,000 Hz	28 dB	28 dB

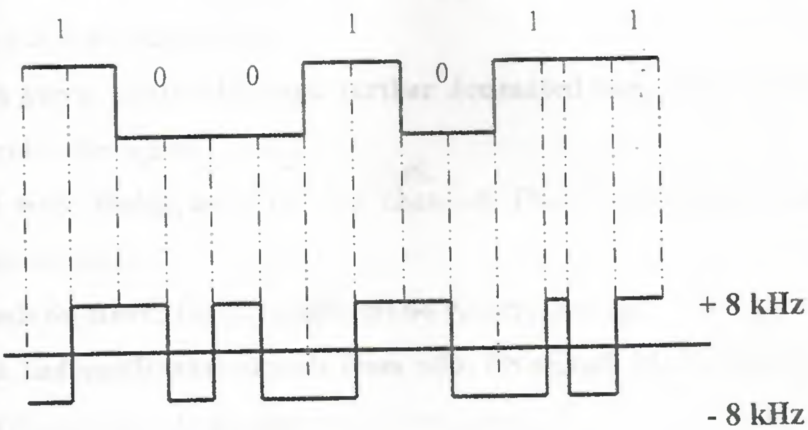


Figure 3.32 Non-Return to Zero (NRZ) to Manchester Encoding

2 CDMA Modulation

CDMA signal transmitted by a RS consists of up to 64 channels, each on a different sh function. Channel 0 is the pilot channel and uses Walsh function 0. Channel 32, sync channel, uses Walsh function 32 and is used to synchronize the receiver. nnel 1-7 (Walsh 1-7) are the paging channels. A maximum of one and a maximum even paging channels are used in order. The remainder of the channels are used as fic channels. Traffic channel data enters the modulation stages as 0.8-8.6 kbs (Rate 1) or 1.05-13.35 kbs (Rate Set 2). It then goes through the following stages (see ures 3.18 and 3.19).

d **one reserved bit (Rate Set 2 only):** This step is needed only for Rate Set2.

d **frame quality indicator:** The number of bits available depend on the data rate. No are added at the highest data rates.

-bit encoder tail per frame: These bits simplify the decoding of the digital .

ne-half convolutional encoding: This step is used for error detection in the er.

l repetition: If the data rate is not the maximum rate, repeat each symbol until a data rate is maintained (19.2 kbs for Rate Set 1 and 28.8 kbs for Rate Set 2).

ure (delete) 2 of 6 bits (Rate Set 2 only): This step lowers the data rate to 19.2

interleave the bits: This step mixes up the transmission of the bits for additional protection capabilities. The data rate remains the same.

sive OR with decimated long code: The long code is different for each user and des privacy of communications.

iplex with power control bits and further decimated long code: This step adds ower control to the signal.

usive OR with Walsh code for the channel: The Walsh function selects the nel used for the traffic.

all channels together: This step adds all 64 Walsh channels.

m in-phase and quadrature signals from pilot PN signal: (At 1.2288 Mcps): This modulates the signal up to the data rate of the channel.

eband filter the signals: This steps prevents modulation components from earing outside of the channel for the CDMA signal.

dulate with the in-phase and quadrature carriers: This step modulates the signals he CDMA channel.

pilot and paging channels enter the rate ½ encoder directly. The receiver uncodes h of these steps. A PS transmits on one Walsh function at a time and uses a rate 1/3 volutional encoder. Other minor differences are shown in Figures 3.22 to 3.24. The signal from the CDMA modulator is a four-phase quadrature signal.

.3 PACS Modulation

CS uses π/4 differentially encoded quadrature shift keying (π/4-DQPSK) with 2 bits r symbol. The transmitter has eight points in the signal constellation. The data rate is 4 kbs or 192 k-symbols/sec and the bandwidth is 288 kHz. The data signals are ferentially encoded and result in a change in phase of transmitted signal. The seband signals are shaped with a linear phase Nyquist 50% bandwidth expansion ot raised cosine spectral shaping. The baseband filter has the frequency with a roll-off factor equal to 0.5 for PACS.

2000 Modulation

[A bursts of PCS 2000 are modulated with a DSSS signal that uses 32 QAM 33) at a rate of 5 Mcps.

MA Modulation

IA protocol uses π/4-DQPSK with 2 bits per symbol. The signal constellation filtering are identical to PACS, but obviously the data rate and bandwidth are

-CDMA Modulation

ration of the W-CDMA system is similar to the operation of the CDMA system hat a PN code rate is higher to account for the wider bandwidth.

NDOFFS

g cellular, the network measures the signal strength of the PS at a variety of RSs cesses a handoff to a channel on a different RS. In analog cellular, the handoff sent on the voice channel by blanking the voice and sending data. The PS then s to the new channel where the network recognizes the presence of the PS and s the network connection; Thus the call continues. Sometimes the network can a bridge connection between the old and new RS and thus minimize the duration handoff. In general, depending on the system design, the handoff process (data ission, channel switching, and network switching) can take 100-200 ms and es a noticeable click in the voice conversation. If the original voice channel is oisy, the click will be imperceptible. However, if the handoff occurs at reasonable quality or is done to balance traffic at an RS, the user may hear the click. Too nt handoffs may cause the user to have a poor impression of the service. With systems, the handoff process is under control of the RS and the PS and can be ed to be imperceptible to the user.

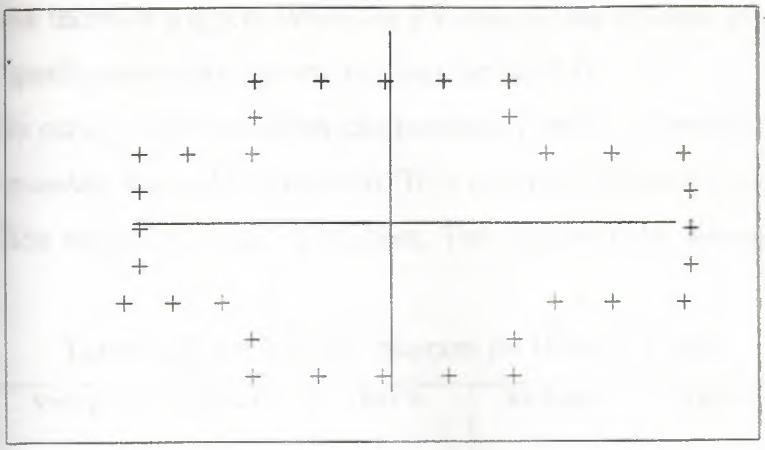


Figure 3.33 PCS-2000 32 QAM Constellation

…andoffs are common in CDMA where both RSs can transmit on the same …ency and Walsh code and the resultant signal is handled by the receiver as just one … multipath to be included in the decoded signal. Soft handoffs are imperceptible to …ser. The soft handoff process requires that all RSs in a network be tightly …ronized so that, when handoffs occur, data synchronization is maintained. The …MA systems support this function by using the Global Positioning System. To …ide a master clock. The IS-136 TDMA system does not currently support this …ion but may in the future. Hard handoffs are the normal handoffs for an analog …lar system. TDMA systems typically process hard handoffs unless special means …aken to synchronize the RSs. CDMA systems forced to hand off to a different …sh function or a different frequency will have hard handoffs. The resultant break in …ection causes speech or data to be lost and is perceptible to the user.

…e 3.25 summarizes the support for each of the air interfaces for soft and hard …offs.The handoff process can be triggered by either the network or the PS (see …e 3.25 for summary of handoff types by air interface)

…**ork-directed handoff trigger:** When the network controls the handoff, signal …ity measurements (error rates, signal strength, and so on) are measured by the …ork. The network then initiates the process to handoff the PS to another RS or …nel at an RS.

…e GPS system is a set of low-orbiting satellites that provides an atomic clock (corrected for relativity effects. …e height. satellite location, and so on).GPS has seen significant civilian use throughout the world. The former …e Union provides a similar capability through its Glossnoss system.)

-directed handoff trigger: When the PS controls the handoff process, it makes e signal quality measurements and initiates the handoff.

handoffs occur, call information can potentially be lost; Therefore, one network nt will maintain the call information. That network element is called the anchor e call. RSs and PCSCs can be anchors. The choice of the anchor is a network n issue.

Table 3.25 Air Interface Support for Handoff Types

Hand- Type	Analog	CDMA	PACS	PCS2000	TDMA	W-CDMA
ard	X	X	X	NA*	X	NA*
oft		X	NA*	NA*	:	X
riggered		X		X	X	X
twork ggered	X	X		X	X	

Not Available

TDMA will support soft handoff in the future

8. SUMMARY

is chapter presented how the analog cellular and each of five digital air interfaces (the -Interface in the reference model) support the functions of basic encoding of the gital data being sent. We also discussed the various air interfaces that are standarized r PCS in the United States. The operation of these air interfaces and the services they pport.

4. EUROPEAN and JAPANESE CELLULAR SYSTEMS

ERVIEW

chapter we present an overview of the Global System for Mobile nications (GSM) as described in the European Telecommunications standards e's (ETSI's) Recommendations. The chapter also addresses a brief description of anese Digital Cellular (JDC) system is also given in this chapter.

SM PUBLIC LAND MOBILE NETWORK (PLMN)

riginally defined GSM as a European digital cellular telephony standard. GSM es defined by ETSI lay the groundwork for a multivendor network approach to mobile communication. GSM offers users good voice quality, call privacy, and k security. Subscriber Identity Module (SIM) cards provide the security nism for GSM. SIM cards are like credit cards and identify the user to the GSM k. They can be used with any GSM handset, providing phone access, ensuring y of appropriate services to that user, and automatically billing the subscriber's k usage back to the home network[16].

Roaming agreements have been established between most GSM network ers in Europe, allowing subscribers to roam between networks and have access to ervices no matter where they travel. Of major importance is GSM's potential for ing enhanced services requiring multimedia communication: Voice, image, and everal mobile service providers offer free voice mailboxes and phone answering s to subscribers. The key to delivering enhanced services is Signaling System er 7 (SS7) [17], a robust set of protocol layers designed to provide fast, efficient, e transfer and delivery of signaling information across the signaling network and port both switched-voice and nonvoice applications. With SS7 on the enhanced s platform and integrating mailbox parameters, subscribers can be notified about mber of stored messages in their mailboxes, time and source of last messages, ge urgency, and type of message-voice or fax. Future applications such as fax nd-forward and audiotext can also use the platform's voice-and data-handling lities.

OBJECTIVES OF A GSM PLMN

SM PLMN cannot establish calls autonomously other than local calls between le subscribers. In most of the cases, the GSM PLMN depends upon the existing ne networks to route the calls. Most of the time the service provided to a riber is the combination of the access service by a GSM PLMN and the service by existing wireline network. Thus, the general objectives of a GSM PLMN with ct to service to a subscriber are:

- To provide the subscriber with a wide range of services and facilities, both voice and nonvoice, that are compatible with those offered by existing networks (e.g., PSTN, ISDN).
- To introduce a mobile RS that is compatible with ISDN.
- To provide certain services and facilities exclusive to mobile situations.
- To give compatibility of access to the GSM network for a mobile subscriber in a country that operates the GSM system.
- To provide facilities for automatic roaming, locating, and updating of mobile subscribers.
- To provide service to a wide range of mobile stations, including vehicle-mounted stations, portable stations, and handheld stations.
- To provide for efficient use of the frequency spectrum.
- To allow for a low-cost infrastructure, terminal, and service cost.

GSM PLMN SERVICES

ecommunication service supported by the GSM PLMN is defined as a group of unication capabilities that the service provider offers to the subscribers. The basic mmunication services provided by the GSM PLMN are divided into three main s: Bearer services, teleservices, and supplementary services [18].

Bearer Services

services give the subscriber the capacity required to transmit appropriate signals en certain access points (i.e., user-network interfaces). The capabilities of the bearer services are the following:

- Rate-adapted subrate information-circuit-switched asynchronous and synchronous duplex data, 300-9,600 bps.
- Access to Packet Assembler/Disassembler (PAD) functions-PAD access for asynchronous data, 300-9,600 bps.