International Telecommunication Union

DISPATCH AND ADVANCED MESSAGING SYSTEMS

Handbook on Land Mobile
(including Wireless Access)

Volume 3
(Edition 2005)

Radiocommunication Bureau

International Telecommunication Union

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Foreword

The Handbook on Land Mobile was started in the late 1990's within the ITU-R Working Party 8A to meet an increasing need by developing countries for a handbook on state of the art technologies covering the various aspects of the land mobile services, technologies and systems. The Handbook is organized in several volumes, two of which are already published. Volume 1: Fixed Wireless Access and Volume 2: Principles and Approaches on Evolution to IMT-2000.

The purpose of the Handbook is to assist in the decision-making process involving planning, engineering and deployment of land mobile systems, especially in developing countries. It should also provide adequate information to assist in training engineers and planners in the regulating, planning, engineering, and deployment aspects of these systems.

Volume 3: Dispatch and Advanced Messaging Systems provides information on state-of-the-art technology in terrestrial land mobile paging and advanced messaging and dispatch systems, applications and technologies, as well as descriptions of typical systems. Dispatch systems refer to a land mobile system where a central location, a dispatcher, communicates with a number of radio mobile units (vehicles or handhelds) in a fleet in a specified area of operation. Paging and advanced messaging technologies continue to grow with advancements like two-way paging and digitized speech paging and it has become a very popular form of mobile communication worldwide. The technical content is intended for use by administrations and operators in both developing and developed countries.

Volume 3 has been developed by a group of experts of Radiocommunication Working Party 8A. I wish to express my appreciation to Ms. Cindy Cook (Canada), Land Mobile Handbook Rapporteur, and Mr. Michel Pierrugues (France) and Mr. Suneil Kanjeekal (Canada) who kindly served as editors of this volume in different stages of the project.

José M. Costa
Chairman, Radiocommunication Working Party 8A
Canada

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CHAPTER 1

INTRODUCTION

1.1 Purpose and scope of the Handbook on Land Mobile

The Handbook on Land Mobile was started in the late 90's within the ITU-R Working Party 8A to meet an increasing need by developing countries for a handbook on state of the art technologies covering the various aspects of the land mobile services; technologies and systems. The Handbook is organized in several volumes, two of which are already published. Volume 1: Fixed Wireless Access and Volume 2: Principles and Approaches on Evolution to IMT-2000. The purpose of the Handbook is to assist in the decision making process involving planning, engineering and deployment of wireless based land mobile systems, especially in the developing countries. It should also provide adequate information that will assist in training engineers and planners in the regulating, planning, engineering, and deployment aspects of these systems. The Handbook covers land mobile applications including, vehicular communications, in-building communication, out-of-building communication, as well as others such as Intelligent Transport Systems (ITS) applications. Systems covered encompass cellular-based systems, messaging systems, dispatch systems, fixed wireless access, as well as ITS.

The users of this Handbook are likely to fall into one of two categories. The first category includes the decision-makers and planners who would like the Handbook to provide them with enough information to aid in decision making on system choices as far as their suitability to meet their requirements. For this purpose, the Handbook provides analysis on the various systems taking into consideration factors such as, traffic estimation and projection, frequency band and spectrum requirements, investments, regulation and policy requirements and experiences, deployment strategies, short and long term implications, as well as other elements that are required for decision making and planning purposes.

For the second category of users – engineers – the Handbook provides more in depth technical information on the characteristics of the various systems and applications, systems design, traffic analysis and estimation, spectrum estimation, channelling plans, cell design and selection, deployment strategy, mobile and base station equipment, as well as other pertinent information.

1.2 Background

The purpose and scope of Volume 3 of the Handbook on Land Mobile, is to provide information on state-of-the-art technology in terrestrial land mobile paging and advanced messaging and dispatch systems, applications and technologies.

Our modern society is greatly dependant on transportation mobility for the provision of critical services to meet our everyday needs and requirements. Fundamental to this is a critical need for fast and highly reliable radio communication providing the required links to ensure robust coordination for the various activities being performed within vast areas of operations. Public safety and emergency operations including police, fire, medical emergency, public utilities, land transportation, delivery, other government operations, etc., would cease to function as we know them today without reliable radio communications for dispatch systems.

1.3 Organization of Volume 3

Volume 3 is organized into a number of chapters providing key information to the reader, with detailed technical, operational, and regulatory information provided in the annexes. The introduction to the volume is provided in Chapter 1. Chapter 2 provides information on the various spectrally efficient digital dispatch systems. Chapter 3 provides brief descriptions and basic characteristics of paging and advanced messaging applications and systems. Detailed technical descriptions of the various systems are provided in Annexes 1 and 2.

CHAPTER 2

DISPATCH SYSTEMS

2.1 Introduction

Dispatch systems refer to a land mobile system where a central location, a dispatcher, communicates with a number of radio mobile units (vehicles or handhelds) in a fleet in a specified area of operation. The fleet may consist of users in police, fire, taxis, delivery, repair, etc., type of operations. The fleet could consist of one type of users or a mix of different users, where an integrated dispatch system is required. The area covered by the mobile units could be quite large and may be served by one cell or multiple cell arrangements. Depending on the size of the fleet, they could share the same channel or may require multiple channels per fleet. Communication could be in the simplex mode (one frequency used for both up and down directions on a push to talk basis), half duplex where two frequencies are used one for up direction and the other for down direction with only one frequency active at a time. The third mode is a full duplex where two different channels are used, one for up direction and the other for down direction communications.

For a small fleet with only a few mobile units, a single individual may be able to handle the dispatch requirements. However, the complexity of the dispatch increases with the number of mobile units. In addition, depending on the type of operation, access to data bases and highway assistance types of information (intelligent transport system (ITS)) has resulted in increasing dependency on data transmission. This is further augmented by the high demand for internet access for information to enhance and support the applications and services provided.

2.2 Background

Dispatch systems have traditionally utilized spectrum very efficiently as several users are able to share a single channel; they are used by different users such as police, fire, taxis, delivery, repair, etc. Until recently, analogue modulation has been the main technology used in dispatch systems. However, the limited spectrum availability for dispatch systems, annual growth and the increasing demands for data based communications[1] have led to the development of more spectrally efficient technologies utilizing digital modulation and trunking techniques.

2.3 Spectrum efficient digital dispatch systems

This section provides a brief description of spectrum efficient digital dispatch systems for international and regional use. Annex 1 contains technical and operational characteristics on eight systems being introduced throughout the world.

2.3.1 Integrated Enhanced Digital Network (iDEN)

iDEN is a communications system in over a dozen countries utilized by over 18 million subscribers based on existing time division multiplex technology. iDEN is a commercial deployment of a digital mobile integrated system (DIMRS) which is described in Report ITU-R M.2014. It offers six communication circuits over a single 25 kHz channel, combining the services and capabilities usually associated with a dispatch system, a cellular telephone system, a packet data system and an advanced messaging and paging network which are accessed using cell phone sized handsets.

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[1] Data communication capabilities stimulating the acceptance of digital wireless include new voice, video and other multimedia communication services.

2.3.2 Integrated digital radio system (IDRA)

IDRA is a dispatch standard and system developed in Japan which incorporates standard two-way radio features with advanced data services in a common infrastructure and subscriber unit. IDRA is an amended version of the Japanese existing digital trunking standard, a time division multiple access (TDMA), 16-quadrature amplitude modulation (QAM) standard which fits six voice conversations into a 25 kHz RF channel.

2.3.3 Project 25

Project 25 is a suite of standards developed through the combined efforts of United States of America local, state and federal government users, in collaboration with the Telecommunications Industry Association (TIA). Project 25 is targeted at enabling public safety agencies to gracefully migrate migration from analogue FM systems to narrow-band and wideband digital systems offering enhanced voice and data functionality. Project 25 consists of two main phases. Phase I is a frequency division multiple access (FDMA) system using compatible FM and quadrature phase shift keying (QPSK) modulations in 12.5 kHz. Phase II has added FDMA at 6.25 kHz bandwidths. Phase II is also addressing TDMA alternatives to achieve 6.25 kHz equivalency, high speed data, and a number of system enhancements. Work involving high speed data was developed under the Wideband Data Standards Project (TIA 902-series documents) for 700 MHz and are currently being prepared as American National Standards. Information also contributed to the Mobility of Emergency and Safety activities (MESA) Broadband Project.

2.3.4 Terrestrial trunked radio system (TETRA)

TETRA is a high performance radio system which has been developed primarily for professional users and which provides a wide variety of voice and data services. TETRA is a TDMA standard using π/4 digital QPSK (DQPSK) modulation to fit four voice circuits into a 25 kHz channel.

2.3.5 TETRAPOL

TETRAPOL is a publicly available specification of a fully operational digital trunked private mobile radio system, developed by a forum of manufacturers for security and emergency services. TETRAPOL uses FDMA technology with Gaussian (filtered) minimum shift keying (GMSK) modulation in a carrier spacing of 12.5 kHz and 10 kHz to allow easy migration from analogue to digital.

2.3.6 Enhanced digital access communications system (EDACS)

EDACS is an advanced two-way trunked radio system operating on 25 kHz or 12.5 kHz channels. EDACS specifications provide backward compatibility and interoperability with the existing base of EDACS equipment and systems. EDACS uses a variety of Gaussian frequency shift keying (GFSK) modulation techniques.

2.3.7 Frequency hopping multiple access system (FHMA)

FHMA has been developed in Israel to provide improved spectrum efficiency for the public access mobile radio market and the private mobile radio market. FHMA operates in 25 kHz channels.

2.3.8 Code division multiple access - public access mobile radio (CDMA-PAMR)

CDMA-PAMR is a spectrally efficient system that utilizes the Voice-Over-IP (VoIP) technology running over a cdma2000 radio network in order to provide voice-based PAMR services to users, in addition to data services with a range of data rates. CDMA-PAMR provides a highly flexible environment for the creation of services and applications, and a powerful combination of PAMR voice and data services. The carrier bandwidth of a CDMA-PAMR system is 1.25 MHz and the system operates with a frequency (cellular) re-use factor of 1.

2.4 Future trends in digital dispatch systems

In addition to the dispatch systems listed above, there are a number of digital dispatch standards that are under development that will further enhance the capabilities of existing land mobile dispatch services. Included in these are TETRA 2, MESA, cdma2000, Wideband Data Standards Project and Project 34.

These digital technologies are being standardized nationally, regionally and internationally. Although a fairly recent phenomena, international digital standards for land mobile radio systems promise a bright future for land mobile radio markets and users. Markets should experience readily available standard digital systems which are cost competitive, multiply sourced, and interoperable.

CHAPTER 3

PAGING AND ADVANCED MESSAGING

3.1 Introduction

Paging was traditionally a method of one-way wireless communications. It uses an assigned radio frequency to contact a paging subscriber anywhere in a service area. The pager, when turned on, constantly scans for signals and receives messages that are uniquely identified for it. The paging message is usually sent by placing a telephone call to the subscriber's designated telephone number. When the pager receives the intended message, it stores it and notifies the owner with an audible tone or vibration.

Today, paging and advanced messaging technologies continue to grow with advancements like two-way paging and digitized speech paging and it has become a very popular form of mobile communication worldwide.

3.2 Background

The first paging systems were initially developed for use in hospitals in the 1950's, and wide area paging systems began to emerge in the late 1960s. Soon, wider area paging systems emerged that linked directly to the public switched telephone network (PSTN).

Paging experienced its worldwide explosive growth in the 1980's and 90's using the POCSAG protocol at speeds of 512, 1 200 or 2 400 bit/s. It has been recognized as radio-paging code No.1 in the Recommendation ITU-R M.584 since 1982. As subscriber growth and requirements for a mix of tone, numeric and alphanumeric capabilities began to reduce total capacity and frequency limitations began to appear, paging operators pressed for development of higher speed protocols.

3.3 High speed paging systems

3.3.1 Flexible wide-area synchronous protocol (FLEX)

First introduced in June, 1993, FLEX is a family of high-speed wireless transport protocols that greatly enhance the channel efficiency and cost of traditional radio paging channels while also enabling new value-added wireless services and applications. FLEX is a multi-speed (1 600, 3 200, 6 400 bit/s) coding format that greatly expands the available subscriber capacity for each RF channel assignment. The FLEX coding format is synchronous which allows for greatly enhanced battery life performance and robust performance under multipath fading conditions.

The FLEX coding format is able to provide service in a mixed system operation with minimum loss of the systems operating efficiency. Because three signalling speeds of 1 600, 3 200 and 6 400 bit/s are available, the introduction of FLEX to an existing low speed operating system infrastructure can be accomplished with minimum added cost. As the service provider builds the systems subscriber base, the system can be gracefully grown up to the highest speed of 6 400 bit/s within reasonable added cost. Thus, the service provider is able to maintain the lowest system cost per subscriber during system growth.

The ability to provide local, regional, nationwide and global roaming operations is also structured within the FLEX coding format. FLEX is compatible with standard VHF, UHF and 900 MHz paging radio air interfaces.

3.3.2 European radio message system (ERMES)

The European Telecommunications Standards Institute (ETSI) has developed an international radio messaging system named ERMES – ETSI standard 300 133. It is capable of offering enhanced paging services both to national subscribers and to those roaming outside their home country where a corresponding network is implemented. Introduction of ERMES started in Europe in 1993. The technical and operational characteristics of ERMES are included in Annex 2 of Recommendation ITU-R M.539-3 – Technical and operational characteristics of international radio-paging systems.

ERMES uses a synchronous protocol at a speed of 6250 bit/s to achieve increased subscriber capacity, improved message performance under signal fading conditions, and improved battery saving operation. The system operates on one or more of the 16 contiguous 25 kHz channels in the band 169.4-169.8 MHz, providing roaming throughout Europe. Error correction capabilities of the transmission protocol have been designed for this frequency band. However, the transmission protocol is not tied to this frequency band and may operate in other frequency bands as indicated in Annex 1 of the ITU-R Recommendation M.539.

3.4 Advanced two-way messaging

New messaging protocols have been developed to service two-way messaging. In two-way messaging, the messaging unit responds back to the system. Simply, the subscriber is carrying a transmitter that operates both automatically and by the subscriber's actions. Two-way messaging offers four progressive levels of service:

- system acknowledgement,
- simple personal acknowledgement,
- multiple-choice response or pre-programmed response,
- message origination.

Two-way protocols provide two-way communications, but not in the same sense as traditional two-way data systems. They are asymmetric, which means that the data transmitted from the messaging units back to the system is limited to a much smaller size. This enables the concentration of the transmitted power into a smaller bandwidth, and thus minimizes the number of receiver sites required to achieve proper coverage.

One of the major applications of two-way messaging is voice messaging. Basically, voice messaging repeats the spoken word of the caller to the subscriber through the messaging communicator. Essentially, voice messaging functions like as an answering machine on the belt. It incorporates all the features of an answering machine, such as message storage and playback, this technology uses a compressed voice protocol again to optimize the service provider's valuable spectrum by increasing system capacity. Voice messaging retains most of the benefits of two-way messaging and the first of these devices is approximately the same size, shape and weight as some of today's one-way alphanumeric pagers.

No narrowband personal communication service (PCS) worldwide standard emerges, although several protocols are receiving favorable attention by paging operators and others interested in entering the narrowband PCS market around the world.

In the United States of America, the trend is towards a *de facto* standard based on the ReFLEX and InFLEXion protocols. These protocols, based on the FLEX protocol are being licensed on an open basis. This allows manufacturers to design and fabricate to a common set of specifications and, therefore, provide critical mass production and lower priced equipment.

ReFLEX protocol will support two-way paging at speeds of 12 800 bit/s and interactive messaging at 25 600 bit/s. ReFLEX two-way paging will be asymmetrical in nature with relatively large amounts of data moved from the network to the subscriber unit, but with minimal amounts of data (<100 bytes initially) coming back to the network from the subscriber unit. This enables low bit rate signalling on the return channel, which minimizes the number of fixed receivers and sites required. The resulting cost structure more closely approximates traditional paging costs rather than higher priced wireless alternatives. User features include message confirmation, message origination, error-free messages, and the ability to forward messages to a personal computer. Network features include localized transmission, frequency reuse, message truncation, registration and location, and transfers at 25 600 bit/s through the network in a subdivided 50 kHz channel.

ReFLEX systems are similar in nature to traditional one-way systems in that they are simulcast within zones (such as a metropolitan area). This means that multiple transmitters activate at the same time, transmitting the same messaging traffic. Between zones, though, re-use of spectrum is possible (which increases subscriber capacity) while providing two-way messaging capabilities

The InFLEXion protocol will allow high-speed voice and data messaging. With data rates exceeding 100 kbit/s, the InFLEXion protocol enables the re-emergence of voice paging in the global paging market as well as value added data services such as wireless facsimile, imaging and multimedia applications. The InFLEXion system does track the individual transmitter, hence provides for frequency re-use based on up to seven sub-channel frequencies within the 50 kHz channel. This creates a cellular-like system design where an individual transmitter site can be active and broadcasting on a given sub-channel, while an adjacent transmitter is active and broadcasting on a different sub-channel. Additionally, transmitters that are properly spaced can be operating on the same frequency, transmitting different messages at the same time.

In Europe, as well as in other countries where GSM has been successfully developed, short message service (SMS) appears to be very popular and may be regarded as a two way messaging service. Obviously, contrary to the other systems, it does not lean on a paging network infrastructure but on a PCS infrastructure.

3.5 Future trends in high-speed paging and advanced messaging

3.5.1 Spectrum efficiency

Two-way protocols provide the best transmitter management, and thus the largest subscriber capacity, for wide-area coverage through the use of automatic registration and locator services. The system does not track the precise transmitter area, or cell, where the subscriber currently is located, but an area encompassing a number of cells called a zone.

This allows the same frequency channels in separate zones to be used simultaneously for different messages. Transmission usage is economized to the maximum level because only the transmitters in the zone of the unit's physical location are used. In one-way systems, all transmitters broadcast every message because the system never knows where the unit is located. All other transmitters in the two-way system can be devoted to other messages, thus giving the entire system peak efficiency and the capacity for many more messages and subscribers.

3.5.2 Acknowledgement of receipt

The paging messaging unit has the ability to confirm the receipt of each message without any subscriber action. This feature has four important advantages:

– When the unit successfully receives a message, it informs the system of the successful reception. The system immediately removes the message from the broadcast queue, making room for other messages.

- If part of the transmitted message was not correctly received, the unit tells the system which parts of the message (known as the data packets) need to be rebroadcast, rather than re-broadcasting the entire message. This provides maximum efficiency in the amount of broadcast time each message can potentially require. In one-way systems, the system is programmed to never rebroadcast the message because the system never knows if there is an error, thus resulting in missed and erroneous messages. Or, the system is programmed to rebroadcast all messages a set number of times, thus wasting valuable transmission time if the original transmission was successful. With the increasing popularity of large messages, such as alphanumeric and voice, this feature becomes critical as the amount of data packets in a single message can climb into the hundreds.
- After the system has confirmation of successful delivery, it can be configured to inform the caller of the transmission's completion. This increases customer satisfaction and broadcasting efficiency because repeat messaging due to caller uncertainty is eliminated.
- Most important, is that all messages are guaranteed to be delivered. If a message cannot be delivered (because, for instance, the messaging unit is turned off), the system holds the message until the messaging unit reregisters its status and location, and then the message is sent.

3.5.3 Two-way voice messaging

Voice messaging is one of the major applications of two-way messaging. Basically, voice messaging repeats the spoken word of the caller to the subscriber through the messaging communicator.

Voice messaging functions as an answering machine on the belt and can incorporate all the features of an answering machine such as message storage and playback. Also, the voice messaging communicator offers low audio volume for private listening or higher volume for hands-free public listening.

Voice messaging includes the following benefits:

- It eliminates operator bureaus when required for alphanumeric messaging. The caller's message is stored directly in the network terminal where it is forwarded to the subscriber's messaging unit. This reduces the operational costs normally associated with alphanumeric paging.
- The sense of message urgency is conveyed, since the message is actually the caller's voice. Voice messaging provides a higher level of communication than numeric or alphanumeric paging.
- In many cases, a voice message does not require a return call because more detail can be conveyed in a voice message.
- Language independence is achieved because the message flows from the caller to the subscriber and therefore translation is not required.

ANNEX 1

DISPATCH SYSTEM DESCRIPTIONS

In this Annex, technical and operational information on various digital dispatch systems are provided for the benefits of engineers, planners and decision-makers to assist them in planning and deploying the appropriate systems in their countries.

1 iDEN and IDRA

The digital dispatch systems submitted to ITU-R, IDRA and iDEN, are fundamentally similar and will, therefore, be discussed together.

1.1 Origin

The Association of Radio Industries and Businesses (ARIB, formerly the RCR or Research and Development Centre for Radio Systems) is an external Ministry of Public Management, Home Affairs, Posts and Telecommunications (MPHPT) affiliate and a recognized standards organization. The first version of Japan's digital dispatch standard, called RCR STD-32, was completed in March 1993. An updated version of this standard which did not alter the basic RF characteristics of the standard, but which did add substantial networking capability to the system, was approved in November 1995, and is referred to as RCR STD-32A. This technology is represented by the IDRA system in Recommendation ITU-R M.2014.

1.2 Description

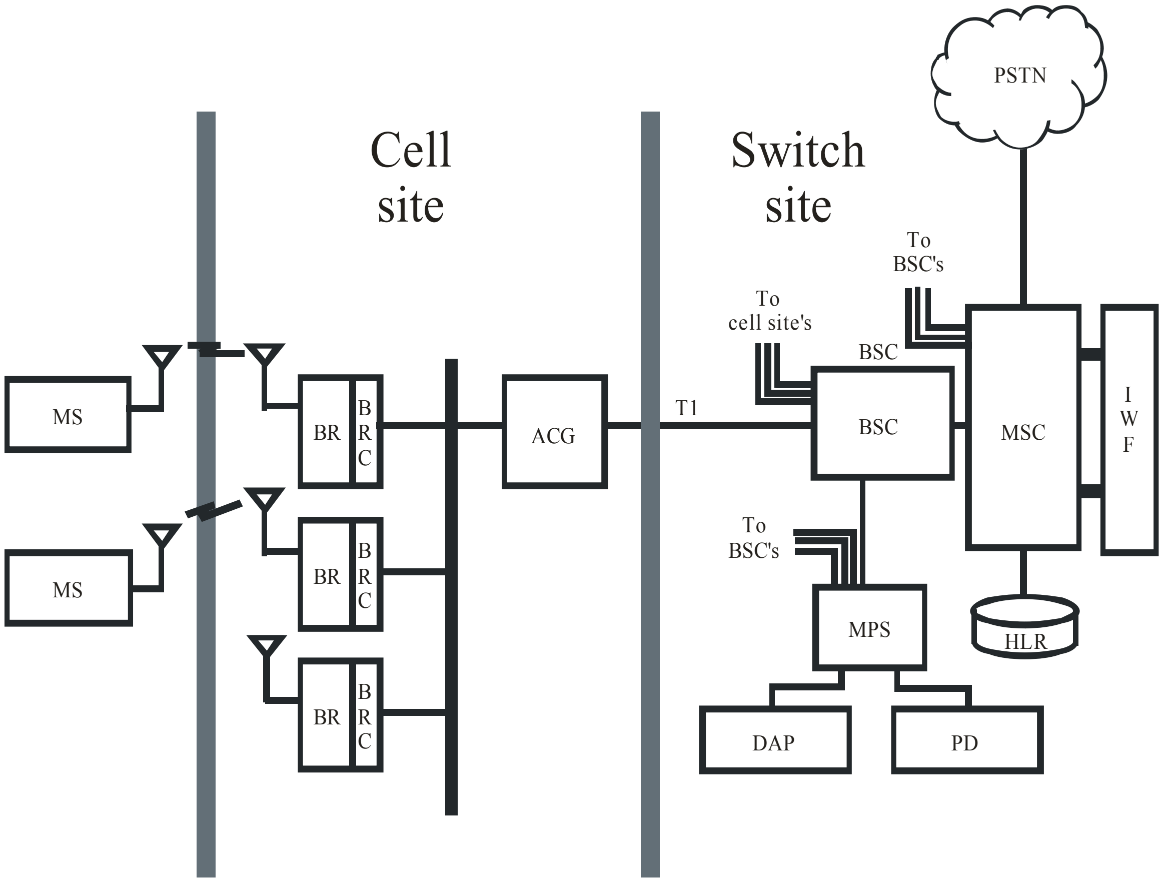
As mentioned above, these two systems are similar, though their focus is somewhat different. IDRA concentrates on offering dispatch service, but provides the capability for more advanced features. iDEN, emphasizes the multi-service aspect of its offering. Both standards describe TDMA systems with six circuits in a 25 kHz RF channel. Both use multi-subcarrier, quad, M16-QAM modulation. This advanced orthogonal frequency division multiplex (OFDM) like modulation provides for a 64 kbit/s gross channel bit rate that makes possible the varied services offered by each of these systems. IDRA operates in the 850-915 MHz and 1 453-1 525 MHz bands, and iDEN is used in the 806-870 MHz and 896-940 MHz bands. For increased packet data transmission rates a variation named WiDEN has recently become available that employs quad 64-QAM modulation and combines four contiguous 25 kHz channels to enable a gross bit rate of 384 kbit/s in a 100 kHz bandwidth.

1.3 System configuration

An example system configuration is shown in Fig. 1. Only one of a number of possible configurations is presented. Specifically, the IDRA system can be configured in several ways. For example, the switch site is not required in all IDRA configurations, so that all call processing applications, dispatch applications, and network connections are performed directly at the cell site. The example configuration is applicable to both the IDRA and iDEN systems.

FIGURE 1

Example IDRA/iDEN system configuration



ACG: Access control gateway	IWF: Interworking function
BR: Base radio	MPS: Metro packet switch
BRC: Base radio controller	MS: Mobile subscriber
BSC: Base site controller	MSC: Mobile switching centre
DAP: Dispatch application processor	PD: Packet duplicator
HLR: Home location register	

LandMob-01

1.4 Key attributes

A critical attribute of both systems is the M16-QAM modulation that enables the 64 kbit/s gross channel bit rate. 16-QAM is part of the class of linear modulations which, because both the amplitude and phase of the symbols are modulated, requires the use of sophisticated linear amplifier technology. This technology not only allows the transmission to be contained within a 25 kHz bandwidth, but to meet strict adjacent channel coupled power criteria which make frequency planning tasks much easier. The division of the transmitted symbols into four sub-channels (M16-QAM) within the 25 kHz bandwidth (another feature common to both systems) enables the systems to be extremely insensitive to delay spreads without the use of an equalizer. Finally, the high bit rate available due to the modulation allows both systems to employ error correction procedures, either trellis coding or convolutional coding, to maintain low received bit error rates even in poor signal conditions.

1.5 Key features

The main priority of each of these systems is to satisfy the users of dispatch applications, with a secondary focus on some cellular-like applications for the mobile work group market. Therefore, the primary feature around which both of the systems are centred is dispatch communication. Each system allows for the definition of talk groups, and for both 1:1, 1:*N*, and *N*:1 communication in the dispatch mode. However, in an effort to provide additional functionality for those who require it, both systems also allow for the possibility of connecting the network to the public telephone network, and allowing users to access that network in the same way as in a typical cellular telephone network. These systems employ advanced handover algorithms which employ information not only from the fixed network side but from the mobile side as well. Handover takes advantage of the feature of a TDMA system that the mobile unit must transmit on one out of six slots, receive on one out of six slots and is idle for the rest of the time. During its idle periods, the mobile unit monitors the signal conditions of nearby sites and maintains a list of preferred handover targets. This type of algorithm provides more consistent coverage throughout the service area. A final voice related feature is a high quality speech coder available in some of these systems which uses two out of six slots in each frame to provide nearly toll quality voice to those customers who need it.

Recognizing the needs of its users, both systems also offer advanced data transmission features. These systems offer circuit data connections for applications like fax or electronic mail. They also offer connection oriented and connectionless packet data capability for applications like database access or other interactive connections that require sporadic transmission of small data messages.

1.6 Key benefits

The primary benefit of each of these systems is the ability to offer a varied suite of services to its users in an integrated system and with a single subscriber unit. Each of these systems can offer dispatch communications, telephone interconnect communications, paging, short message service, circuit data, and packet data services, and these services are all supported on the same basic infrastructure. The design of the infrastructure, too, points to a significant benefit of the systems. The telephony related networking and switching functions are based on the GSM mobile telephone system architecture. This globally recognized architecture with its interfaces incorporates all of the network management functions, including billing, which the provider of the system services will need. Also, the architecture allows for flexible scaling, so that a service provider can begin with a small system and a select set of services. The system size can grow, as well as add services, as the provider's capabilities allow.

2 Project 25

2.1 Origin

In the United States of America, the original Association of Public Safety Officials (APCO) Project 25 Committee brought together representatives from local, state, and federal government agencies to evaluate advanced technologies for private land mobile dispatch radio. A user-driven steering committee guides the process, making all functional decisions. Consisting exclusively of users, the steering committee includes representation from many agencies. Project 25 was initiated in 1989. A Memorandum of Agreement was signed with the TIA in 1991, to utilize the resources of the TIA, as an ANSI-accredited and ITU-recognized Standards Development Organization, to supply the necessary technical support (developed in TIA Engineering Committee TR-8 as the 102-series of technical documents). Phase I of the Project 25 standard was completed and presented at the August 1995 APCO Convention. Phase II commenced shortly thereafter, with the development of standards for 6.25 kHz FDMA (improved spectrum utilization), a TDMA solution, with high-speed data[2], and a number of system enhancements as its key points of focus. Attention is also paid to interoperability with legacy equipment, roaming capacity and spectral efficiency/channel reuse. In addition, Phase II is undertaking activity involving console interfacing, fixed station interfacing and intersystem interfacing labeled inter sub-system interface (ISSI), and man-machine interfaces for console operators that would facilitate centralized training, equipment transitions and personnel movement. Project 25 is included in Report ITU-R M.2014.

2.3 Description

Project 25 systems are operating today in the VHF, UHF, and 800 MHz bands utilized by public safety communications, and have been for a number of years. A key element of the Project 25 technology is its ability to coexist with existing analogue systems, enabling a graceful migration from analogue to digital, while maintaining an emphasis on interoperability and compatibility among conventional and trunked system implementations. Primary channel characteristics were chosen early in the process in favour of band splitting current 25 kHz operation to 12.5 kHz for Phase I and then to 6.25 kHz equivalency as part of Phase II. These decisions were weighted in consideration of the United States of America Federal Communications Commission (FCC) refarming plan. The TIA-905 series of standards are related to Project 25 Phase II, 2-Slot TDMA series, meeting the 6.25 kHz FCC Mandate.

A modulation was selected from the QPSK family because of its unique ability to support graceful migration. A four-level FM modulation known as C4FM was selected for Phase I operation in 12.5 kHz, which offers interoperability with C-QPSK linear modulation for Phase II FDMA operation at 6.25 kHz. Phase I and Phase II operate with identical channel formats at 9 600 bit/s. Phase II will use the same vocoder and encryption as Phase I, thus assuring compatibility and interoperability. The modulation selection for Phase II TDMA is in process, involving Project 25 Phase II, 2-Slot TDMA and 2-Slot TDMA Voice Coder or Codec differentiator options.

In the year 2000, FCC spectrum allocation provided public safety in the United States of America with 24 MHz of spectrum in the 700 MHz band. As part of this spectrum allocation, the FCC specified the Project 25 Phase I Common Air Interface as the interoperability mode for 700 MHz. This band will support 6.25 kHz bandwidth that can be aggregated based upon user requirements to 25 kHz for integrated voice and data. It also supports a bandwidth of 50 kHz that can be aggregated to 100 kHz, and 150 kHz bandwidths for high-speed data applications. Project 25 documents define the various interfaces (such as common air interface, data interface, ISSA, network management interface, telephone interface, etc.).

Recognizing the need for high-speed data for public safety use, as expressed in the Public Safety Wireless Advisory Committee (PSWAC) final report[3], among others, the P25 steering committee established the P25/34 Committee to address proposed Phase III implementation. Similarly to the P25 approach, the steering committee established the P25/34 user forum to address this issue. Phase III activities are addressing the operation and functionality of new terrestrial and aeronautical wireless digital wideband/broadband public safety radio standards that could be used to transmit and receive voice, video, and high-speed data in a ubiquitous, wide-area, multiple-agency and

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2 Related to Wideband Data Standards Project (TIA 902-Series). Currently preparing for ANS and in addition has been a contribution to the MESA broadband project.

3 URL: http://www.fcc.gov/Bureaus/Wireless/News_Releases/nrwl6043.txt

multiple-vender network. On 1 June 1999, the P25/34 committee released its Statement of Requirements for a wideband aeronautical and terrestrial mobile digital radio technology standard for the wireless transport of rate intensive information. These Project 34 activities relate to the international Project MESA work.

Wideband Data Standards Project (TIA 902-series documents): Recognizing the need for common public safety land mobile radio standards that allow for higher data rates than previously available, TIA TR-8 has and continues to develop digital radio wideband data standards. Both Project 25 and Wideband Data are open standards, intended for multiple vendor availability. The 902-series standards are currently being prepared for publications as ANSs.

In the United States of America, regulatory decisions and plans helped to spur development of LMR wideband standards, including the dedication, by the FCC, of spectrum in the 700 MHz frequency band for wideband data. The channels are at 50 kHz, and can be aggregated to 150 kHz, allowing users data rates as high as 700 kbit/s. The TIA-902 series of standards for this technology are mainly expected to handle data, however voice traffic is also supported. Interoperability at this point primarily involves the over-the-air interface. The FCC has mandated the use of the Project 25 standards for the interoperability channels in the spectrum at 700 MHz.

2.3 System configuration

Project 25 is intended to cover a broad range of system configurations, spanning conventional unit-unit direct communications, and repeater based communications, as well as trunking system configurations covering anything from single site through nationwide networking. A generic system model was adopted depicting the maximum complexity, although systems may embody all elements at one extreme, or just a simple conventional base station at the other extreme. An alternative system model is also described for infrastructure-less unit-to-unit direct communications. Project 25 is designed to serve an environment of many different Project 25 compliant systems operating either independently, or in concert.

2.4 Key attributes

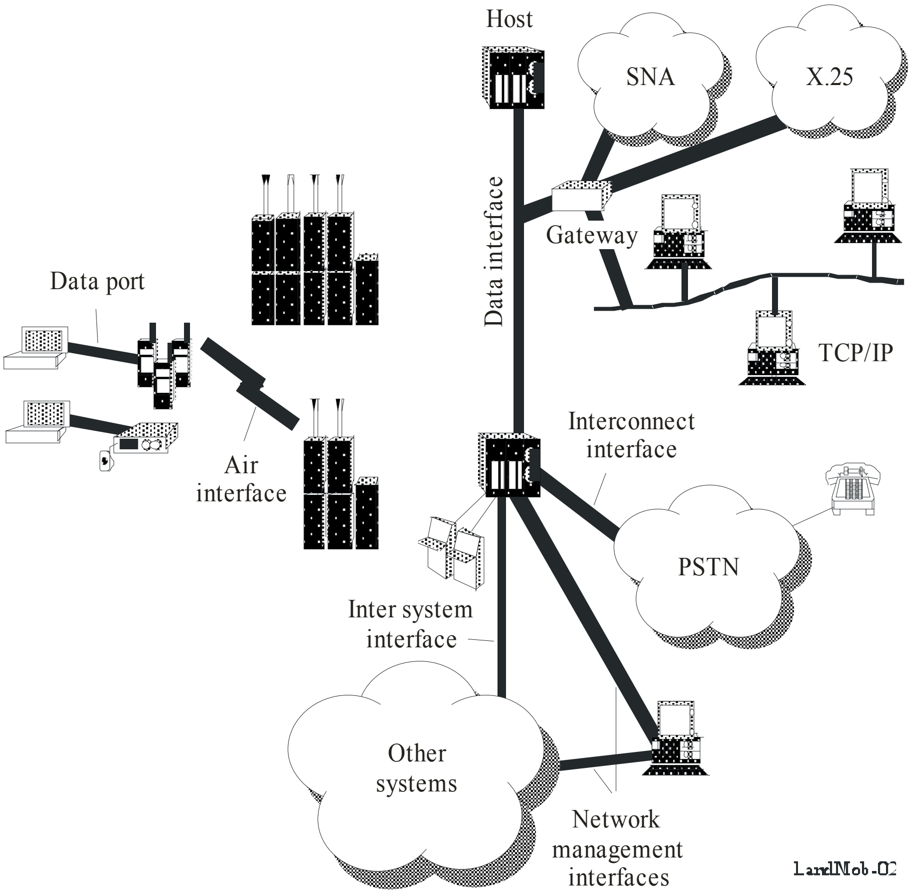
The fundamental key attribute of Project 25 is its common channel formats for a wide-range of configurations and applications. Formats for conventional operation are common with trunking operation. Formats for data are common with trunking control or conventional control. Encryption mechanisms are applicable to any service format, should it be data, voice, or control. This commonality yields the “integration” desired for the system definition. Another key attribute of Project 25 is in the hierarchical addressing and inter-subsystem organization which will permit multiple Project 25 systems to either coexist, or offer coordinated service at the discretion of the system operators.

2.5 Key features

The main priority of each of these systems is to satisfy public safety users of conventional, trunking, and data applications. The system allows for the definition of voice communications in a group call, private call, or interconnected call. The system further allows for the definition of circuit data or packet data bearer services. Any voice or data service may be digitally coded although clear, or encrypted. Further, the digital control necessary for coordinating trunking system operation may be digital clear or encrypted. A hierarchical system design provides for the ability to connect RF subsystems together to provide roaming and or wider-area communications.

FIGURE 2

Example Project 25 system configuration



The services provided by Project 25-compliant systems are intended to be utilized 24/7 with ubiquitous coverage, including inside buildings and structures (with most newer systems), and throughout the geographical areas of responsibility. Current Project 25 systems, TIA wideband data standards and future P34/Project MESA capabilities and technology will share many compatibility requirements and functionalities.

Project 25-compliant systems are being increasingly adopted and deployed, allowing for a high degree of equipment interoperability, compatibility and economy of scale. Currently, over 50 countries utilize Project 25-interoperable equipment or networks. Additional information on Project 25 is available at: http://www.tiaonline.org/standards/project_25/.

2.6 Key benefits

The primary benefits achieved by the public safety community can be summarized in several key areas:

– Competitive procurement of equipment allows public safety agencies to bid common equipment from multiple manufacturers. Forty-eight manufacturers have signed the Project 25 Intellectual Property Rights (IPR) Memorandum of Understanding (MoU), representing infrastructure, terminal and test equipment suppliers.

– Interoperability ensures that public safety agencies may communicate over-the-air and through infrastructure, or directly among subscriber units, with equipment from different suppliers.

– Spectrum efficiency is achieved as users gracefully migrate to digital technology, simultaneously moving from 25 kHz channels, through 12.5 kHz, and on to 6.25 kHz equivalency.

– User-friendly operation delivers capabilities to today's public safety market that are common across bands, system configurations, services (voice, secure, data and trunking) and manufacturers.

3 TETRA

3.1 Origin

TETRA is the ETSI standard for digital trunked land mobile radio addressing traditional private mobile radio (PMR) and public access mobile radio (PAMR) user needs in Europe. TETRA is now widely deployed in Europe in the Far East, Middle East, Africa and South America. TETRA is also a standard in the People's Republic of China. TETRA has already standardized and harmonized and will expand the market for digital LMR in the same way GSM has for cellular systems.

Work on the TETRA standard was started in late 1989. The TETRA standard was produced by voluntary work within ETSI from both users and manufacturers, funded by ETSI and the EU Commission.

The core TETRA standards were voted full European Telecommunication Standard (ETS) status on 29 December 1995. The first TETRA compliant systems became operational during 1997. Because TETRA has been designed to meet the needs of traditional PMR and PAMR users, TETRA is in operation by public safety, transportation, utility, defence and industrial organizations. Also, TETRA is used by a number of commercial operators to provide PAMR services.

The TETRA MoU Association was founded in 1994 as a joint effort of users, manufacturers, operators and regulators to support ETSI in the standardization process, to advice in spectrum matters and to provide forum for discussion. Currently the MoU has close to 100 members and is also managing the interoperability specification and testing process that certifies interworking between different manufacturers' products.

3.2 Description

TETRA operates with a TDMA channel access method, using four logical channels per carrier. The data rate for one carrier is 36 kbit/s, and a carrier occupies 25 kHz.

Each channel can support speech using an algebraic code exited linear prediction (ACELP) codec running at 4.567 kbit/s, plus error correction, making an aggregate of 7.2 kbit/s. Alternatively a channel can support user data at up to 7.2 kbit/s. Multiple channels can be combined in a single communication to provide bandwidth on demand, giving a maximum data throughput of 28.8 kbit/s to a single user or group.

A TDMA system such as TETRA consists of frames, each containing four slots. A group of 18 frames is known as a multiframe, of which one complete frame is reserved for signalling, making it possible to signal a radio that is engaged in a call. The framing system is broadcast by the system, and so known by both infrastructure and mobile stations.

Together with the introduction of TETRA standards the European PMR spectrum regulation took major steps towards harmonization. A sharing agreement with NATO and the resulting European Radiocommunications Committee (ERC) Decision (96) 01 opened a new harmonized European allocation for emergency services between 380 MHz and 395 MHz allowing cross-border cooperation. Accordingly, the ERC Decision (96) 04 opened common allocations for TETRA users outside the emergency services, of these the one between 410 MHz and 430 MHz has turned out to become the most popular. Outside Europe the 800 MHz band has become the *de facto* TETRA allocation at higher frequencies with products introduced from multiple suppliers.

3.3 System configuration

LMR systems themselves may be privately owned for the exclusive use of the owner, or owned by a third party referred to as PAMR or shared systems.

The core standard is a voice and data trunked standard, where the individual time slots can be allocated as, control, voice, circuit mode or packet mode data. There is also a direct mode standard, allowing mobile subscribers to talk directly to each other without the need of network infrastructure. A direct mode operation (DMO) standard supports gateway and repeater options to extend DMO coverage and to interwork with the network communication.

A generic system model was developed. It should be noted that the interfaces standardized make it possible to design from a very small single site system to large nationwide systems, as there are no artificial boundaries within the infrastructure.

3.4 Key attributes

The advantages of TETRA are its features such as fast group calls with many different user groups, priority calling facilities, high spectral efficiency – and telephony features offered by cellular telephone systems. A prime attribute of TETRA is its very thoroughly defined TDMA air interface, where also the services in terms of voice, messaging, data and supplementary services have been fully defined to ensure true interoperability in multi-vendor environment.

TETRA supports two levels of confidentiality, at the air interface and end-to-end, as well as an authentication mechanism. Certain groups of users will require a higher degree of security than is available by just protecting the air interface alone, providing an end-to-end encryption mechanism at all points in the infrastructure.

TETRA systems have to permit users to operate across several sites. Indeed, another of the strengths of TETRA is that groups can have members distributed across sites without interruption or degradation of communications up cross-border talk groups.

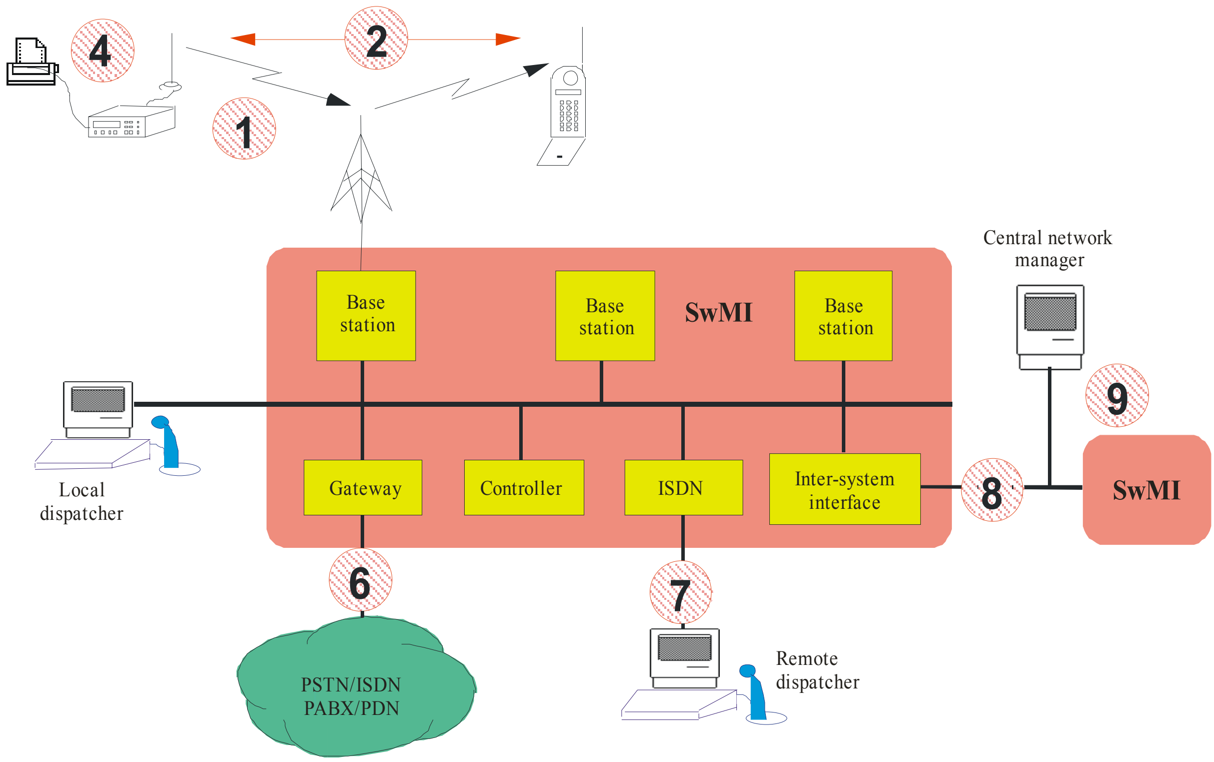
3.5 Key features

Many services and supplementary services are provided, including: individual, group, broadcast, acknowledged group call, status and short data messaging, protected and unprotected data in various bit rates as well as IP packet data. For supplementary services the first set selected for standardization and implementation is comprised of: priority call, late entry, pre-emptive priority call, discrete listening, ambience listening, area selection, access priority, call authorized by dispatcher as well as supplementary services related to one-to-one calls and telephony.

TDMA provides two easy key features full duplex communication, and allocation of bandwidth on demand. Another primary feature is its dispatch communication with the definition of talk groups, and for both one to one, one to many, and many to one communication in the dispatch mode.

FIGURE 3

TETRA Interfaces



SwMI: switching and management infrastructure

1. System air interface
2. Direct mode operation (DMO) air interface
4. Peripheral equipment interface (PEI)
6. Gateway to external network
7. Line station (LS) interface
8. Inter-system interface (ISI)
9. Network management interface

LandMob-03

3.6 Key benefits

The primary benefits of TETRA are:

– One open standard covering all traditional PMR and PAMR user needs.

– Competitive procurement of equipment in a multi-vendor environment for user organizations.

– Interoperability of equipment supplied by different independent manufacturers.

– Greater occupied bandwidth spectrum efficiency (4 logical channels in a 25 kHz channel spacing) thereby reducing spectrum needs for a given grade of service.

– Technology evolution through the provision of new services in TETRA Release 2 to meet future user needs.

4 TETRAPOL

4.1 Origin

The development of the publicly available specifications for TETRAPOL has been carried out by the manufacturers of the TETRAPOL Forum and the TETRAPOL Users' Club. The TETRAPOL specifications aim to satisfy primarily the public safety sector and could be used by other large private networks.

More than 50 TETRAPOL systems have been in operation worldwide since 1994, among them eight national networks. They offer a large range of services, excellent call confidentiality, uniform transmission quality and improved utilization of spectrum.

4.2 Description

TETRAPOL is a fully digital voice and data system, which can operate in the radio bands from 70 MHz to 933 MHz, using FDMA access technology. FDMA allows frequency efficient direct mode and simulcast operation. The carrier spacing is 12.5 kHz or 10 kHz to allow easy migration from analogue to digital. GMSK modulation is used, which is similar to the form of modulation used in GSM. TETRAPOL is fully compatible with ETS 300.113 for radio and ETS 300.279 for EMC (ETSI standards for radio equipment type acceptance).

The TETRAPOL codec uses a robust noise protected RPCELP encoding algorithm running at 6 kbit/s with a speech frame of 20 ms. Convolutional or BCH codes plus interleaving and error detection are used to protect information against various types of burst errors. Logical channels are split into control, traffic and data channels, which can be extended according to traffic demand. One channel can support circuit mode data at 3.2 kbit/s protected against errors, or up to 7.6 kbit/s without protection.

4.3 System configuration

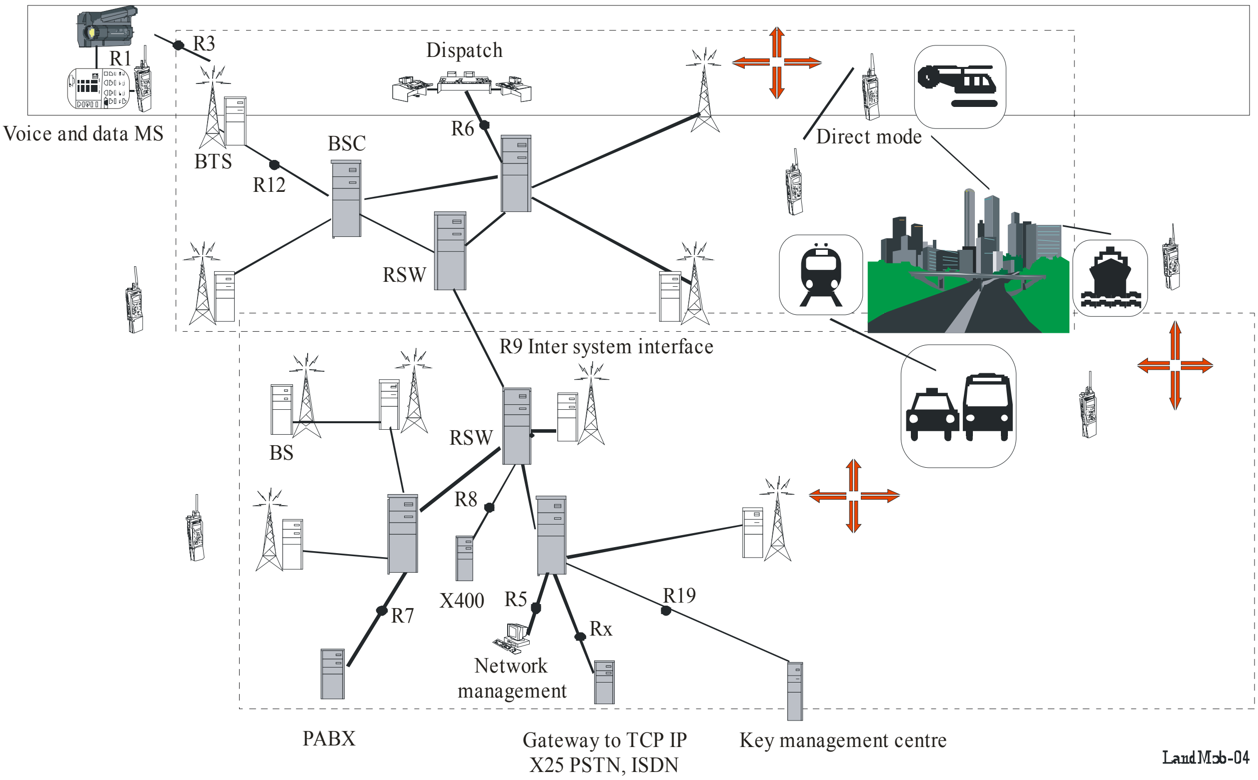
TETRAPOL open interfaces are defined and described in the Publicly Available Specification at each reference point of the model as shown in Fig. 4. There are 21 enumerated reference points for the interfaces. They include the Dispatch interfaces, the key management centre interface, the network management centre interface, and the PABX interface. All of these interfaces are specific to such systems that offer the corresponding services.

Subsystem modules are offered depending of the size of the network including the switches and the base stations. This allows TETRAPOL to cover one cell, a county, a region, or even up to a whole country. The inter system interface allows roaming between several networks.

TETRAPOL Specifications apply to three different modes: a network mode, a direct mode, and a repeater mode. The network mode occurs where the mobile is under the coverage and control of the infrastructure. The network mode includes a trunking mode of operation and an open channel mode. The direct mode occurs when the mobile is directly in communication with another terminal. The repeater mode occurs when the mobile communicates with another terminal through a repeater.

FIGURE 4

Example TETRAPOL system configuration



4.4 Key attributes

The TETRAPOL choices of FDMA and GMSK modulation are key attributes for the market of professional users.

The high receiver sensitivity offered by the TETRAPOL modulation enables large cells and less sites necessary to cover an area, inducing an economy of cost for group calls and talk groups. The possible use of simulcast over large synchronized areas enhances frequency efficiency and reduces infrastructure with very large cell coverage. Paging can be offered.

The absence of a synchronization time allows easy and frequency efficient direct mode operation.

Efficient voice coding allows good performance in a noisy environment.

Security mechanisms have been designed into the protocol from the start to provide high levels of security without compromising performance.

Compression techniques allow TETRAPOL to efficiently transfer images, slow moving pictures, or maps, which can be combined with GPS.

4.5 Key features

As private users have many different requirements and applications, TETRAPOL is like a toolbox from which the user chooses to build the services he wants to get. A core of teleservices are offered like open channel, group calls and talk groups, and emergency calls. A large set of supplementary services can be chosen like pre-emptive priority call, call authorized by dispatcher, or ambience listening. Bearer services for data include circuit mode, packet mode and connectionless mode.

Applications are supported like Messaging through Internet or intranet services through TCP/IP protocols. Short data message protocols such as SMS, status, and paging are offered.

Management of fleets, groups and sub-groups allows dynamic control of the users' call privileges.

The dispatch centre and key management centre interfaces are open for those systems that use dispatch consoles or key management centres.

Different levels of security are offered including authentication and end-to-end security, with an optional SIM.

Existing standards are used whenever possible, such as QSIG for the Inter System Interface, and CMIP or SNMP for network management. The tactical, technical, and operational management, are separated.

Modularity applies to all levels:

– at the coverage level from cell to a region to a country to multiple countries;

– at the addressing level from user to sub-fleet and group.

To address new user needs and evolution to higher data rates, TETRAPOL is designed to allow migration to full IP core for voice and data, wideband radio and new codecs.

4.6 Key benefits

Security and emergency users, which are the primary targeted users, with very stringent requirements, are assured to get:

– robust and efficient systems, with open interfaces from operational systems;

– systems proven in the field, demonstrating the adaptation to users' needs and performance; and

– interoperable equipment from multiple sources.

5 EDACS

5.1 Origin

EDACS was developed by Ericsson and M/A-COM purchased the EDACS system product rights in 2001. It provides advanced digital land mobile radio services for private radio communications and services for industrial, SMR, federal, public safety, and utility organizations, on all levels, including local, state, and national. The first of the EDACS family of documents was published by the TIA, in 1998. By 2000, the TIA-69 Suite of Standards (developed in TIA TR-8 Engineering Committee) for EDACS digital radio systems included four Telecommunications Systems Bulletins (i.e., System and Standards Definition for a Digital LMRS; Digital Air Interface for Channel Access, Modulation, Messages and Formats, and LMRS Packet Data), one Interim Standard (IMBE Implementation) and work on transceiver methods of measurement. The EDACS family of documents is backward compatible and interoperable with existing installed EDACS systems, for the defined services.

5.2 Description

The EDACS family of documents applies to land mobile equipment licensed under National Telecommunications and Information Administration (NTIA) and FCC rules and regulations. They are suitable for 12.5 kHz or 25 kHz channels designed for VHF, UHF, 800 and 900 MHz frequency bands. EDACS supports the following communications modes: digital voice, digital data, encryption of digitized voice, and analog FM for mutual aid capability. The digital voice mode supports the following call types: group calls, group emergency calls, individual calls and system all-calls.

EDACS operates with an FDMA channel access method, running at a carrier data rate of 9 600 bit/s. The EDACS system utilizes a digital modulation technique for all communication including the control channel, digital voice and data modes. This is accomplished through binary modulation of a carrier frequency with two states via a non-return to zero (NRZ) signal. A pre-modulation Gaussian filter is used to reduce the carrier occupied bandwidth. The modulation technique is a form of binary FSK known formally as GFSK. It is a continuous phase, binary FSK modulation with a Gaussian pulse-shaping function. Each channel can support speech using an advanced multi-band encoding (AME) voice encoder running at a gross rate of 9 099 bit/s, which includes error correction.

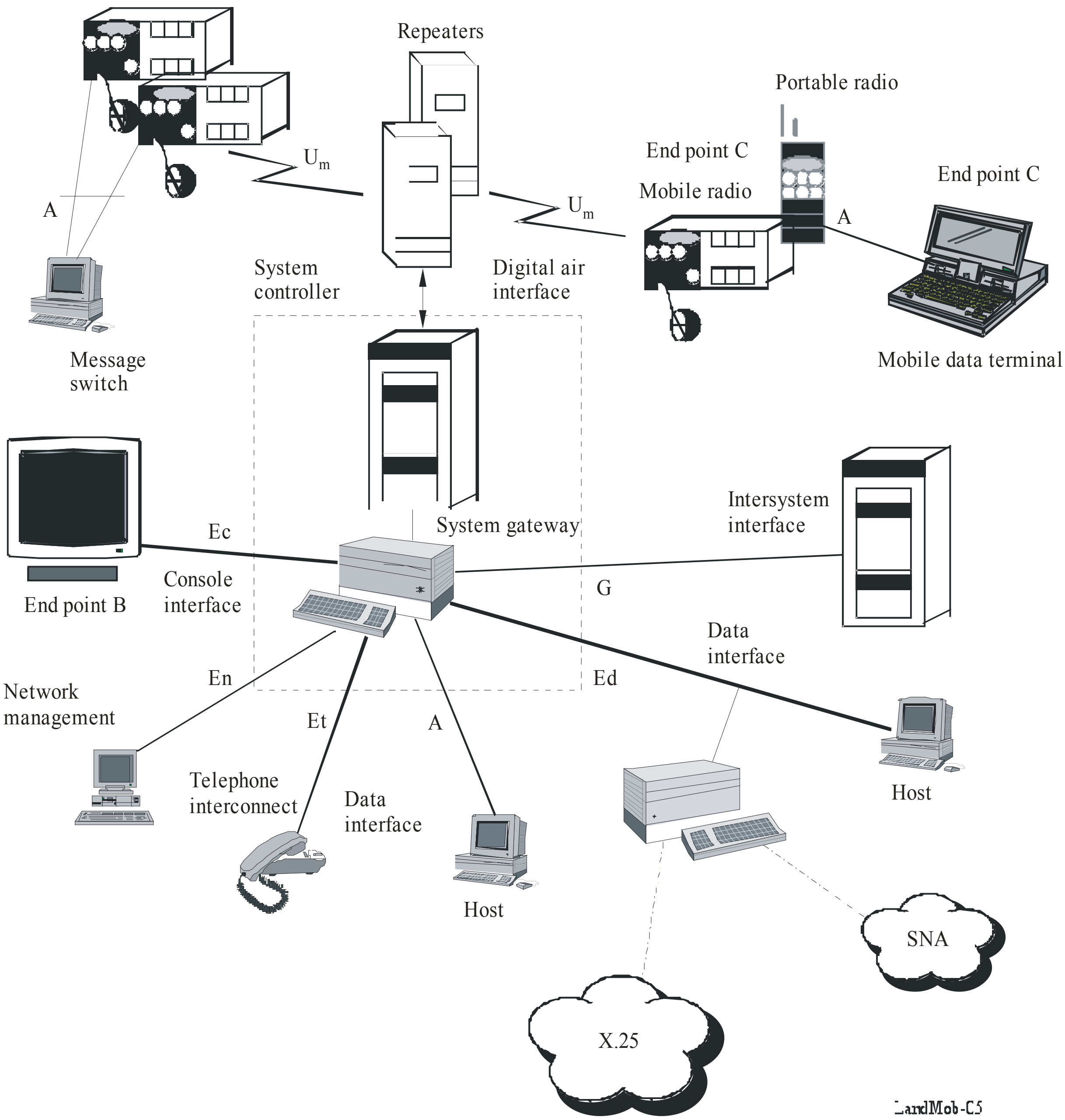
5.3 System configuration

The EDACS system model describes a trunked land mobile radio system utilizing digital voice technology. The system is made up of interconnected elements, acting together. These elements are represented by their physical and architectural attributes, and together describe the entire system, to provide several services for private land mobile radio communications.

Figure 5 displays the physical elements which are mapped into functional groups, such as mobiles or portables, base stations, system controller equipment, and mobile data terminals. Each of these functional groups performs specific functions that are necessary for the operation of the system, and some or all of these functional groups may be present in a specific system. This Figure also identifies a total of seven system interfaces that will be defined by the EDACS documents.

FIGURE 5

Example EDACS system configuration



5.4 Key attributes

A fundamental attribute in meeting the communication needs of today as well as in the future is the EDACS proven migration path. EDACS products and services are designed to be compatible with past, present and future technologies. As an extended life technology, EDACS continues to evolve to accommodate new features and services that are compatible with systems sold since 1987 as well as provide a migration plan to integrate this technology with future, spectrally efficient EDACSIP and OpenSky F-TDMA prism systems.

5.5 Key features

EDACS teleservices include group calls, emergency group calls, individual calls and system all-call. A group call can be a subfleet, fleet or agency call, depending on the group ID. Group calls may operate in one of three different modes: digital voice, digital data or analog mutual aid.

Supplementary services include features such as: fast channel access, automatic call sign, transmit prompt tone, caller ID display, group scan, call queuing, encryption, telephone interconnect, 8 priority levels and an alarm subsystem, etc. These supplementary services change or improve the abilities of the bearer services and teleservices of the system.

5.6 Key benefits

The development of specifications based on EDACS technology provides backward compatibility and interoperability with the large existing base of EDACS equipment and systems, globally.

6 FHMA

6.1 Origin

The FHMA system has been developed in Israel, where a test bed is operating for validation of system evolution. The prime incentive for developing FHMA has been spectral efficiency. The level of spectral efficiency achieved makes it a viable solution for PAMR/PMR services, even when the spectral assignment is extremely small (e.g. 30 frequencies of 25 kHz for unconstrained service coverage). FHMA systems are primarily focused on the PAMR market. FHMA tries to address challenges posed by commercial users. FHMA has been specified and developed to comply with the US FCC regulations (e.g. Part 15, Part 68, Part 90, Part 94).

6.2 Description

FHMA is primarily an advanced digital radio technique, which yields an optimal spectrum efficient mobile radio system. The underlying communication technique is a combination of TDMA (3:1) and frequency hopping multiple access (a CDMA method). Error protection codes, together with interleaving provide excellent protection against deteriorated channel conditions, either due to low received signal power or to interference.

Hopping parameters were selected for accomplishing the objective of high spectrum efficiency for the mobile and to operate in mobile interfered channels. The robustness of the physical layer of the FHMA technology is utilized for capacity enhancement by implementing a cellular reuse pattern with a low frequency reuse factor. The system enables trading reuse for capacity and vice versa i.e. reuse of 1 with smaller capacity per topological unit or opt for a reuse of 3, with higher capacity for same topological unit (base station, sector). The FHMA air interface defines traffic channels and control channels (bi-directional), of which only traffic channels are hopping.

6.3 System configuration

The FHMA system is configured with the FHMA air interface, intersystem signalling through an SS7-MAP standard interface, connectivity through a PSTN telephone interface, and standard Internet connectivity through a line-station interface.

6.4 Key attributes

An effort was made in defining the services and applications so as to provide the community of the mobile fleets with all their communications and control needs by a single system. This includes voice telephony, voice dispatch (individual and group), data bearer services, and data specific applications (e.g. AVL, Manifest).

Teleservices provided by the system include, among other service, mobile to mobile telephony and dispatch speech communications, mobile to group voice communications, selective access to services including optional secure communications, and telephony communications between a mobile unit and a PSTN.

6.5 Key features

FHMA hopping parameters together with error protection codes and interleaving provides excellent resistance to channel degradation from interference.

7 CDMA-PAMR

7.1 Origin

CDMA-PAMR was introduced in Europe to meet the substantial demand for digital PAMR systems and services, including for high-speed data as well as low-medium speed data and PAMR voice services. CDMA-PAMR is a technology that is able to satisfy these requirements effectively, in particular for national and regional PAMR networks, as well as provide a range of other benefits that were previously not provided by other PAMR systems but are beneficial to PAMR users (and operators) for a wide range of applications. The carrier bandwidth of a CDMA-PAMR system is 1.25 MHz and the system operates with a frequency (cellular) re-use factor of 1, implying that the spectrum efficiency for CDMA-PAMR is very efficient.

Although, today there are only limited CDMA-PAMR networks deployed globally, more networks are expected to be deployed in the near future as demand for efficient PAMR voice and high-speed data grows globally.

7.2 Description

CDMA-PAMR utilizes the voice-over-IP technology running over a CDMA radio network in order to provide voice-based PAMR services to users, in addition to data services with a range of data rates. This is implemented by means of a PAMR application running on a server connected to the CDMA radio network, which utilizes features and services of the underlying CDMA radio network (i.e., cdma2000). This flexible approach provides a powerful combination of PAMR voice and data services with convenient interfaces for the development of a wide range of user PAMR applications and solutions.

CDMA-PAMR technology is designed for use for PAMR networks, in particular in the following frequency bands:

– 410-420/420-430 MHz

– 450-460/460-470 MHz

– 870-876/915-921 MHz.

The carrier bandwidth of CDMA-PAMR transmissions is 1.25 MHz, and a spacing of 1.25 MHz is employed between the centre frequencies of adjacent CDMA-PAMR carriers. Unlike FDMA and TDMA systems, CDMA-PAMR does not require a cellular frequency re-use pattern. With CDMA-PAMR, the same carrier frequency can be used by all the base stations in a network (i.e. the re-use factor is 1). This is achieved through the use of “codes” to distinguish between the channels used by different mobiles, as opposed to frequencies and/or timeslots.

Voice and data applications co-exist within the same carrier. One CDMA-PAMR carrier can support up to 35 voice users. Assuming an Erlang B traffic model, that equates to 24.6 Erlangs/carrier at 1% blocking or 26.4 Erlangs/carrier at 2% blocking. The fundamental data rate per user starts at 9.6 kbit/s and can reach a maximum of 153.6 kbit/s on the Uplink. The total capacity for both voice and data is intermediate depending upon the mix of the two services.

7.3 System configuration

The system essentially consists of the following two parts:

– CDMA radio access network, together with the associated IP data network and components.

– PAMR application, consisting of a mobile client part running on a mobile terminal, and a network server part running on servers connected to the network.

Figure 6 shows the architecture of a CDMA-PAMR system.

The push-to-talk (PTT) server/media controller, a key element in the architecture, provides coordination of the PTT call based on the originating member’s requests and the associated response from the subscriber database. The functionalities provided by the PTT server/media controller include: subscriber registration; call processing via SIP; PTT applications, including both point-to-point and point-to-multipoint services; sending out packets with the proper destination IP addresses of each available member for the call in progress; and dynamic activation and deactivation of group members during an active call. The associated subscriber database provides subscriber profile provisioning, group list administration, mobile based administration for end user updates to group lists, and web-based administration for updates to group lists.

The interface between the CDMA radio access network and the IP packet data network/WAN is provided by a packet data serving node (PDSN), which is a standard product for such purposes. This node supports the use of a standards-based protocol that provides header compression to improve the efficiency of over-the-air traffic transmission and, therefore, to provide better voice quality.

In addition to the above-mentioned network elements, PTT subscriber mobiles must be equipped with appropriate client software. The software allows the mobile to interface with corresponding software at the PTT server to effect PTT features/functionality key features CDMA-PAMR provides are a highly flexible environment for the creation of services and applications, and a powerful combination of PAMR voice and data services. Services available using CDMA-PAMR technology include, among others:

– PTT voice services

– Group calls

– Dispatch services

– Prioritization and queuing

– Status and short data messages

– Packet data/IP services

– Simultaneous voice and data

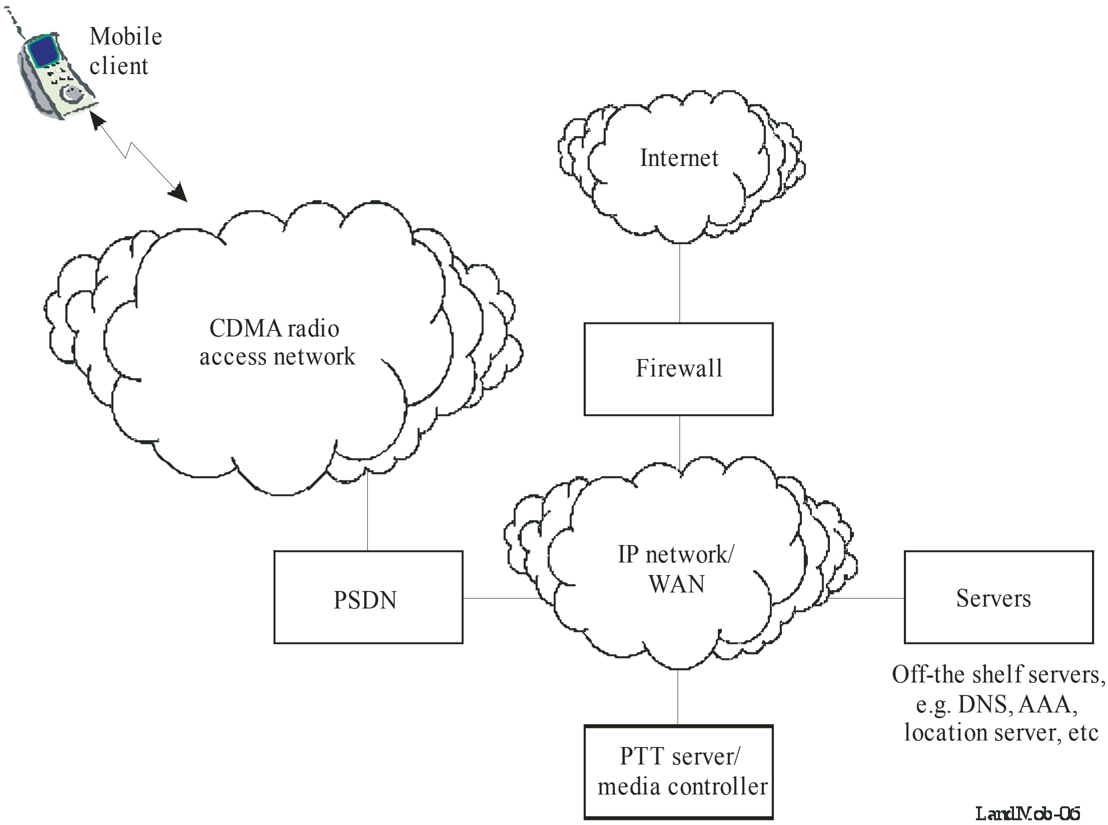
– Dynamic group management

– Over-the-air reprogramming of terminals

– Location services.

FIGURE 6

Example CDMA-PAMR system configuration



In addition to providing such services and features that are traditionally required by PMR/PAMR users, CDMA-PAMR also provides a wide range of other services and features that have not generally been provided by PAMR or PMR systems in the past, but are likely to be beneficial to PAMR users (and operators) for a wide range of applications. Some examples include:

– Flexibility in setting up numbering/addressing schemes for user organizations (and efficient use of scarce network numbering resources).

– Integration with/use of IP-based services such as instant messaging, presence services, intranets, voice-over-IP, end-to-end encryption for voice and data, Web-based services, etc.

– Ability to replay dispatch messages (voice and data) as required.

- Automatic storage and retry of high priority (voice and data) messages until received, with acknowledgement and guaranteed delivery.
- Ability to rapidly set-up an ad-hoc group on a temporary basis, based on a variety of possible parameters (including location), e.g. at a particular site for a particular situation involving all those users who happen to be in the vicinity at that time.

ANNEX 2

PAGING AND ADVANCED MESSAGING SYSTEMS

This Annex provides detailed technical and operational characteristics of various paging and messaging systems and associated codes.

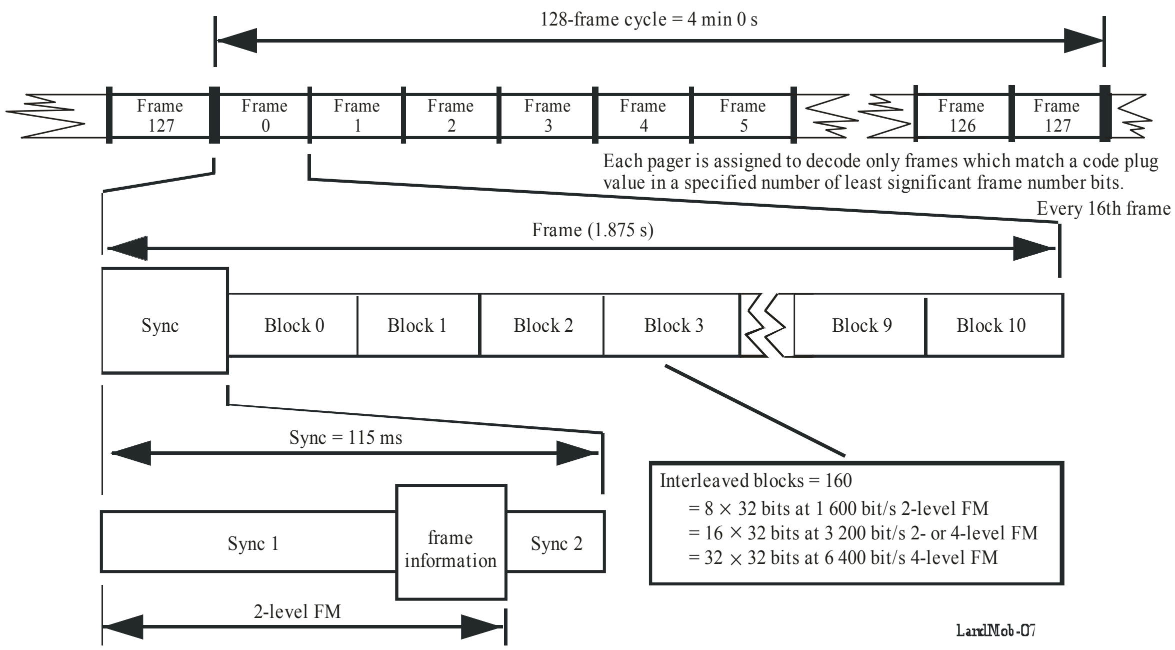
1 Technical characteristics of FLEX

1.1 FLEX coding format

The FLEX coding format operates as a synchronous code and has the added signal fade protection in its data field (addressing, vectoring and messaging field) of sending the bits at an interleaved depth of 8. From the details in Fig. 7, the structure of the repetitive frames each at 1.875 s long, allows for eleven data blocks each containing 8 codewords. The first codeword in Block 0 is assigned to be a Block information word which contains frame and system structure information. This leaves 87 available codewords to be utilized for data delivery. The FLEX coding format has 3 signalling speeds, in 4 formats, to allow its implementation into existing infrastructure with a 1 600 bit/s, 2-level FM system; new infrastructure setups for 3 200 bit/s, using either 2- or 4-level FM (FSK) and the highest speed of 6 400 bit/s, using 4-level FM (FSK). Each of these choices allows the system operator to add on subscribers and infrastructure when system capacity increases are deemed appropriate. The multiple speed feature of FLEX is accomplished by multiplexing one, two or four 1 600 bit/s channels of traffic. These 1 600 bit/s channels of traffic are referred to as “phases”. As a result, we have available data field codewords totalling 87, 174 and 348 for each of the three speeds.

FIGURE 7

FLEX coding format



FLEX is designed to run concurrently with existing paging systems worldwide, including that of the Post Office Code Standardization Advisory Group (POCSAG). System operators need not move to the ultimate FLEX speed of 6 400 bit/s one single step. They can add FLEX 1600 to their existing 1 200 bit/s system by upgrading existing network terminals and continuing service to existing subscribers.

1.2 FLEX key benefits

Faster paging speed

FLEX raises paging speeds up to 6 400 bit/s. This is achieved through multiplexing up to four data streams into one 6 400 bit/s transmission. Each data stream or phase operates independently and pagers (paging services subscriber's receivers) will only decode a single phase. This helps prevent long messages from blocking or delaying other messages.

Higher channel capacity

FLEX supports up to one billion individual addresses and up to 600 000 numeric pagers per channel (based on a typical traffic rate per subscriber). For 10-digit numeric pagers, FLEX has more than four times the capacity of the most advanced POCSAG systems running at 1 200 bit/s. For 40-character alphanumeric pagers, FLEX has a capacity that is five times greater than POCSAG 1200.

Low system cost per user

Higher capacity allows for carriers to add subscribers to existing channels resulting in the lowest cost per delivered bit and lowest cost per user of any pager.

Efficient mixing of services

Today's POCSAG systems quickly run out of capacity when mixing numeric, alphanumeric and information services on the same channel. With FLEX, all of these services can efficiently mix with no penalty to numeric users. This is accomplished by dedicating phases to a single service since these phases operate independently of each other.

Compatibility with existing codes

FLEX is effective alone or when mixed with existing codes. This means that an existing POCSAG system, that is not fully loaded, can migrate to FLEX and initially use only 3.1 % of the existing air time. And, in this 3.1 %, FLEX supports from 5 000 (1 600 bit/s operating) to 20 000 (6 400 bit/s operation) numeric subscribers.

Robust and reliable code

FLEX gives pager users exceptional protection against signal fading which translates to improved page reliability for all paging services, especially alphanumeric and information services. When there is a variation in signal strength, FLEX is able to withstand a 10 ms fade at all speeds and still correctly decode the information.

FLEX improves reliability through checksum validations which is another error detection mechanism; message numbering to allow for missed message retrieval; and positive end of message control by specifying the length of a message. This means that fades will have to be longer in duration to corrupt a word.

Improved page delivery

With it greater reliability, FLEX offers improved page delivery capacity. This means busy hour delays are reduced and has the effect of reducing or eliminating redials into the paging network terminal and subsequent over the air re-transmissions. Not only does this lead to increased customer satisfaction, it also makes for more efficient use of infrastructure resources, like telephone inputs and air time.

Graceful growth

FLEX makes efficient use of existing infrastructure systems by building on the current POCSAG 1200 system. The FLEX system is flexible, running at 1 600, 3 200 and 6 400 bit/s to enable service providers to match their system capacity to market requirements. FLEX enables service providers to dynamically alter the transmission speed to match traffic patterns.

Improved battery performance and smaller pagers

With FLEX, a pager's battery can last up to 10 times longer than pagers running on POCSAG. This is because FLEX has enhanced synchronization capabilities, meaning that the pager can search for its cap code more efficiently therefore saving power. This improved battery life will allow for the use of smaller batteries and the design of smaller, uniquely shaped pagers.

Foundation for future enhanced paging services

FLEX protocol will accommodate the ever-increasing sophistication of the paging marketplace, allowing growth to enhanced services such as two-way paging and nationwide roaming. The FLEX protocol was designed to allow future variations to co-exist on the same system.

2 Technical characteristics of ERMES

2.1 The ERMES architecture

To cater for ERMES services regarding the international traffic handling and roaming capability, the different national networks should be connected to provide extended coverage. The general functional structure of the ERMES system and of the different interfaces is shown in Fig. 8. The ERMES system is divided into two main parts, the telecommunication part and the operation and maintenance part. This architecture follows the M Series ITU-T Recommendations and is similar to that of any international telecommunication networks.

As far as the telecommunication part is concerned, each network is under the control of a paging network controller (PNC) which is detailed in the following paragraph. Paging area controllers (PAC) and base stations (BSs) ensure the radio coverage into one or several paging areas and together form the radio sub-system.

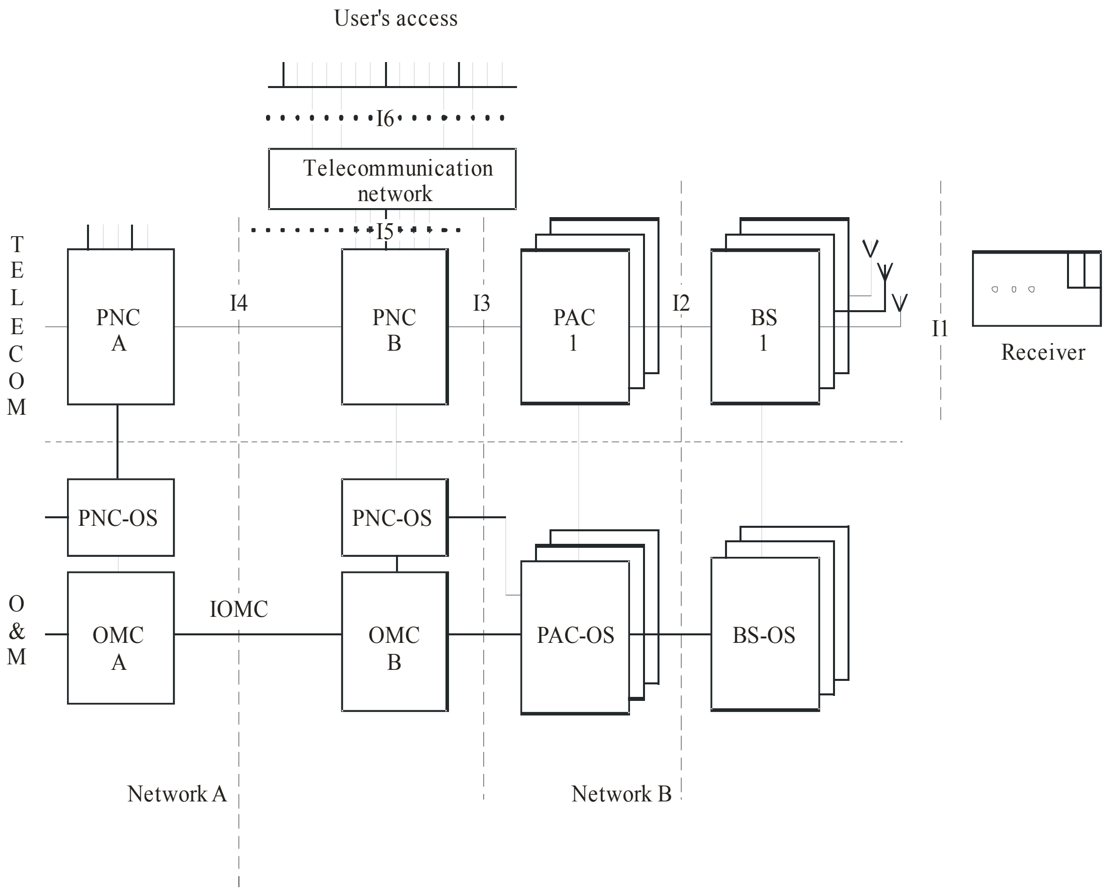
The PNC is the central call processing unit of the network. One PNC is normally in charge of one network and is linked to every other PNC of the ERMES system using the I4 interface in order to provide international paging calls and the roaming facility.

The PNC is responsible for the call processing. The PNC will proceed with a call acceptance mechanism for each call attempt in order to guarantee the offered quality of service. For this purpose, the PNC cooperates with the operation and maintenance centre (OMC), which delivers status information.

Access to the service is gained through the I6 interface of the PNC, which handles the user's dialogue. When the connection between the user's terminal and the PNC is made through a telecommunication network, I5 is the interface between the network and the PNC.

FIGURE 8

ERMES architecture



I1, I2, I3, I4,
I5 and I6: functional interfaces between entities
O&M: operation and maintenance part of the network
OS: operation system
TELECOM: telecommunication part of the network
User's access: set of possibilities offered to users to gain access to the system (telephone, data terminal, telex, ISDN, etc.)

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The PAC, which controls one paging area, organizes the message queuing and batching according to the level of priority and the format of the transmission within the paging area under its responsibility.

The BS consists of one or more transmitters and the associated control and timing equipment. The transmission should be one of the 16 radio channels and organized in a coordinated manner in order to facilitate the receiver synchronization, whatever is its situation, either under its home network or roaming under a visited network.

Interfaces between PNC/PAC and PAC/BS are respectively called I3 and I2. They are internal to the operator's network. External interfaces are I1 (air interface), I4/IOMC (inter-network interworking) and I6 (user interface). I5 is considered external to the paging operator's network but it need not be harmonized with other paging operators.

I1, the radio interface, is based on the following characteristic:

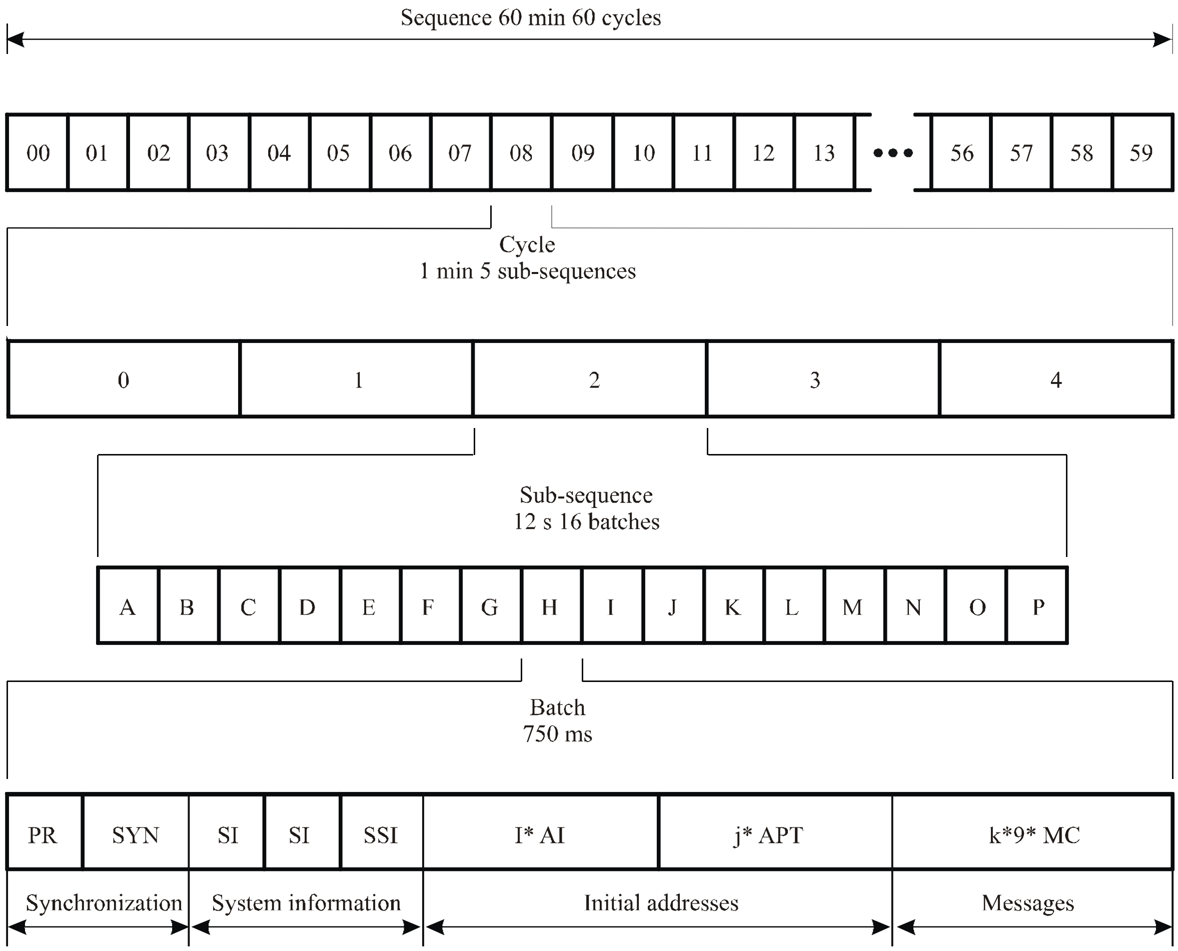
– frequency band: 169.4-169.8 MHz;

Error correction capabilities of the transmission protocol have been designed for this frequency band. However, the transmission protocol is not tied to this frequency band and may operate in other frequency bands as indicated in Annex 1 of ITU-R Recommendation M.539-3. It should be necessary that at least 1 of the 16 channels is common in the network that offers the roaming service. The common channel(s) is not necessarily the same in every network.

– 25 kHz channel spacing,
– modulation method = 4P AM/FM,
– symbol rate = 3.125 kBd (6.25 kbit/s bit rate),
– transmission protocol as described in Fig. 9,
– frequency agile receiver (16 channels).

FIGURE 9

Transmission protocol



AI: initial addresses
APT: address partition terminator
MC: message codewords
PR: bit synchronization word
SI: system information
SSI: supplementary system information
SYN: frame synchronization word

LandMob-09

2.2 ERMES Key Benefits

Faster paging speed

Thanks to a 4PAM/FM modulation, the transmission rate is 6.25 kbit/s per channel. Forward correction is specified using a BCH code. This results in an effective data rate of approximately 3750 bit/s.

Higher system capacity

ERMES operates with one or more of 16 frequencies in one radio band. It is designed so that the receiver is able to receive its calls when they are transmitted on any of these frequencies. It provides for an environment in which there are simultaneous transmissions on these frequencies. Furthermore ERMES can operate with frequency and time divided networks and with simulcast transmission. Receiver numbering uses a 35-bit address split in 5 parts and gives a single number for each receiver with parts indicating the country code and the operator code.

Efficient mixing of services

ERMES provides first basic paging functions with tone only, numeric and alphanumeric services. It also allows advanced services such as unformatted data transmission, closed user group calls, long messages, priority levels, security facilities, roaming, etc.

Robust and reliable code

In addition to forward error correction interleaving is used in order to minimize burst error seriously affecting data reception.

Spectrum efficiency

In terms of users per Hz in the spectrum ERMES has improved upon POGSAG by about a factor of four. This ensures that ERMES can provide a very cost-effective radio link for operator and customers alike.

Improved battery performance and smaller pagers

ERMES is designed to ensure very effective battery economy with a broad range of techniques. First of all, the synchronization of the radio interface on the standard time UTC enables the receivers to determine when any information can be expected. It is possible to tailor the battery needs of different products up to 10 000 times less than existing POGSAG receivers. For instance, a watch style product could be designed for minimum battery consumption.

Open system

ERMES is an open system developed by consensus with operators and manufacturers. It allows a multi-operator environment with multi-sourcing supply.

ANNEX 3

LIST OF ACRONYMS

AM	Amplitude modulation
ANS	American National Standard
ANSI	American National Standards Institute
APCO	Association of Public Safety Officials
ARIB	Association of Radio Industries and Businesses
BS	Base station
CDMA	Code division multiple access
C-QPSK	Compatible quadrature phase shift keying
DIMRS	Digital integrated mobile radio systems
DMO	Direct mode operation
DQPSK	Digital quadrature phase shift keying
EDACS	Enhanced digital access communications system
EMC	Electromagnetic compatibility
ERC	European Radiocommunications Committee
ERMES	Enhanced radio messaging system
ETS	European Telecommunications Standard
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission
FDMA	Frequency division multiple access
FHMA	Frequency hopping multiple access
FLEX	Flexible wide-area synchronous protocol
FM	Frequency modulation
GFSK	Gaussian frequency shift keying
GMSK	Gaussian (filtered) minimum shift keying
GPS	Global positioning system
GSM	Global system for mobile communications
iDEN	Integrated enhanced digital network
IDRA	Integrated digital radio system
IPR	Intellectual Property Rights
ITS	Intelligent transport system

LMR	Land mobile radio
MESA	Mobility of Emergency and Safety Activities
MoU	Memorandum of Understanding
MPHPT	Ministry of Public Management, Home Affairs, Posts and Telecommunications
NRZ	Non-return to zero
NTIA	National Telecommunications and Information Agency
OFDM	Orthogonal frequency division multiplex
OMC	Operation and maintenance centre
PABX	Private automatic branch exchange
PAC	Paging area controller
PAMR	Public access mobile radio
PCS	Personal communication service
PMR	Private mobile radio
PNC	Paging network controller
POCSAG	Post Office Code Standardization Advisory Group
PSDN	Packet data service node
PSTN	Public switched telephone network
PTT	Push-to-talk
QAM	Quadrature amplitude modulation
QPSK	Quadrature phase shift keying
RCR	Research and Development Centre for Radio Systems
RPCELP	Regular pulse code excited linear prediction
SMS	Short messaging service
SNMP	Simple network management protocol
TCP/IP	Transport control protocol / Internet Protocol
TDMA	Time division multiple access
TETRA	Terrestrial trunked radio system
TIA	Telecommunications Industry Association
VoIP	Voice over Internet Protocol
WAN	Wide area network

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