CAPACITY AND SPECTRUM EFFICIENCY ANALYSIS OF AN ASYMMETRIC PMR SYSTEM WITH DAB DOWNLINK

A THESIS

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MASTER OF SCIENCE

By

Ersin Şengül

August, 2003

I certify that I have read this thesis and that in my opinion it is fully adequate, in scope and in quality, as a thesis for the degree of Master of Science.

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ABSTRACT

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M.S. in Electrical and Electronics Engineering

Supervisor: Prof. Dr. Y. Ziya İder

August, 2003

Different trunked Private Mobile Radio (PMR) systems have been designed over the last several decades, all of which have symmetric downlink and uplink channel capacities. Due to this symmetry, those systems may not be spectrally efficient in case of different types of services which are only supported by PMR systems, such as group (acknowledged or unacknowledged) and broadcast calls, either voice or data. In this thesis, a new asymmetric trunked PMR system comprising an OFDM based broadband, wide-area downlink and a narrowband cellular uplink, is proposed to achieve a higher capacity and higher spectral efficiency than current digital trunked PMR systems have. This thesis concentrates on the system capacity analysis of the proposed system associated only with the downlink part for voice communications, as well as the spectrum efficiency comparison of the proposed system with the Terrestrial Trunked Radio (TETRA) system, which is accepted as the spectrally most efficient PMR system. In this study, we study the performance and capacity of the proposed system using Digital Audio Broadcasting (DAB) downlink. In particular, we study the capacity of such a system for voice calls using voice activity detection and statistical multiplexing. Moreover, we show that, the capacity of the system can significantly increase, if the incoming calls, which cannot find an available channel, are allowed to wait a certain amount of time before occupying a channel. The system is shown to have high trunking efficiency since all users are assumed to use the pool of channels available in the wideband downlink. Spectral efficiency of the proposed system and a standard TETRA system are compared using numerical case studies against different traffic loads, cell sizes and number of clusters. The optimum point, with respect to number of clusters, up to which the proposed system is more efficient, is determined. It is shown that for a realistic PMR scenario the proposed system is more efficient up to 5 clusters, i.e. 35 cells, and therefore it can be concluded that the proposed system can be used efficiently in realistic situations.

Keywords: private mobile radio (PMR), spectrum efficiency, digital audio broadcasting (DAB), orthogonal frequency division multiplexing (OFDM), broadband downlink.

ÖZET

DAB UYDU-YER BAĞI OLAN ASİMETRİK BİR PMR SİSTEMİNİN KAPASİTE VE SPEKTRUM VERİMLİLİĞİ ANALİZİ

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Son birkaç on yıl içerisinde değişik PMR (Private Mobile Radio) sistemleri dizayn edilmiştir ve bunların hepsi simetrik uydu-yer ve yer-uydu kanal kapasitesine sahiptir. Bu simetri yüzünden, yukarıda bahsedilen sistemler, grup (onaylanmış ya da onaylanmamış) ve yayın konuşmaları gibi sadece PMR sistemleri tarafından desteklenen değişik tipte servisler göz önüne alıdığında, spektrum bazında verimli olmayabilirler. Bu tezde, günümüzdeki dijital demetlenmiş PMR sistemlerinin sahip olduğu kapasite ve spektrum verimliliğinden daha iyi kapasite ve spektrum verimliliğine ulaşmak için, OFDM (Orthogonal Frequency Division Multiplexing)'e dayalı, geniş bantlı, geniş alan kaplamalı uydu-yer bağı ve dar bantlı yer-uydu bağından oluşan yeni bir asimetrik PMR sistemi önerilmiştir. Bu tez, önerilen sistemin spektrum verimliliği açısından en verimli sistem olduğu kabul edilen TETRA (Terrestrial Trunked Radio) sistemiyle spektrum verimliliği bazında karşılaştırılmasının dışında, önerilen sistemin sadece uydu-yer bağı konuşmaları açısından sistem kapasitesi analizi üzerine yoğunlaşmıştır. Bu çalışmada, uydu-yer bağı olarak DAB (Digital Audio Broadcasting) kullanılarak önerilen sistemin performans ve kapasitesini inceledik. Ayrıca, öyle bir sistemin kapasitesini, ses aktivite algılaması ve istatistiki çoğullama kullanılarak da inceledik. Üstelik, eğer hali

hazırda kanal bulamayan konuşmaların kanal işgal etmeden önce belirli bir zaman beklemelerine izin verildiğinde, sistemin kapasitesinin önemli bir ölçüde arttığını gösterdik. Sistemin demetleme verimliliğinin yüksek olduğu gösterildi, zaten tüm kullanıcıların geniş bantlı uydu-yer bağında hali hazırda bulunan kanal havuzunu kullanabildiği varsayılmıştır. Önerilen sistemin ve standart TETRA sisteminin spektrum verimliliği, değişik trafik yükleri, hücre büyükleri ve topak sayısı bazında nümerik örnek çalışmalarıyla karşılaştırılmıştır. Önerilen sistemin, topak sayısına göre, daha verimli olduğu optimum nokta belirlendi. Gerçekçi bir PMR senaryosu için, önerilen sistem 5 topağa kadar, yani 35 hücreye kadar daha verimli olduğu gösterildi ve dolayısiyle sistemin gerçekçi ortamlarda verimli bir şekilde kullanabileceği sonucuna varılabilir.

Anahtar kelimeler: PMR, spektrum verimliliği, dijital ses yayını (DAB), OFDM, geniş bantlı uydu-yer bağı.

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It is a great pleasure to express my special thanks to my family who brought me to this stage with their endless patience and dedication. Finally, I would like to thank to my fiancée, Ahu, for her unconditional love and interest she has shown during all the stages of this thesis; without her this thesis would not have been possible.

To my family and my fiancée Ahu

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Chapter 1

Introduction

Professional or Private Mobile Radio (PMR) systems are operated by a wide range of users, such as the Police and Fire Services, the utility (Gas, Water and Electricity) and transportation companies, and for on-site applications such as in factories and airports [1]. Several PMR system standards have been developed and adopted in Europe, North America and Japan. Seven well known digital trunked PMR systems are Terrestrial Trunked Radio System (TETRA), Association of Public-Safety Communications Officials (APCO-25), Integrated Dispatch Radio System (IDRA), Digital Integrated Mobile Radio System (DIMRS), TETRAPOL system, Enhanced Digital Access Communications System (EDACS), and GEOTEK-FHMA system. Out of these PMR systems, APCO25, TETRAPOL and EDACS are based on FDMA technology; TETRA, IDRA and DIMRS are based on TDMA; and GEOTEK-FHMA is based on frequency hopping. All of these systems have a cellular architecture and they can use the allocated spectrum as efficiently as possible by making use of frequency reuse. They also have symmetric downlink and uplink channel capacities, having disadvantages in terms of spectral usage in some PMR applications, such as group calls. To illustrate, in APCO25 both uplink and downlink channels have a bandwidth of 12.5 KHz. In TETRA, each channel is one fourth of a 25 KHz TDMA channel. One can refer to [2]-[4] for the details of the current PMR systems.

Although the present systems have been developed using different technologies, and for either general or more specific applications, they share a number of common features and objectives. Some of the common features of PMR systems, which also make them fundamentally different from Public Access Communications Systems (PACS), and which are especially important with respect to spectral efficiency considerations are:

- Typical lengths of calls are very short compared to the public telephone calls in a PACS and they serve a smaller population. Therefore PMR systems have considerably lower traffic per user.

- There are a number of voice and data applications, which are not supported by a PACS such as GSM. Most important of these are broadcast calls and group calls (acknowledged and unacknowledged) for both voice and data communications.

In this thesis, we propose a new PMR system, which has better spectral efficiency especially regarding broadcast and group calls. Spectrum efficiency is highly dependent on the channel assignment schemes and the proportion of different types of calls, since all call types have different channel allocation schemes. When a point-to-point individual call is initiated in a digital trunked PMR system, channel allocations are very similar to those in PACS. When a broadcast call is initiated, a downlink channel must be allocated in all cells in that PMR network. Similarly, when a group call is initiated, either data or voice, at least a downlink channel is allocated in all cells in which there exists a group member. If group members are distributed in more than one cell, spectral efficiency decreases. However, if an asymmetric system as proposed in this study, where the downlink has wide area coverage, i.e. it covers the whole service area, is used, spectrum efficiency of the system is maintained even with group and broadcast calls, because for each broadcast or group call a single channel in the downlink is allocated. In PMR systems, group calls and broadcast calls constitute about 50% of all calls [5]. Therefore an asymmetric system as described above should provide significant increase in spectral efficiency.

Cellular PMR systems can increase their spectral efficiency by making use of frequency reuse. However for reasons as explained above regarding the channel requirements of broadcast calls and group calls, wide area coverage for the downlink part offers opportunity for additional spectrum efficiency. On top of this, a wide area downlink provides trunking efficiency. However this architecture, i.e. cellular uplink and single cell downlink, may not be efficient if the number of cells in the system is increased and more frequency reuse is employed, because in such a system there is no provision for frequency reuse in the downlink part. In this thesis we are aiming at determining the optimum point up to which the proposed system is more efficient than conventional digital PMR systems. We have particularly investigated this problem in reference to realistic PMR systems, which have significantly lower traffic, compared to a PACS.

A new digital trunked PMR system comprising an Orthogonal Frequency Division Multiplexing (OFDM) based broadband, wide-area downlink and a narrowband cellular uplink, is proposed in this study to achieve a higher spectral efficiency than the TETRA system, which is accepted to be the most efficient system among digital trunked PMR systems in terms of spectral usage. OFDM is a spectrally efficient multi-carrier digital modulation scheme, offering reliable reception under hostile reception conditions such as multipath propagation [6], [7]. In addition, OFDM allows for intentional multipath, where all the transmitters within a Single Frequency Network (SFN) operate at the same frequency, thereby achieving wide area coverage. These multipaths are constructive within the Guard Interval, strengthening the signal level [6], [7]. As the OFDM downlink, we have considered in this study the use of a Digital Audio Broadcasting (DAB) system, which has well-established standards in terms of frequency allocations and protocols and reasonable cost [8].

This thesis concentrates on the system capacity analysis of the proposed system associated only with the downlink part for voice communications, as well as the spectrum efficiency comparison of the proposed system with the TETRA system [41],[42]. In the capacity analysis part, the number of subscribers that can be supported by the proposed system is calculated under several constraints such as grade of service (GOS) and bit error rate. In fact, this analysis can be seen as downlink capacity analysis because all capacity calculations are independent of the uplink part of the proposed system. The uplink part will certainly have a cellular architecture and have a capacity much larger than the downlink part by making use of frequency reuse. Therefore the system capacity is upper bounded by the downlink part and the downlink capacity can be seen as the system capacity of the overall proposed system. In particular, we study the efficiency of such a system for voice that would be obtained using silence detection. Moreover, we show that, if the incoming calls are allowed to wait a certain amount of deterministic time before occupying a channel, capacity can significantly increase.

For the spectrum efficiency analysis part, both the proposed system and the standard TETRA system are compared to demonstrate the advantages of the proposed system. In order to compare the spectral efficiencies of these PMR systems, a metropolitan area is assumed to be covered by either one of the systems considering equal traffic patterns. Spectrum requirements in order to achieve the same Quality of Service (QoS), e.g. the same call delay probability, are calculated. In spectrum efficiency calculations, only individual and group calls are considered

Recently, there has been an increasing interest for high bit rate multimedia applications for mobile users, in the general framework of the convergence of cellular and broadcast networks [9], [10]. An asymmetric system with a wideband downlink for higher data rates is desirable, because it is generally accepted that wideband is needed more for downlink. Studies for such asymmetric systems have been undertaken in several projects in which the public access communications system, GSM, is complemented with DAB [11] or Digital Video Broadcasting (DVB) [12]. These proposals consider the broadband downlink only for multimedia applications, and voice calls are still carried by the narrowband channels. However if a broadband downlink is available then it may be possible to achieve spectrum efficiency for speech signals as well, if they are carried by the broadband downlink channel via a suitable trunking protocol. Asymmetric PMR systems have not yet been proposed.

The outline of this thesis is as follows. Chapter 2 contains an overview of PMR environment, including PMR user communities, service requirements such as reliability, fast call setup, speech and data transmission capability, group and broadcast calls, centralized and decentralized operation etc., some PMR configurations, which are used in realistic situations and finally the analogue and digital PMR standards. In Chapter 3, a brief overview of the DAB system is presented, including basic system description and features as well as the basic properties of the modulation with OFDM. Chapter 4 covers the details of the capacity analysis of the proposed system. Both analytical models and simulation algorithms as well as the numerical results are presented and discussed. In Chapter 5, the analysis made for the spectrum efficiency comparison of the proposed system with the standard TETRA system is described and discussed. Chapter 6 concludes this thesis and briefly describes future work.

Chapter 2

Overview of Private Mobile Radio (PMR)

2.1 Introduction

Private (or professional) mobile radio (PMR) systems are systems set up by a company or group of users to provide mobile radio services for that group of users alone. By this way PMR systems differ from public access communication systems.

‘Walkie-talkie’ systems were the simplest form of PMR in which mobile stations communicate each other directly and there is no need for repeaters, base stations or a controlling network. While such systems are simple to set up and cheap to run, they are not very flexible since mobiles have to be within range of each other and calls to other networks is not possible. Later, a new service called PMR446 [13] in which eight 12.5 kHz channels were harmonized across Europe at 446 MHz in the PMR band was set up and replaced walkie-talkie systems.

At the other end of the PMR spectrum, several systems rival or even exceed the complexity of public communication systems. Companies or users may group together and use joint systems that are also called public access mobile radio (PAMR). PAMR systems are shared by several different user groups. PAMR or PMR systems with common protocols and interworking plan have the advantage of enabling users from different systems communicate each other directly. While such interworking can be done by routing calls through a fixed network, this adds delays which should be as little as possible in the PMR field. These interworking constraints have caused a movement towards open standards in the PMR environment.

An important question facing PMR users is which options will provide them with the most efficient service in terms of cost and spectrum. Conventionally, potential users were encouraged to use PMR systems for reasons such as to guarantee sufficient coverage, reasonable cost and to provide supplementary services.

2.2 The PMR User Profiles

There is a wide range of users of PMR systems. The major sectors that are using PMR systems can be summarized as [1]:

- *Public safety*: Emergency services such as ambulance, police, fire, etc.
- *National government*: Governmental agencies such as customs, non-emergency health.
- *Local government*
- *Transport*: Railways, airports, buses, taxis etc.
- *Other utilities*: Electricity, gas, water, coal.
- *On site-PMR*: General-purpose businesses operating in local areas or within their own premises.

- *Other PMR*: Operating over larger areas
- *PAMR*: PMR systems that are jointly used by different users or companies.

A good example of a mature market for PMR is provided by the UK, having the second largest number of PMR users throughout the Europe. The breakdown of the UK PMR market is shown in Fig 2.1.

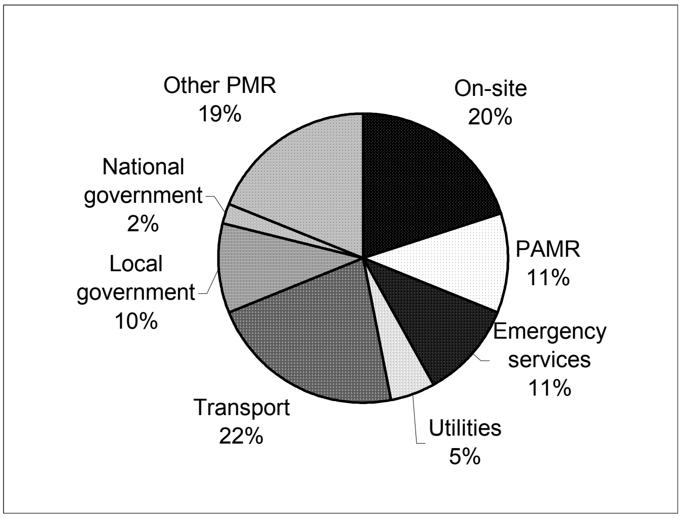


Figure 2.1: Breakdown of PMR user community in the UK [1].

PAMR operators provide many users with many different requirements. Consequently, they want to set up rich systems in terms of features. Furthermore, their interest is to use a spectrally efficient system, since capacity constraints will limit the number of customers.

2.3 PMR Service Requirements

There are several requirements of a private mobile radio system, most important of which provide users with the ability to communicate with each other reliably. To be more specific, it is possible to point out several key requirements of PMR users. In no particular order, these requirements are [1]:

- ***Reliability.*** In the PMR market, many services are used in safety important systems. The user involving in the operation of the service has the advantage of ensuring reliability and being independent from other PMR or PAMR operators. The lack of public access cellular communications systems to ensure quality of service (QoS) or grade of service (GoS) in all circumstances, may force customers to use a PMR system. A survey found that service availability was classed as "extremely important" by two-thirds of those questioned [14].

- ***Speech and data transmission capability.*** In addition to speech services, mobile data services are increasingly used for information updating services, telemetry, or tracking. To illustrate, some companies are conducting trials on systems in which one can transmit medical telemetry, including still images, video, text messages and GPS data to support safety operations. As more data services develop, the applications that make use of them will also develop. Consequently, a flexible data service provision is critically crucial. In the survey, about 80% of users classed data communications as "important" [14].

- ***Point-to-point, group calls and broadcast calls.*** When a PMR system is used, a flexible group call structure should be maintained where users share any information directly rather than having to relay it via others. As a result, in addition to the point-to-point (single terminal to single terminal) calls, group calls, calls involving a number of predefined users, and broadcast calls, where the calls involves all terminals in the network, are required.

- ***Centralized and decentralized operation.*** In many business sectors, a PMR network is used to organize users, therefore a centralized dispatch point is required. However, in some circumstances, it may also be important that users are able to communicate each other in the absence of a dispatcher or even any infrastructure at all. The survey [14] shows that about 80% of users classed direct mode operation as “important”, with over half of these saying that it was “very important”.

- ***Fast call set-up.*** Almost all PMR systems have a pressel or “push-to-talk” button to activate any call to the user group or dispatcher without an answering procedure, rather than dialing a number to set up a call with called party answering a phone. Users look forward to being connected to the desired terminal(s) without delay since calls usually consist of a sentence or two. This situation is extremely important in the emergency services where the system may be used to give urgent commands. Therefore, dropping several words of the message due to call set up delay may have serious results.

- ***Good coverage.*** PMR users usually have less choice as to where to make a call compared to a public cellular user. The area where the user is undertaking the work often limits the available locations. On the other hand, a mountain rescue service may require coverage in areas where public cellular systems are not needed. Sometimes, as well as overall area coverage, lack of blackspots within the covered area is also very essential.

- ***Long battery life.*** In the PMR environment, users maintenance costs some price in terms of lost work time, and reliability of service is also very important. This compares with public cellular systems where users are responsible for battery charging.

- ***Flexibility.*** Flexibility with regard to services has already been covered, but another aspect of flexibility is the ability of the system to change with the developing needs of the operators and users. In particular, the system should be scalable, so that the growth can be supported and adaptable to allow new services, which were not foreseen at the time of system installation.

- ***Low cost.*** Companies using PMR systems will consider costs over the entire life of the equipment, including the capital costs for the infrastructure and maintenance costs. About 95% of the users classify cost as “important” [14].

There are also some other requirements which may not be necessary in all circumstances but will be needed by a large number of companies or users. These are security, call priorities, communication between networks, ease of licensing, and in-house control of system. Many PMR users expect high levels of security, with different forms, in terms of reliability of operation and protection of transmitted information from tampering and interception. Call priorities may be important for example in emergency services, where an emergency call should be able to pre-empt any other calls in order to access the network. In many circumstances, communications with general phone or data networks is a useful feature.

One requirement not mentioned above is efficient use of radio resources. In fact, capacity is normally not an issue to PMR users due to the length of the call and since licenses are relatively cheap in most countries. PMR users are concerned at a more general level as channels become scarce and have to be reused more frequently in high traffic areas such as cities. Division of available spectrum between different PMR users is of general concern since it means little trunking efficiency. As a result, there will be fewer channels than required in most large urban areas. The standardization authorities are likely to insist on spectrally efficient PMR systems with a narrow carrier spacing.

2.4 Operational Scenarios

There are some PMR configurations [1] which are used in realistic scenarios and which enable us to understand basic operation types of the PMR systems. The simplest PMR configuration is point-to-point direct terminal communication. As seen from Fig.2.2, there is no need for an infrastructure, and in most cases, all terminals are within range of each other. Either single frequency is used or different frequencies can be used for different call groups.

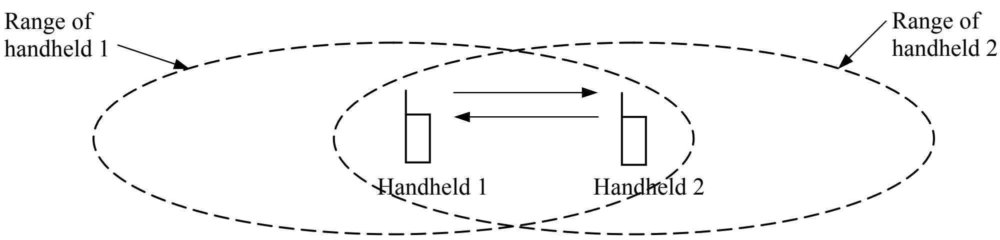


Figure 2.2: Direct mode PMR configuration where communication takes place without any base station or repeater.

One of the most common PMR configurations is dispatch operation. At least two channels are used, one for uplink communications from mobile stations and one for downlink to the mobile stations. Messages from the dispatcher on the downlink can be received by all stations although individual addressing is possible. Fig.2.3 shows the dispatch operation where mobile-to-mobile communication is possible via the dispatcher and links with the public switched telephone or data networks are possible, again via the dispatcher.

If extended coverage is provided and central dispatcher or PSTN network is not necessary, base station can be used as repeater. This effectively increases range of mobile stations as seen from Fig.2.4. If there were no repeater, mobile 1 would not be able to communicate with mobile 2 and 3.

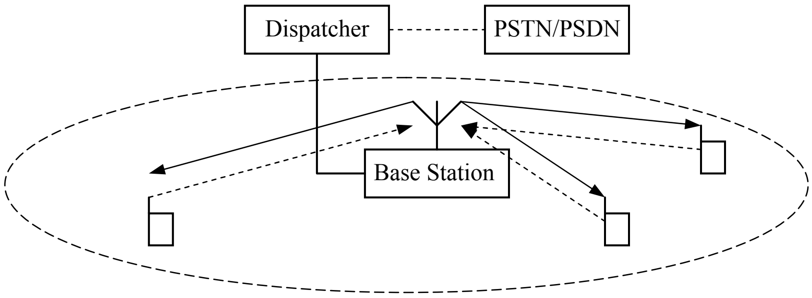


Figure 2.3: Dispatch mode PMR configuration where communication takes place via a base station.

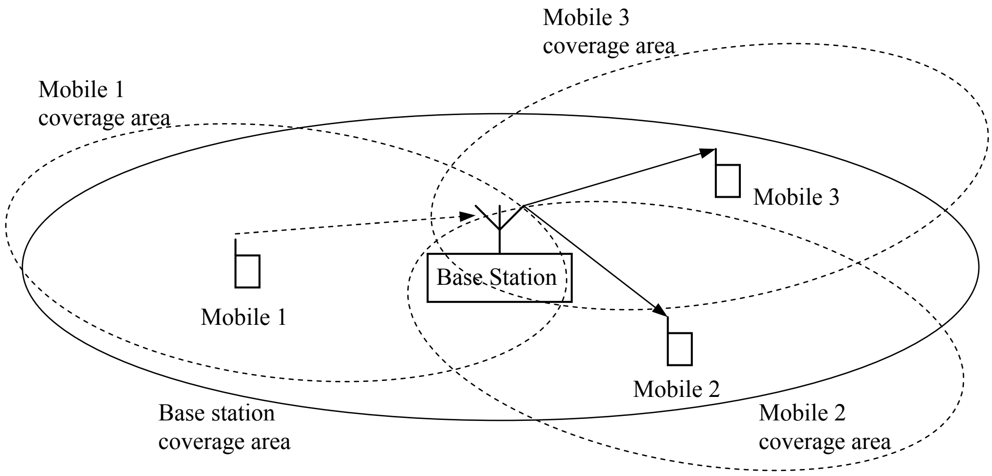


Figure 2.4: Talkthrough repeater operation.

A better option, although it requires more complexity, is trunked operation. In such a case, a number of channels are available and pooled between different PMR operators. This provides trunking efficiency and increases the probability of finding a free channel.

In many circumstances, a single base station will not be able to cover the entire service area. Remote radio ports with smaller powers can be used to cover blackspots. Hand-held terminals usually have lower power than mobile terminals mounted in vehicles especially due to battery and safety conditions, and therefore mobiles can receive signals at greater ranges than handheld terminals. Portable vehicle mounted repeaters can therefore be used to provide handheld coverage to users working near to their vehicles. In Fig.2.5, this mode of operation is shown and commonly used by emergency services.

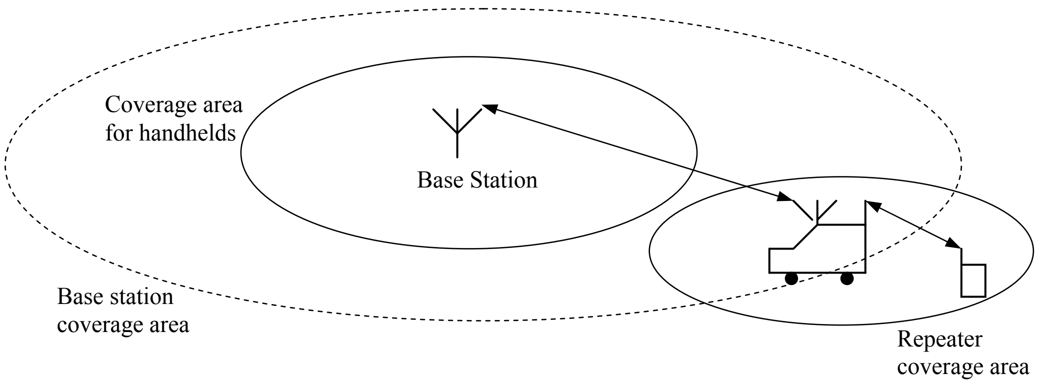


Figure 2.5: Vehicle mounted repeater to increase the local hand-held coverage.

If the service area to be covered is large, several base stations have to be used. If only a relatively low capacity is required, all these base stations can transmit the same signal in a system which known as “simulcasting”. In such a case system behaves in the same way as one large cell. In analogue systems the frequencies used in the different cells vary by a few hertz which reduces problems in overlap regions that receive signals from two or more cells. In a digital system this is more complex and systems must be carefully designed to ensure that terminals can receive an adequate signal in the overlap regions. [15] demonstrates that TETRA, which is digital trunked system, can support simulcast up to a point.

Systems that have to support large capacities require the use of cellular schemes. Such systems have considerably more complex infrastructures than other configurations, supporting switching between base stations and handover of mobiles between different cells. Large PMR and PAMR operators need to use cellular configurations, which are shown in Fig. 2.6, to give them the required capacity. However, even large PMR and PAMR systems do not have as much traffic as public cellular systems have, and have a relatively flat architecture compared to the complex hierarchical network architecture of GSM [1].

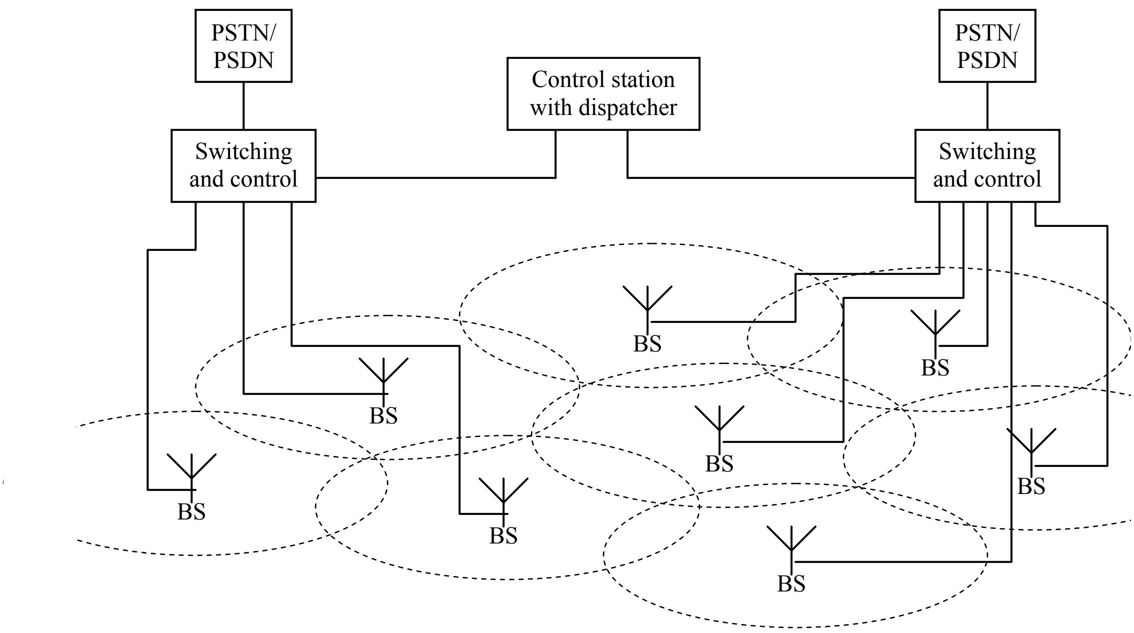


Figure 2.6: Cellular PMR configuration that is used in large PMR and PAMR systems to increase the total capacity.

2.5 PMR Standards

With the move towards digital PMR systems, there has been a trend away from proprietary systems toward a public standard to which equipment must conform, therefore allowing equipment from different manufacturers to be used together. Moves towards public standards have come from manufacturers and operators, as in the case of TETRA, or the user community, as in the case of APCO25.

Public standards have a number of advantages, as shown by the success of GSM in the digital mobile radio environment. Public standards enlarge the market, allowing an economy of scale as well as opening the market for more specialized areas. All the various standards include defined interface points allowing users to use different parts of the system from different suppliers. As well as forcing more competition between providers, it means that suppliers are no longer required to produce all the components of the system. Public standards also give users more freedom to move equipment between networks. This is less of an advantage than it would be in the case of public cellular systems, where some users want a high degree of mobility and roaming between networks. However, it can still be seen to be an advantage to many PMR users, especially those, such as the emergency services, who co-ordinate each other.

2.5.1 Analogue PMR

Early PMR systems were analogue and proprietary. However, a wish to share infrastructure costs and spectrum led to the development of trunked radio systems, with this development came the need for standards so that equipment could be sourced from different suppliers. The earliest such system, which is still available from a number different suppliers, is LTR (Logic Trunked Radio). A number of these systems are still in operation worldwide.

Another major analogue PMR standard is MPT1327 (Ministry of Post and Telecommunications), which was developed in the UK in the late 1980s, but has been adopted by manufacturers and implemented worldwide [1]. Although MPT1327 is more complex and expensive than simpler analogue systems, it is relatively efficient in terms of spectrum usage and offers some capabilities as well as the conventional PMR voice features such as group calls, fast call set-up and priorities. Manufacturers of LTR and MPT1327 equipment are still promoting their analogue systems, in particular for use in countries or areas without capacity constraints.

2.5.2 Digital PMR Systems

In late 1980s, success of wireless communications systems, particularly public access cellular systems, pointed out the need for new types of PMR systems. As PMR systems provide the required services to more customers, the systems have to expand their capacity to accommodate them. There exist two main reasons for the move towards new digital trunked PMR systems, one of which is to provide less costly ways to expand the system and the other is to be capable of supporting many new services.

Digital systems, the systems in which the modulation scheme is digital, offer a large number of benefits over conventional analogue systems. A major advantage is the ability to receive the signal completely as long as the noise level is below a particular threshold value. There is also a disadvantage that when the noise level reaches the threshold of a digital system, the system performance falls off very rapidly, while the quality falls off steadily in an analogue system, as seen in Fig 2.7 [1].

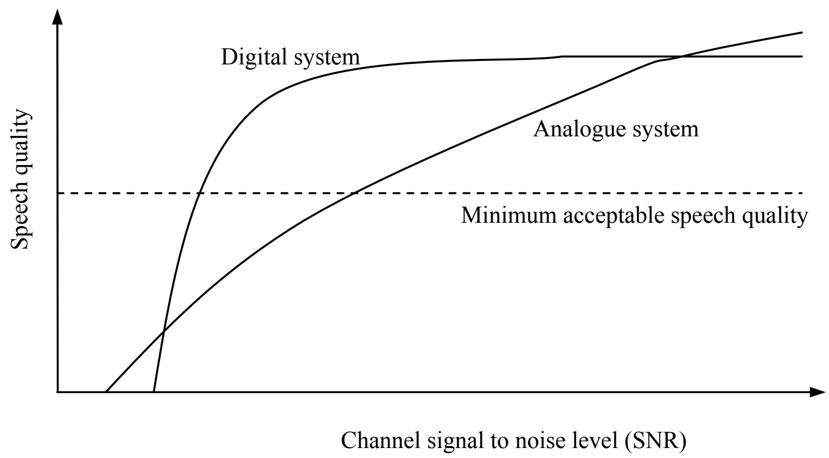


Figure 2.7: Comparison of analogue and digital speech quality with same noise threshold levels.

The basic advantages of digital PMR systems over analogue systems can be summarized as [4]:

- A significant increase in system capacity.
- New services (e.g. data messaging, video services, priority access)
- Improved quality of service
- Added security and voice privacy
- Economic benefits.

Another advantage related to transmission of data, which can be sent directly in a digital system without the need for a modem and for trunking, as a digital signal can be manipulated more easily than a conventional one. Although digital modulation schemes are more complex than analogue modulation schemes, transmission of a speech signal as a digital signal allows for the use of efficient compression algorithms so that the spectral bandwidth needed for a speech signal is lower with digital modulation schemes.

Digital PMR systems are more complex and therefore more expensive. On the other hand, the increased capacity and flexibility, availability of new services, increased spectrum efficiency and quality of service move PMR market to the digital systems as in the case of public access communication systems.

A number of digital trunked mobile radio systems have recently been developed in Europe and North America, Although these systems have been developed for either general-purpose applications or more specific users, they share a number of common features and objectives. There are seven well known Professional or Private Mobile Radio systems; Enhanced Digital Access Communications System (EDACS), GEOTEK-FHMA system, Integrated Dispatch Radio System (IDRA), Digital Integrated Mobile Radio System (DIMRS), TETRAPOL system, Association of Public-Safety Communications Officials (APCO25) and Terrestrial Trunked Radio System (TETRA) [3], [4].

EDACS is a proprietary trunked radio system from Ericsson, featuring distributed processing for enhanced reliability. First systems were installed in late 1980s, finding applications in the military field. When the system is introduced, a major selling point was its data services, which were not used in the PMR systems of that time. The system operates at 9600 baud for 25 kHz channel and 4800 baud for 12.5 kHz channels [4]. Although it was frequency division based, the system evolved to a three-time-slot TDMA system by 1994.

Geotek-FHMA is a digital system, which uses slow frequency hopping as well as FDMA scheme. The technology introduced is novel for the civil mobile radio users, being more common in secure military communications. The primitive incentive for developing FHMA has been spectral efficiency and FHMA systems are primarily focused on the PAMR market.

IDRA system and its standards were developed by the Association of Radio Industries and Businesses in Japan. The technical requirements of the specification aim at satisfying the needs of the users over a wide range of profession, from emergency services to industrial organizations. It has a TDMA based structure and 64 kbit/s transmission rate per RF carrier allocating a bandwidth of 25 kHz.

DIMRS is one of the methods being used especially in North America to provide integrated dispatch services and increase spectrum efficiency. It has a TDMA based structure and 64 kbit/s transmission rate per RF carrier allocating a bandwidth of 25 kHz.

TETRAPOL is a digital FDMA based PMR system. TETRAPOL offers a solution to the PMR environment at lower costs than systems like TETRA, albeit with slightly lower spectral efficiency and a more limited range of services, by combining digital modulation with FDMA technology. The TETRAPOL system uses a 64 kbit/s transmission rate per RF carrier allocating a bandwidth of 12.5 kHz.

APCO25 standard is a combination of conventional and trunked digital PMR technology. APCO25 compliant systems are primarily used for public safety applications. Furthermore APCO25 compliant stations can operate as conventional radios with or without the use of repeaters or as digital trunked radios. There are three basic communication channel types supported by the APCO system; 12.5 kHz digital, 6.25 kHz digital and 12.5 kHz analog for backwards compatibility. APCO system is an FDMA system and all radio channels can be set up as conventional and trunked radio channels. When set up as conventional, the same radio channel is used for both call setup and voice communications. When used as a trunked system, one of the radio channels in a base station is dedicated as a control channel.

2.6 The TETRA System

Terrestrial Trunked Radio System (TETRA) [1],[16] has been developed to allow migration from analogue PMR systems and therefore radio parameters have been adapted. The key air interface parameters are shown in Table 2.1, which indicates that the specific technology adopted for the TETRA system, and are driven by the requirements of PMR environment. As shown Table 2.1, the carrier spacing is 25 kHz, which allows direct replacement of two conventional 12.5 kHz analogue FM channels or a single conventional 25 kHz analogue FM channel.

Table 2.1: Main parameters for TETRA system operation

Parameter	**Value**
Carrier Spacing	25 kHz
Modulation	π/4-QPSK
Carrier data rate	36 kbit/s
Vocoder data rate	ACELP (4.56 kbit/s net, 7.2 kbit/s gross)
Access method	TDMA with 4 time slots/carrier
User data rate	7.2 kbit/s per time slot
Maximum data rate	28.8 kbit/s
Protected data rate	Up to 19.2 kbit/s

Operation of the TETRA system is intended for existing VHF and UHF PMR frequencies. The bands allocated for TETRA are 380-400 MHz for emergency services, parts of 410-430 MHz, 450-470 MHz and parts of 870-933 MHz for civil applications. The duplex spacing is 10 MHz except in 900 MHz band, where it is 45 MHz.

Each carrier in the TETRA system provides four time slots which represent the physical channels. These physical channels are shared between several logical channels which carry both traffic and control signaling information. The four physical channels are derived from a single transmitter, without need for combiners or splitters. Both mobiles and handhelds can operate in full duplex mode without the need for a duplex filter.

TETRA provides the common trunked radio features including; broadcast calls, direct mobile-to-mobile calls, mobile used as a repeater, group calls (with dynamic assignment), encrypted speech, circuit mode data, short messages, conference calls, call diversion, mail box, automatic callback, include call, discreet listening, call number ID, call me back and telephone access. For high security applications, some channels in the system can be removed from the pool and assigned to specified users. The target call setup time is 300ms. Among the claims for TETRA, is the possibility that a multimode TETRA/GSM/DCS1800/ERMES terminal could be made available.

The modulation method chosen is $\pi/4$ quadrature phase shift keying (QPSK), which allows a 36 kbit/s transmission rate on a carrier. The modulation scheme has four symbols that transmit two bits per symbol period. The signaling rate permits 19.2 kbit/s data throughput after error protection coding is employed. Three rates of circuit mode data are available, with the data rate depending on the level of operation. Unprotected data rate is 28.8 kbit/s per 25 kHz of spectrum. When higher protection is required, the throughput is reduced to 9.6 kbit/s for the full 25 kHz bandwidth.

The TETRA system can operate in a quasi-synchronous mode, especially in low traffic regions, in order to increase spectrum utilization. This mode allows use of the same frequencies from different cells operating effectively in parallel. When using this, it is necessary to identify the area where the field strength will be within ± 6dB, from two adjacent sites. In this overlap area the arrival of the two signals needs to be kept within a difference of less than one-fourth of a symbol period, which is 13.9 μs or equivalent to a distance of 4 km. In very low-density areas, TETRA permits time-shared control channels (one frequency can serve as a control channel of a number of cells) and traffic channel assignment on demand.

Chapter 3

Overview of Digital Audio Broadcasting (DAB)

3.1 Introduction

The Eureka DAB system is designed to provide reliable, multi-service digital sound and data broadcasting for reception by mobile, portable and fixed receivers using a simple, non-directional antenna. It can be operated up to 3 GHz for mobile reception, higher for fixed reception and may be used on terrestrial, satellite, hybrid (satellite with complementary terrestrial), and cable broadcast networks. In addition to supporting a wide range of sound coding rates, it is also designed to have a flexible, general-purpose digital multiplex which can support a wide range of source and channel coding options, including sound-programme associated data and independent data services. The detailed specification of the Eureka DAB system is given by the European Telecommunications Standards Institute (ETSI) in final draft ETS 300 401 [8].

The Eureka DAB system is a rugged, highly spectrum and power-efficient sound and data broadcasting system. It uses digital techniques to remove redundancy and continuously irrelevant information from the audio source signal, and then it applies controlled redundancy to the signal to provide the desired error protection. The information to be transmitted is spread in both frequency and time domains so that the imperfections of channel distortions and fades may be removed from the recovered signal in the receiver, even in critical conditions, i.e. in motion and in the presence of signal reflection phenomena due to obstacles between transmitting and receiving point.

Efficient spectrum utilization is achieved by interleaving multiple programme signals and by a special feature of frequency re-use, which allows broadcast networks to be extended, virtually without limit, by running additional transmitters on the same frequency. The latter feature is known as Single Frequency Network (SFN) [8]. Nonetheless, the relatively low co-channel protection ratio of the system also allows adjacent local coverage areas to be planned on a continuously extending basis, with as few as four different frequency blocks.

3.2 System Description and Features

The DAB system provides a signal that carries a multiplex of several digital services simultaneously. The total system bandwidth is approximately 1.5 MHz, supplying a total transmission bit-rate capacity of 2.4 Mbit/s in a complete transmission frame. The amount of error protection provided is adjustable for each service independently, with a coding overhead ranging from 33% to 300%, depending on the requirements of the broadcaster such as reception quality and transmitter coverage. The available net bit-rate for services range between about 0.6 Mbit/s to 1.7 Mbit/s.

The services may consist of either audio or data services where data services may be independent from audio services. Data services, whereby each service can be a separately defined continuous stream, are segmented into 24 ms frames, or can be further transmitted by means of a packet structure. In general, the capacity available for independent data will be limited by the capacity requirements of the audio programme services. Conditional Access (CA) is applicable to each individual service and to each individual packet mode data. The number and bit-rate of each service is flexible and in general receivers are able to decode several services simultaneously [17].

A conceptual block diagram of the DAB system is shown in Fig.3.1. Fig.3.1 (a) shows a conceptual transmitter structure in which each service signal is coded individually at source level and then error protected and time interleaved and then multiplexed into the Main Service Channel (MSC).

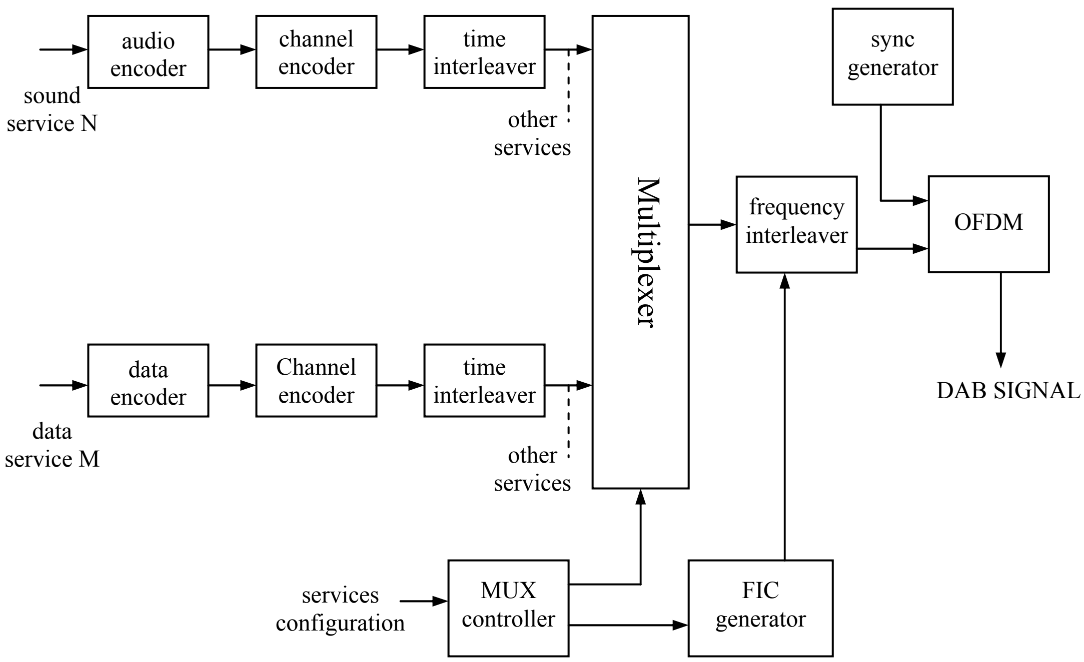


Figure 3.1 (a): A conceptual DAB signal generator

The output of the multiplexer is frequency interleaved and combined with Fast Information Channel (FIC). Finally, very rugged synchronization symbols are added before applying Orthogonal Frequency Division Multiplexing (OFDM) and differential QPSK modulation onto large number of carriers to form the final DAB signal. Fig.3.1 (b) shows a conceptual receiver structure, which performs transmitter operations of Fig.3.1 (a) in reverse order, having selected the wanted DAB ensemble and acquired synchronization.

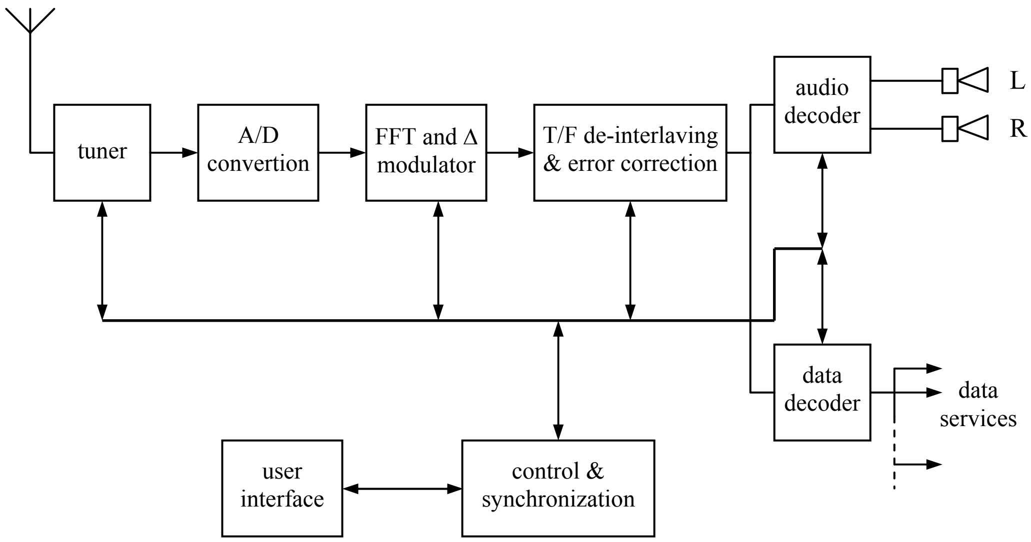


Figure 3.1 (b): A conceptual DAB receiver structure

3.2.1 Transmission Coding and Time Interleaving

The data representing each service being broadcast are subjected to energy dispersal scrambling, convolutional coding and time-interleaving. If needed, greater protection is given to some source encoded bits than others, following a pre-selected pattern known as the Unequal Error Protection (UEP) profile. The convolutional encoding process contains adding redundancy to the service data using a code with a constraint length of 7. The average code rate, defined as the ratio between the number of source bits and the number of the encoded bits after convolutional encoding, may take a value from 0.35 (highest protection level) to 0.75 (the lowest protection level). Different average code rates can be applied to different sources, subject to the protection level required and bit-rate of the source-encoded data. General data services are convolutionally encoded using one of the uniform rates while data in the FIC are encoded at the highest protection rate (1/3).

3.2.2 Main Service Multiplex

The encoded and interleaved data are fed to the Main Service Multiplexer (MUX) where for each 24 ms, the data are combined in sequence into the multiplex frame. The combined bit stream is known as MSC, which has a gross capacity of 2.4 Mbit/s. Depending on the chosen convolutional code rate, which can differ from one application to another, this supports a net bit-rate ranging from 0.6 to 1.7 Mbit/s, accommodated in a 1.5 MHz bandwidth DAB signal. The Main Service Multiplexer is the point at which synchronized data from all of the services using the multiplex are brought together.

3.2.3 The Guard Interval

Multipath propagation represents a severe problem for mobile receivers, because its characteristics can change very rapidly with motion of the vehicle. This can cause problems in the receiver because it has to analyze the received signal over a time window corresponding to the symbol period, and the optimum positioning of this time window varies as the receiver moves. Incorrect positioning of the time window can cause ISI in the received signal.

DAB, in common with other OFDM systems, overcomes this problem by adding Guard Interval to each symbol to be transmitted. The duration of the receiver's symbol window is called the "active" symbol period, which is the reciprocal of the carrier spacing and thus maintains orthogonality, which will be described in Section 3.3. The guard interval extends the total length of the symbol by about one quarter. The DAB system generates the guard interval by inserting it before active symbol and using data identical to that at the end of the active symbol, which avoids discontinuity at the boundary between "active symbol" and guard interval. Fig.3.2 illustrates the use of the guard interval.

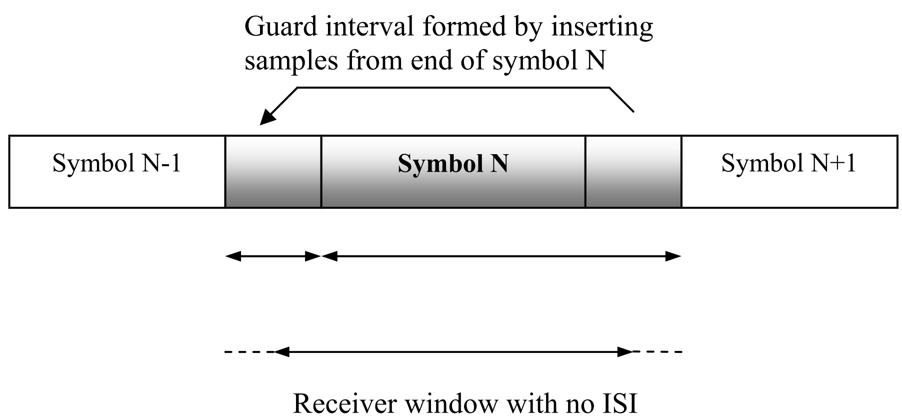


Figure 3.2: Symbols with a guard interval

If two identical signals are transmitted from a nearby and a distant transmitter, the receiver would receive two contributions, one of which is delayed compared to the other. However, it will not be able to determine that the delayed signal is coming from a distant transmitter or from a genuine long-delay echo from the nearby transmitter. The receiver will be able to position its time window correctly and decode the received signal successfully, provided that the delay does not exceed the guard interval. Therefore, a number of transmitters can transmit the same signal on the same frequency at the same (or nearly same) time. The maximum spacing of these transmitters is limited by the guard interval. Thus DAB allows the use of SFNs to provide wide area coverage.

3.2.4 Transmission Frame and Modes

The actual content of the multiplex is described by the "Multiplex Configuration Information" (MCI) and this is transmitted in a specific part of the "Fast Information Channel" (FIC), since it does not experience the inherent delay of time interleaving which is applied to the "Main Service Channel" MSC. The frame structure of DAB ensemble is shown in Fig.3.3.

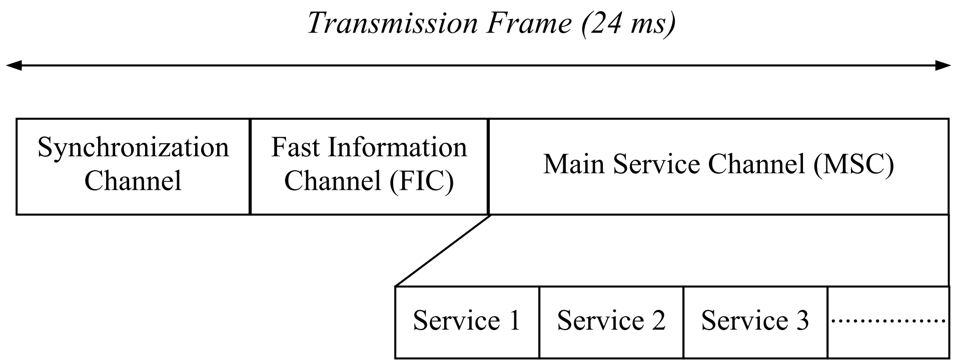


Figure 3.3: An example of a DAB Frame structure

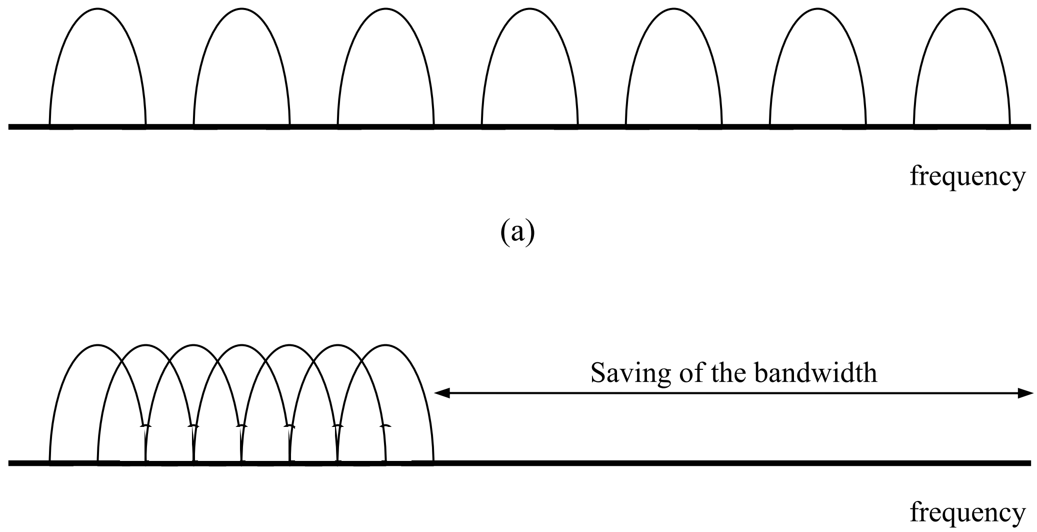
The DAB system provides four different transmission mode options which permits the use of a wide range of transmitting frequencies, upto 3 GHz for mobile reception for different purposes. These transmission modes, which are listed in Table 3.1, have been designed to cope with Doppler spread [18] and delay spread [18], for mobile reception in the presence of multipath echoes. It can be seen from Table 3.1 that all transmission modes have the same spectral occupancy. The carrier seperation is a major factor for the immunity of the system to the effects of Doppler spread in the mobile receivers. In general, while Mode I and IV are suitable for terrestrial transmission and for single frequency networks, Mode II and III are suitable for satellite transmission and cable networks.

Table 3.1 : The four different transmission modes of the DAB system

Mode	I	IV	II	III
Number of carriers	1536	768	384	192
Carrier seperation	1 kHz	2 kHz	4 kHz	8 kHz
Guard interval duration	246 μs	123 μs	62 μs	31 μs
Nominal maximum transmitter seperation for SFN	96 km	48 km	24 km	12 km
Nominal frequency range (for mobile reception)	≤ 375 MHz	≤ 750 MHz	≤ 1.5 GHz	≤ 3 GHz

3.3 Modulation with OFDM

OFDM is a special case of multicarrier transmission, where a single data stream is transmitted over several lower rate subcarriers. OFDM can be seen as either a modulation technique or a multiplexing technique as a solution to the problem of transmitting data over channels with large delay spread [19]-[23]. The major reason for using OFDM is to increase the robustness against frequency selective fading or narrowband interference. In a single carrier system, a single fade or interferer can cause the entire link to fail, whereas in a multicarrier system, only a small portion of the subcarriers will be affected. The concept of using parallel data transmission by means of frequency division multiplexing (FDM) was published in mid 1960s [24], [25]. The idea was to use parallel data streams and FDM with overlapping subchannels, which are spaced *b* apart in frequency, each carrying a signaling rate *b*, to avoid the use of high-speed equalization and to combat impulsive noise and multipath distortion, as well as fully use the available bandwidth. Fig.3.4 demonstrates the difference between conventional non-overlapping multicarrier technique and overlapping multicarrier modulation technique. As shown in Fig.3.4 orthogonal multicarrier modulation technique saves almost 50% of bandwidth. In OFDM, each carrier is orthogonal to all other carriers, and therefore there is no crosstalk between any subcarrier.



(b)

Figure 3.4: Concept of OFDM signal: (a) conventional multicarrier technique (b) orthogonal multicarrier modulation technique.

For large number of subchannels, the arrays of sinusoidal generators and demodulators required in a parallel system make the system to be unreasonably expensive and complex. The receiver needs precise phasing of the demodulating carriers and sampling times in order to keep crosstalk between subchannels acceptable. Weintein and Ebert [26] applied discrete Fourier Transform (DFT) to parallel data transmission system as part of the modulation and demodulation process. In addition to eliminating the banks of subcarrier oscillators and coherent modulators required by FDM, a completely digital implementation could be built around special-purpose hardware performing the fast Fourier transform (FFT), which is an efficient implementation of DFT. Recent advances in very large scale integration (VLSI) techniques enable the making of high speed chips that can perform large size FFT at affordable price. In 1990s, OFDM has been exploited for wideband data communications over mobile radio FM channels, high-bit-rate digital subscriber lines (HDSL, 1.6 Mbps), asymmetric digital subscriber lines (ADSL, 1.536 Mbps), very-high speed digital subscriber lines (VDSL, 100 Mbps), digital audio broadcasting and HDTV terrestrial broadcasting [20], [27]-[30].

3.3.1 OFDM Signaling and Orthogonality

After the qualitative description of the system, mathematical definition of the OFDM signal allows us to see how the signal is generated and gives us a tool to understand the effects of imperfections in the channel.
Mathematically, each carrier can be described as a complex wave:

$$s_c(t) = A_c e^{j[\omega_c t + \phi_c(t)]} \tag{3.1}$$

The actual signal is the real part of $s_c(t)$, and A_c and ϕ_c are the amplitude and phase of the carrier. The values of these parameters are constant over the symbol duration period τ. OFDM contains many carriers, therefore the final signal $s_s(t)$ can be represented by:

$$s_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_N e^{j[\omega_n t + \phi_n(t)]} \tag{3.2}$$

where $\omega_n = \omega_o + n\Delta\omega$. If the signal is sampled using a sampling frequency of 1/T, then the resulting signal can be represented by:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_N e^{j\phi_n(t)} e^{j(n\Delta\omega)kT} \tag{3.3}$$

where $\tau=NT$, $\omega_o=0$ without loss of generality.

Now (3.3) can be compared with the general form of the inverse Fourier Transform [29]:

$$g(kT) = \frac{1}{N}\sum_{n=0}^{N\text{-}1} G\left(\frac{n}{NT}\right) e^{j2\pi k/N} \tag{3.4}$$

Equations (3.3) and (3.4) are equivalent if

$$\Delta f = \frac{\Delta\omega}{2\pi} = \frac{1}{NT} = \frac{1}{\tau} \tag{3.5}$$

This is the same condition that is required for orthogonality, and therefore maintaining orthogonality enables us to make use of Fourier transform procedures.

The "orthogonal" part of the OFDM name indicates that there is a precise mathematical relationship between frequencies of the carriers in the system. In an OFDM signal, it is possible to arrange the carriers in an OFDM signal so that sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference. In order to achieve this, the carriers must be mathematically orthogonal. If the symbol period is τ and the carrier spacing is 1/τ, then all carriers are linearly independent (i.e. orthogonal).

Suppose s_p, s_q are the signals of different carriers, then the signals are orthogonal if

$$\int_b^a s_p(t)s_q^*(t) = \begin{cases} K & for \;\; p = q \\ 0 & for \;\; p \neq q \end{cases} \tag{3.6}$$

where interval [a, b] is a symbol period.

3.3.2 OFDM in Digital Audio Broadcasting

The digital modulation scheme used in Eureka 147 DAB system is based on 4-differential QPSK OFDM, since it meets the transmission needs of high bit-rate digital signals to vehicular, portable and fixed receiver even under critical reception conditions due to the presence of multipaths [8]. This scheme combines the advantages of wideband and narrowband modulation.

As described in the previous section, the information is divided into a large number of bitstreams, having low bitrates individually, which are then used to modulate individual orthogonal carriers, such that the corresponding symbol duration becomes larger than the delay spread of the transmission channels. By inserting a temporal guard interval between successive symbols, channel selectivity and multipath propagation will not cause inter-symbol interference (ISI). As seen from Table 3.2, *N* orthogonal carriers, which can be conveniently generated by an FFT process, is known collectively, as a “DAB Ensemble”.

The spectrum of the DAB Ensemble signal is approximately rectangular, Gaussian noise-like, and occupies a bandwidth of about 1.5 MHz. In the presence of multipath propagation, some of the carriers are enhanced by constructive signals, while others suffer destructive interference (frequency selective fading). Therefore, DAB system provides frequency interleaving by a re-arrangement of the digital bitstream among the carriers, such that successive source samples are not affected by selective fade. When the receiver is stationary, the diversity in the frequency domain is the prime means to ensure successful reception. The time diversity provided by time interleaving provides further assistance to a mobile receiver.

Table 3.2: OFDM parameters for different transmission modes of DAB system

Mode	I	IV	II	III
Frame duration	96 ms	48 ms	24 ms	24 ms
Null symbol duration	1297 μs	648 μs	324 μs	168 μs
Guard interval duration	246 μs	123 μs	62 μs	31 μs
Useful symbol duration	1 ms	500 μs	250 μs	125 μs
Total symbol duration	1246 μs	623 μs	312 μs	156 μs
Number of radiated carriers	1536	768	384	192

3.3.3 Multicarrier Allocation in OFDM

OFDM used in DAB assigns the same number of bits and an equal amount of power to all subchannels, regardless of their individual characteristics. This is expected, since OFDM is used in DAB for only broadcasting audio services and related data, without considering the large number of listeners' conditions. However, if OFDM is intended to be used for a multiuser wireless communications system, for example, the proposed asymmetric PMR system, the problem of adapting the transmit power and bit allocations is crucial. In this way, one may utilize the resources efficiently by adapting the power level and the number of bits to be transmitted according to the channel characteristics of the user.

In a multiuser OFDM based communication system, each of the users' signals may undergo independent fading because users may not be in the same locations. Therefore, the probability that all the user's signals on the same subcarrier are in deep fading is very low. Hence for a specific subcarrier, if a user's signal is in deep fading, the others may not be in deep fading and the user in a good channel condition may be allowed to transmit data on that subcarrier yielding multiuser diversity effects [31]. Therefore, in a multiuser OFDM system, the multiuser diversity as well as the spectral diversity can be exploited, if the transmit power for each user and for each subcarrier is appropriately adapted to the channel conditions.

Several papers have concentrated on the problem of transmit power allocation for the multiuser OFDM system in a downlink transmission [32], [33]. In [32], the authors' aim is to minimize the total transmit power under a fixed performance requirement and a given set of user data rates. They focused on the algorithms that can support real-time multimedia applications whose data rate are generally fixed. In [33], dynamic subchannel and power allocation were performed to maximize the minimum capacity of all the users under the total transmit power constraint and zero delay constraint. The authors of [33] restricted their focus on exclusive assignment of each subcarrier to only one user in order to avoid the error propagation and complexity of successive decoding in the multiuser detection. However, [34] has provided a theorem which states that, the subcarrier assignment strategy for multiple users to maximize the data rate of a specific subcarrier in a downlink multiuser OFDM system is that the subcarrier should be assigned to only one user who has best channel gain for that subcarrier and the transmit power should be distributed over the subcarriers with the well known water-filling policy [35], which is known as optimal to maximize data rate under the constraint of total transmit power.

Chapter 4

Capacity Analysis of the Proposed System

4.1 Introduction

In this study, a new PMR system comprising an OFDM based wideband downlink and a narrowband uplink is proposed. Digital Audio Broadcasting system is considered as the OFDM based wideband downlink. This chapter focuses on the system capacity and grade of service (GOS) associated only with the downlink channel for voice communications. All capacity calculations are independent of the uplink part of the proposed system. The uplink part will certainly have a cellular architecture and have a capacity much larger than the downlink part making use of frequency reuse. Therefore the system capacity is upper bounded by the downlink part and the downlink capacity can be seen as the system capacity of the overall proposed system.

The Main Service Channel (MSC) of DAB transmission frame is of our concern. Since MSC carries the useful payload including the services supported by the system, the whole capacity of MSC is considered to be used for voice communications. In particular, we study the efficiency of such a system for voice that would be obtained using silence detection. The use of voice activity detection is modeled by a continuous-time Markov chain, which is called "ON-OFF Level Characterization". Using on-off level characterization, the maximum number of simultaneous voice calls, which can be supported by the proposed system for a given Frame Loss Rate (FLR) value, is calculated. Later call arrivals to the proposed system is modeled by another continuous-time Markov chain, which is called "Call Level Characterization". Using call level characterization, the maximum number of subscribers, which can be supported by the proposed system for a given GOS value, is calculated. The number of subscribers supported by the system is taken as system capacity.

Moreover, we show that, if the incoming calls are allowed to wait a certain amount of deterministic time before occupying a channel, the capacity can significantly increase. The effects of a certain amount of deterministic waiting time on the system capacity and blocking probability are presented.

The outline of this chapter is as follows. Markovian model to calculate the system user capacity and grade of service as well as the simulation method and its details are presented in Section 4.2. Numerical results of the analytical model and simulations are discussed in Section 4.3. Finally we conclude and briefly discuss the capacity analysis in Section 4.4.

4.2 Analysis Methods

In this section, methods for calculating the maximum number of users the system can accommodate are developed for a given average frame loss rate (FLR), which is the ratio of the average number of the lost frames to the total number of the frames and Grade of Service (GOS), which is the probability of blocking.

It is well known that voice contains an alternating sequence of ON and OFF times (talkspurts and silence gaps) [36]. It is assumed in this study that a voice user generates typically 9.6 kbits/s data during the ON time, including the overhead coming from error correction coding and packet headers. This data rate corresponds to ≈ 230 bits/slot where one slot corresponds to 24 ms of DAB transmission frame duration. This amount of information per slot per user is called a "frame" hereafter. The whole data transmission capacity, 55296 bits/slot, of DAB is assumed to be used for voice transmission in this study. Therefore 55296 / 230 ≈ 240 users can be in the ON state simultaneously if no loss of information is desired. However, $n_{oc} > 240$ voice calls can be admitted at the same time if a certain amount of FLR is allowed. In the proposed system, the maximum number of admitted calls, N_{oc}, is limited by the allowable FLR. At any slot, if $n_{on} \leq N_{oc}$, which is the number ongoing calls in the ON state, is larger than 240, then n_{on}-240 frames are assumed to be lost.

Once N_{oc} is determined for a specified (given) value of FLR, the total number of users which can be accommodated by the system is calculated by considering call-level statistics of the users. The whole system is modeled as a continuous time Markov chain. In the following, two approaches are used for obtaining numerical results. The first approach is based on the analytical solution of the Markov model, whereas the second one uses simulations.

4.2.1 Markov model of the system and analytical solution

The system is modeled as two Markov chains, one for call-level characterization and the other for ON-OFF level characterization. It is assumed that calls arrive at a rate λ (calls/hour) and the average call duration is $1/\mu$ (hour). Arrivals are assumed to be Poisson distributed and the call duration is assumed to be exponentially distributed. Moreover, both ON and OFF durations are assumed to be exponentially distributed with mean $1/\alpha$ (hour) and $1/\beta$ (hour) respectively.

With respect to ON-OFF level characterization, the state of the system can be described by the number of calls that are in talkspurts $i \le j$ as seen in Fig. 4.1.

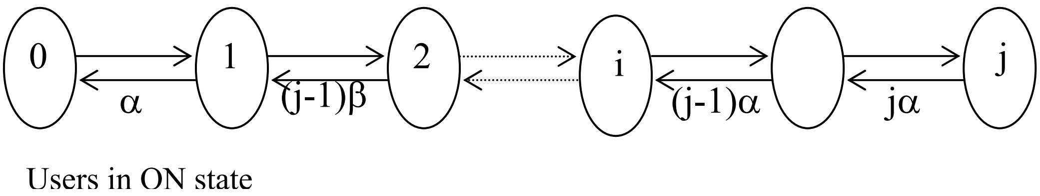


Figure 4.1: Markov model of the ON-OFF level characterization.

It is straightforward to show that the steady-state probability of having i calls out of j calls being in the ON state is given by [37];

$$B_i^j = \frac{\binom{j}{i}\left(\beta/\alpha\right)^i}{\left(1+\beta/\alpha\right)^j}, 0 \le i \le j \tag{4.1}$$

FLR is calculated by the ratio of the expected number of lost frames to the expected number of total frames using

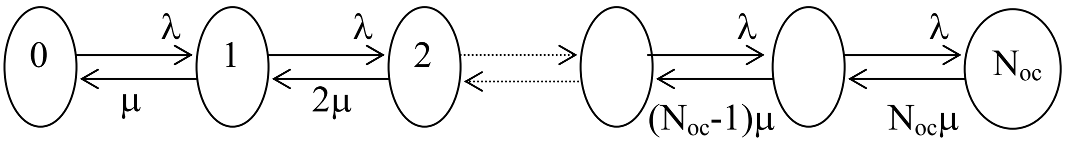
$$FLR = \frac{\sum_{i>240}^{N_{oc}} (i-240)B_i}{\sum_{i=0}^{N_{oc}} iB_i} \tag{4.2}$$

With respect to call level characterization the system can be described by an M/M/c/c queuing model in which the state of the system is the number of ongoing calls. To find GOS, we must now analyze the call level Markov chain in Fig. 4.2 where N_{oc} is the maximum number of calls that can be supported simultaneously. The probability of having j ongoing calls at any observation time is [37];

$$P_j^{N_{oc}} = \frac{\left(\lambda/\mu\right)^j \Big/ j!}{\sum_{k=0}^{N_{oc}} \left(\lambda/\mu\right)^k \Big/ k!}, \; 0 \le j \le N_{oc} \tag{4.3}$$

The call blocking probability can be derived using the P_j values from (4.3) which is the well known Erlang-B formula [18],

$$P_B = \frac{\left(\lambda/\mu\right)^{N_{oc}} \Big/ N_{oc}!}{\sum_{k=0}^{N_{oc}} \left(\lambda/\mu\right)^k \Big/ k!}, \; 0 \le j \le N_{oc} \tag{4.4}$$



Ongoing calls

Figure 4.2: Markov model of the call level characterization.

For a given value of FLR, the maximum number of users that can be supported by the system simultaneously, N_{oc}, is calculated from (4.2) by a binary search algorithm. Later, λ is calculated from (4.4) for a given values of

P_B, N_{oc} and μ. Since $\lambda = \lambda_u.N_{pop}$ where λ_u is number of calls per hour per user, N_{pop}, which is the total number of users accommodated by the system, i.e. the number of subscribers, can be calculated by

$$\boldsymbol{N}_{\boldsymbol{pop}} = \lambda \big/ \lambda_{\boldsymbol{u}} \tag{4.5}$$

We also study a variant of M/M/c/∞ queuing model with reneging. In this model, if a user is not directly admitted to the system, that user is willing to wait a certain amount of deterministic time. The user waits until this maximum waiting time, and then he reneges if service has not yet been provided. We analyzed such a system by simulations, since there exist little work on queuing models with deterministic waiting time in the literature.

4.2.2 Solution by simulation

The analytical models to find the system capacity are based on continuous time Markov chains. On the other hand, the proposed system is a discrete-time system since the DAB system transmits its data once every slot, which is 24 ms. Therefore, we have also made Monte Carlo simulations to test the validity of the analytical solution. The simulation method includes three steps:

4.2.2.1 Finding N_{oc} using ON-OFF level simulation

For each of n_{oc} ongoing calls, 3 hour-long ON-OFF patterns are generated. These ON and OFF periods are exponentially distributed with mean $1/\alpha$ and $1/\beta$.

The simulation period, 3 hours, is divided into slots; lasting 24 ms each. During the steady state region, for each slot, n_{on} is determined. If $n_{on} > 240$, then n_{on} -240 frames are assumed to be lost since a maximum of 240 calls in ON state can be supported by the system. If $n_{on} \leq 240$, then all voice information is sent successfully. FLR is calculated as the ratio of total lost frames to total frames sent during the steady state region. N_{oc} is taken to be the value of n_{oc} for which desired FLR is obtained.

For any given value of n_{oc}, the 3 hour-simulation is repeated 20 times to find 20 FLR's which are averaged to find the FLR corresponding to that n_{oc}. For n_{oc}=500, FLR was found to be 1.07e-04 with a 95 % confidence interval of ± 5.8e-06. Therefore 20 repetitions of 3-hour simulations were found to be sufficient.

4.2.2.2 Finding N_{pop} using call-level simulation without queuing

N_{pop} is the maximum number of users i.e. subscribers that can be supported by the system. This step of the simulation is performed for a longer period, typically 150 hours. Call arrivals are Poisson distributed; therefore interarrival time is exponentially distributed. The steps in the call level simulation without queuing are:

1) For the 150-hour simulation period, random call arrival times are generated using an exponentially distributed random number generator with mean $1/\lambda$.

2) For each arrival point, that's to say, for each call, call duration is assigned using the same random number generator with mean $1/\mu$.

3) Total simulation time is divided into 24ms slots. When a new call arrives in a certain slot, if there are already N_{oc} ongoing calls in that slot, then that call is blocked. Otherwise the system can support that new call and the call is admitted. In other words "blocked calls cleared" strategy is employed.

4) Call blocking probability, i.e. Grade of Service (GOS), is determined as the ratio of the number total blocked calls to the number total calls in the 150-hour period.

For a given value of GOS, the above procedure is repeated using a simple search algorithm to find the corresponding λ. N_{pop} is then found using (5).

4.2.2.3 Finding N_{pop} using call-level simulation with queuing

This method is similar to the previous one except for the blocking mechanism. Again call arrivals are Poisson. In this simulation, blocked calls are put into a FIFO queue and stay in that queue for a maximum of WT (waiting time) slots before they enter the system. If they cannot enter the system within WT time slots they are blocked.

The steps of this simulation are the same as the simulation without queuing except for the third step. In the third step, for a new call arrival, if there are already N_{oc} ongoing calls, then that call is placed in a FIFO queue where that call can maximally stay for WT time slots. If there are less than N_{oc} ongoing calls, the system can support a new call and that call is admitted. The system accepts the calls waiting in the queue in each time slot if there are lower than N_{oc} ongoing calls. This strategy is called "reneging".

4.3 Results

The analytical and simulation methods explained in the previous section are applied for two scenarios. In the first scenario, the system is assumed to be used for GSM type voice calls while in the second scenario the system is assumed to be used for PMR type voice calls. The simulation input parameters are shown for each scenario in Table 4.1.

Table 4.1: System parameters for GSM and PMR [5] scenarios

Parameters	GSM	PMR
Average call arrival rate (λcalls/hour/user)	2	10
Average call duration (1/μ sec)	180	20
Average duration in ON state (1/α sec)	1	1
Average duration in OFF state (1/β sec)	1.35	1.35

4.3.1 Number of Ongoing Calls, N_{oc}

The variation of FLR with N_{oc} as calculated by both the analytical and simulation method is given in Fig. 4.3. The results of the two methods do not agree. The reason for this difference is as follows: The Markov chain given in Fig. 4.1 assumes that only transitions between neighboring states are possible. However in our system because of the 24ms slot period, we have observed in the simulations that n_{oc} can change more than ±1 in each slot.

If a frame loss rate of 10^{-4} is accepted, the maximum number of the users that can simultaneously be supported by the system is found to be approximately 500 from simulations for both GSM and PMR scenario, since ON-OFF patterns are assumed to be the same for both scenarios.

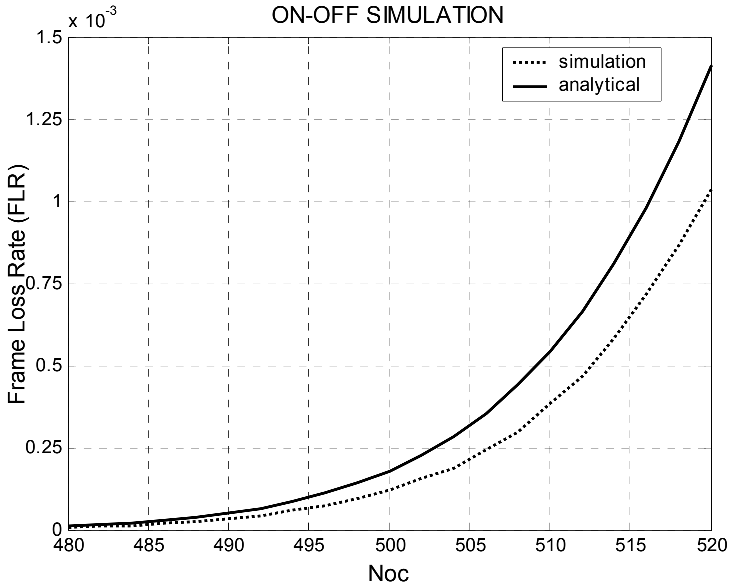


Figure 4.3: Frame loss rate as a function of number of ongoing calls.

4.3.2 Number of Users Supported by the System, N_{pop}, Without Queuing

Using the call level simulation without queuing, N_{pop} is found for both GSM and PMR scenarios. In Figs. 4.4 and 4.5, call-blocking probability vs. N_{pop} is plotted for both scenarios with analytical and simulation results. For a typical GOS of 2%, and frame loss rate of 10^{-4}, population that is supported by the system is found to be 4865 for GSM and 8765 for PMR scenario.

4.3.3 N_{pop} with Queuing

In the simulations, which also consider a maximum waiting time, user population is taken as 4865 for GSM scenario, and 8765 for PMR scenario. In Figs. 4.6 and 4.7, call-blocking probability is plotted as a function of waiting time. As expected, call blocking probability decreases as the waiting time increases. While the total number of users that generate traffic does not change, for GSM scenario, maximum waiting time of 2.16 sec (90 slots) provides nearly zero call-blocking probability. Since for the PMR scenario call durations are much smaller than GSM call durations, the decreasing rate of call blocking probability is higher than that of GSM as seen in Fig. 4.7. A maximum waiting time of 0.48 sec (20 slots) supports nearly zero call-blocking probability in PMR scenario.

For a given fixed call blocking probability, the system user capacity increases as the waiting time increases. As seen in the Fig. 4.8, for GSM scenario of 2% GOS, system user capacity is 4865 when maximum waiting time is zero. When approximately 2.2 sec maximum waiting time is allowed, the system capacity, in terms of population, increases as much as three times. For PMR scenario with 2% GOS, system user capacity is 8765 when maximum waiting time is zero. As shown in Fig. 4.9 if 2.88 sec maximum waiting time is allowed, the system capacity increases to approximately 20000.

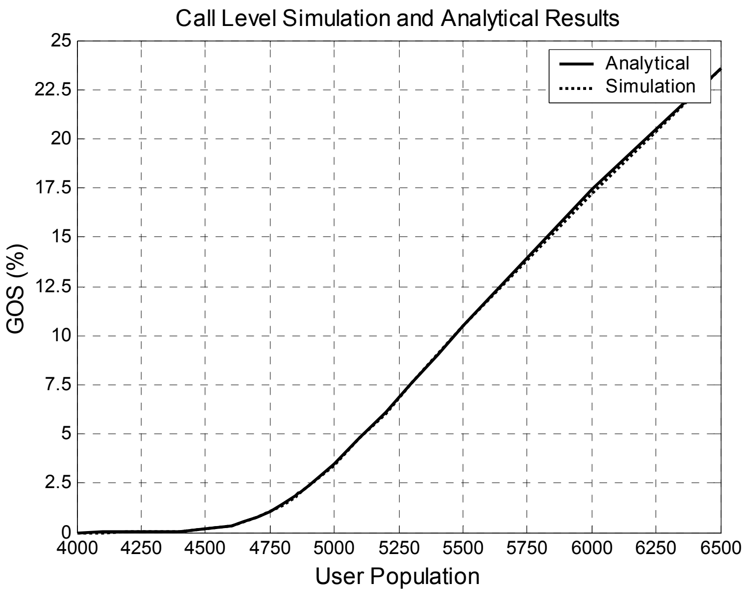


Figure 4.4: GOS: Grade of service providing on the average 10^{-4} FLR for GSM scenario. For GOS of % 2, optimum N_{pop} is found to be 4865.

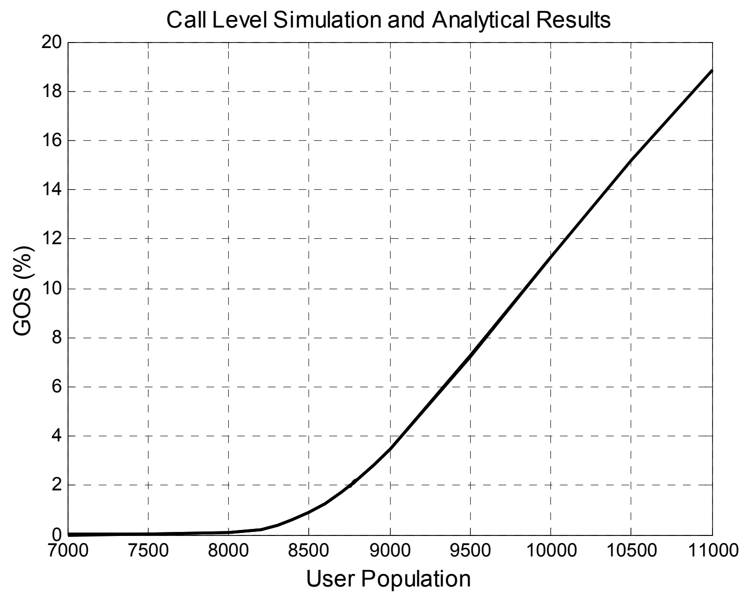


Figure 4.5: GOS: Grade of service providing on the average 10^{-4} FLR for PMR scenario. Simulation results are very close to analytical results. For GOS of % 2, optimum N_{pop} is found to be 8765.

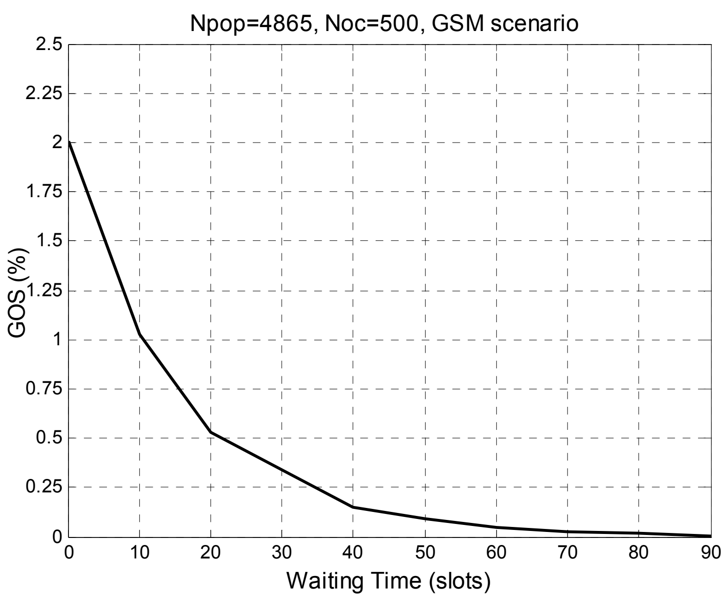


Figure 4.6: GOS as a function of maximum waiting time for GSM scenario.

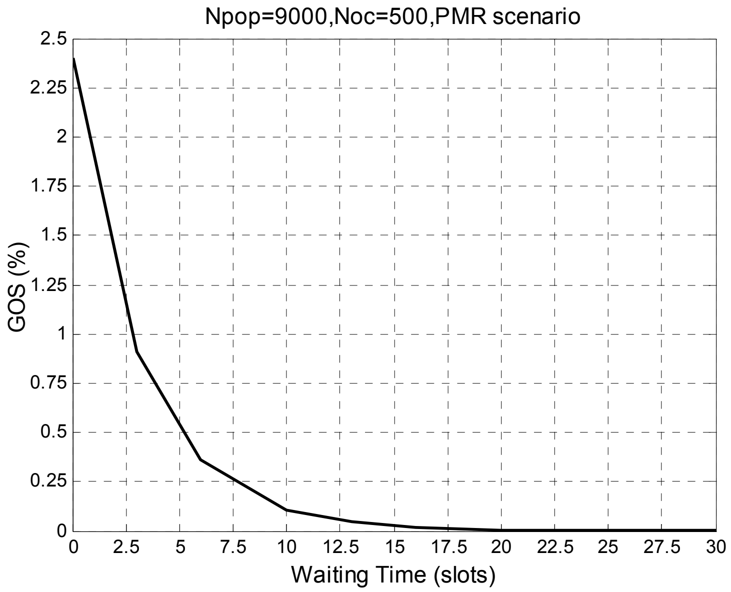


Figure 4.7: GOS as a function of maximum waiting time for PMR scenario.

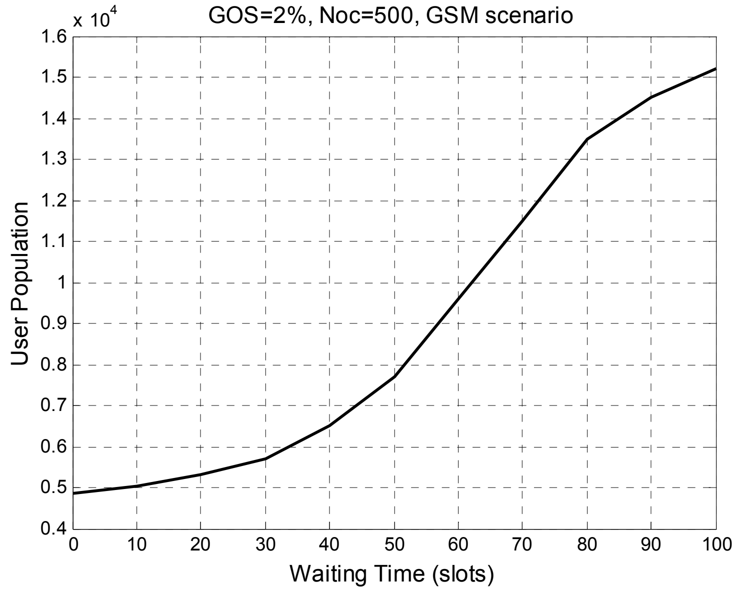


Figure 4.8: The maximum number of users that can be supported by the system as a function of maximum waiting time for GSM scenario for 2% GOS and 10^{-4} frame loss rate.

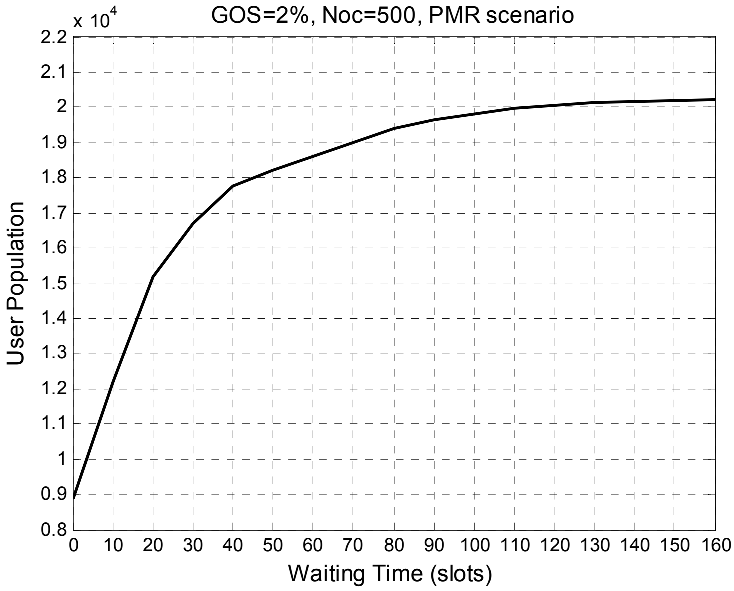


Figure 4.9: The maximum number of users that can be supported by the system as a function of maximum waiting time for PMR scenario for 2% GOS and 10^{-4} frame loss rate.

4.4 Discussion

We have found that for 2% GOS, 10^{-4} FLR and 2.88 sec maximum waiting time approximately 20,000 PMR users can be supported by the proposed system. It is obvious that the specifications of PMR system can be relaxed to for example 5% GOS, 10^{-3} FLR and 4 sec maximum waiting time. In such a case, a considerable increase in the number of users will be achieved. In general, the number of subscribers of a PMR system is low compared to the number of subscribers of a GSM system. For example, in a big metropolis like Istanbul there are about 8000 users of the police PMR system. We can conclude from these observations that only a small proportion of an existing DAB system's capacity needs to be allocated to the downlink of a PMR system.

Therefore, the proposed system offers new service opportunities for existing DAB operators.

A PMR system as proposed in this study also has the capability of supporting high data rate downlink applications such as multimedia services.

In this study, an acceptable value for FLR is taken to be 10^{-4}. In speech communications typically a BER of 10^{-3} is considered to be appropriate. We have used a lower FLR because from the point of view of transient behavior (the frames lost have 230 bits each) a more conservative loss rate would be appropriate.

Chapter 5

Spectrum Efficiency of the Proposed System

5.1 Introduction

Spectrum congestion in dense urban areas is a reality not only in public access communications bands but also in conventional PMR bands, which in most European countries are around 80 MHz, 160 MHz and 450 MHz. In major metropolitan cities, assignment of frequencies for new users or extension of the capacity of existing networks is becoming a real challenge in the communications area. Data transmission and trunked networks are introduced in order to deal with the spectrum congestion. Data communications can be more spectrum efficient than voice communications. Trunking refers to the fact that different users have access to a pool of channels. Spectrum resources and infrastructure are shared with channels being assigned on demand. However these are proved to be insufficient in many cases and other ways need to be found of optimizing the usage of the limited spectrum dedicated to PMR applications.

In this study, we propose a new PMR system, which has better spectral efficiency especially regarding broadcast and group calls. The factors which are very important in accessing the spectrum efficiency of all types of communication systems are:

- Geographical reuse factor of a given radio channel
- Type and quantity of information per traffic channel
- Number of RF carriers in a known amount of spectrum
- Number of traffic channels per RF carrier

Some of the common features of PMR systems, which also make them fundamentally different from Public Access Communications System (PACS), and which are important with respect to spectral efficiency considerations are:

- Typical lengths of calls are very short compared to the public telephone calls in PACS and they serve a smaller population. Therefore PMR systems have considerably lower traffic per user.

- There are a number of voice and data applications, which are not supported by a PACS such as GSM. Most important of these are broadcast calls and group calls (acknowledged and unacknowledged) for both voice and data communications.

In a PACS, such as GSM, the number of channels to be allocated for a call is equal to the number of mobiles engaging that call, and therefore independent of the number and size of cells. In contrast to the full duplex service offered by PACSs, in a PMR system the number of channels is equal to the number of cells engaging in the coverage of the call and independent of number of participating mobiles. Therefore the type of the traffic or the mode of operation also has an important influence on the spectrum efficiency. If mobile-to-mobile links are compared to the mobile-to-multipoint links, which are to be found in a high percentage of the total traffic within a PMR system, the latter show a considerable spectrum efficiency improvement. The main reason is that in such cases more than one subscriber is served in parallel. However, if subscribers engaging in a call spread to more cells, then the achieved spectrum efficiency will eventually decrease. However, if an asymmetric system as proposed in this study, where the downlink has wide area coverage, i.e. it covers the whole service area, is used, spectrum efficiency of the system is maintained even with group and broadcast calls, because for each broadcast or group call a single channel in the downlink is allocated. In PMR systems, group calls and broadcast calls constitute about 50% of all calls [5]. Therefore an asymmetric system as described above should provide significant increase in spectral efficiency.

As mentioned before, a new digital trunked PMR system comprising an OFDM based broadband, wide-area downlink and a narrowband cellular uplink, is proposed in this study to achieve a higher spectral efficiency than the TETRA system, which is accepted to be the most efficient system among digital trunked PMR systems in terms of spectral usage. As the OFDM downlink, we have considered in this study the use of a DAB system. Since for spectrum efficiency comparison process, we should specify a structure for the uplink part as well, we have made use of the standards of a TETRA system for the cellular uplink part of the system. In particular in going from number of channels to necessary frequency band we are using the TETRA standards [16]. Therefore, in this chapter from now on, the proposed system is called as TETRA-DAB system.

Cellular PMR systems can increase their spectral efficiency by making use of frequency reuse. However for reasons as explained above regarding the channel requirements of broadcast calls and group calls, wide area coverage for the downlink part offers opportunity for additional spectrum efficiency. On top of this, a wide area downlink provides trunking efficiency. However this architecture, i.e. cellular uplink and single cell downlink, may not be efficient if the number of cells in the system is increased and more frequency reuse is employed, because in such a system there is no provision for frequency reuse in the downlink part. In this study we are aiming at determining the optimum point up to which the proposed system is more efficient than conventional digital PMR systems. We have particularly investigated this problem in reference to realistic PMR systems which have significantly lower traffic compared to a PACS.

Both the proposed TETRA-DAB system and the standard TETRA system are compared to demonstrate the advantages of the proposed system in terms of spectral efficiency. In order to compare the spectral efficiencies of these PMR systems, a metropolitan area is assumed to be covered by either one of the systems considering equal traffic patterns. Spectrum requirements in order to achieve the same Quality of Service (QoS), e.g. the same call delay probability, are calculated. In spectrum efficiency calculations, individual and group calls are considered. Broadcast situation regarding broadcast calls is discussed in the discussion section.

The outline of this chapter is as follows. Starting with a brief description of the standard TETRA system in Section 5.2, a brief overview of the proposed TETRA-DAB system is made in Section 5.3. Section 5.4 covers the details about analyses for spectral efficiency comparison of the systems. In Section 5.5 numerical results for the spectrum requirements of both systems are given under the same constraints and assumptions.

5.2 The TETRA System

The TETRA system is similar to PACS networks, difference being especially in cell sizes and service types. The system can be used as a single or multi-cell network. The maximum cell radius in rural areas is limited to 25 km. TETRA uses $\pi/4$-DQPSK modulation scheme and offers a gross bit rate of 36 kbit/s in a single 25 kHz channel. Each carrier in the network has a bandwidth of 25 kHz and supports 4 TDMA channels [1], [16]. In this study we especially deal with the channel allocation schemes for different types of calls from the spectral efficiency point of view.

In the TETRA system, there is a capability that uplink and downlink channels can be allocated independently, which corresponds to TETRA's best mode of operation [16]. In this mode, when a group call is initiated in a cell, an uplink channel is allocated in that cell and a downlink channel is allocated in all cells in which there exists a group member. When a group call is initiated in another cell, there exists a probability that one or more members of that group are in the cell of interest. Therefore the total group call traffic offered in a cell is not only due to the group calls initiated in that cell, but also due to the group calls initiated in other cells which have members in the cell of interest. In case of individual calls, again assuming that uplink and downlink channels can be allocated independently, when the two users engaged in a call are in the same cell, the TETRA network allocates one uplink-downlink channel pair in that cell. On the other hand, if the two users engaged in a call are in different cells, then the system will allocate an uplink channel in the initiator cell and a downlink channel in the destined cell.

5.3 The Proposed Asymmetric PMR System (TETRA-DAB system)

The proposed TETRA-DAB system is composed of an OFDM based broadband, wide-area downlink which is supposed to be supported by a standard DAB system and a narrowband cellular uplink. In the TETRA-DAB system, the downlink part of all types of call requests for both voice and data calls are transmitted by the DAB transmitter whose transmission bandwidth is approximately 1.5 MHz. This transmitter serves the whole service area using a Single Frequency Network (SFN) using COFDM modulation. On the other hand, a cellular network, e.g. TETRA in our study, handles the uplink part of any offered call request.

DAB system has a capacity of 1.5 MHz [8], which corresponds to a data rate of 2.4 Mbit/s to serve the coverage area of the SFN. If the bit rate needed to handle a voice call is assumed to be 9.6 kbit/s, including the coding overheads, DAB system is able to handle 240 simultaneous voice calls. As a result, DAB system can be assumed to have 240 traffic channels available to the whole service area covered by the system. (In fact, all channels in the TETRA-DAB system are logical channels. There exist 240 logical traffic channels, but if a user requests a higher data rate application, more than one traffic channel can easily be allocated for that user on demand. To illustrate, when a user wants to download a file from a server, total available capacity of the downlink part of the TETRA-DAB system can be allocated for that user.) In this study, we assume that a DAB network is already available and only a part of the whole capacity of the DAB downlink is used by the proposed TETRA-DAB PMR system. In the TETRA-DAB system, when a call is initiated in a certain cell, whether a group call or an individual call, one uplink channel is allocated in the initiator cell and one downlink traffic channel is allocated in DAB downlink part. This scheme significantly increases the spectrum efficiency, as it will be demonstrated in the section on numerical results.

5.4 Comparative Results for the Spectrum Requirements of the Proposed TETRA-DAB and TETRA Systems

5.4.1 Basic definitions and assumptions

Our aim is to determine the optimum point, with respect to number of cells, up to which the proposed TETRA-DAB system is spectrally more efficient than the TETRA system. In order to compare the spectral efficiencies of TETRA-DAB and TETRA, a metropolitan area is assumed to be covered by either one of the systems under equivalent traffic conditions as explained in the next section. The spectrum requirements in order to achieve the same QoS, i.e. the same delay probability, are compared. The desirable delay probability is assumed to be less than 0.05 for both systems [5].

All mobile stations are assumed to use transmission trunking mode in which a user presses the push-to-talk button to initiate a call, thereby allocating an uplink channel in his/her cell and downlink channels in other cells in which there exists a group member. Therefore systems are assumed to allocate uplink and downlink channels independently, which corresponds to TETRA's best mode of operation. The user releases the button after completing the call, deallocating the channels allocated for that call. Average number of members in a group is assumed to be equal to 7 [38], [39].

Spectrum requirement of any one of the systems is ultimately related to the traffic load in the system. In the following, first, the relation between traffic load and spectrum requirement of the TETRA-DAB system is formulated, and then the same is done for TETRA system.

5.4.1.1 Spectrum requirement of the TETRA-DAB system

The following definitions are used for the TETRA-DAB system with N cells:

$A\_ind^u{}_j$**:** The uplink traffic initiated in the jth cell due to individual calls.

$A\_group^u{}_j$**:** The uplink traffic initiated in the jth cell due to group calls.

$A\_total^u{}_j$**:** Total uplink traffic initiated in the jth cell.

A_total^d**:** Total downlink traffic in the DAB downlink of the system.

The total uplink traffic initiated in the jth cell can be expressed by

$$A\_total^u{}_j = A\_ind^u{}_j + A\_group^u{}_j \tag{5.1}$$

For any given individual or group call initiated in a cell, a single downlink channel must be allocated in the DAB downlink. Therefore total DAB downlink traffic is the sum of uplink traffic loads of all cells, i.e.

$$A\_total^d = \sum_{j=1}^{N} A\_total^u{}_j \tag{5.2}$$

From A_total^d one can find the total number of required DAB downlink channels using Erlang-C [37] formula for the desirable delay probability. The use of Erlang-C theory means that call arrivals are Poisson and call durations are exponentially distributed. These assumptions seem to be valid for PMR systems [40]. Assuming 9.6 kbit/s bitrate for each channel one can then find the total spectrum to be used from the DAB downlink. From $A\_total^u{}_j$, one can find the required number of uplink channels of the jth cell using Erlang-C formula for the desirable delay probability. Since a TETRA-like structure is assumed to be used in the uplink part of the system, 25 kHz spectrum is assumed to be allocated for each 4 uplink channels. The total uplink spectrum requirement of the system is the sum of spectrum requirements of all cells if there is no frequency re-use in the system. However if there is frequency re-use in the system, then the cluster structure must be taken into consideration as explained in the numerical results section.

In this study it is assumed that for an N cell system all cells have the same individual and group uplink traffic loads, i.e. $A\_ind^u{}_j = A\_ind^u$ *and* $A\_group^u{}_j = A\_group^u$**.** In this case**,**

$$A\_total^u{}_j = A\_ind^u + A\_group^u \tag{5.3}$$

$$A\_total^d{}_j = N \cdot (A\_ind^u) + N \cdot (A\_group^u) \tag{5.4}$$

5.4.1.2 Spectrum requirement of the TETRA system

The following definitions are used for the TETRA system with N cells:

***$A\_ind^u{}_j$*:** The uplink traffic initiated in the jth cell due to individual calls.

$A\_group^u{}_j$: The uplink traffic initiated in the jth cell due to group calls.

$A\_total^u{}_j$: Total uplink traffic initiated in the jth cell.

***$A\_ind^d{}_j$*:** The downlink traffic in the jth cell due to individual calls.

***$A\_group^d{}_j$* :** The downlink traffic in the jth cell due to group calls.

$A\_total^d{}_j$: Total downlink traffic in the jth cell.

Total uplink traffic in the j[th] cell is

$$A\_total^{u}{}_{j} = A\_ind^{u}{}_{j} + A\_group^{u}{}_{j}, \tag{5.5}$$

and the total downlink traffic in the j[th] cell is

$$A\_total^{d}{}_{j} = A\_ind^{d}{}_{j} + A\_group^{d}{}_{j}. \tag{5.6}$$

The downlink traffic components of the system can be related to the uplink traffic components. Assuming individual uplink traffic in all cells are the same, i.e. $A\_ind^{u}{}_{j} = A\_ind^{u}$ for j = 1,…, N, one can show that $A\_ind^{d}{}_{j}$ is also equal to A_ind^{u} as explained in appendix-A. Similarly, if uplink group call traffic in all cells are assumed to be the same, i.e. $A\_group^{u}{}_{j} = A\_group^{u}$ for j = 1,…, N, $A\_group^{d}{}_{j}$ can be related to A_group^{u}, as explained in appendix-B, as follows:

$$A\_group^{d}{}_{j} = M \cdot A\_group^{u} \tag{5.7}$$

where M is the average number of group members in a group call. As shown in the appendix-B, this formula is valid for $N \geq M$ and under certain assumptions.
In summary, (5.5) and (5.6) are simplified to

$$A\_total^{u}{}_{j} = A\_ind^{u} + A\_group^{u} \tag{5.8}$$

$$A\_total^{d}{}_{j} = A\_ind^{u} + M \cdot (A\_group^{u}) \tag{5.9}$$

Both the uplink and downlink spectrum requirements of the TETRA system are calculated similar to how the spectrum requirement of the uplink part of the TETRA-DAB system is calculated.

5.4.1.3 Cluster size

The total number of cells used in both TETRA and TETRA-DAB system is another important variable to be considered for making a comparison in terms of spectral efficiency, since traffic calculations are based on per cell basis. In the TETRA system, both uplink and downlink are based on a cellular structure. On the other hand, the TETRA-DAB system has cellular uplink and single cell downlink. In order to find the number of cells, considering the carrier to interference ratio, appropriate cluster size should be calculated which is applicable to both systems. When hexagonal cells are assumed to be used and the service area is assumed to be flat, cluster size is determined from the desired carrier to interference ratio from the following simple formula [18];

$$\frac{C}{I} = \frac{(3N)^{\gamma/2}}{i_o} \tag{5.10}$$

where N, γ, and i_o stands for cluster size, path loss exponent and number of interferer cells respectively. In this study, path loss exponent is assumed to be 4 and number of interferer cells is assumed to be 3 [39]. Since the minimum carrier to interference ratio is 19 dB in the TETRA system, the appropriate cluster size is found to be 7.

5.4.2 Numerical Results

In order to compare the proposed TETRA-DAB system with the standard TETRA system, one needs to define a common reference with respect to the traffic carried by both systems. It is natural to assume that both A_ind^u and A_group^u are the same for both systems. In this study as a starting point we first specified the amount of total capacity in the DAB downlink that is used by

PMR services in the TETRA-DAB system. This means that we are specifying A_total^d for the TETRA-DAB system. From this specification one can find A_ind^u and A_group^u from (5.3) and (5.4), provided that we assume a certain ratio for these variables. In this study we have assumed that individual call traffic and group call traffic initiated in a cell are the same. From A_ind^u and A_group^u values, i.e. uplink traffic loads, one can proceed to find the spectrum requirement of both the TETRA-DAB and TETRA systems. At first, the whole capacity of DAB, which is 1.5 MHz, is assumed to be used for PMR services, and the spectrum requirements for both systems to handle the corresponding traffic are calculated.

When the total capacity of the TETRA-DAB system, 2.4 Mbits/s is allocated to the PMR services, there are 240 available traffic channels. Using Erlang-C formula, with 240 channels and probability of delay of 0.05, TETRA-DAB system capacity is calculated as 213.8 Erlang for the downlink part. The TETRA-DAB system allocates one downlink channel for each uplink channel. Therefore the proposed system has equal uplink and downlink traffic,

$$A\_total^u = A\_total^d = 213.8\ Erlang$$

where A_total^u stands for the total uplink traffic and A_total^d stands for the total downlink traffic in the network.

The spectrum requirements for different DAB capacities, from 100 KHz to 1.5 MHz, as well as for different cell sizes are examined. The service area is assumed to be covered by 1 to 15 clusters each having 7 cells.

5.4.2.1 Single-Cluster Case (7 cells) using the Whole Capacity of DAB Downlink

Using full capacity of DAB system, for the 1-cluster case with 7 cells, traffic for each cell can be calculated by

$$A\_total^{u}_{j} = 213.8 / 7 = 30.54 \text{ Erlang}$$

where $A\_total^{u}_{j}$ stands for the total uplink traffic per cell

When TETRA-DAB is used, $A\_total^{u}_{j}$ stands for the sum of A_ind^{u} and A_group^{u} in (5.3). Therefore the total uplink traffic per cell can be taken as 30.54 Erlang. Again using Erlang-C formula with traffic 30.54 Erlang and P (D) less than 0.05, 41 channels per cell are needed to handle uplink traffic. In the TETRA system, a single carrier supports 4 channels, therefore 11 carriers per cell are required. This corresponds to a spectrum of 1.925 MHz (11 carriers/cell*25 KHz/carrier* 7 cells) for uplink purposes in addition to 1.5 MHz allocated for the downlink part. Summing up those values, the total spectrum needed to handle 213.8 Erlang is found to be 3.425 MHz by TETRA-DAB.

When the TETRA system is used, total downlink traffic per cell can be calculated by (5.9). If the proportions of individual and groups are the same, then (5.9) is simplified to

$$A\_total^{d}{}_{j} = (M+1) \cdot A\_group^{u} \tag{5.11}$$

where A_group^{u} is equal to 30.54/2 Erlang and M is taken as 7 in our calculations.

The total downlink traffic per cell is found to be 106.9 Erlang from (5.11), which is significantly larger than 30.54, the uplink traffic of a TETRA-DAB cell. There is an increase in the traffic due the difference in the group call schemes. For that traffic and P (D) less than 0.05, 143 channels per cell are needed to handle downlink traffic. This corresponds to 36 carriers per cell and a spectrum of 6.3 MHz (24 carriers/cell*25 KHz/carrier*7 cells) for downlink purposes. The spectrum needed for uplink is the same as in the TETRA-DAB system, i.e. 1.925 MHz, since (5.3) and (5.9) are equal to each other. Therefore the total spectrum needed by the TETRA system to handle the same traffic offered to the TETRA-DAB system is found to be 8.225 MHz.

5.4.2.2 Two-Cluster Case (14 cells) using the Whole Capacity of DAB Downlink

For the two-cluster case with 14 cells, traffic for each cell can be calculated by

$$A\_total^{u}{}_{j} = A\_total^{d}{}_{j} = = 213.8 / 14 = 15.27 \ Erlang$$

When TETRA-DAB is used, the total traffic per cell can be taken as 15.27 Erlang. Using Erlang-C formula with traffic 15.27 Erlang and P (D) of 0.05, 23 channels, therefore 6 carriers per cell are required. This corresponds to a spectrum of 1.050 MHz (6 carriers/cell*25 KHz/carrier* 7 cells) for uplink purposes in addition to 1.5 MHz allocated for the downlink part. Summing up those values, the total spectrum needed to handle 213.8 Erlang is found to be 2.55 MHz by the TETRA-DAB system.

When TETRA is used, total traffic per cell can be calculated by (5.9). Again, if the proportions of individual and groups are same, *A_ind* and

A_group will be the same. Then (5.9) is simplified to (5.11) as in the single cluster case.

The total downlink traffic per cell is found to be 61.08 Erlang from (5.11). For this traffic and P (D) less than 0.05, 76 channels per cell are needed to handle uplink or downlink traffic. This corresponds to 19 carriers per cell and a spectrum of 3.325 MHz (13 carriers/cell*25 KHz/carrier*7 cells) for downlink purposes. Therefore the total spectrum needed by the TETRA system to handle the same traffic offered from the TETRA-DAB system is found to be 4.375 MHz.

5.4.2.3 Multiple Cluster Cases using Different Capacities of DAB Downlink

The service area is assumed to be also covered by 3 clusters (21 cells), 4 clusters (28 cells), 5 clusters (35 cells) or up to 15 clusters (105 cells). The spectrum requirement calculations are similar to those made for single and two-cluster cases. Decreasing the used capacity of DAB downlink, and increasing the number of clusters i.e. decreasing cell sizes, spectrum requirements for both TETRA-DAB and TETRA systems are calculated and shown in Table 1 and 2 respectively.

As expected, as the number of clusters increases, spectrum requirements of both systems decrease. This is mainly due to frequency reuse in the uplink part of the TETRA-DAB system, and in both uplink and downlink parts of the TETRA system. In order to compare the two systems for a given number of clusters and allocated DAB capacity, we used the ratio of the spectrum requirement of the TETRA-DAB system to the spectrum requirement of the TETRA system. If the ratio is less than 1, then the TETRA-DAB system is spectrally more efficient than the TETRA system, otherwise the reverse is true.

TABLE 5.1

SPECTRUM REQUIREMENTS FOR THE TETRA-DAB SYSTEM WITH DIFFERENT CLUSTER CASES AND TRAFFIC

Capacity allocated in DAB	1-cluster case (MHz)	2-cluster case (MHz)	3-cluster case (MHz)	4-cluster case (MHz)	5-cluster case (MHz)	6-cluster case (MHz)	10-cluster case (MHz)	15-cluster case (MHz)
100 kHz	0.450	0.275	0.275	0.275	0.2750	0.275	0.275	0.275
200 kHz	0.550	0.550	0.375	0.375	0.375	0.375	0.375	0.375
300 kHz	0.825	0.650	0.650	0.475	0.475	0.475	0.475	0.475
400 kHz	1.100	0.750	0.750	0.750	0.750	0.575	0.575	0.575
500 kHz	1.200	1.025	0.850	0.850	0.850	0.850	0.675	0.675
600 kHz	1.475	1.125	0.950	0.950	0.950	0.950	0.775	0.775
700 kHz	1.750	1.225	1.225	1.050	1.050	1.050	1.050	0.875
800 kHz	1.850	1.500	1.325	1.325	1.150	1.150	1.150	0.975
900 kHz	2.125	1.600	1.425	1.425	1.250	1.250	1.250	1.075
1000 kHz	2.400	1.875	1.525	1.525	1.525	1.350	1.350	1.350
1100 kHz	2.500	1.975	1.800	1.625	1.625	1.450	1.450	1.450
1200 kHz	2.775	2.075	1.900	1.725	1.725	1.725	1.550	1.550
1300 kHz	2.875	2.350	2.000	1.825	1.825	1.825	1.650	1.650
1400 kHz	3.150	2.450	2.100	2.100	1.925	1.925	1.750	1.750
1500 kHz	3.425	2.550	2.375	2.200	2.025	2.025	1.850	1.850

TABLE 5.2

SPECTRUM REQUIREMENTS FOR THE TETRA SYSTEM WITH DIFFERENT CLUSTER CASES AND TRAFFIC

Capacity allocated in DAB	1-cluster case (MHz)	2-cluster case (MHz)	3-cluster case (MHz)	4-cluster case (MHz)	5-cluster case (MHz)	6-cluster case (MHz)	10-cluster case (MHz)	15-cluster case (MHz)
100 kHz	0.875	0.525	0.525	0.525	0.350	0.350	0.350	0.350
200 kHz	1.400	0.875	0.700	0.525	0.525	0.525	0.350	0.350
300 kHz	1.925	1.225	1.050	0.700	0.700	0.525	0.525	0.525
400 kHz	2.450	1.400	1.050	1.050	0.875	0.700	0.525	0.525
500 kHz	2.975	1.750	1.225	1.050	1.050	0.875	0.525	0.525
600 kHz	3.500	1.925	1.400	1.225	1.050	1.050	0.700	0.525
700 kHz	4.025	2.275	1.750	1.400	1.225	1.050	0.875	0.525
800 kHz	4.550	2.625	1.925	1.575	1.225	1.225	0.875	0.700
900 kHz	5.075	2.800	2.100	1.750	1.400	1.225	1.050	0.700
1000 kHz	5.600	3.150	2.275	1.925	1.575	1.400	1.050	0.875
1100 kHz	6.125	3.500	2.450	1.925	1.750	1.400	1.050	0.875
1200 kHz	6.650	3.675	2.625	2.100	1.750	1.575	1.050	0.875
1300 kHz	7.000	4.025	2.800	2.100	1.925	1.750	1.225	1.050
1400 kHz	7.700	4.200	2.975	2.450	1.925	1.750	1.225	1.050
1500 kHz	8.225	4.375	3.325	2.625	2.100	1.925	1.225	1.050

Fig. 5.1 shows the spectrum requirement ratio for the single cluster case, versus the total capacity used in the DAB downlink. For the single cluster case, the spectrum requirement ratio is between 0.52 and 0.41, which means that the TETRA system requires 2-2.5 times the spectrum required by the TETRA-DAB system. The dependence of spectrum ratio to used DAB capacity has noisy behavior due to mainly two reasons, one of which is the quantization effect due to integer number of channels, and the other is the quantization effect due to channel capacity of TETRA carriers. A TETRA carrier employs multiples of 4 channels. Therefore, if 5 channels are needed in a TETRA cell, 8 channels will be allocated to that cell with redundant 3 channels. Figs. 5.2-5.6 display the dependence of spectrum requirement ratio to allocated DAB capacity for 3, 5, 6, 10 and 15 clusters respectively. When the service area is covered with 5 or less clusters, the TETRA-DAB performs better. With 6 clusters, TETRA becomes more efficient as the allocated DAB capacity is increased beyond 500 kHz. The reason is that both uplink and downlink parts of TETRA system are cellular, and therefore when higher number of clusters is used, the frequency reuse capability increases, thereby increasing the spectral efficiency. On the other hand, only the uplink part of the TETRA-DAB system is cellular, therefore when higher number of clusters is used, the frequency reuse capability increases for only uplink part. For 10 or 15 clusters, the spectrum requirement ratio is predominantly larger than 1 except for very small DAB capacities. Fig. 5.7 shows the dependence of the spectrum requirement ratio to the number of clusters for 500, 1000 and 1500 kHz of DAB capacities. It is also clear from Fig. 5.7 that below 5 clusters i.e. 35 cells, TETRA-DAB is more efficient.

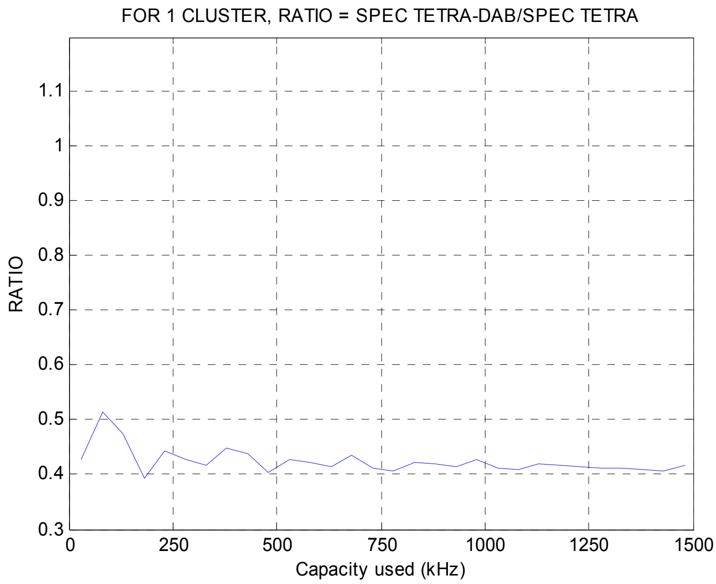


Figure 5.1: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus total traffic capacity of DAB system for 1-cluster case.

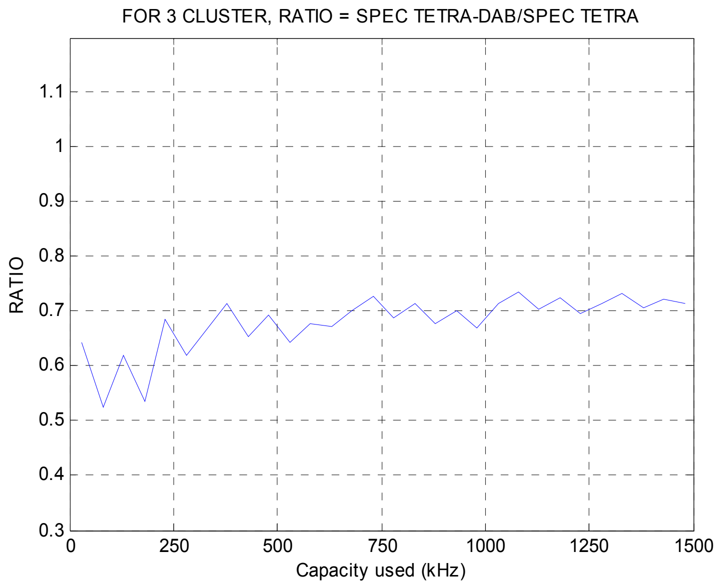


Figure 5.2: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus total traffic capacity of DAB system for 3-cluster case.

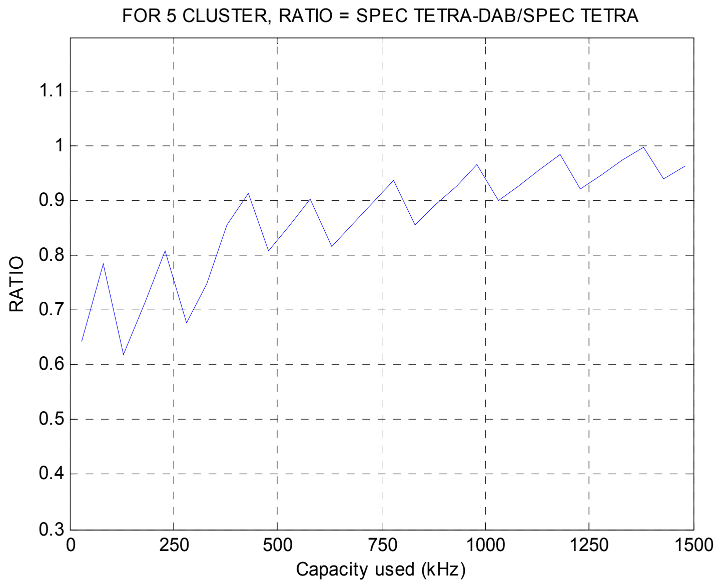


Figure 5.3: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus total traffic capacity of DAB system for 5-cluster case.

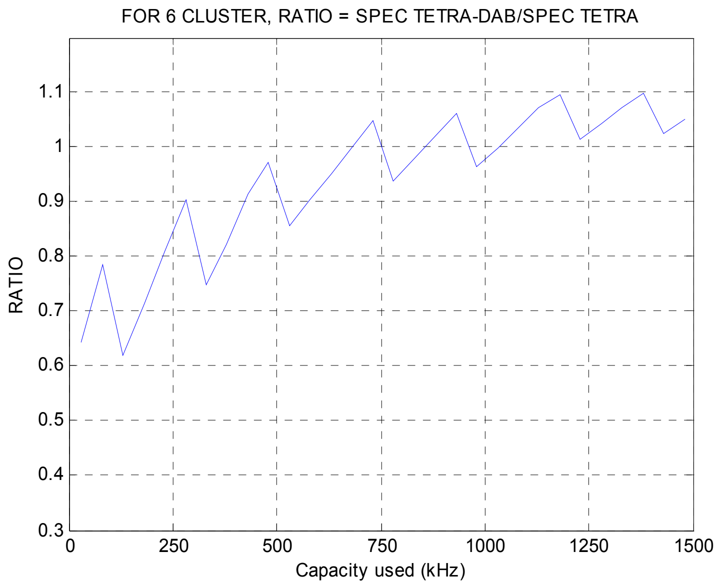


Figure 5.4: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus total traffic capacity of DAB system for 6-cluster case.

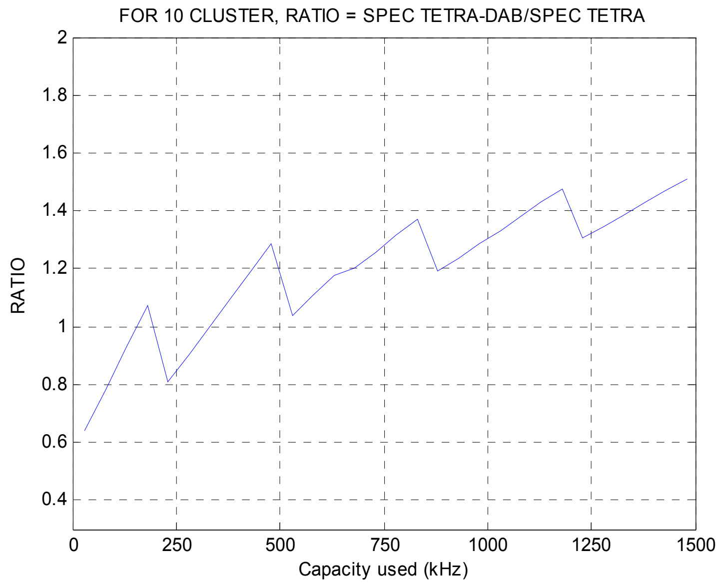


Figure 5.5: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus total traffic capacity of DAB system for 10-cluster case.

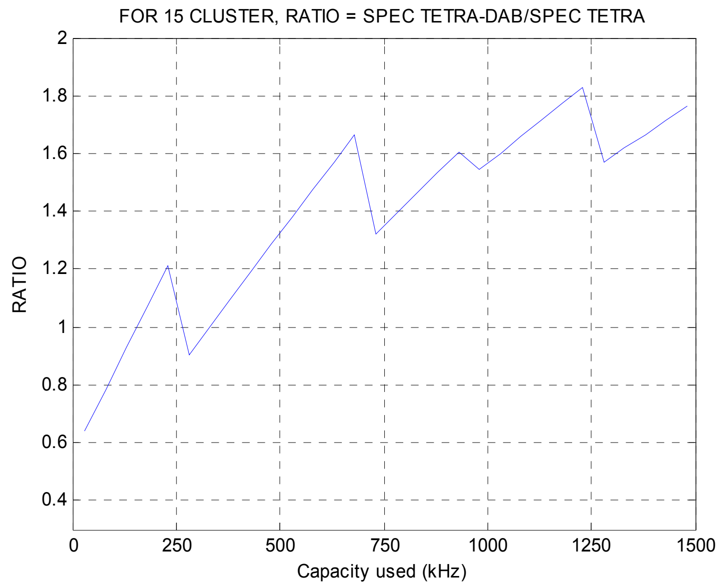


Figure 5.6: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus total traffic capacity of DAB system for 15-cluster case.

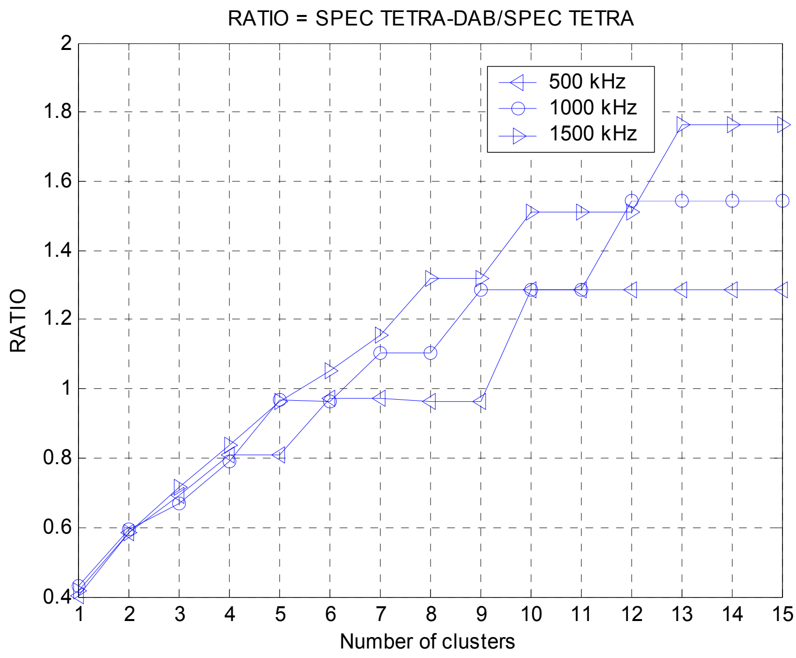


Figure 5.7: Ratio of the total spectrum requirement of TETRA-DAB to conventional TETRA system versus the number clusters in the system.

5.5 Discussion

A new asymmetric PMR system, which comprises a single broadband and wide area downlink cell and narrowband cellular uplink structure, is proposed in order to achieve higher spectral efficiency. It has been shown by numerical case studies that the proposed system is more efficient spectrally for less than or equal to 5 clusters. Due to relatively low traffic density in PMR applications, a PMR network with 5 clusters i.e. 35 cells can be assumed to adequately cover an average sized metropolitan area. In other words, if there should be 35 or fewer cells to cover a service area, our proposed system is shown to perform better than the TETRA system. Therefore our proposed system would be useful to achieve spectral efficiency in realistic system installations. As the number of clusters increases, the spectral advantage of the proposed TETRA-DAB system is lost and this has been shown to be due to the fact that a symmetric standard PMR system can achieve more spectrum efficiency through frequency reuse.

In this study, the average number of users in a group is taken as 7. Furthermore, we assume that the average number of cells engaging in a group call is also equal to the average number of users in a group. This assumption may be valid especially when there are large number cells in the network, since the probability that all members are located in different cells increases as the number cells increases in the service area, i.e. as the cell size decreases. However, this assumption may not be valid if the cell sizes are large, i.e. the number of cells in the network is relatively small. In such a case, the probability that all members are located in different cells decreases and therefore the average number of cells engaging in a group call will probably less than 7. As a result, the advantage of the proposed TETRA-DAB system will be lowered. In order to exactly find the average number of cells engaged in a group call one has to know the probabilities of having one , two , etc. cells engaged in a group call. Furthermore these probabilities are dependent on the particular properties of groups which are using the PMR system. These probabilities can also be calculated from the steady state solution of the mobility model of groups using the system. Since the probabilities mentioned above and/or the mobility models for groups are not in general available we have simply assumed a fixed number for the number of cells engaged in a group call.

We have shown that the presence of group calls is the main reason for achieving spectral efficiency through the proposed new PMR system. We have not included broadcast calls in our numerical case studies, however if broadcast calls were to be included, then the proposed system would have been shown to achieve even more spectral efficiency.

There are other factors that are to be considered for spectrum efficiency determinations. For example, in TETRA, almost 40% of both uplink and downlink capacity should be reserved for the traffic due to handoffs. However, in the TETRA-DAB system, spectrum reservations for handoff must be made only in the uplink part. Therefore, from the handoff point of view, the TETRA-DAB system has additional advantage. Still another factor which influences the spectrum requirement of a PMR system is the need for additional control channel assignments for the purpose of call establishment and other control procedures. Again, in TETRA-DAB there is need for control channels only for the uplink part. On the other hand, the DAB downlink will use some additional spectrum due to coding overheads which are needed for packet identification and multiplexing. Detailed inclusion of these additional factors could be the subject matter of further studies.

If there are already DAB services installed in a metropolitan area, then a part of the DAB capacity can be allocated to PMR services, and thus significant economic advantages can be obtained. Even if there are no already installed DAB services, the proposed system would still have economical advantages due to relatively low cost of DAB equipment. The proposed system has an additional advantage in that, high data rate multimedia applications can also be implemented because of the availability of the broadband downlink.

For practical implementations of the proposed PMR system, one has to adapt the DAB standards in order to meet the specific requirements of a PMR system. For example, channel allocations in the DAB downlink must be achieved in less than 300ms, whereas in present DAB systems due to time interleaving, much more delays are inevitable.

Chapter 6

Conclusions and Future Work

A new asymmetric PMR system, comprising a broadband, wide area, single downlink cell and a narrowband cellular uplink structure, is proposed in this study to achieve higher capacity and higher spectral efficiency than conventional digital trunked systems have.

Single cell downlink is useful to achieve higher efficiency especially regarding different types of calls, such as group calls and broadcast calls, which are supported by only PMR systems. Due to their channel allocation schemes and their asymmetric characteristics, group calls and broadcast calls require more downlink channels as the number of cells engaging in a call increases. However, in a single cell structure, i.e. the whole service area is covered by a single transmitter, the number channels required for any type of call is not dependent on the number and location of users engaging in a call. In this study, an OFDM based structure is accepted to be the best scenario for downlink transmission, since OFDM based systems easily achieves wide area coverage by the use of Single Frequency Networks (SFN). On the other hand, large number of channels should be available to the service area when downlink transmissions are made by single transmitter. Therefore, OFDM based downlink structure must be broadband. Digital broadcasting systems are the best candidates for a broadband OFDM based structure, since they have well established standards and are widely used all over the world. In this thesis, we have taken into account the DAB system for the downlink communications. The uplink part should have a cellular structure and probably be narrowband.

Our work concentrates on the system capacity analysis of the proposed system associated only with the downlink part for voice communications, as well as the spectrum efficiency comparison of the proposed system with the TETRA system. In the capacity analysis part, the number of subscribers that can be supported by the proposed system is calculated under constraints such as grade of service (GOS) and bit error rate. In particular, we make use of voice activity detection that increased the system capacity significantly. Moreover, we show that, if the incoming calls are allowed to wait a certain amount of deterministic time before occupying a channel, capacity significantly increases as well. We have found that 20,000 PMR users, which is very large considering that PMR environment has relatively low traffic compared public access systems, can be supported by the proposed system if DAB is used as downlink, under reasonable constraints. Consequently, the proposed system can be used even in metropolitan cities as well as smaller cities using a portion of an existing DAB system's capacity for PMR services.

Spectrum efficiency is the most important parameter in today's communications world where spectrum congestion in dense urban areas is becoming a big problem. Therefore we have also investigated the performance of the proposed system in terms of spectrum usage. Our system is compared with the TETRA system which is generally accepted as the most efficient system among digital trunked systems. In the comparison process, the uplink structure of the proposed system should also be specified and we have made use of standards of TETRA. Therefore, the proposed system became a hybrid system making use of TETRA and DAB standards. It has been shown by numerical case studies that the proposed system is more efficient than the TETRA system when the service area is covered by less than or equal to 5 clusters, i.e. 35 clusters. In other words, if there should be 35 or fewer cells to cover a service area, our proposed system is shown to perform better than the TETRA system. As the number of clusters increases, the spectral advantage of the proposed TETRA-DAB system is lost and this has been shown to be due to the fact that a symmetric standard PMR system can achieve more spectrum efficiency through frequency reuse. We have shown that the presence of group calls is the main reason for achieving spectral efficiency through the proposed new PMR system. We have not included broadcast calls in our numerical case studies, however if broadcast calls were to be included, then the proposed system would have been shown to achieve even more spectral efficiency.

There are other factors that are to be considered for spectrum efficiency determinations. For example, in TETRA, when there are large number of cells in the network, about 40% of both uplink and downlink capacity should be reserved for the traffic due to handoffs. This reduces the available capacity for actual traffic. However, in the proposed system, spectrum reservations for handoff must be made only in the uplink part, meaning that the spectrum allocated for handoff in the proposed system will be roughly 20%. Therefore, from the handoff point of view, the TETRA-DAB system has additional advantage and the optimum found in this study will certainly go beyond 5 clusters to 6 or more. In the discussion part of Chapter 5, it was mentioned that taking the average number cells engaged in a group call as 7 overestimates the spectrum efficiency of the proposed system when there are relatively less number of cell. However, the advantages of the proposed system due to handoffs will probably surpass the overestimation of spectrum efficiency, and therefore not only will the optimum point go beyond 5 clusters, but also the proposed system will be more efficient even in scenarios where the cell sizes are large.

Still another factor which influences the spectrum requirement of a PMR system is the need for additional control channel assignments for the purpose of call establishment and other control procedures. Again, in the proposed system, there is need for control channels only for the uplink part. On the other hand, the DAB downlink will use some additional spectrum due to coding overheads which are needed for packet identification and multiplexing. Detailed inclusion of these additional factors could be the subject matter of further studies.

If there are already DAB services installed in a metropolitan area, then a part of the DAB capacity can be allocated to PMR services, and thus significant economic advantages can be obtained. Even if there are no already installed DAB services, the proposed system would still have economical advantages due to relatively low cost of DAB equipment. The proposed system has an additional advantage in that, high data rate multimedia applications can also be implemented because of the availability of the broadband downlink. For practical implementations of the proposed PMR system, one has to adapt the DAB standards in order to meet the specific requirements of a PMR system. For example, channel allocations in the DAB downlink must be achieved in less than 300ms, whereas in present DAB systems due to time interleaving, much more delays are inevitable.

Statistical data [38] show that PMR traffic is increasing, since more companies and users will make use of PMR systems. Therefore, if traffic reaches to such a large value that DAB system capacity is not enough to support PMR services, Digital Video Broadcasting (DVB) system, which supports 15 times more capacity than the DAB system does, is a good solution for high density urban areas. On the other, designing a new OFDM based system will probably be more efficient since DAB and DVB technologies are designed especially for broadcasting audio and video respectively. In fact, OFDM used in DAB assigns the same number of bits and an equal amount of power to all subchannels, regardless of their individual characteristics. This is expected, since OFDM is used in DAB for only broadcasting audio services and related data, without considering the large number of listeners' conditions. However, if OFDM is intended to be used for a multiuser wireless communications system, for example, the proposed asymmetric PMR system, the problem of adapting the transmit power and bit allocation is crucial. In this way, one may utilize the resources efficiently by adapting the power level and the number of bits to be transmitted according to the channel characteristics of each user. In a multiuser OFDM based communication system, each of the users' signals may undergo independent fading because users may not be in the same locations. Therefore, the probability that all users' signals on the same subcarrier are in deep fading is very low. Hence for a specific subcarrier, if a user's signal is in deep fading, the others may not be in deep fading and the user in a good channel condition may be allowed to transmit data on that subcarrier yielding multiuser diversity effects. Therefore, the problem of transmit power allocation for the multiuser OFDM system in downlink transmission is an important subject of matter for future studies.

APPENDIX A

The relation between $A\_ind^d_j$ and A_ind^u in a cell

Assume there exist N cells in the network and let $A\_ind^u_j = A\_ind^u$ denote the total individual uplink traffic in each cell in the TETRA network, where j denotes the j[th] cell and it is assumed that individual uplink traffic in all cells are the same and equal to A_ind^u. Let a be the probability that an individual call in a cell requests a downlink channel in the same cell and $(1-a)$ represent the probability that any other cell in the network is requested. Furthermore we assume that an individual call requests a downlink channel from other cells with equal probability $p=1/(N-1)$. Then, total downlink traffic in a TETRA cell can be represented by

$$\begin{aligned} A\_ind^d{}_j &= a \cdot A\_ind^u{}_j + (1-a) \cdot \sum_{\substack{i=1 \\ i \neq j}}^{N} p \cdot A\_ind^u{}_i \\ &= a \cdot A\_ind^u + (1-a) \cdot (N-1) \cdot p \cdot A\_ind^u \\ &= a \cdot A\_ind^u + (1-a) \cdot (N-1) \cdot \frac{1}{(N-1)} \cdot A\_ind^u \\ &= A\_ind^u \end{aligned} \tag{A.1}$$

APPENDIX B

The relation between $A\_group^d_j$ and A_group^u in a cell

Assume there exist N cells in the network and let p_{ji} be the probability that the i^{th} cell's group calls have a group member in the j^{th} cell. We assume that a group call engages M group members. Also the distribution of the group members in the service area is assumed to be uniform.

<u>**Case I:**</u> $N \geq M$

If N is larger than or equal to M, then we assume that M cells will be engaged in a group call, i.e. each group member is located in a different cell. Therefore, the probability p_{ji} that the i^{th} cell's group call has a group member in the j^{th} cell, i.e. j^{th} cell is engaged in the group call, can be described by

$$\begin{aligned} p_{ji} = P\{&\text{there exist a group member in } j^{th} \text{ cell}, \\ &\text{given that a group call arrives in } i^{th} \text{cell}\} \end{aligned} \tag{B.1}$$

Hence,

$$p_{ji} = 1 - P\{cell\ j\ is\ not\ engaged\ in\ cell\ i's\ group\ call, \quad \text{(B.2)}$$
$$given\ that\ there\ is\ a\ group\ call\ in\ cell\ i\}$$

$$= 1 - \frac{\binom{N-1}{M}}{\binom{N}{M}} = \frac{M}{N} \quad \text{(B.3)}$$

The relation between $A\_group^d_j$ and $A\_group^u_j$ in the j[th] cell can be represented by

$$A\_group^d{}_j = \sum_{i=1}^{N} p_{ji} A\_group^u{}_i \quad \text{(B.4)}$$

which is also the expected value of the total downlink group call traffic of in the j[th] cell.

If we assume uniformly distributed uplink group call traffic in all cells, i.e. $A\_group^u{}_i = A\_group^u$ for i = 1,…, N, the total downlink traffic in the j[th] cell can be described by

$$\begin{aligned} A\_group^d{}_j &= A\_group^u \sum_{i=1}^{N} p_{ji} \\ &= A\_group^u \sum_{i=1}^{N} \frac{M}{N} = M \cdot (A\_group^u) \end{aligned} \quad \text{(B.5)}$$

<u>Case II:</u> *N*<*M*

When *N* is smaller than *M*, group calls will engage smaller number of cells. We assume that if a group call is initiated, the number of cells engaging that call is equal to the number of cells in the system, *N*, when *N*≤*M*. That is to say,

$$\begin{aligned} A\_group^{d}{}_{j} &= A\_group^{u}\sum_{i=1}^{N} p_{ji} \\ &= A\_group^{u}\sum_{i=1}^{N}\frac{N}{N} = N\cdot(A\_group^{u}) \end{aligned} \tag{B.6}$$

Therefore, total group call downlink traffic of a TETRA cell can be computed as,

$$A\_group^{d}{}_{j} = \left\{ \begin{matrix} M\cdot(A\_group^{u}) & for\ \ N \geq M \\ N\cdot(A\_group^{u}) & for\ N < M \end{matrix} \right\}. \tag{B.7}$$

In this thesis, all numerical calculations fall into the *N*≥*M* case.

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