Capacity Analysis of a PMR System with DAB Downlink

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Abstract

Several trunked Private Mobile Radio (PMR) systems have been designed over the last decade, most of which have symmetric downlink and uplink channel capacities. These systems may not be spectrally efficient in case of group or broadcast-based voice and data calls, a common feature of PMR systems. We propose a new asymmetric PMR system comprising a wideband OFDM-based downlink and a narrowband uplink, which not only achieves a better spectral efficiency but also can support high bit rate multimedia applications. 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. In this paper, we study the performance and capacity of a private mobile radio system using a Digital Audio Broadcasting (DAB) downlink. In particular, we study the efficiency of such a system for voice calls using voice activity detection and statistical multiplexing. Moreover, we show that, the efficiency of the system can significantly increase, if the incoming calls, which can not find an available channel, are allowed to wait a certain amount of time before occupying a channel.

1. Introduction

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; 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. Three of the PMR systems, APCO25, TETRAPOL and EDACS, are based on the FDMA technology; however the other three, TETRA, IDRA and DIMRS, are based on TDMA. Current PMR systems are overviewed in [1].

All of the above PMR systems have symmetric downlink and uplink channel capacities. For example, 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 [2].

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 [3, 4]. An asymmetric system with a wideband downlink for higher data rates is desirable, because it is generally accepted that wideband is needed more for the downlink. Studies for such asymmetric systems have been undertaken in several projects in which the public access communications system, GSM, is complemented with DAB [5] or Digital Video Broadcasting (DVB) [6]. These proposals consider the wideband downlink only for multimedia applications, and voice calls are still carried by the narrowband channels. However if a wideband downlink is available then it may be possible to achieve spectrum efficiency for the speech signals as well if they are carried by the wideband downlink channel via a suitable trunking protocol. Asymmetric PMR systems have not yet been proposed.

In PMR systems, there are a number of voice and data applications which are not supported by public access communications systems such as GSM [2]. These are group (acknowledged or unacknowledged) and broadcast calls for both voice and data. In a cellular symmetric PMR system, in regions where offered traffic is high, cell sizes must be reduced. This in turn, increases the number of calls between users in different cells. If the members of a group call are in different cells, each one of them must be assigned uplink and downlink channels individually. However, if the group members are located in the same cell, then they use the same uplink and downlink channels. This spectrum efficiency in a PMR system is therefore lost if the group members are in different cells. However, in an asymmetric system where the downlink has wide area coverage, spectrum efficiency in a PMR system is maintained even with group calls. In PMR

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systems, group calls constitute a major proportion of all calls (50 %) [13]. Therefore, an asymmetric system is also advantageous when group and broadcast calls in a PMR system are considered.

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 paper focuses on the system capacity and grade of service (GOS) associated with the downlink channel for voice communications. 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, the efficiency can significantly increase.

Starting with a brief description of DAB in Section 2, a 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 3. Numerical results of the analytical model and simulations are discussed in Section 4. Finally we conclude and briefly describe our proposed future work.

2. A brief overview of the DAB system

The DAB system has been developed within the European Eureka 147 Project [8] and standardized by the European Telecommunications Standards Institute (ETSI), providing the means to broadcast high quality audio services to the users. Through the use of digital communications, the DAB system provides important potential benefits and exciting opportunities to the operators and users. Some of these benefits and opportunities are [7] reliable interference-free reception due to high resistance to the effects of multipath propagation and interference, efficient use of the limited RF spectrum by establishing single-frequency networks (SFNs), and flexibility of services.

The DAB system contains three main signal-processing sub-systems [8]: Audio coding**,** Transmission coding & multiplexing and COFDM Modulation [9]. At the transport layer, data is transmitted as blocks of Transmission Frames (Figure 1).

Especially 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 the proposed private mobile radio approach.

3. 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.

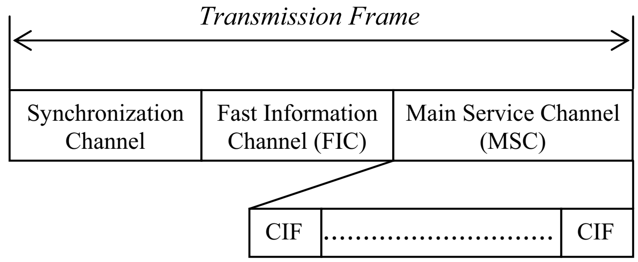

Figure 1. The structure of the DAB transmission frame

It is well known that voice contains an alternating sequence of ON and OFF times (talkspurts and silence gaps) [10]. 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 BER 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.

3.1. 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 $i \leq j$ as seen in Figure 2.

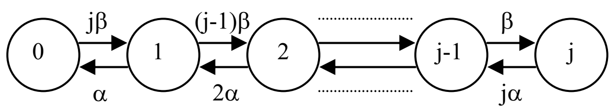

Figure 2. Markov model of the ON-OFF level characterization.

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 [12];

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

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

$$FLR = \frac{\sum_{i>240}^{N_{oc}} (i-240)B_i}{\sum_{i=0}^{N_{oc}} iB_i} \qquad (2)$$

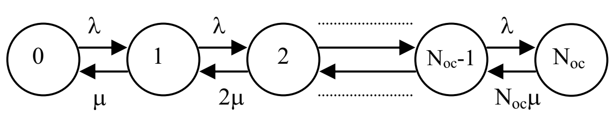

Figure 3. Markov model of the call level characterization.

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 Figure 3 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 [11];

$$P_j^{N_{oc}} = \frac{\left(\lambda/\mu\right)^j / j!}{\sum_{k=0}^{N_{oc}} \left(\lambda/\mu\right)^k / k!}, \ 0 \leq j \leq N_{oc} \qquad (3)$$

The call blocking probability can be derived using the P_j values from (3) which is the well known Erlang-B formula [11],

$$P_B = \frac{\left(\lambda/\mu\right)^{N_{oc}} / N_{oc}!}{\sum_{k=0}^{N_{oc}} \left(\lambda/\mu\right)^k / k!}, \ 0 \leq j \leq N_{oc} \qquad (4)$$

For a given value of FLR, the maximum number of users that can be supported by the system simultaneously, N_{oc}, is calculated from (2) by a binary search algorithm. Later, λ is calculated from (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

$$\boldsymbol{N}_{\boldsymbol{pop}} = \lambda / \lambda_{\boldsymbol{u}} \qquad (5)$$

We also study a variant 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.

3.2. Solution by simulation

The simulation method includes three steps:

3.2.1 Finding N_{oc} using ON-OFF 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.

3.2.2 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).

3.2.3 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. 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 1.

Table 1. System parameters for GSM and PMR scenarios

Parameters	GSM	PMR
Average call arrival rate (λcalls/hour/user)	2	10
Average call duration ($1/\mu$ sec)	180	20
Average duration in ON state ($1/\alpha$ sec)	1	1
Average duration in OFF state ($1/\beta$ sec)	1.35	1.35

4.1. 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 Figure 4. The results of the two methods do not agree. The reason for this difference is as follows: The Markov chain given in Figure 2 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.

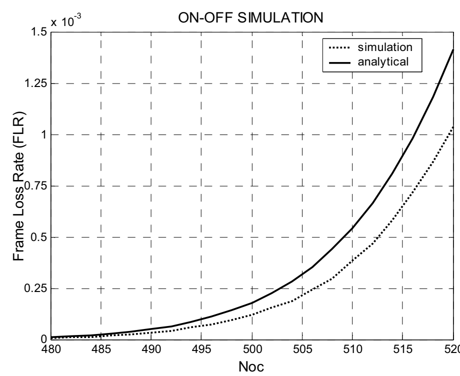

Figure 4. Frame loss rate as function of number of ongoing calls.

4.2. 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 Figures 5 and 6, 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 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 Figures 7 and 8, 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 Figure 8. A maximum waiting time of 0.48 sec (20 slots) supports nearly zero call-blocking probability in PMR scenario.

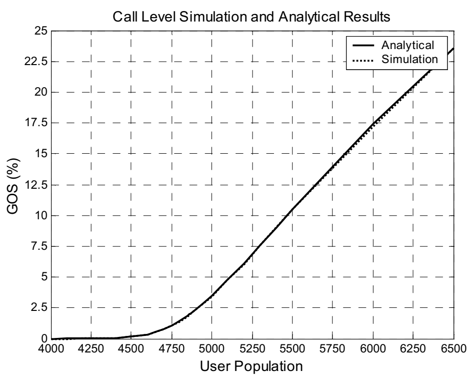

Figure 5. GOS: Grade of service providing on the average 10^{-4} FLR for GSM scenario. For GOS of % 2, optimum N_{pop} is found to be 4865.

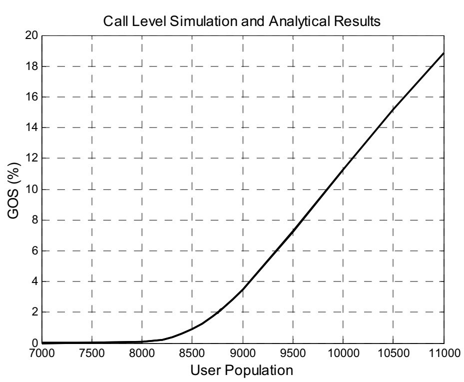

Figure 6. GOS: Grade of service providing on the average 10^{-4} FLR for PMR scenario. Simulation results are very close to analytical results. For GOS of % 2, optimum N_{pop} is found to be 8765.

For a given fixed call blocking probability, the system user capacity increases as the waiting time increases. As seen in the Figure 9, 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 Figure 10 if 2.88 sec maximum waiting time is allowed, the system capacity increases to approximately 20000.

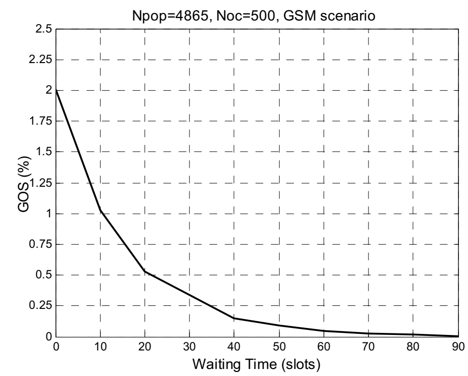

Figure 7. GOS as a function of maximum waiting time for GSM scenario.

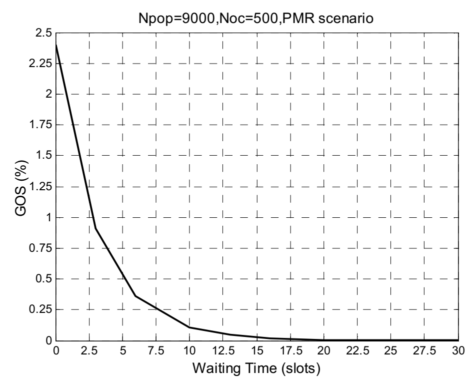

Figure 8. GOS as a function of maximum waiting time for PMR scenario.

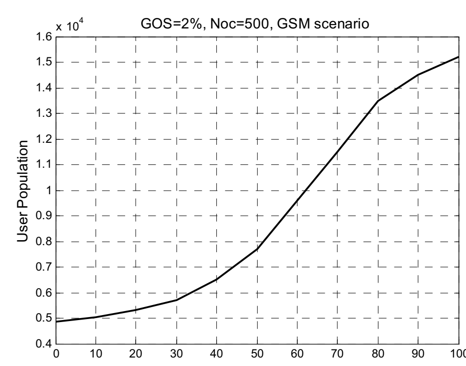

Figure 9. The maximum number of users that can be supported by the system as a function of maximum waiting time for GSM scenario for 2% GOS and 10^{-4} frame loss rate.

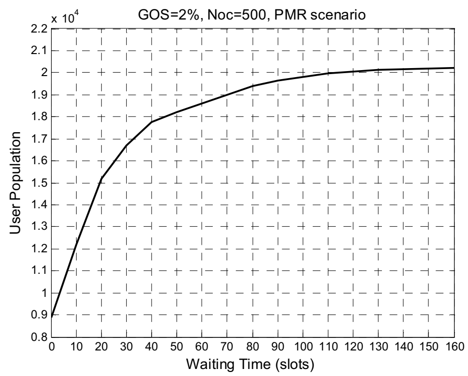

Figure 10. The maximum number of users that can be supported by the system as a function of maximum waiting time for PMR scenario for 2% GOS and 10^{-4} frame loss rate.

5. Conclusion and 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 [2]. 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.

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