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SPECTRAL EFFICIENCY OF CELLULAR LAND MOBILE RADIO SYSTEMS

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ABSTRACT

A rigorous and comprehensive approach to the definition and evaluation of spectral efficiency of cellular land mobile radio systems is presented. The method accounts for all pertinent system parameters within a cellular land mobile radio network. Subjective tests to evaluate the protection ratio for various systems is thought to be imperative to the evaluation of spectral efficiency. The paper concludes with a comparison of a number of current and proposed mobile radio schemes based upon the spectral efficiency evaluation package developed.

1. INTRODUCTION

The radio spectrum is a finite resource, and it is important that it is exploited efficiently by all users. The predicted growth in the UK is such that the bands allocated to mobile radio will become congested in the foreseeable future unless steps are taken to deploy modulation techniques and systems which significantly improve spectrum utilization.

Nowadays, a wide variety of modulation and multiple access techniques are offered as a solution to spectral congestion in the land mobile radio environment. Amongst the modulation systems suggested are: Wideband and narrowband digital systems (TDMA and FDMA based), spread spectrum and ACSSB, besides conventional FM analogue systems. Voice channel spacings vary from 5KHz for ACSSB systems up to 300KHz or more for spread spectrum systems. Furthermore, each multiple access technique: TDMA, CDMA, FDMA and hybrid multiple access techniques, is claimed to have the highest spectral efficiency when applied to mobile radio systems.

To date, many methods have been employed in an attempt to evaluate and compare these different modulation and multiple access techniques in terms of their spectral efficiency. These methods range from pure speculation, mathematical derivations, statistical estimations, to computer simulations, as well as methods based on laboratory measurements.

Unfortunately, none of the above methods can be said to be rigorous or conclusive. Mathematical methods, for instance, have been used to predict the co-channel protection ratio, yet this is highly a subjective system parameter. Other approaches, such as the statistical one, are difficult for the practicing engineer to apply in general. Results based on computer simulations must be treated with a degree of suspicion when the basis of such simulations, is not revealed.

Thus, it is essential to establish a rigorous and comprehensive set of criteria with which to evaluate and compare between different combinations of modulation and multiple access techniques in terms of their spectral efficiency in the cellular land mobile radio environment. Such a method is the subject of this paper, in which, only an outline of the method is presented. A thorough treatment will be published in the near future.

2. MEASURES OF SPECTRAL EFFICIENCY IN CELLULAR LAND MOBILE RADIO SYSTEMS

In order to assess the efficiency of spectrum usage in cellular land mobile radio networks, it is important to agree upon a measure of spectral efficiency which accounts for all pertinent system variables within such networks. A precise definition of spectral efficiency is the first step towards the resolution of the contemporary conflicting claims regarding the relative efficiencies of existing and proposed cellular land mobile radio systems. A precise measure of spectral efficiency will also permit the estimation of the ultimate capacity of various, existing and proposed, cellular land mobile radio systems as well as setting minimum standards for spectral efficiency.

Hatfield [1] surveys various proposed measures of spectral efficiency in land mobile radio systems, reviewing the advantages, disadvantages and limitations of each. As a result, the measure of spectral efficiency as:

$$\text{Erlangs/MHz/Km}^2$$

proves to be adequate, comprehensive and appropriate for cellular land mobile radio systems. In this definition, the Erlang as a measure of traffic intensity is used. It measures the quantity of traffic on a voice channel or a group of channels per unit time, and as a ratio of time, it is dimensionless.

An alternative and conceptually simpler definition of spectral efficiency in cellular land mobile radio systems is presented in terms of:

$$\text{Voice Channels/MHz/Km}^2$$

The above two measures of spectral efficiency are directly related. It can be shown that the conversion from Channels/MHz/Km^2 to Erlangs/MHz/Km^2 is easily obtained given equivalent blocking probability and holding time on the channels in the system.

Measures of Spectral Efficiency and Quality of Cellular land Mobile Radio Systems

For the above measures of spectral efficiency to be

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successful, the quality of service offered by different cellular land mobile radio systems, has to be included. However, when we talk about quality in terms of cellular land mobile radio systems, typically, three kinds of quality requirements are considered:

1) The degree of coverage in terms of traffic or area. That is to say, the percentage of the total area in which the service is available.

2) The grade of service in terms of blocking probability or waiting times, when the service is required.

3) Interference levels. This is judged by the protection ratio of a given modulation system which gives rise to a particular voice quality.

Of the above three quality requirements, only 2 and 3 are relevant to our spectral efficiency measures. In general, the grade of service and voice quality can be fixed to a given standard, which all cellular mobile radio systems in comparison have to comply with, and hence, a uniform quality is maintained throughout the comparison.

3. SPECTRAL EFFICIENCY OF MODULATION SYSTEMS

In cellular land mobile radio systems, there are two major parameters which govern spectral efficiency: the modulation system employed and the multiple access technique used to trunk the signals in the system. For the sake of convenience and flexibility, we propose to calculate the efficiency of modulation systems and the efficiency of multiple access techniques in isolation. The overall spectral efficiency of a particular cellular land mobile radio scheme is then obtained by combining the two types of efficiency.

Adopting the measure of spectral efficiency in cellular land mobile radio systems as: Channels/MHz/Km^2, the spectral efficiency of a modulation system can be mathematically interpreted by the following equations:

$$n_m = \frac{\text{Total number of channels available to the system}}{\text{Total available bandwidth . Cluster area}} \qquad ...(1)$$

$$n_m = \frac{B_t/B_c}{B_t.(N.A)} \qquad ...(2)$$

$$n_m = \frac{1}{B_c.N.A} \qquad ...(3)$$

Where:

n_m : Modulation efficiency in Channels/MHz/Km^2.
B_t : Total bandwidth available to the system in MHz.
B_c : Voice channel spacing in MHz.
N : Number of cells per cluster.
A : Cell area in Km^2.

From equation *(3)*, the spectral efficiency of a modulation system is inversely proportional to the channel spacing B_c and the cluster area *(N.A)*. It is independent of the total bandwidth B_t allocated to the cellular land mobile radio system, excluding trunking efficiency considerations, which are considered in section 6.

For two competing modulation systems x and y, the relative spectral efficiency n_r is given by:

$$n_r = \frac{(n_m)_x}{(n_m)_y} \qquad ...(4)$$

$$n_r = \frac{(B_c)_y.N_y.A_y}{(B_c)_x.N_x.A_x} \qquad ...(5)$$

The cell area, A, is independent of the type of modulation used and the amount of power transmitted, since a cellular land mobile radio system is interference limited. Hence, equation *(5)* becomes:

$$n_r = \frac{(B_c)_y.N_y}{(B_c)_x.N_x} \qquad ...(6)$$

Ultimate Capacity of Cellular Land Mobile Radio Systems

The ultimate capacity of a land mobile radio system using a particular modulation system is achieved by employing the minimum possible cell size through cell splitting. This, in turn, is limited by some practical considerations such as the hand-off rate, cell site tolerance, acceptable co-channel interference, power control and paging mobiles within the system [2]. Cells as small as 1.5 km in radius appear to be practical [3]. In our opinion, a cell radius as small as 1 km ($A \approx 3$ Km^2) is possible with the present technology. This will be used in the efficiency comparison of various cellular systems.

4. CALCULATION OF THE NUMBER OF CELLS PER CLUSTER (N)

In a cellular land mobile radio system, the number of cells per cluster, N, influences its capacity. Nevertheless, N itself depends on the cell shapes as well as the model used to calculate the co-channel interference in the system. In this respect, we need to look at the cellular geometry used to model the cellular land mobile radio system.

The Cellular Geometry

The main reason for defining cells in a cellular system is to outline areas in which specific channels and specific cell sites are used. Visualizing all cells as having the same geometrical shape helps to ease the assessment of spectral efficiency of various cellular land mobile radio systems especially to calculate the co-channel interference in the system. For many reasons, the regular hexagon is favoured and widely used by system designers. Correction factors can be introduced to allow for practical considerations. Using the hexagonal geometry, the following relations are obtained [2]:

$$\text{Co-channel reuse ratio} = \frac{\text{Min. co-channel cell separation}}{\text{Cell radius}} \qquad ...(7)$$

$$Q = \frac{D}{R} \qquad ...(8)$$

$$\frac{D}{R} = [3N]^{1/2} \qquad ...(9)$$

where N can only take restricted values, e.g. $N = 1, 3, 4, 7, 9, 12, 13,$ ...etc.

5. RELATIONSHIP BETWEEN CO-CHANNEL REUSE RATIO (D/R) AND PROTECTION RATIO (C/I) IN A CELLULAR SYSTEM

The spectral efficiency of a cellular land mobile radio system employing a particular modulation system is a function of three main system parameters: channel spacing, cell area and the co-channel reuse ratio. It is of great importance, however, to relate the spectral efficiency of modulation systems to speech quality experienced by the users in the cellular land mobile radio system. The speech quality is influenced by the signal to co-channel interference protection ratio determined by the modulation system used. To establish a relationship between the protection ratio of a modulation system and the co-channel reuse distance, it is necessary to model the cellular land mobile radio system in such a way that the propagational effects on the radio signal are accounted for. It is also necessary to model the relative geographical locations of the transmitters and receivers in the system so as to be able to predict all the significant co-channel interference affecting the desired signal.

Co-channel Interference Models

In general, two main categories of co-channel interference models can be visualized. The first category is a geographical one, where the models are constructed by considering the relative geographical locations of the transmitters and receivers, considering different possible numbers of interferers in the system. The second category is a statistically based group of models, in which the propagational effects, mainly fading and shadowing, are included in a statistical fashion.

A thorough comparative study of six different models were conducted. Three geometrical models: considering one interferer, six interferers [4] and several tiers of interferers. Also, three statistical models: fading only [5], shadowing only [6] and both fading and shadowing [6]. The overall result is shown in Fig.1 and a brief comparison is given.

Comparison of the Various Models

1. In general, the geometrical models are easier to develop and use.

2. The geometrical model with six interferers is a good compromise compared with the two other geometrical models. The geometrical model with one interferer is not a realistic one, on the other hand, considering several tiers of interference does not significantly improve the model.

3. Although fading and shadowing effects are not considered when the geometrical models are developed, their effect on the signal can be included in the value of protection ratio by performing subjective measurements for various modulation systems, under fading and shadowing conditions.

4. As for the statistical category, the model with fading and shadowing accounts for the general situation which characterizes the mobile radio channel. The shadowing only model is useful when means of diversity reception are adopted in the system. The fading only model is useful when a line of sight reception is available.

5. The major drawbacks in the statistical models are their complexity and in the unrealistic assumption of one interferer only. Attempts to account for the interference from the first tier of co-channel cells [7] adds further complexity to the statistical models making them more difficult for application.

6. Furthermore, the way in which the statistical models are used is not very obvious. If the protection ratio for a particular modulation system is to be subjectively assessed under fading and shadowing conditions and the statistical model with fading and shadowing is used, then large values of reuse distances appear to be necessary. For example, the protection ratio for 25/30kHz FM cellular land mobile radio systems is in the range of 18dB; according to the statistical model with 90% fading and 6dB shadowing the number of cells per cluster required for quality voice reception is 21, which does not comply with established cellular systems such as AMPS and TACS, which use only 7 cells per cluster. On the other hand, the statistical models inherently assume that fading and shadowing cause identical deterioration to the signal regardless of the modulation system employed, which is not the case in practice.

As a result, although the statistical models appear attractive, they do not resemble the practical situation and hence are not favoured. We feel strongly in favour of the geometrical model with six interferers. It is a useful tool in assessing the spectral efficiency of cellular systems, provided that the values of protection ratio used are subjectively evaluated for various modulation systems under fading and shadowing conditions.

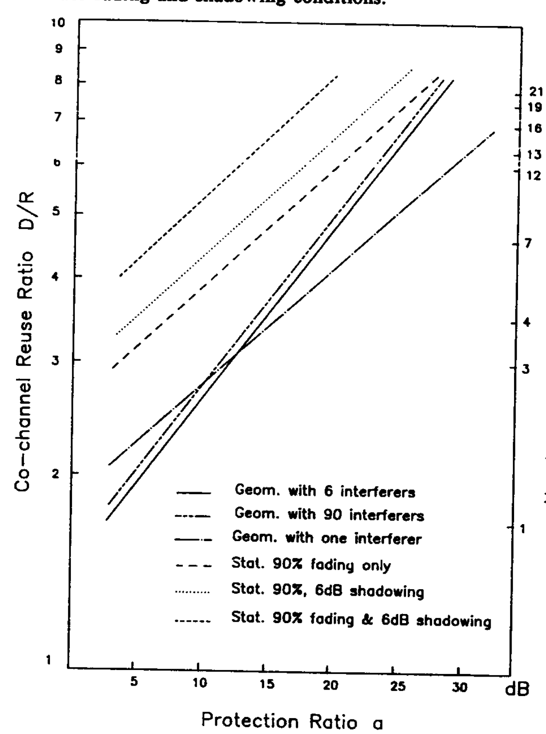

Fig.1 Comparison of Various Co-channel Interference Models

6.SPECTRAL EFFICIENCY OF MULTIPLE ACCESS TECHNIQUES

The aim of multiple access techniques is to combine signals from different sources on a common transmission medium in such a way that, at their destinations, the different signals or channels can be separated without mutual interference. In other words, multiple access techniques permit many users to share a common communication band, presumably, in the most efficient way.

Basic Multiple Access Techniques

Basically, there are three multiple access techniques:

a) Frequency Division Multiple Access (FDMA): In this technique, the users share the radio spectrum in the frequency domain. The user is allocated part of the frequency band to use throughout the time of his communication.

b) Time Division Multiple Access (TDMA): In this technique, the users share the radio spectrum in the time domain. The user is allocated a time slot during which, he has access to the whole frequency band allocated for the system (wideband TDMA), or only part of the band (narrowband TDMA).

c) Code Division Multiple Access (CDMA): This technique combines FDMA and TDMA. With the CDMA technique, each user in the system is assigned a 'unique' pseudorandom user code. Then each user can access the time-frequency domain at any time in a 'unique' manner. Frequency hopping is one example of this technique.

Theoretical and Practical Efficiency of Multiple Access Techniques

Theoretically, all multiple access techniques described above have the same spectral efficiency, provided that signals transmitted from different users in the cellular system, are orthogonal. However, this is not the case in practice, where implementation limitations reduce the theoretical efficiency of different multiple access technique, each in a different way. In FDMA, for instance, the multiple access spectral efficiency is reduced by the need of guard bands between channels to reduce filter roll-off requirements and to accommodate frequency shifts. Similarly, in TDMA technique, the spectral efficiency is reduced by the inclusion of guard time and synchronization preamble ...etc. In general, multiple access efficiency is reduced by the need of control channels, set up channels ...etc.

Multiple Access Efficiency Factor

We define a multiple access efficiency factor as: "The ratio of the total time-frequency domain dedicated for voice transmission to the total band available to the system".

From the above definition, the multiple access efficiency factor is dimensionless, and has an upper limit of one. Mathematically, the multiple access efficiency factor n_t, t for trunking, has to be defined for different multiple access techniques. Furthermore, since multiple access techniques are defined in both the time and frequency domains, n_t is given by :

$$n_t = \textit{(Its efficiency in the frequency domain)} \cdot \textit{(Its efficiency in the time domain)} \qquad ...(10)$$

n_t is evaluated for different multiple access techniques:

1. **FDMA**

$$n_t = \frac{B_c.M_a}{B_t} \leq 1 \qquad ...(11)$$

where :

B_c and B_t have their usual notations.
M_a : The total number of voice channels available to the system.

2. **Wideband TDMA**

$$n_t = \frac{\tau.M_t}{T} \leq 1 \qquad ...(12)$$

where :

τ : Time slot duration in seconds (for voice transmission).
T : Frame duration in seconds.
M_t : Number of time slots for voice transmission in a frame (time channels).

Alternatively :

$$n_t = \frac{\textit{Voice channel (in b/s)} . M_d}{\textit{Total bandwidth (in b/s)}} \qquad ...(13)$$

3. **Narrowband TDMA**

$$n_t = \frac{\tau.M_t}{T} \cdot \frac{B_u.M_u}{B_t} \qquad ...(14)$$

where :

τ, M_t, T and B_t have their usual notations.
B_u : Bandwidth which a user has access to, during his time slot.
M_u : Number of users sharing the same time slot in the system, but having access to different frequency bands.

4. **CDMA**

$$n_t \leq \frac{\tau.M_t.B_u.M_u}{T.B_t} \qquad ...(15)$$

Some studies show that the theoretical efficiency of asynchronous CDMA technique in utilizing the frequency domain is as low as 70% compared with FDMA and synchronous TDMA techniques [8].

7. OVERALL SPECTRAL EFFICIENCY OF CELLULAR LAND-MOBILE RADIO SYSTEMS

The overall spectral efficiency of a cellular land mobile radio scheme employing a modulation system of spectral efficiency n_m and a multiple access technique of efficiency factor n_t, is given by :

$$n = n_m \cdot n_t \qquad ...(16)$$

Using the geometrical model with six interferers, n and n_r are given by :

$$n = \frac{3n_t}{B_c.[6a]^{2/\alpha}.A} \qquad ...(17)$$

$$n_r = \frac{(B_c)_y.(a_y)^{2/\alpha}}{(B_c)_x.(a_x)^{2/\alpha}} \cdot \frac{(n_t)_x}{(n_t)_y} \qquad ...(18)$$

where n is in Channels/MHz/Km^2 and n_r is dimensionless; α is the propagation power law.

Table.1 shows a comparison of a number of current and proposed cellular land mobile radio schemes perform under the above spectral efficiency package. Parameters being obtained from the open literature, $A = 3\text{Km}^2$ and α is 4.

8. SPECTRAL EFFICIENCY OF DIGITAL CELLULAR MOBILE RADIO SYSTEMS

Although the previous spectral efficiency evaluation method is mainly designed for analogue cellular land mobile radio systems, the method can easily be adapted for digital modulation systems. The two main parameters which need to be evaluated for digital modulation systems are: a) The bandwidth occupancy, since the channel is defined in terms of kb/sec. b) The protection ratio in a cellular environment. In this paper, two approaches to evaluate the two parameters are presented:

1. Practical approach: This is a simple method used to combine the voice channel bit-rate in kbits/sec with the practical speed of the digital modulation system in bits/sec/Hz. In this way the equivalent channel spacing (or bandwidth occupancy) is mathematically given by:

$$\textit{Equivalent ch. spacing} = \frac{\textit{Voice channel bit-rate (kb/s)}}{\textit{Modem speed (b/s/Hz)}} \qquad ...(19)$$

The protection ratio required for quality voice reception is then subjectively assessed under fading and shadowing conditions, in exactly the same way as for analogue modulation systems. The attraction of this approach is in its simplicity as well as resembling the true practical situation. Furthermore, the effect of coding on the value of the protection ratio will automatically manifest itself in the subjective assessment.

2. Theoretical approach: A more formal approach is found in [9]. The protection ratio of a digital modulation system is calculated in terms of its bandwidth utilization in bits/Hz and the energy per bit per noise density which characterizes the type of digital modulation employed. The following assumptions were made:

a) Optimum processing and performance can be achieved for all digital modulation systems.

b) A 16 kb/sec digitized speech channel with the quality maintained.

c) Random bit error rate of less than 1 in 10^{-2} is adequate for voice quality.

d) Only one bit error occurs per symbol in multi-level digital modulation systems.

For multilevel systems, the protection ratio is given by :

$$S/I = (E_b/N_o).\log_2 M \qquad ...(20)$$

where:

E_b : The energy per bit.
N_o : The interference density.
M : The number of levels in the M-ary digital modulation system. Also, the system bandwidth efficiency is given by $\log_2 M$ (bits/Hz).

For M-ary bi-orthogonal signaling using QPSK as the carrier, the protection ratio is given by [10]:

$$S/I = [2k/2^{(k-1)}].(E_b/N_o) \qquad ...(21)$$

Parameters / Cellular scheme	Channel spacing or Equivalent Channel spacing (kHz)	Protection ratio (dB)	Modulation efficiency (Ch/MHz/Km^2)	Relative efficiency (to 25kHz/FM)	Trunking efficiency	Overall efficiency (E/MHz/Km^2)
TACS/FM	25	18−19 (Measured)	1.82	1.0	>0.95	1.47
AMPS/FM	30	17−18 (Measured)	1.52	0.84	>0.95	1.19
ACSB/5kHz	5	20 (Measured)	7.10	3.90	>0.95	6.57
CD900 TDMA/Digital	72	11−12 (Measured)	1.47	0.81	~0.80	0.94
SFH900 Spread Spectrum Digital	75	7 (Derived)	1.42	0.78	~0.75	0.83
NB/FDMA/Digital (AT&T Proposal)	10	15 (Some tests)	7.96	4.37	>0.95	6.21
DMS−90 NB/TDMA/Digital (Proposal for Europe)	30	10−12 (Derived)	3.54	1.95	~0.80	2.58

Table 1. Spectral Efficiency of Some Cellular Schemes

where:

E_b and N_o as above.
k : The number of orthogonal levels employed. Also, the theoretical bandwidth efficiency for bi-orthogonal QPSK signaling is given by $[2k/2^{(k-1)}]$.

This theoretical approach has some drawbacks due to the assumptions made earlier. For instance, although coding can be accounted for in terms of bandwidth, the above equations do not show its effect on the value of protection ratio. Nevertheless, this method is particularly appealing in the absence of subjective protection ratio figures.

9. ALTERNATIVE SPECTRAL EFFICIENCY MEASURE: ERLANGS/MHZ/KM^2

An alternative measure of spectral efficiency is in terms of Erlangs/MHz/Km^2. Following the definition of an Erlang as the quantity of traffic on a voice channel or a group of channels per unit time and using equation *(2)*:

$$n_m\ (Erlangs/MHz/Km^2) = \frac{\text{Traffic carried by } (B_t/B_c) \text{ channels}}{B_t \cdot N \cdot A} \quad ...(22)$$

A more accurate definition is to treat the traffic in each cell in the system independently. Hence, n_m becomes:

$$n_m\ (Erlangs/MHz/Km^2) = \frac{\text{Traffic carried by } [(B_t/B_c)/N] \text{ channels}}{B_t \cdot A} \quad ...(23)$$

Furthermore, the trunking efficiency can be included in the following way:

$$n\ (Erlangs/MHz/Km^2) = \frac{\text{Traffic carried by } [(B_t/B_c)n_t/N] \text{ channels}}{B_t \cdot A} \quad ...(24)$$

Basically, there are two traffic models by which the amount of traffic available to a cell in the system can be calculated:

1. A 'pure loss' or blocking system model in which blocked calls are cleared. That is to say, if a call arises when all channels are busy, the call is immediately cleared with zero holding time. This model is best described by the Erlang-B distribution, which is widely used in traffic theory.

2. Queuing system model in which blocked calls wait. In this model, a blocked call can wait until a voice channel becomes free. The Erlang-C distribution is a good representative of this model. It gives the probability of delay greater than 't' seconds in terms of the number of channels available to the system.

In general, traffic models are very much dependent upon the behaviour of users in the system and hence, the choice of a traffic model for cellular systems must follow a detailed study of such behaviour within mobile systems. Nevertheless, the blocking system model suffices for cellular systems. In our opinion, adopting a queuing system model is unlikely to achieve any substantial increase in spectrum efficiency. This is because more control channels are required to accommodate for the queue, which can considerably lower the trunking efficiency of the system.

From equation *(24)*, the following are noted :

1. The spectral efficiency in Erlangs/MHz/Km^2 is influenced by the quality of the cellular system in terms of the grade of service as blocking probability or waiting time, depending on the traffic model.

2. The voice quality is represented by the value of N, since N is a function of the protection ratio of the modulation system employed.

3. The spectral efficiency in Erlangs/MHz/Km^2 depends on the total bandwidth allocated to the cellular system B_t. This is due to the non-linear relation between the number of channels available to a cell, and the amount of traffic in Erlangs which can be carried by the channels.

4. The capacity of a cellular system in terms of Users/MHz/Km^2 can be derived given the spectral efficiency in Erlangs/MHz/Km^2, and the average traffic per user in the system (e.g. 0.05 Erlangs/user, in the busy hour).

Table.1 shows the spectral efficiency in Erlangs/MHz/Km^2 of a number of current and proposed cellular land mobile radio schemes operating in a 10MHz bandwidth, using Erlang-B distribution and 2% blocking probability.

10. METHODS OF OBTAINING THE VALUE OF THE PROTECTION RATIO FOR DIFFERENT MODULATION SYSTEMS

Cellular land mobile radio systems are interference limited, and the protection ratio plays an important role in assessing their spectral efficiency. Unfortunately, although many protection ratio figures appear in the literature every now and then, some methods by which such figures are obtained are not very convincing. It is necessary, therefore, to agree upon a standard systematic method by which the protection ratio can be evaluated for different cellular systems. Such a standard method should account for all propagational effects on the radio signal. Also, any technique which is believed to enhance the performance of any particular system could be included. First, the protection ratio needs to be defined.

Definition of Protection Ratio

The world Administrative Radio Conference, Geneva, 1979, defined the protection ratio as the minimum value of the wanted-to-unwanted signal ratio, usually expressed in decibels, at the receiver input determined under specified conditions such that a specified quality of the wanted signal is achieved at the receiver output [11]. This ratio may have different values according to the type of modulation system used. A more precise definition of protection ratio is [2]: "The level at which 75% of the users state that the voice quality is either good or excellent in 90% of the service area". Other definitions exist, however, it is necessary to have a standard definition of the protection ratio so that consistent values of protection ratios can be obtained.

The following is a survey of possible methods of obtaining the protection ratio.

a) Mathematical Derivation: To mathematically derive a value for the protection ratio requires a great deal of involvement in the various parameters of a modulation system as well as the the conditions which affect the radio signal. The mathematical model must also be able to allow for any technique which can improve the signal quality. Furthermore, the hardware used such as the modems and transceivers etc. may need to be realized mathematically. In all, this method can prove very tedious with a doubtful outcome of protection ratio figures.

b) Objective Measurements (SINAD Method): In this method, a drop in the SINAD value of the desired signal, usually from 18 to 12dB's, is set as an indication of the modulation system performance under co-channel interference conditions. In the test, the desired signal is replaced by a modulated tone, usually at 1000Hz, and the interference signal is either a voice shaped noise or a voice recording played back. Note that with the desired signal replaced by only a modulated tone, propagational conditions such as frequency selective fading, will not have the same practical effects. Also, large uncertainties in the SINAD readings can result due to fading, which can make this method impractical.

c) Subjective Measurements: In cellular systems, the protection ratio as a measure of voice quality is certainly a matter for subjective assessment. The subjective assessment of the signal to co-channel protection ratio involve human judgment of system performance. Such judgment may be based on the quality, intelligibility or the general acceptability of the received signal under realistic conditions. Details of this method can be found in [12].

d) Field Measurements: The protection ratio can be subjectively assessed in the real cellular mobile environment. This can be done using one desired transmitter site and six undesired transmitters, as in practice. Then, with the undesired transmitters playing back typical conversations, the desired signal is recorded at a roaming mobile station for a range of S/I values. The protection ratio is then subjectively assessed by a group of listeners. This method has the advantage of a real mobile environment but it can prove very expensive to carry out. It is an option in hand for established systems which already exist.

11. CONCLUDING REMARKS

This paper combines a global approach to the definition and evaluation of spectral efficiency which accounts for all system parameters in cellular land mobile radio systems and the ease of a practical applicability to all existing and proposed, digital and analogue, cellular systems. Hence such systems can be set in a ranking order of spectral efficiency.

This study also demonstrates the crucial importance of the protection ratio in the evaluation of the spectral efficiency of modulation systems. It is also argued that since the protection ratio of a given modulation system inherently represents the voice quality under varying conditions, it is imperative that such a parameter is evaluated subjectively. Furthermore, the evaluation of the protection ratio should be performed under various simulated conditions, e.g. fading and shadowing, in such a way that the effect of these conditions is accounted for in the overall value of the protection ratio. In addition, any technique which improves signal quality or overcomes hazardous channel conditions in the system should also be included in the test. Consequently, the effect of amplitude companding, emphasis/de-emphasis, coding, etc. will influence the overall value of the protection ratio.

Unfortunately, because of the space limitation, a thorough treatment cannot be given. However, this will be the subject of a full paper which to be published soon.

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RADIO CHANNEL STRUCTURE FOR SCPC/FDMA DIGITAL MOBILE SYSTEMS

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ABSTRACT

This paper describes SCPC/FDMA digital mobile radio system channel structure supporting ISDN services. Three kinds of radio channels are proposed offering ISDN services with efficient spectrum utilization. They are an information channel, control channel and packet channel.

The effects of diversity and FEC are discussed for the improvement of digital transmission quality and frequency spectrum utilization.

Some studies are presented regarding quality objectives for bearer services under multipath fading environments.

1. INTRODUCTION

ISDN services for a fixed network are now being planned for world-wide utilization. In Japan, they will be initiated this year (1988) in major cities.

In the future, it should be possible to offer the ISDN related services and facilities to public land mobile and portable radio systems.

In addition to the various services provided for in ISDN, the information bit rate is considerably higher than data communications in analog telephone systems (using baseband modem). Thus present analog mobile radio systems may no longer be applicable for ISDN services.

The severe multipath fading in radio transmission make it difficult to establish highly reliable digital signal transmission. The channel structure for mobile ISDN should be designed while bearing in mind the differences in wireline and mobile radio environment characteristics.

Efficient spectrum utilization is also an important subject to accommodate the tremendous growth in numbers of mobile and portable radio subscribers within spectrum resource limitations.

Digital mobile radio systems are being developed to meet these requirements(1)~(3).

In this paper, logical and practical (radio) channel structures are proposed to realize various mobile ISDN services with efficient spectrum utilization.

Effects of diversity and forward error correction (FEC) for the digital transmission reliability and frequency spectrum utilization are investigated. The studies are also given for the quality of mobile ISDN.

2. ISDN SERVICES

ISDN offers circuit switched service and packet service. Circuit switched service bearer rates are 64kb/s, 384kb/s, 1,536kb/s, 1,920kb/s. In addition, 40Mb/s and 150Mb/s rates are now being discussed.

For mobile radio, however, it is desirable to provide lower bearer rate (sub-rate) services from the viewpoint of efficient spectrum utilization.

CCIR IWP 8/13 proposed low bit rate bearer service of 8kb/s, 16kb/s and 32kb/s.

Two types of packet services are provided in ISDN. One is B-channel packet service and the other is D-channel packet service. B-channel packet is transmitted/received using circuit switched radio channel. There are two ways to transmit and receive D-channel packet signal.

(1)A circuit switched radio channel (dedicated channel) is assigned to each packet user.

(2)Several packet users share one radio channel (radio packet).

From the viewpoint of frequency spectrum utilization, (2) is preferable. However, the through-put of (1) is higher than that of (2). Therefore, both types of radio channels may be necessary.

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3. RADIO CHANNEL DESIGN

3.1 Channel Implementaion Structure

Since available radio spectrum is very limited, it is difficult to offer mobile ISDN service with data rate of 144kb/s (2B+D interface) or more. Instead, the following three kinds of channel structures are proposed.

(a) Information Channel:

-The information channel, I, corresponds to ISDN B (Bearer) channel.

-The I-channel access rate consists of m_1 or $m_2 \times I_0$, where I_0 is the basic building block rate.

-The building block rate I_0 should be determined considering bearer service rates of for mobile radio, voice coding rate and frequency spectrum economy. I_0=8kb/s was recommended in CCIR IWP8/13.

-Plural information channels are combined as $m_1 \times I_0 + m_2 \times I_0$, where m_1 and m_2 are chosen individually.

(b)Control Channel:

-The control channel, C, corresponds to ISDN D (Data) channel.

-C-channel consists of the following two channels:

* C_1 (Common channel):
  - Paging and access control.
  - D-channel packet.
* C_2 (Dedicated channel):
  - Assocated to I-channel for supervisory control.
  - D-channel packet.

-The assocoated mode C_2-channel bit rate should be as low as possible within the confines of good service quality.

In dedicated D-channel mode, C_2 bit rate should be variable according to the through-put and the amount of information. So, channel structure of $C_2 = \ell \times C_0$ is desirable where C_0 is the basic building block rate of C_2.

(c) Packet Channel :

B-channel packet signals are transmitted/received in I-channel. D-channel packet signals are transmitted/received in C_1 channel or C_2 channel as written in (b). From the view point of efficient spectrum utilization, it is effective to use C_1-channel for user packet service. However, it may be difficult to achieve high through-put in C_1 channel because paging and access control signal

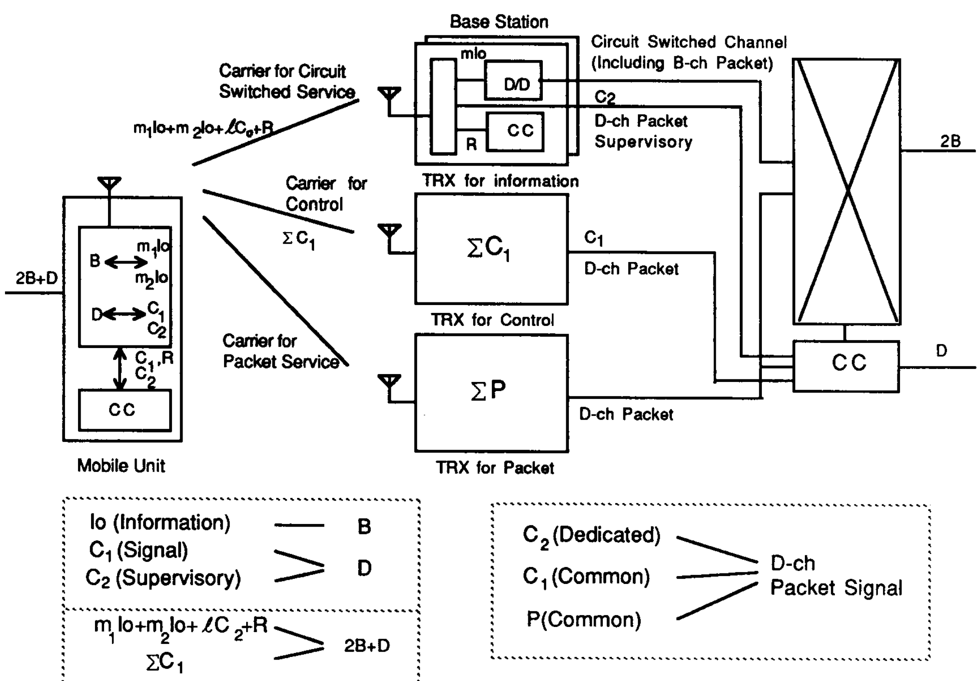

Fig.1 Radio Channel Structure

take priority over packet signal. So, in order to provide high through-put packet services, it is necessary to provide the user packet channels (P) which are independent of C_1-channel.

(d) House-Keeping Channel for Radio Link :

In order to accommodate the supervisory functions (radio signal attenuation etc.) for the radio link, the house-keeping channel, R, should be provided. House-keeping channels are terminated at the base station.

3.2 Multipe Access Techniques

Basically, there are two ways to achieve multiple access for the digital mobile radio system : FDMA and TDMA. From the view point of service quality and spectrum efficiency, FDMA is preferable because of radio channel allocation flexibility. Co-channel interference can be avoided by re-allocating the carrier frequency. Bandwidth can be optimized in proportion to the information bit rates. Transmitter power can be varied according to the required quality.

Another feature of FDMA is that it is not sensitive to selective fading. Because of the spectrum limitation, higher frequency bands (1~3GHz) will be applied to mobile radio in the near future. For such higher frequency bands, fading frequency becomes 100~400Hz. It becomes more difficult to equalize the unwanted time dispersion at such a high speed fading.

4. QUALITY AND IMPROVEMENT TECHNIQUES FOR DIGITAL TRANSMISSION

Good quality digital signal transmission is necessary for mobile ISDN service as well as efficient frequency utilization. Diversity and FEC are effective ways to overcome severe multipath fading and realize high quality digital transmission. Following is a study of diversity and FEC on bit error rate (BER) improvement and spectrum utilization.

4.1 Diversity :

BER performance experiments were carried out both with diversity and without diversity under the following conditions.(4)

-Modulation : GMSK (BbT=0.25)

-Bit rate : 16kb/s

-Diversity algorithm : Baseband switching

For same BER, average CNR decreases considerably as compared with that of no diversity as shown in Fig.2.

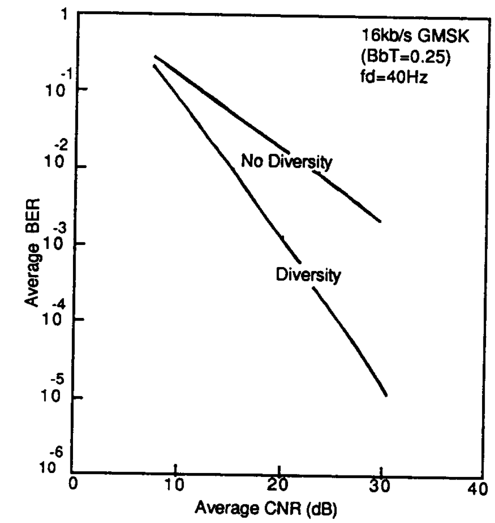

Fig. 2 BER vs. CNR

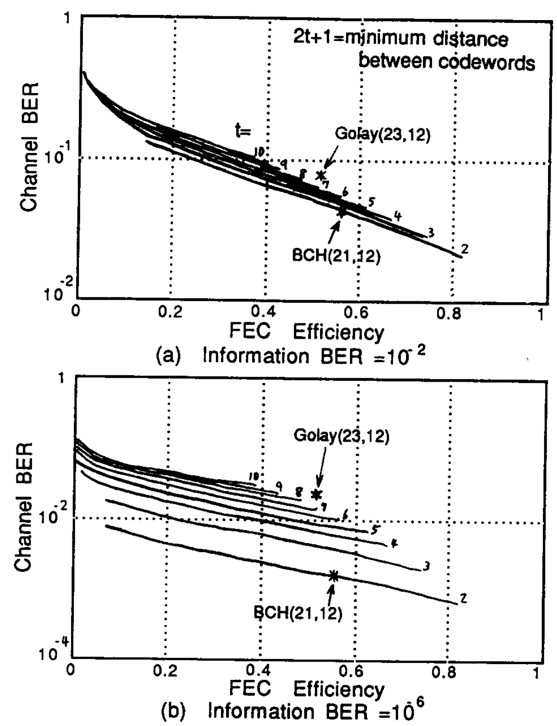

Fig.3 Information BER vs. Channel BER

4.2 FEC :

Figure 3 shows simulation results about the relation between channel BER and information BER as a function of FEC efficiency. As the FEC efficiency decreases, BER also decreases.

Since additional bandwidth is necessary for FEC to reduce BER, the spectrum efficiency degrades. However, considering co-channel interference resistivity, FEC improves spectrum efficiency. Accordingly, frequency utilization factor should be evaluated considering both increased bandwidth and a decreased frequency reuse zone number.

Figure 4 shows the flow chart used to obtain spectrum efficiency. Calculations were carried out assuming the following conditions.

-FEC : BCH(21,12) or Golay(23,12) with soft decision[5]

-Channel separation : 25kHz for no FEC

-Diversity : Baseband switching diversity

-Fading : Rayleigh, fading frequency=40Hz

-Outage probability : 5% (corresponds to CIR margin=11dB)

Spectrum efficiency factor A (Number of channels/Bandwidth(1 MHz)/1 zone) is derived from Fig 5. in the case of uniform traffic distribution. Figure 5 shows that FEC is effective to improve spectrum utilization efficiency for high quality transmission (average BER is about 10^{-4}~10^{-6}). As ISDN bearer service requires maintaining the 10^{-6} BER (as shown in 4.3), FEC is effective. But FEC does not improve spectrum utilization efficiency for relatively low quality transmission (average BER is 10^{-2} or more). This means that FEC is not necessarily effective for voice channel communication, because voice channel average BER is prescribed to be around 10^{-2} (for zone boundary).

Relations between FEC efficiency and spectrum utilization factor for tapered traffic distribution are different from those for uniform traffic distribution. To evaluate spectrum efficiency factor G (erlang/MHz) under practical traffic distribution conditions in a metropolitan area, the following assumptions are imposed.

-Traffic Distribution : $10^{-(r/20)}$ (r is the distance from traffic peak)

-Zone Radius : 1km

Figure 6 shows that spectrum efficiency degrades by using FEC even for BER=10^{-2}. The more the traffic concentrates, the more the spectrum efficiency degrades.

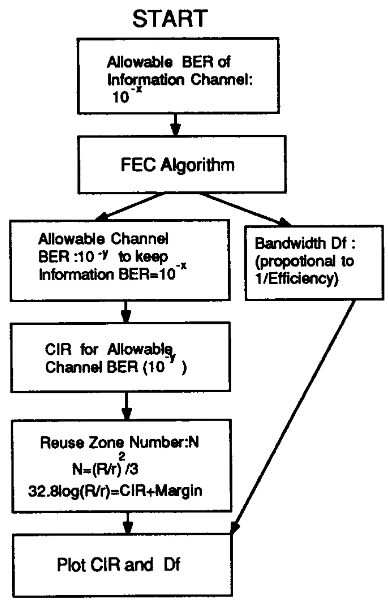

Fig. 4 Flow Chart for Estimating Spectrum Efficiency

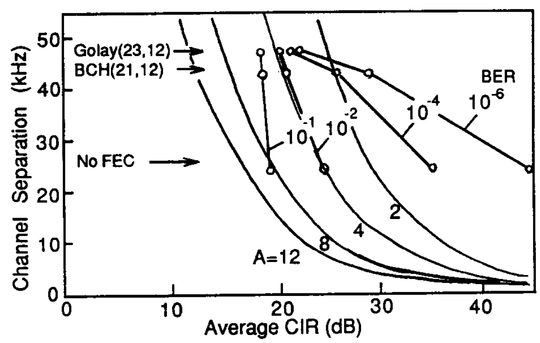

Fig.5 Spectrum Efficiency with FEC
(Uniform Traffic Distribution)

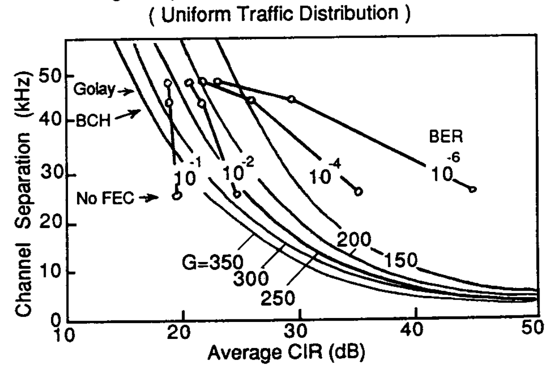

Fig.6 Spectrum Efficiency with FEC
(Tapered Traffic Distribution :$10^{-r/20}$)

From these results, it must be noted that spectrum efficiency should be taken into account when FEC is applied, especially for voice transmission.

4.3 Study for Mobile ISDN Service Quality:

End-to-end BER quality objectives are defined according to CCITT recommendation G.821, for ISDN 64kb/s bearer service as shown in table 1.

It is difficult for mobile radio systems to fulfill these objectives. So, it is important to discuss what quality grades are necessary and what kind of quality objectives are practical for mobile ISDN services, considering spectrum efficiency and economy.

Simulations were carried out in order to estimate the radio link parameter (i.e. the required average CNR) which is necessary to fulfill bearer services quality. Several assumptions are imposed to evaluate the average CNR roughly. They are,

(1)Time domain outage probability can be replaced by the outage probability of area.

(2)RF signal amplitude variation in receiving during one second is Rayleigh distributed.

(3)Outage probability of %DM is replaced by the average outage probability that BER exceeds 10^{-6} under Rayleigh (short term) and log normal (long term: σ=6dB) distribution.

(4)BER for 1 radio link is assumed to be limited within 1/4 of end-to-end quality mentioned in table 1.

(5)Information bit rate is assumed to be 8kb/s.

(6)Bit error occurs at random.

Figure 7 and 8 show static and short term averaged BERs, respectively. Figure 9 shows the relations between outage probability and average CNR margin. From (1), (2), (4) and (5), short term average BER should be lower than 1.25×10^{-4} for 98% area (%ES), and be lower than 1×10^{-3} for 99.95% area (%SES). From (1), (3) and (4), BER should be lower than 1×10^{-6} for 97.5% area (%DM). Table 2 shows the

Table 1 ISDN Service Quality Objectives

Item		Outage time
%ES	More than 1 bit error during 1 second	< 8%
%DM	Average BER during 1 minute > 10^{-6}	<10%
%SES	Average BER during 1 second > 10^{-3}	< 0.2%

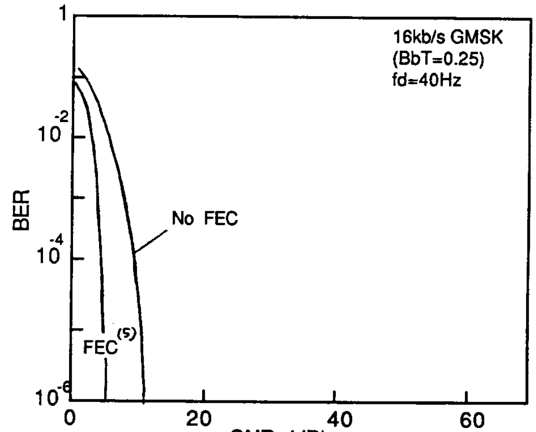

Fig.7 Static BER Characteristics

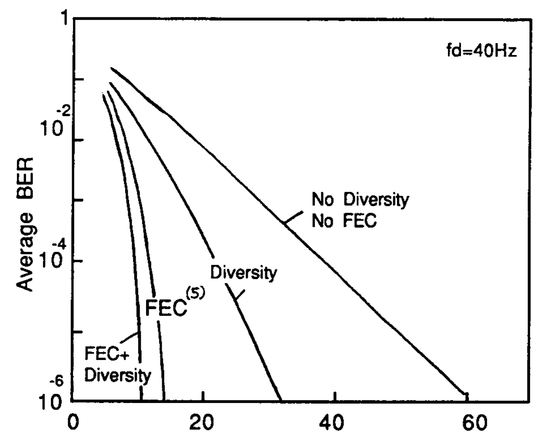

Fig.8 Short Term Average BER under Fading

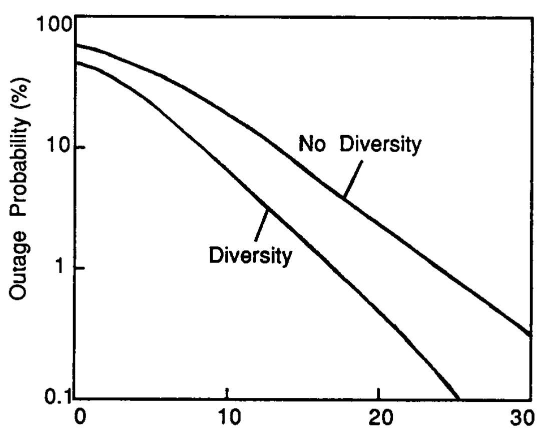

Fig.9 Outage Probability vs. CNR Margin

required average CNR. %SES requires the highest quality so that average CNR value is determined by %SES. Regarding voice service, greater than 7dB output power is necessary to offer ISDN bearer service even if diversity and FEC are applied. If diversity were not used, an additional 5dB margin should be added. If FEC were not used, an additional 9dB margin would be necessary.

These results show that not only are diversity and FEC quite effective for ISDN bearer services, but also increased output power might be necessary to fulfill ISDN service quality requirements. CIR should also be maintained as high level as CNR. Frequency reuse distance may increase as compared with that of voice service.

Quality estimations were based on the propagation model for usual analog mobile systems. CNR margin seems to be underestimated. More detailed investigation based on practical digital transmission field data is necessary to define mobile ISDN service quality objectives.

5. CONCLUSION

SCPC/FDMA digital mobile systems channel structures for ISDN services are proposed. Radio channels are classified into three types : an information channel, control channel and packet channel. ISDN bearer and packet services can be offered with efficient spectrum utilization by using these channels .

It should be noted that FEC has an effect on spectrum efficiency. For high quality services, FEC is effective both in reliability and spectrum efficiency. However, FEC is not advantageous for spectrum efficiency in relatively lower quality services such as voice.

To offer ISDN bearer services, diversity and FEC may be indispensable. Moreover, much more RF output power will be necessary as compared with voice service. Further study is necessary for mobile ISDN service quality design objectives.

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Table 2 An Example of the Objectives for Allowed CNR of ISDN Bearer Service

	%ES	%DM	%SES	(Voice)
BER	1.25×10^{-4}	10^{-6}	10^{-3}	10^{-2}
Outage Probability	2%	2.5%	0.05%	5%
No Diversity No FEC	48dB	32dB	49dB	27dB
Diversity	31dB	27dB	36dB	20dB
FEC	21dB	26dB	32dB	—
FEC + Diversity	18dB	20dB	27dB	—

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SPECTRAL EFFICIENCY IMPROVEMENT TECHNIQUES FOR NONLINEARLY AMPLIFIED MOBILE RADIO SYSTEMS

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Abstract

We study 'double-jump' Nyquist channel filters which lead to reduced spectral spreading and an improved Bit Error Rate (BER) performance in nonlinearly amplified (NLA) mobile radio systems. In linearly amplified systems our filter retains the InterSymbol-Interferemce free properties of Nyquist filters and has the advantage of lower overshoot than the conventional Raised Cosine (RCS) filters. This Reduced Overshoot Power (ROP) filter leads to a 1dB system gain over conventional RCS filtered 16-QAM systems in NLA channels.

1. Introduction

Efficient bandwidth utilization is very important in mobile radio channels because of the steadily increasing demand for mobile radio services. However, in mobile radio channels, a filtered data signal having a compact spectrum may have to be amplified by a nonlinear class-C power amplifier, in order to achieve a better power efficiency. In this case spectral restoration occurs [1]. Althourgh Nyquist raised cosine filters can limit the bandwidth of digital signal in linearly amplified channels, they may not be desirable for the digital mobile system, because they lead to more envelope fluctuation and more senstivity to the nonlinearities of mobile radio channels. In this paper, we study 'double-jump' Nyquist channel filters which lead to reduced spectral spreading and an improved bit error rate performance in nonlinearly amplified digital mobile radio systems.

To improve the digital signal transmission performance in linear channels, Franks [2] proposed and studied, and later Tugbay [3], and Oshita [4] also investigated double-jump filters which have a specified excess bandwidth in the frequency domain. They all optimizesd their filters based on their own optimization criteria respectively. Franks selected the rolloff shape to minimize the sensitivity to phase jitter, Tugbay to maximize the fraction of signal energy in a given time interval, and Oshita to minimize bit error rate under timing error uncertainties. Here, we modify their filters to apply them to nonlinearly amplified channels. Our filter is simillar to Franks' filter, but its jump step is different from his filter's. The jump step factor C of our filter is optimized to minimize the average overshoot power of the impulse response in the time domain, i.e., to minimize the envelope fluctuation of filtered digital signals, so that a digital signal filtered by our modification of the Franks' filter becomes less sensitive to nonlinear distortion of mobile radio channels. Because of its reduced overshoot power (ROP), the new filter leads to a better BER performance and less spectral spreading than the conventional raised cosine filter in nonlinearly amplified channels.

2. ROP Filter and its Optimization

The reduced overshoot power (ROP) filter we propose has a linear-decaying slope combined with double-jump transfer function in the frequency domain [2]. The filter's spectrum and its corresponding impulse response are written as follows:

$$H(f) = \begin{cases} 1 & |f| \leq \frac{1-\alpha}{2T_s} \\ \frac{1}{2} - \frac{T_s(1-C)}{2\alpha}\left(f - \frac{1}{2T_s}\right) & \frac{1-\alpha}{2T_s} < |f| \leq \frac{1+\alpha}{2T_s} \\ 0 & \text{elsewhere} \end{cases} \quad (1)$$

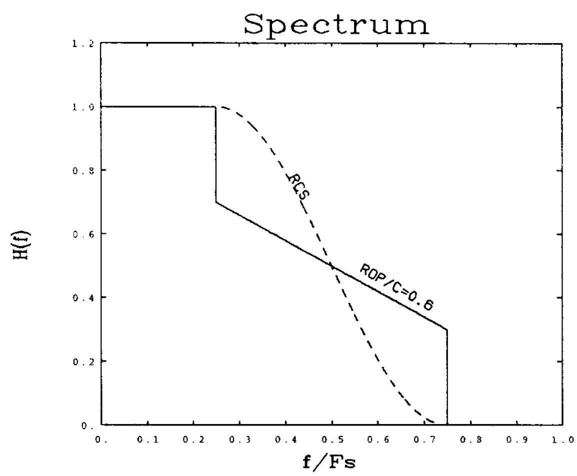
$$h(t) = \frac{C}{T_s}\frac{sin(\frac{\pi t}{T_s})}{\frac{\pi t}{T_s}}\left[cos\left(\frac{\alpha\pi t}{T_s}\right) + \frac{(1-C)}{C}\frac{sin(\frac{\alpha\pi t}{T_s})}{\frac{\alpha\pi t}{T_s}}\right] \quad (2)$$

where

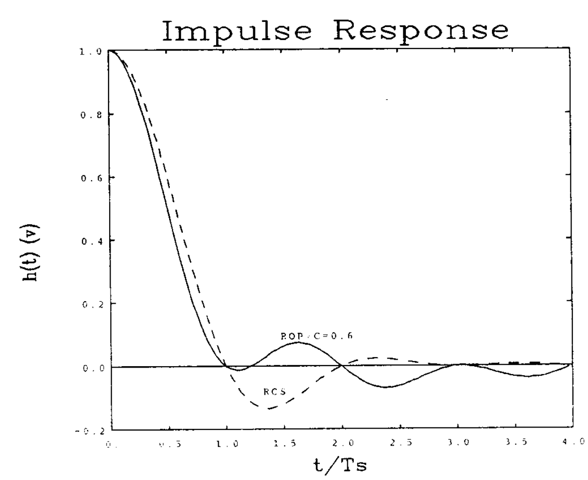
T_s: the symbol duration;

$\frac{1}{2T_s}$: Nyquist band (minimum theoretical band [7]);
α: rolloff factor or excess bandwidth factor above the minimum Nyquist band.

The parameter C is defined as the jump depth of the spectrum. The impulse response satisfies the Nyquist I criterion, i.e., $h(kT_s) = 0$ for $k = \pm 1, \pm 2$ .... The amplitude transfer function and the impulse response of the ROP filter and raised-cosine (RCS) filter are shown in Fig.1 (a) and (b) respectively. Linear phase is assumed throughout our studying.



(a)



(b)

Fig.1 ROP filter with $C = 0.6$ and RCS filter both in frequency (a) and time (b) domain, with $\alpha = 0.5$.

The Nyquist shaping filter always operates in a limited band width, and its impulse response spreads over many symbol instants. Therefore, when a random data sequence enters such a filter, an overshoot must take place which leads to increased spectral spreading in nonlinearly amplified systems. *Peak factor* [4] is a relative parameter of overshoot and is indirectly related to spectral spreading. Here, we introduce a new parameter, the *overshoot factor* F_{over} which is more directly related to spectral spreading than the peak factor.

In our analysis, we take a 16-QAM signal as an example, but the results could be extended to other modulation schemes. A 16-QAM signal is presented in a complex format :

$$s(t) = \sum_{k=-\infty}^{+\infty} a_k h(t - kT_s) + jb_k h(t - kT_s) \qquad (3)$$

where

$a_k, b_k = \pm 1, \pm 3$ for 16-QAM systems.

And its overshoot signal is defined as follows:

$$S_{over} = \begin{cases} |s(t)| - m & t \subset \{x : |s(x)| > m\} \\ 0 & elsewhere \end{cases} \qquad (4)$$

where

$m = 3\sqrt{2}$ for 16-QAM systems.

Fig.2 (a) and (b) show 16-QAM overshoot signal diagrams filtered by RCS and ROP filters respectively. If we define P_s as the average power of a 16-QAM signal and P_{over} as its average overshoot power, then the overshoot factor mentioned above is calculated as follows:

$$F_{over} = 10log\left(\frac{P_{over}}{P_s}\right) \qquad (5)$$

The overshoot factor is a function not only of the rolloff factor α but also of the jump depth factor C. For our ROP filter, we optimized the factor C by minimizing the overshoot factor F_{over} versus different α by means of numerical computations. The relationship curve is shown in Fig.3. To compare ROP filters with conventional raised cosine filters, we show their overshoot power factors versus different α in Fig.4. We note that the ROP filter has a 6.4dB lower F_{over} than the square root Nyquist RCS filter and 4.1dB lower than the Nyquist RCS filter at $\alpha = 0.5$.

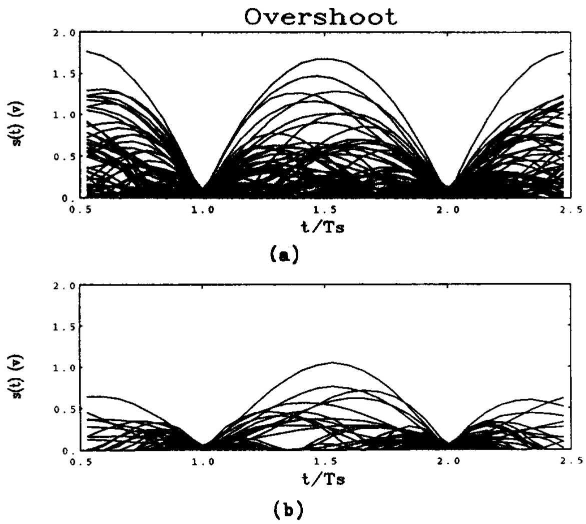


Fig.2 Overshoot diagrams of 16-QAM signals, in part (a), filtered by RCS filter, and in part (b), filtered by ROP filter.

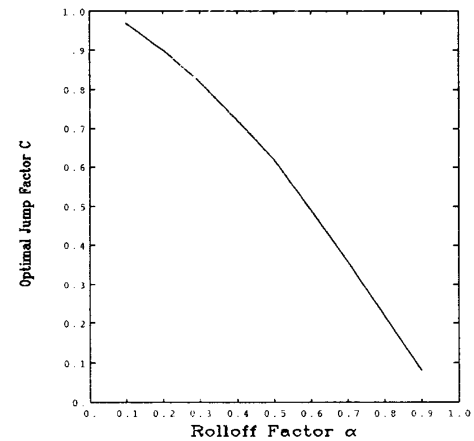


Fig.3 Optimized jump factor C of ROP filter versus different rolloff factor α.

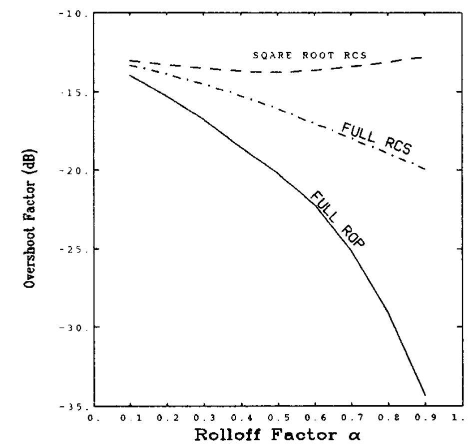


Fig.4 Overshoot power factors F_{over} of three types of filters versus different rolloff factor α.

3. Nonlinearly Amplified Channel Model

A simplified block diagram of the equivalent baseband model of a nonlinear channel model is shown in Fig.5. In a linear channel, the best theoretical combination of shaping filters is when both the transmit filter and the receive filter have the shape of square root of Nyquist shaping filter characteristics. The only exception is that in the transmit filter an $\frac{x}{sin(X)}$ shaping section is added [7]. However, in a nonlinear channel, this matched filtering is not an optimal selection because of ISI and quadrature crosstalk [6]. The Nyquist shaping filter may be locaded at the transmit side or receive side only, i.e., it is not equally divided.

In this paper, two types of transmission models are considered. Referring to Fig.5, for the convetional 16-QAM system, the square root of Nyquist raised cosine filters ($\alpha = 0.5$) are used in transmitter (with $\frac{x}{sin(x)}$ aperture equalizer) and receiver. And for our new filtering 16-QAM system, the full Nyquist ROP filters ($\alpha = 0.5$ and $C = 0.6$) are used in the transmitter (with $\frac{x}{sin(x)}$ aperture equalizer), and a brick wall filter (cut off frequency of $f_N(1+\alpha)$) is used in the receiver. Because of the mismatch, the ROP filtering 16-QAM systems the in linearly amplified channels are 0.7dB lower to RCS filtering systems. See the BER performance shown in Fig.11.

4. Performances of Nonlinaerly Amplified 16-QAM System Filtered by ROP Filters

The variable of our analysis is the output power of the outermost signaling states defined as follows:

$$P_{out} = km^2 \tag{6}$$

where m is defined in (4), and k is a constant defined by the choise of OBO. This is the peak output power to the HPA taken over the 16-QAM constellation , and it also corresponds to the peak power at the detector at sample times.

The assumptions used in our computer simulation are

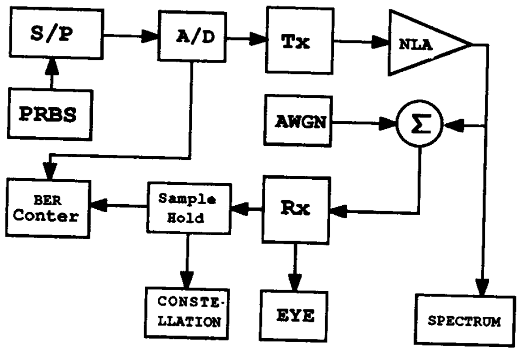


Fig.5 Block diagrm of computer simulattion model — equivalent baseband model of 16-QAM modem is illustrated.

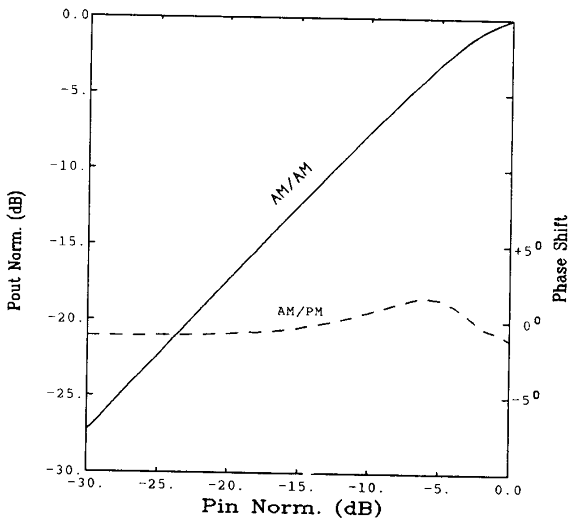


Fig.6 AM/AM and AM/PM characteristics of the FET amplifier.

summarized as follows:

a. As an illustrative example, a 16-QAM system with f_s of 1 MBaud (or $f_b = 4Mbit/s$) is used.

b. The nonlinearly amplified channel has an amplifier such as the Fujitsu linearized solid state GaAs FET amplifier. Its characteristics of AM/AM and AM/PM conversions are shown in Fig.6 [9].

c. As an illustrative rolloff factor, α of 0.5 is used in all simulations.

4.1 Power Spectrum of the ROP Filter in a Nonlinear Channel

The spectrum of 16-QAM signals filtered respectively by ROP filters and square root of Nyquist raised-cosine filters are shown in Fig.7. In part (a), a linearly amplified channel is considered, and In part (b), a nonlinearly amplified channel is considered. Note that in the comparison with RCS filter, ROP filter spectrum has lower spectral spreading in the nonlinearly amplified channels with the same OBO of 2dB, and that an 2dB OBO improvement of FET amplifier is shown in the same spectrum spreading. If the power spectral dencity of the spread out of band signal beyond the cut off frequecy must be 35dB less than the inband spectrum, , then the HPA amplifier can operate at an OBO of 2dB in a ROP filtering 16-QAM system, but it has to operate at an OBO of 4dB in a conventional RCS filtering 16-QAM systtem.

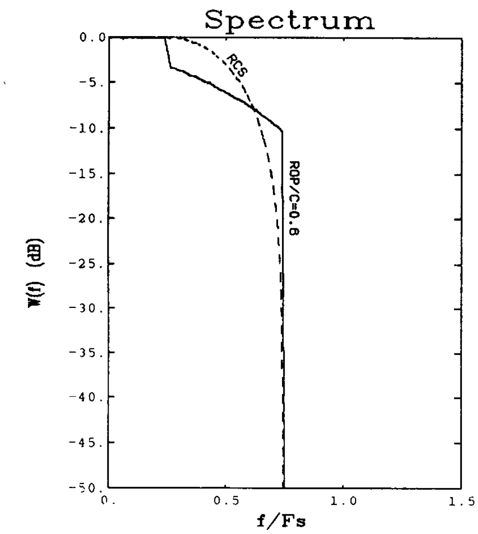
4.2 The Eye Diagrams and Constellations of 16-QAM Systems

The 16-QAM eye diagrams of I-channel in linear channels are shown in Fig.8. Fig.8 (a) shows the conventional raised cosine filter's response, and Fig.8 (b) the ROP filter's. Comparing the Fig.8 (a) and (b), we note that the 16-QAM signal filtered by ROP filter has less overshoot and an increased horizontal eye opening. The 16-QAM eye diagrams in the nonlinearly amplified channel are illustrated in Fig.9. In part (a), the signal is filtered by the raised cosine filter, and the FET amplifier in the nonlinear channel operates at an OBO of 4dB. In the part (b), the signal is filtered by the ROP filter, and the FET amplifier in the nonlinear channel operates at an OBO of 2dB. The 16-QAM constellations corresponding to the environments in the Fig.9 (a) and (b) are illustrated in Fig.10 (a) and (b) respectively.

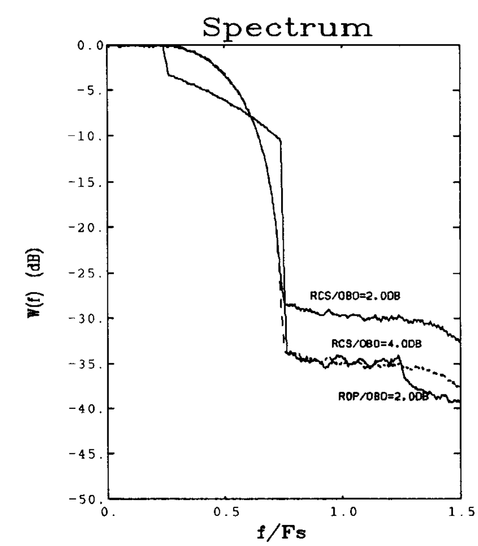
4.3 Bit Error Rate of 16-QAM Signal Filtered by ROP Filter

P(e) performance results of 16-QAM signal filtered by ROP filter in linear channels are shown in Fig.11. We found that 16-QAM signal filtered by a ROP transmit filter ($\alpha = 0.5, C = 0.6$) and a brick wall receive LPF (cut off frequency of $f_N(1+\alpha)$ has about 0.7dB degradation of Carrier to Noise Rate (CNR) at P(e)=10^{-6} from theoretical performance.

Fig.12 shows the BER performance of 16-QAM system filtered respectively by the ROP filter with an OBO of 2dB of HPA and the RCS filter with an OBO of 4dB of HPA. In ROP filtered system a 1dB degradation of CNR from the RCS filtered system is found. However, considering the reduced OBO of 2dB, we still get 1dB system gain from ROP filtered systems. The SNR degradations of RCS and ROP filtering 16-QAM systems versus the different OBO is illustrated in Fig.13. Referring to Fig.13, when the OBO of the FET amplifier is less than 4dB, the ROP filtering system indicates 0.5dB more improvement in



(a)



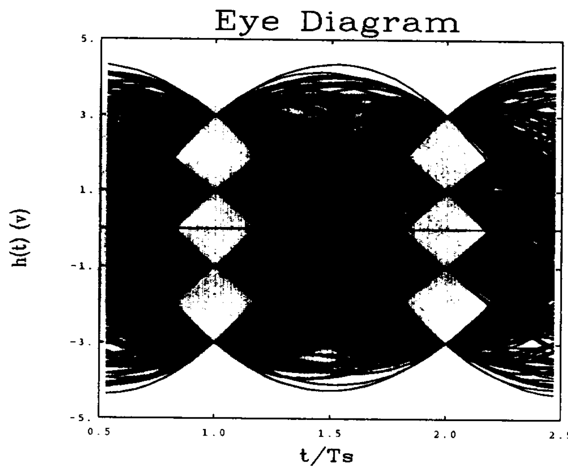
(b)

Fig.7 Power spectrum densities of 16-QAM signals filtered by ROP filter and RCS filter respectively. A linear channel (a) and a nonlinear channel (b) are illustrated.

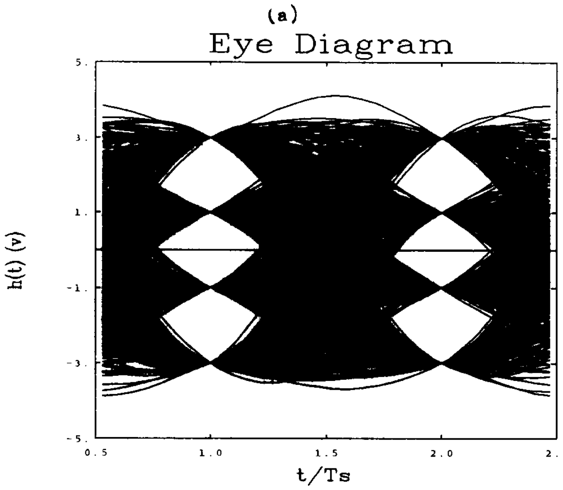
P(e) performance than the RCS filtering system. However, when HPA operates nearly linearly, that is, the OBO is much enough, the square root Nyquist raised cosine filter has a better performance. This is because for the optimal matched filtering in the linear channel. In this studying the adjacent channel interference (ACI) caused degradation was not fallen into account. Combined with ACI our propsed ROP filtering system is expected to lead to some of limit system gains.

5. Conclusions

A reduced overshoot power filtering strategy was proposed and analyzed. Without need for additional spec-



(a)



(b)

Fig.8 Eye diagrams of 16-QAM signals filtered by RCS (a) and ROP (b) filters in the linearly amplified channels.

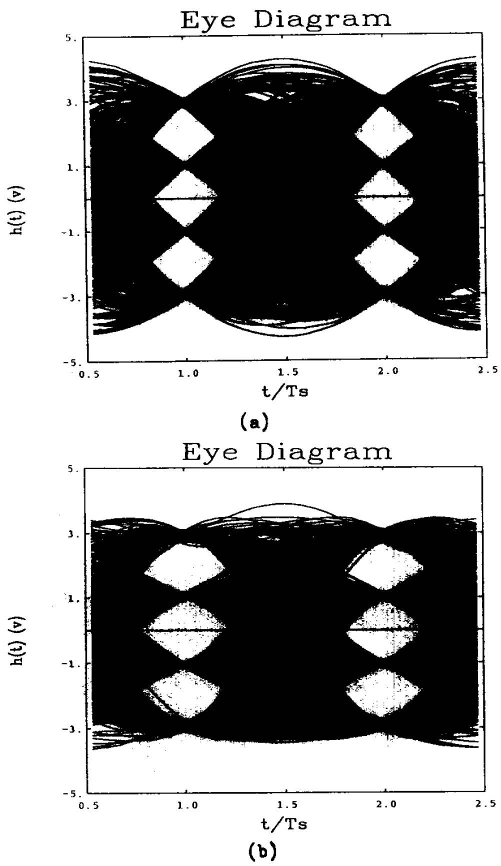


Fig.9 Eye diagrams of 16-QAM signals in the nonlinearly amplified channels, which are filtered by RCS (a) with an OBO 0f 4 dB and ROP (b) filters with an OBO of 2dB.

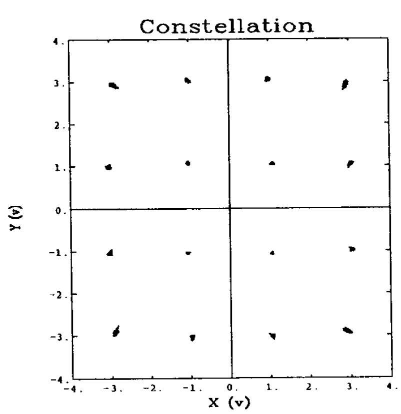


Fig.10 (a)

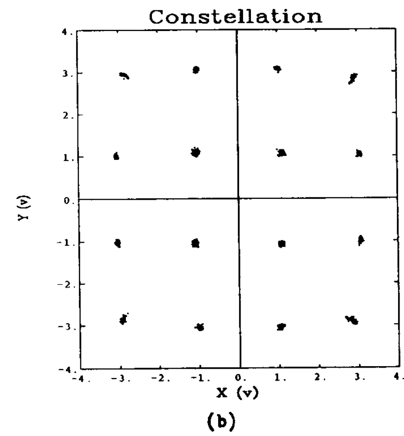


Fig.10 Constellation diagrams of 16-QAM signals in the nonlinearly amplified channels which are filtered by RCS (a) with an OBO 0f 4dB and ROP (b) filters with an OBO of 2dB.

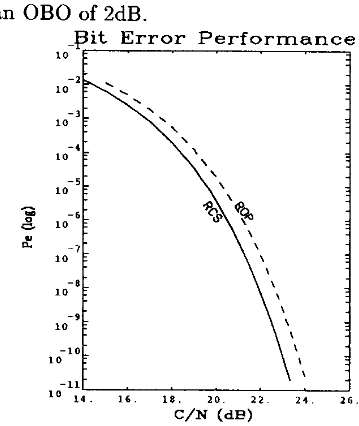


Fig.11 P(e) performances of 16-QAM filtered by RCS and ROP filters in linearly amplified channels.

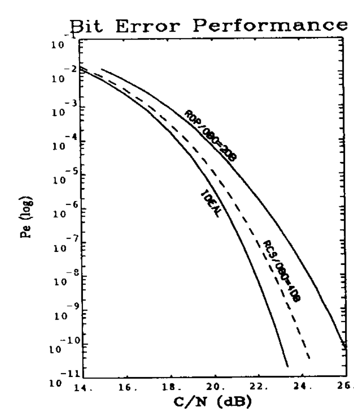


Fig.12 P(e) perpormances of 16-QAM filtered by RCS and ROP filters in nonlinearly amplified channels.

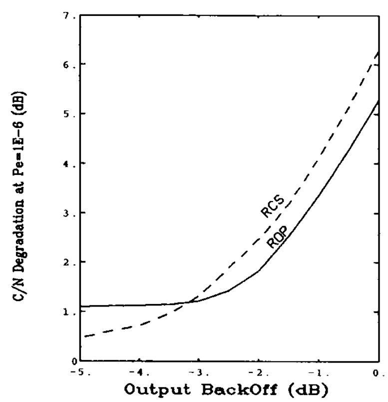


Fig.13 SNR degradation of 16-QAM filtered by RCS and ROP filters in nonlinearly amplified channels versus different rolloff factor α.

tral shaping filters after nonlinear amplification, the 16-QAM signal filtered by the ROP filter has smaller spectral spreading than the conventional raised cosine filter. Our study results reveal that when the spectral spreading is below 35dB, the OBO of a linearized FET amplifier can be reduced by 2dB through application of a ROP filter and that a 1dB system gain can be achieved considered the reduced OBO and the CNR degradation at $P(e)=10^{-6}$.

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VIABILITY OF PACKET RADIO FOR FUTURE MOBILE COMMUNICATIONS

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ABSTRACT

Future Public Land Mobile Communication systems will be characterized by digital transmission of both voice and data services, flexibility, portable outdoor and indoor applications and very high user penetrations. These large capacity systems would need large amont of spectrum, a precious resource. Efficient access techniques are therefore required to best utilise the available spectrum along with efficient network techniques such as microcellular structures.

This paper discusses the viability of packet radio access techniques for Public Land Mobile Systems. A comparative analysis is developed which compares the efficiency of packet radio techniques with the efficiency of TDMA. It is shown the efficiency of packet radio depends on the proportion of data and voice users and the access technique used.

INTRODUCTION

Over the last few years several trends in communications such as increasing use of digital technology in both switching and transmission systems, and increasing need for non-voice services such as interactive data transfer,facsimile, file transfer and electronic mail have emerged which will have tremendous impact on the nature of future mobile communications. Most significant however,is the ever increasing number of people wanting portability and flexibility out of their communication devices both indoors and outdoors. This implies future mobile systems will need to have a very large capacity and be capable of supporting non-voice services along with traditional voice telephony placing great demand on the radio spectrum, a scarce commodity. Therefore, the transmission and network techniques to be used in such systems have to be spectrally efficient. The use of cellular systems and digital transmission techniques is a first step towards this end. Microcellular techniques have been suggested wherein the radio frequency is reused more frequently over a geographical area by employing much smaller cells than in conventional cellular radio leading to improved spectral efficiency.

Along with these techniques, the spectrum utilization can be enhanced using efficient channel access methods of which packet radio techniques appear to be most useful given the integrated nature of services being proposed for the future mobile communications. Apart from improved efficiency, packet radio techniques offer flexibility, integrated transmission, variable bit rate transmission and possible simplicity of handover. In packet radio, the digitized information along with essential overhead bits is sent over a common radio channel in form of small packets by use of suitable access protocol. At the receive end, the packets are buffered and sequenced if necessary, to reconstruct the information. Packet radio techniques exploit the bursty nature of most information to efficiently and dynamically multiplex large number of users on a single wideband channel. This paper describes a study undertaken to investigate the efficiency of Packet Radio when used for Public Mobile Communication in the integrated scenario, described earlier, leading to a Public Packet Radio Network (PPRN).

PACKET RADIO OVERVIEW

In general,each packet is composed of a preamble, a header field, information field, an error detection field and an end flag (Fig.[1]). All except the information field constitute the overhead of a packet and should be kept small relative to the information field for high framing efficiency. The preamble containing a known sequence of bits is used to extract the timing information and for AGC locking. Normally, the header field contains the following information-

-the address of the source and the address of the destination

-type of the packet. For example, whether its a voice packet or a data packet.

-control information such as routing, length of the packet, delay information or time stamp etc.

The information carried by header is extremely important and it may be necessary to protect it in the radio environment by methods such as forward error correction (FEC). Depending on the type of packet the length of the header varies and is between 30 to 300 bits long (1,2). The error detection field is a cyclic redundancy checksum (CRC), usually 16 bits long and is needed for Automatic Repeat Request (ARQ) for services requiring high error performance. The end flag is about 8 bits long.

The length of a voice packet is a crucial design parameter and usually is a compromise between obtaining a high framing efficiency (long packet) and obtaining small packetization delay and less susceptibility to fading (small packet) (3). A voice packet length of ~ 400 bits with 16 kbit/s speech codecs and at channel rate between 50-500 kbit/s appears satisfactory as most packets will be transmitted in the interfade intervals (3). For data packets packet lengths up to ~ 800 bits are also acceptable as more delay can be tolerated. At higher bit rates, since fading is frequency selective causing delay spread, packet systems would require methods such as diversity, adaptive equalization and coding to combat the effect of delay spread, just as in TDMA

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PUBLIC PACKET RADIO NETWORK

A typical PPRN would consist of one or two (two in case of mobile to mobile call) radio links and a number of fixed high speed cable links between base stations, mobile switching centers and PSN nodes and could extend typically up to 3000 km (Fig. [2]). In this study, 6 cable links and 1 radio link having a total 3000 Km length have been considered. The channel speed for the radio path is 280 kbit/s while the fixed cables support 2 Mbit/s. Connection to a satellite network is not considered.

Network Architecture

Several possibilities exist for the radio link architecture in PPRN. The network can operate in full duplex or in half duplex mode with full duplex implemented either in frequency or in time. In frequency division duplex (FDD) the transmit and receive frequencies are separated by a fixed amount while in time division duplex (TDD) the transmit and receive functions are at same frequency but are separated in time by fixed amount. On the the other hand, half duplex mode is like TDD with no fixed time relationship. Half duplex and TDD would allow use of the full bandwidth and should provide higher channel rates but radio multipath conditions give rise to excessive delay spread limiting the channel rate and thus making these two schemes less attractive than FDD. Also, since radio medium is potentially a broadcast medium it is desirable to have a separate frequency for broadcast to derive full benefits of the broadcast property. Thus if a star topology is used the downlink traffic has a dedicated channel. This broadcast channel can also be used for base station signalling, uplink status flagging (as in some access protocols), supervisory network management etc.

A packet radio network can have star topology or fully connected mesh or even a partial mesh. In the latter case the packet radio terminals backup as repeaters for packets from their connected neighbours thus enabling packet transmission from a source to an unconnected destination. This type of network can extend to a very large distances obviating the need for a cellular structure. However, for voice like traffic such a network would introduce intolerable delays and is not suitable for a PPRN. Thus packet radio with a suitable cellular structure appears to be the proper network architecture for PPRN. The handover function in the cellular PPRN appears to be simpler than in conventional cellular systems as each cell has one single wideband channel instead of a group of channels. Also, the mobiles can themselves implement handover upon losing connectivity with a base station. This distributed handover function can lead to simpler switching requirements at the Mobile Switching Centre if a common logical channel addressing is used in the entire system.

Radio Channel Access in PPRN

The performance of packet radio systems is very much dependent on the multiple access protocol used. Detailed description on multiple access is available in references (4,5). Briefly, multiple access schemes can be broadly classified into three classes. Namely,

- random access
- centrally scheduled
- Fixed assigned

In the random access class, the pure and slotted ALOHA schemes, Carrier Sense Multiple Access (CSMA) and the Busy Tone Multiple Access (BTMA) schemes are applicable to mobile packet radio although performance of CSMA can degrade considerably due to the hidden terminal problem (4,5). The centrally scheduled schemes applicable to packet radio are reservation ALOHA, Token Passing Bus and Polling of which last two are very similar for packet radio because in case of token passing, the token has to be transferred back to a central scheduler after the use as the next user may not be in radio connectivity. The last class, namely the fixed assigned class include the familiar FDMA, TDMA and CDMA. These fixed assigned schemes are contention free but have poor channel utilization if the incoming information is bursty. Moreover, these schemes are very rigid and can not effectively cater for changing number of users and require sufficient orthogonality between the subchannels. Demand assigned schemes can overcome some of these problems. Figure [3] shows the throughput and delay curves for some of these protocols (4,5).

SERVICE REQUIREMENTS AND SERVICE CHARACTERISTICS IN PPRN

Service Requirements For Data And Voice

Data services require virtually an error free channel and BER of the order of E-6 or better are essential for faithful reproduction of data at the receive end. To achieve this, packet systems use ARQ protocol and an acknowledgment scheme. For mobile radio channel FEC with bit interleaving is also usually recommended to combat bursty errors caused by impulse noise, interference and fading irrespective of the access protocol used.

Data services in most cases are not critically affected by delays and moderate delays and delay variability may be tolerable.

Voice on the other hand, is a real time service and has stringent delay requirements. Subjective tests (1) have indicated that if end to end delay over the entire network is less than 250 ms for 99 % of speech packets the user does not perceive any degradation in quality. The variability in delay should also be kept minimum so that an optimum playback time may be selected at the receiver without causing loss of packets due to excessive delay.

The packet loss in the voice systems should be small. In reference (1) it is noted that 5 % packet loss is acceptable without significantly affecting the quality. The packet loss in packet radio systems is due to the following -

- Excessive delay
- Packet collisions if contention access schemes used
- Errored packet due to fading or interference

The packet voice systems do not implement any error detection scheme and ARQ because voice is a real time service and the information due to retransmission will be too late to be useful.

Service Characteristics

Most data services of transaction style are bursty in nature. The average packet generation rate is very low for services such as interactive computer communication, electronic messaging, dispatch and facsimile, with packets arriving in short bursts only. Even speech is bursty although the burstiness is not as high as in data services. The burstiness in speech arises due to the fact that speech is active only 33 % of the duration of a talk, rest of it being pauses between speech and silence while listening. Table [1] shows some of the services, their peak generation rate, average generation rate and burstiness which is defined as the ratio of the first two. It can be seen from the table that for data services burstiness can typically be 100 whereas for voice it is about 2.5 (6).

PERFORMANCE ANALYSIS OF PPRN

In this section the efficiency of Packet Radio Networks is compared with the efficiency of TDMA systems as latter is the current state of the art technology. The ultimate figure of merit is the number of users the system can support per MHz. per square kilometer and is related to the spectral efficiency. For comparative purposes, it is however convenient to define the network efficiency of the network-

$$\eta = \eta_M * \eta_F * \eta_A \qquad \text{--(1)}$$

where, η is the network efficiency, η_M is the multiplexing efficiency, η_F is the framing efficiency and η_A is the access efficiency. These quantities will be defined subsequently. It is assumed that the modulation efficiency and cellular reuse factor are same for both packet and the TDMA systems.

The data packet arrival process can usually be modeled as Poisson distributed. The same is not strictly true for voice packet arrival because packets arrive periodically in a talkspurt and the arrival times have strong correlation (7). However, when there are a large number of active voice sources, the superposition packet arrival process loses much of this correlation and can be approximated by a Poisson distribution (8). Assuming this, let λ_D denote average arrival rate for aggregate data sources and λ_V that for aggregate active voice sources.

$$\lambda_D = N_D * \lambda_{D1} \qquad \lambda_V = N_V * \frac{C}{b} * \alpha$$

Where N_V is the number of active voice sources, N_D the number of active data sources, λ_{D1} the average arrival rate from a single data source, C the speech codec speed, b the number of bits in the packet and α is the voice activity factor. Thus in the integrated situation assuming independent arrivals the aggregate arrival process will also be Poisson with average arrival rate = $\lambda_D + \lambda_V$

The following three cases are considered :-

Case (a): There are equal number of active voice and data sources

Case (b): The integrated mix consists of 80 % active voice sources and 20 % active data sources

Case (c): The integrated mix consists of 20 % active voice sources and 80 % active data sources.

A PPRN will perhaps initially be represented by case (b), settling down to case (a) in future. Case (c) is a representative of a wireless business office.

Multiplexing Efficiency

This quantity represents the number of simultaneous active users possible. In packet radio, this depends on the burstiness of the arrival of the bits and is equal to it, relative to the TDMA system. From table [1] we can infer that typical value of burstiness for data is between 10 and 100 and for voice about 2.5 with standard speech codecs. Following values for multiplexing efficiency are obtained for typical values of data burstiness.

Data Burstiness	10	100
	Multiplexing Efficiency	
Case (a)	3.16	4.88
Case (b)	3.05	3.49
Case (c)	3.53	10.04

For TDMA systems multiplexing efficiency is equal to 1 as the system is designed on the basis of peak arrival rate from individual user.

Framing Efficiency

Framing Efficiency can be defined as the ratio of number of information field bits to the total number of the bits in the packet for packet radio and in the TDMA slot for TDMA systems. For data packets framing efficiency can be increased by having a long information field as packetization and transmit delays are relatively unimportant. However, as discussed in ref.(3) radio fading will restrict the packet length to ~ 800 bits. The packet overheads would normally be about 80-100 bits if no routing information is required as is the case in PPRN, giving a framing efficiency of 80 to 90 % . Similar framing efficiencies are obtainable in case of voice packets also if abbreviated addresses using logical channel address, exclusion of CRC bits (as error detection is not required) and end flag are implemented.

TDMA system also do not have 100 % framing efficiency because of need to provide guard bands at the boundaries of slots and preamble in each slot. However framing efficiency upto 90% are obtainable. In this paper, this value will be taken for TDMA and 80 % for packet radio systems.

Access Efficiency

Access efficiency is measured in terms of the throughput available from the access protocol used. TDMA systems being circuit switched and contention free have a throughput of 1 at a fixed delay assuming that errors due to fading, noise and interference are kept small by suitable system design. Under the same assumptions, packet radio systems have variable delay which increases with increasing throughput. In an integrated packet network, the delay criterion for voice packets along with the voice packet loss due to collisions (if any) will decide the maximum throughput available. As mentioned earlier 99 % of voice packets should have total delay less than 250 ms. With reference to the reference configuration of figure [2]

Total delay D =

packetization delay D_P + transmit delay D_T
+ switching delay D_S + access delay D_A
+ depacketization and buffering delay D_B
+ propagation delay across the network D_N

Of these, all except access delay can be taken as constant. The packetization delay with 16 kbit/s speech codecs and 480 bits packet length is 30 ms and the transmit delay, D_T is 1.71 ms @ 0.28 Mbit/s channel rate. Switching architectures are available which introduce 1 mS or less delay and with 6 nodes in the reference configuration 6 ms would be required for switching. If we assume that packet processing and buffering is done only at the destination then about 10 ms should be budgeted for D_B (9). The end to end propagation delay over a length of 3000 kms. of cable is ~ 20 ms. This then leaves 99% of the access delay to be less than 182 ms. i.e.,

$$D_{99} = 68 \text{ ms} + D_A(99\%) < 250 \text{ ms}$$

Each fixed link of the PPRN can be modelled as a M/M/1 queue. The delay distribution over 6 such fixed links connected in tandem can be described by a 6 stage Erlang distribution having density $f_c(d)$

$$f_c(d) = \frac{\lambda_c^6 d^5 e^{-\lambda_c d}}{5!}$$

where λ_c is the Erlang parameter and $6/\lambda_c$ its expected value.

Denoting the delay density over the radio path by $f_R(d)$ and assuming independence of the two probability densities the overall delay density is given by,

$$f(d) = f_R(d) * f_c(d)$$

The average delay in a n-stage tandem fixed link is a function of system loading and is given by (9)

$$D_C = \frac{b * n}{[R * (1-\rho)]}$$

Where b is the packet length, ρ the channel utilization and R its speed. At loading of $\rho = 0.7$, D_C is = 7.4 ms. Higher loading than this can not be accepted as overall 99 % delay may become more than the design value. Now the 99% delay distribution of overall delay is given by

$$\int_0^{D_A(99\%)} f(y).dy = 0.99 \qquad \text{---- (2)}$$

Analytic expressions for delay distribution and density for most of the radio link access protocols considered are not obtainable. However, in some cases approximate expressions for average delay and delay variance are available. For instance, in case of CSMA type of protocols the average delay and delay variance can be written as (10)

$$\text{mean} = \frac{(r+1)}{2p}$$

$$\text{coefficient of variation} = \left[\frac{p\,(r-1)}{3\,(r+1)} + q\right]$$

where, q= 1-p, r = rescheduling delay range, p = S/G, S = throughput and G = offered channel traffic.

These results are based on CSMA models which include packet retransmissions due to either channel being busy or as a result of packet collisions. As mentioned earlier, the packet voice tranmission protocol does not have any retransmission if packets collide. Thus the estimation of overall delay obtained from the following delay analysis which uses above moments will be worse than the value in actual practice.

Using the moments described above for different values of the throughput, a suitable distribution can be fitted for the overall delay. In this analysis, the Weibull distribution given by

$$F(X > x) = \exp(-[(ax)^{**}b])$$

is used. Here a and b are the two parameters found from the moment matching method.

Table [2] shows the average delay, delay variance and the parameters a and b of the Weibull distribution for BTMA protocol for 1 km cell radius and at a bit rate of 280 kbit/s. Using this distribution, equation (2) can be solved for 99 % delay for various values of the throughput. The last column in the table [2] lists these for BTMA. As can be seen throughput upto 0.5 are achievable keeping the 99% delay below 182 ms. If the cell radius is increased to 3 Km, bit rates upto 1 Mbit/s would be required to obtain same value of the throughput for same delay objective as can be seen from table [2a]. This is because the delay for these packet protocols is sensitive to increase in the propagation delay increase requiring a reduction in the transmit time. However, for achieving high throughput from these protocols, the ratio of propagation delay to transmit time should not be more than 0.05.

In case of contention protocols another limit is set by the packet loss due to collisions. It is possible to work out average collision rate for BTMA and CSMA using expressions for throughput and offered traffic. These are

$$1-\text{Exp}(-\nu m[0,T]) * 100 \quad \text{(BTMA)}$$

$$[(H-S)/H] * 100 \quad \text{(CSMA)}$$

where,

$\nu = G/D$

m[.] is a function of inhomogeneous arrival rate

H = Actual rate of transmission per transmit time

Table [3] gives the average packet loss for CSMA and BTMA. It is should be noted that this collision rate is for the aggregate users. It is clearly seen that to meet the loss objective of < 5% the throughput of BTMA is limited to 0.5 . The stability considerations also prohibit higher throughputs than this value. Capture effects existing in packet systems due to differing arrival times or differing signal strengths of the colliding packets would however, improve throughput further. Although collisions are not a problem in token bus and reservation ALOHA protocols delays criterion limits the throughput in these cases also to about 0.5.

The network efficiency can now be calculated using the values of η_M, η_F and η_A in equation (1). The network efficiency of TDMA scheme is about 0.9. Thus the efficiency advantage of packet radio over TDMA can be evaluated as

Efficiency advantage,

$$\text{E.A.} = \frac{\text{Network efficiency of packet radio}}{\text{Network efficiency of TDMA}}$$

RESULTS

The relative figure of merit, the efficiency advantage, was calculated for the three cases mentioned above. The results are plotted in figures [4], [5] and [6]. A range of throughput values and multiplexing efficiencies have been used to show the regions for which packet radio will be more efficient than

TDMA. For values of efficiency advantage less than 1, TDMA is more efficient than packet radio. This situation can occur if voice burstiness is less than 2 and proportion of data users is small, as shown in figure [5]. But if voice burstiness is 2 or better, packet radio shows considerable increase in efficiency as the proportion of data users increases and should be preferred. In this situation, even in case (b) packet radio shows an increased efficiency albeit small. The efficiency advantage in packet radio essentially stems from its ability to dynamically use the silence periods in a voice call and the burstiness in the data packet arrival for other users. However, the limitation on efficiency advantage comes about due to the limits on access efficiency imposed by packet delay, packet loss and protocol stability requirements. While stability of the protocol can be improved by flow control means, better access protocols are required giving low delay and negligible packet loss due to collisions. Figure [6] shows the region where packet systems would be required to operate for a given E.A. Packet radio would perform better than TDMA in the shaded region.

CONCLUSIONS

In this paper, application of packet radio to public land mobile radio and indoor wireless communications were discussed. Spectral efficiency has been used as the figure of merit for comparison of packet radio techniques and more conventional digital transmission techniques. This study has shown the limit on achievable access efficiency in packet radio to be 0.5. TDMA, on the other hand, is limited in multiplexing efficiency. Given the limitation on the access efficiency, the relative performance of packet radio depends on the burstiness of the information. In the integrated voice and data environment packet radio would be viable and would perform better than TDMA if the proportion of data users is high and / or the burstiness of voice is 2 or better. For this value, packet radio performs better when the proportion of data users is about 40-50% for low data burstiness and about 20% for high data burstiness. If the voice burstiness is less than 2, it would be advantageous to use TDMA. The results in figure [6] show the regions where packet techniques are superior to conventional TDMA techniques and vice versa.

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PREAMBLE	HEADER	INFORMATION FIELD	CRC	FLAG
16 BITS	~88 BITS	~1000 BITS	16- 32 BITS	8 BITS

TYPICAL DATA PACKET

PREAMBLE	HEADER	INFORMATION FIELD	CRC
16 BITS	~ 40 BITS	~ 400 - 500 BITS	8 BITS

TYPICAL VOICE PACKET

Figure 1. Packet Structure

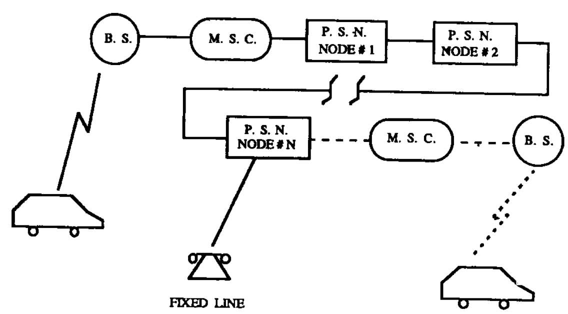

Figure 2. Public Packet Radio Network

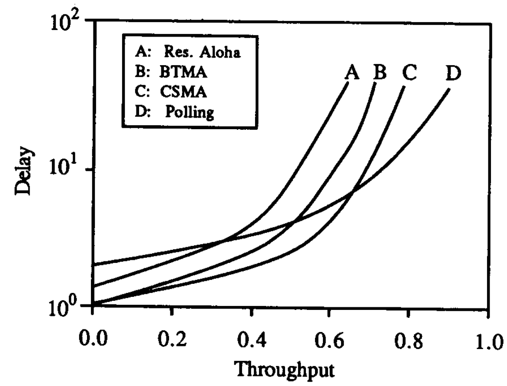

Fig. 3 Performance Of Some Packet Radio Access Protocols (Ref. 4)

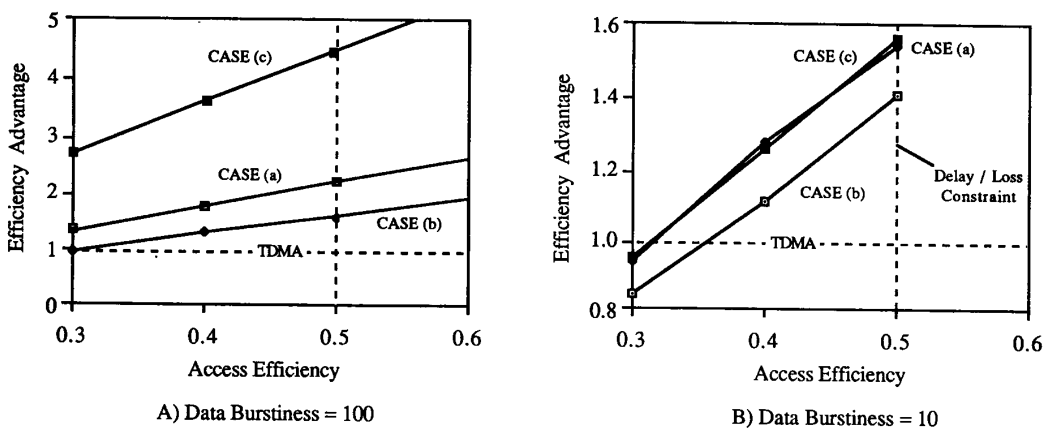

Fig 4: Efficiency Advantage of Packet Radio vs. Access Efficiency

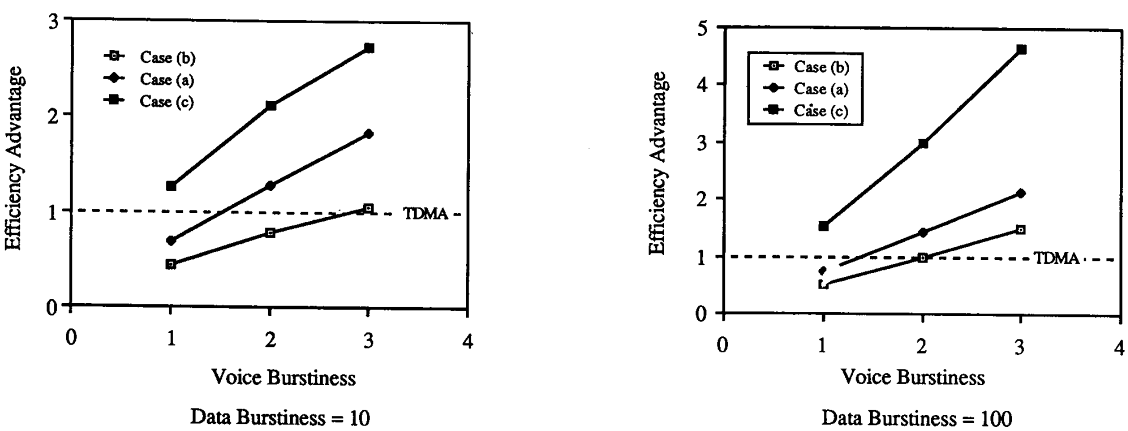

Figure 5. Sensitivity To Voice Burstiness

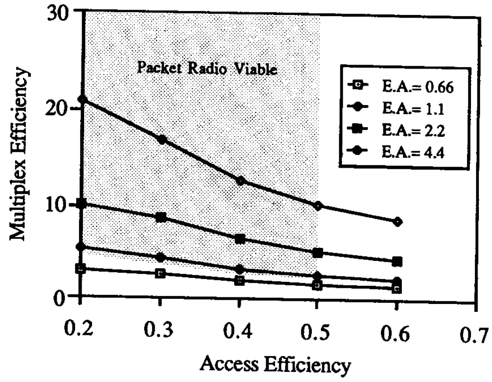

Figure 6. Operating Region For Packet Radio

TABLE 1 : SERVICE CHARACTERISTICS

Service	Peak Arr. rate (kbit/s)	Average Arr. rate (kbit/s)	Burstiness
Interactive terminal to computer	.1	.001	100
Computer to terminal	10	.01	1000
Remote job entry	10	.1	100
Computer to computer	1000	10	100
Digitized speech	64,32,16	25.6,12.8,6.4	2.5
Image video	64-2000	6.4-200	10
Video	140,000	28,000	5

TABLE 2 ACCESS DELAY FOR PPRN WITH BTMA PROTOCOL

Through -put	Mean delay mS	Coeff. of variation	$a * 10^{+3}$	b	99 % delay mS
At 280 kbit/s					
0.1	3.95	1.6	12.19	0.224	74.0
0.2	4.52	1.45	10.65	0.224	85.2
0.3	5.39	1.29	8.93	0.224	101.5
0.4	5.93	1.23	8.12	0.224	111.7
0.5	7.22	1.16	6.67	0.224	136.0
0.6	11.09	1.01	4.34	0.224	208.0
At 1 Mbit/s					
0.1	5.24	1.278	11.77	0.217	95.0
0.2	6.03	1.171	10.23	0.217	109.9
0.3	7.26	1.085	8.50	0.217	132.32
0.4	8.02	1.047	7.69	0.217	146.17
0.5	9.825	0.994	6.28	0.217	179.07
0.6	12.24	0.821	5.03	0.217	223.09

Table 3 : AVERAGE PACKET LOSS DUE TO COLLISIONS

Throughput	CSMA %	BTMA %
0.1	0.29	0.30
0.2	0.56	0.86
0.3	0.93	1.59
0.4	1.38	2.36
0.5	2.21	3.66
0.6	3.15	6.92

SPECTRUM EFFICIENCY AND DIGITAL CELLULAR
by

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A spectrum efficiency formula for cellular systems has been derived and used to compare digital and analog systems. Since the digital system is always less susceptible to interference, the tolerable interference level can be higher, in other words, the co-channel reuse distance can be less. This less reuse distance proves that digital cellular is a spectrum efficient system.

In order to take advantage of digital nature and implement digital systems in cellular structures, several schemes can be introduced to further increase the spectrum efficiency. A theoretical analysis is carried out to indicate the advantages of developing digital cellular systems.

I. INTRODUCTION

Today, cellular systems have recognized that with this continued growth, present cellular technology will not allow systems to accommodate subscriber demand in the near future. To ensure that cellular can grow to meet future demand, a more spectrum-efficient technology digital cellular has been studied.(1)

In this paper, the question has to be answered, why digital cellular is more spectral efficient? We have evaluated many analog systems such as 30 kHz-FM,(2) 15 kHz-FM(3) and 7.5 kHz single-sideband,(2) and found that they are all more or less the same in spectrum efficiency. Then, the need of increasing spectrum efficiency leads us to search for other kinds of technologies.

The digital cellular becomes a new candidate. In this paper, we would like to understand the spectrum efficiency measured in cellular systems and the improvement of spectrum efficiency from a digital cellular system.

II. THE MEASUREMENT OF SPECTRUM EFFICIENCY IN CELLULAR SYSTEMS

Since the cellular system is a high capacity system which utilizes the frequency reuse concept, then

$$\text{spectrum efficiency} \neq \text{channel efficiency} \quad (1)$$

where spectrum efficiency is measuring the maximum number of customers being served in a unit geographical area using a given allocated spectrum bandwidth while channel efficiency is counting the maximum number of channels provided in a given allocated spectrum bandwidth.

Here we may derive the spectrum efficiency formula for digital cellular systems based on a maximum co-channel interference in cellular systems. The most interference of a cell is from the first tier of 6 co-channel cells(4) as shown in Fig. 1. There are two situations; one is the interference received at the mobile unit in the home cell; and the other is the interference received at the cell site of the home cell. Since the required carrier-to-noise ratio has to be a fixed value depending on the subjective acceptance*, we denote it, $(C/I)_s$. Both cases would end up like the following equation(3)

$$(C/I)_s = q^4/6 \quad (2)$$

where q is the ratio of co channel-cell separation to cell radius.

$$q = D/R$$

Also the following relation is always held due to the geographical configuration

$$q = \sqrt{3K} \quad (3)$$

where K is the cell reuse number.

Then substituting EQ.(3) into EQ.(2) yields:

$$(C/I)_s = \frac{3}{2} K^2 \quad (4)$$

In cellular systems, the spectrum efficiency is measured by the number of channels per cell, m.

*The required $(C/I)_s \geq 18$ dB is for a 30 kHz FM channel, and $(C/I)_s$ is around 10 dB for a 30 kHz digital channel.

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$$m = \frac{B_t}{B_c \cdot K} = \frac{B_t}{B_c \sqrt{2/3\,(C/I)_s}} \tag{5}$$

where B_t is the total allocated bandwidth and B_c is the channel bandwidth. m is called the radio capacity. EQ.(5) is used to evaluate in both analog cellular systems and digital cellular systems. Comparison of different values of m is shown in Fig. 2.

III. CONCEPT OF SPECTRUM EFFICIENCY IN CELLULAR

The radio capacity m is used to measure the spectrum efficiency. The m shown in EQ.(5) is a function of B_c and $(C/I)_s$. Increasing m, leads to reducing either B_c or $(C/I)_s$ or both. However, the voice quality based on the subjective test has to be kept in a certain level, i.e. the level $(C/I)_s$ is fixed at a given channel bandwidth B_c. IF lowering $(C/I)_s$, B_c must be increased. The relationship between $(C/I)_s$ and B_c can be deduced from EQ.(5) as

$$\frac{(C/I)_s'}{(C/I)_s} = \left(\frac{B_c}{B_c'}\right)^2 \tag{6}$$

The parameters with prime indicate a different set. EQ.(6) indicates that in order to maintain a same voice quality, reduce the channel bandwidth by half, increase $(C/I)_s$ by 4 times.

$$(C/I)'_s = 4\ (C/I)_s \tag{7}$$

Furthermore, the radio capacity m in cellular systems has a similar nature as channel capacity derived by C. E. Shannon (5)

$$\hat{C} = B_c \log_2 \left(1 + \frac{C}{I}\right) \tag{8}$$

where $\hat{C}$ is the channel capacity. Comparing EQ.(5) and EQ.(8), we find both radio capacity m and channel capacity $\hat{C}$ are the functions of B_c and C/I. From EQ.(8), we may derive the following relationship

$$\frac{\log_2 (C/I)}{\log_2 (C/I)'} = \frac{B_c'}{B_c} \tag{9}$$

In EQ.(9) if the B_c reduces by half, C/I increases by squaring itself.

$$(C/I)' = (C/I)^2 \tag{10}$$

Comparing EQ.(6) and (9), a closed analogy can be seen between radio capacity in cellular and channel capacity in Gausian noise.

IV. CONCEPT OF SPECTRUM EFFICIENCY IN DIGITAL CELLULAR

In a digital cellular system, the following relationship holds

$$E_b \cdot R_b = E_s \cdot R_s = C \tag{11}$$

where E_b is the energy per bit at a baseband information rate R_b. E_s is the energy per bit at a transmission rate R_s. C is the carrier power.

A. Coding and subjective test

In EQ.(11), we may emphasize that R_b is related to speech coding and R_s is related to channel coding, as

$$\underbrace{E_b \cdot R_b}_{\text{speech coding}} = \underbrace{E_s \cdot R_s}_{\text{channel coding}} = \underbrace{C}_{\text{subjective test}} \tag{12}$$

In EQ.(12), we may see that even though the received carriers power C are the same, the evaluation of voice quality can be different based on different techniques of speech coding and channel codings. EQ.(12) also has to be tested in different kinds of environment with different vehicle speeds.

B. Relationship between E_s and B_c

Substituting EQ.(11) into EQ.(6) yields

$$\frac{E_s' R_s'}{I'} \Big/ \frac{E_s R_s}{I} = \frac{B_c^2}{B_c'^2} \tag{13}$$

The interference level I is the same due to the environment regardless of different modulation schemes,

$$I = I' \tag{14}$$

The linear relationship between R_s and B_c is as follows:

$$R_s = kB_c, \quad R'_s = kB_c' \tag{15}$$

where k is a constant which can be determined when a modulation scheme is known. Substituting EQ.(14) and EQ.(15) into EQ.(13) yields

$$E_s'/E_s = B_c^3 / B_c'^3 \tag{16}$$

For designing a digital cellular system, the relationship in EQ.(16) is important. It means that if B_c is reduced by half, the energy per bit will increase eight times.

V. DEFINITION OF CELLULAR CAPACITY

We may have to clearly define the term cellular "capacity" in cellular operation. There are two definitions of capacity. One is call radio capacity which is shown in EQ.(5) as m, m is the number of channels (radios) per cell. The radio capacity is the radio engineers used to evaluate the digital cellular system on the radio side. However, the marketing or system engineering people are using traffic capacity, i.e. how many users can be served in a system. Traffic capacity are calculated based not only on radio capacity but also trunking efficiency and different spectrum efficiency schemes.(6) Therefore, traffic capacity is always larger than radio capacity. If new radio capacity is increased by four, for example, the new traffic capacity can be increased by six or seven with comfortable design effort.

VI. SPECTRUM EFFICIENCY SCHEMES IN DIGITAL CELLULAR

Some schemes for spectral efficiency has been suggested in Reference 6 such as (1) multiple channel bandwidth systems, (2) one-third channel offset scheme. These schemes can be very useful in digital cellular systems. Besides, the following schemes are also needed to be addressed:

(1) Variable channel bandwidths
An operational digital channel can be assigned with different bandwidths based on different circumstances.

The operational channel is formed from a physical channel (or basic channel). If the physical channel bandwidth is say 10 kHz, the operational channel can be 5 kHz or a multiple of 10 kHz.

A. Vary channel bandwidth with distance — When the mobile unit is close to the cell site, the channel bandwidth can be reduced; far from the cell site, the channel bandwidth can be increased.

B. Vary channel bandwidth with data speed — If using operational channels for transmitting data; the higher the data rate, the wider the operational channel bandwidth.

(2) Compandor

In digital cellular systems, the syllabic compandor may be still a good device to improve voice quality especially when the speaker's voice is loud. The compandor is placed before the voice being digitalized.

(3) Vehicle location

In the future cellular systems, the vehicle location device can be very helpful for the operator to assign a new channel based on where the vehicle is. Thus the interference can be reduced, and the spectral efficiency increases.

VII. CONCLUSION

The concept of spectrum efficiency in digital cellular is described in this paper. The radio capacity is a function of carrier-to-interference ratio and bandwidth. It is very useful to measure the spectral efficiency. The radio capacity and Shannon's channel capacity have similarity in nature. In digital cellular systems, some issues related to the increase of spectrum efficiency are mentioned. The radio capacity is different from the traffic capacity in a cellular system. If the radio capacity is increased by four, the traffic capacity should be increased by more than four, say six or seven. However, the radio capacity has a unique formula but the traffic capacity has not.

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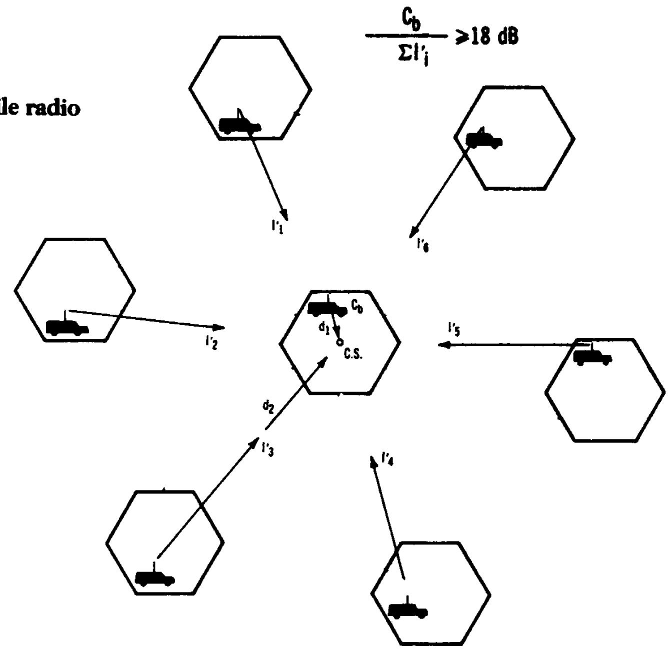
FIG 2 M AND C/I WITH K (GIVEN $B_t/B_c = 416$)

M	$(C/I)_s$ (IN DB)	D/R	K
64	18	4.6	7
114	13	3.31	4
161	10	2.78	3
203	8	2.45	2

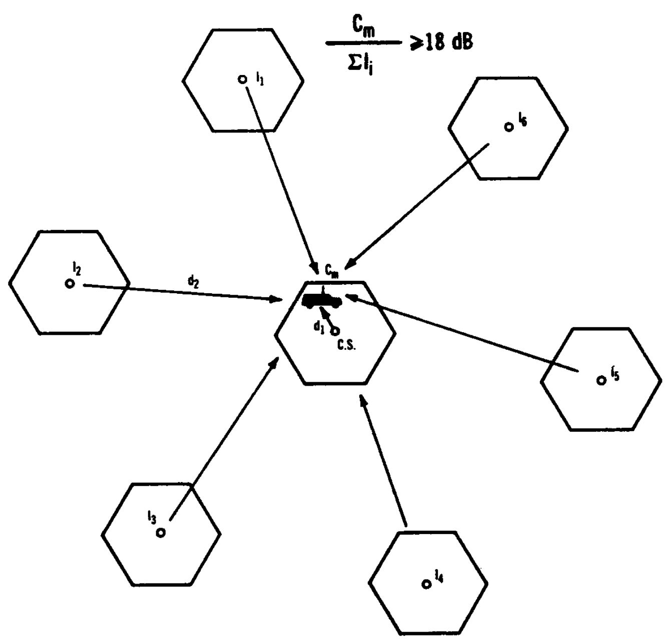
LOWEST IN C/I, HIGHEST IN SPECTRUM EFFICIENCY

Figure 1.

Co-channel interference in a mobile radio environment with six interferers.



(A) Case 1: At a desired cell site.



(B) Case 2: At a desired mobile unit.