# ANALYSIS OF NEW METHODS FOR BROADCASTING DIGITAL DATA TO MOBILE TERMINALS OVER AN FM-CHANNEL

Marion J. de Ridder-de Groote, Ramjee Prasad, Senior Member IEEE, Jan H. Bons
Telecommunications and Traffic-Control Systems Group
Delft University of Technology
P.O. Box 5031, 2600 CA Delft
The Netherlands

**ABSTRACT**

Two new methods using an FM-radio channel for transmission of digital data to mobile terminals are examined:

1. A modification of the radio data system (RDS)

In RDS, additional digital information is multiplexed with a stereo sound signal. A new system is suggested where the data signal can be multiplexed with a mono audio signal. This causes extension to the bandwidth available for the data signal, and therefore the RDS bitrate can be increased. Error calculations are performed both for the original RDS system and for the new system.

2. Orthogonal Frequency Division Multiplex (OFDM)

OFDM is used in the Digital Audio Broadcasting System (DAB), which is designed to transmit digital audio in the FM band. In OFDM a signal is divided over a large number of 2- or 4-PSK modulated orthogonal subcarriers. The subcarriers of 6 different programmes are multiplexed in one beam to reduce the effects of frequency selectivity of the transmission channel.

A new system based on OFDM is proposed, in which the carriers of each programme are transmitted in one FM-channel with a bandwidth of 200 kHz instead of multiplexed with the carriers of other programmes. Error calculations are performed for the subcarriers used in the OFDM modulation method.

## I. INTRODUCTION

In our society distribution of information has become very important, so it is also important to find new ways to transmit information to a large number of people. One way of information distribution is broadcasting digital data to a large number of receivers, where the data is processed and displayed on a monitor or TV-screen. Teletext is a system that performs this kind of data distribution. But since teletext can only use part of the TV bandwidth, it has a limited capacity. Already companies, like the insurance company "Centraal Beheer" in the Netherlands, are hiring teletext capacity to transmit information to their customers. And if more companies will start doing this, the capacity will soon be too small. That is why we have to find new ways of transmitting the information.

Other companies that could be interested in data broadcasting are Railway Companies. They could use the system to transmit information about arrivals, departures and delays of the trains to terminals on each station or even to terminals in the trains, to inform the passengers.

In this paper new systems are investigated, that could use the present VHF-FM system for broadcasting digital information to mobile terminals. The VHF-FM system has the advantage that mobile FM-receivers are already available, and the VHF-FM signal can easily be received in a mobile environment.

Two possible systems are suggested:

1. a modification of the Radio Data System (RDS).

In present VHF-FM systems the RDS signal with a bandwidth of 4.8 kHz is multiplexed with the stereo radio signal. If only the mono radio signal is transmitted and the pilot tone is suppressed, a larger bandwidth is free for data transmission. Then the bandwidth of the RDS signal can be extended, and the bitrate of the RDS data stream can be higher.

2. a system which uses the full bandwidth of an FM-channel (200 kHz) for data transmission.

Here a new system is proposed, which uses a modified version of OFDM (Orthogonal Frequency Division Multiplex). The OFDM technique is also used in Digital Audio Broadcasting (DAB), for modulation of the data signal. The DAB system is developed for the transmission of digital sound.

The outline of this paper is as follows. The modified RDS system is described in section II. In this section also some error calculations are performed for both the present RDS signal and for the modified version. Before the modified OFDM system is described, first the DAB system and OFDM are reviewed in section III. Then section IV describes the system which uses a modified version of OFDM for transmission of data over an FM-channel. Also, in section IV some error calculations are performed. Finally, conclusions are given in section V.

0018-9316/94$04.00 

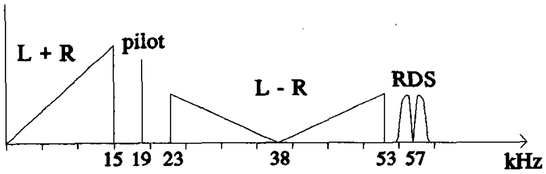


**Figure 1** The spectrum of the FM stereo signal with additional RDS data

## II. NEW RADIO DATA SYSTEM

In the present radio systems the Radio Data System (RDS) is used to transmit additional digital information in the FM-band: traffic information, programme information etc. [1]-[3].

RDS has a bitrate of 1.1875 kbit/s and the bandwidth of the modulated signal is 4.8 kHz. In the RDS system the digital RDS data modulates a 57 kHz subcarrier, which is added to the stereo multiplex signal. The spectrum of the stereo signal in combination with the RDS data is given in fig. 1 [2].

The RDS signal is modulated in three steps: first differential coding of the RDS data stream, followed by biphase coding and finally filtering to reduce the bandwidth of the generated pulses. Figure 2 shows the three coding steps and the resulting signal [2].

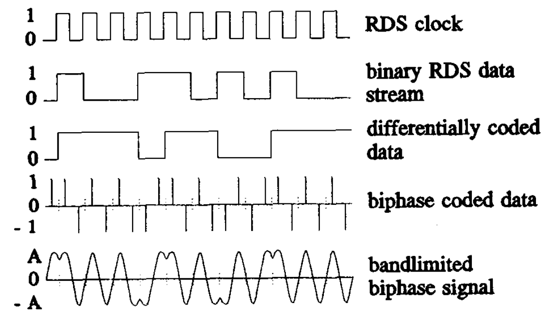


**Figure 2** Coding of the RDS signal

### A. MEAN ERROR PROBABILITY OF THE RDS SIGNAL

The noise at the output of the FM receiver has a quadratic spectrum:

$$G(f) = \frac{\eta f^2}{2S_R} \tag{1}$$

where $\eta$ is the positive-frequency power density of the noise, f is the operating frequency, and $S_R$ is the power of the received signal [4]. Equation (1) holds for noise-limited networks. However, FM-broadcast networks have become more and more interference limited. In our computations the interfering signals are assumed to behave as bandlimited white Gaussian noise with bandwidth $B_T \approx 200\text{-}300$ kHz. The corresponding signal-to-noise ratio $S_R/N_i$ is set equal to the carrier to interference ratio (C/I-ratio or protection ratio) $\gamma_0$ [5]. For a given $\gamma_0$ in the channel the equivalent spectral density of the interference signal is

$$\eta_i = \frac{N_i}{B_T} = \frac{S_R}{\gamma_0 B_T} \tag{2}$$

where $N_i$ is the noise power. After detection this leads to a noise spectrum in the form of equation (1).

The FM noise spectrum is approximated as a flat spectrum in the relevant range of the RDS sub-signal (54.6-59.4 kHz). The post detection noise spectrum for the RDS signal is then approximated as

$$\eta_{RDS} \triangleq G(f_{sc}) = \frac{\eta_i f_{sc}^2}{2S_R} = \frac{1}{2\gamma_0 B_T} f_{sc}^2 \tag{3}$$

where $f_{sc}$ is the frequency of the RDS subcarrier (57 kHz).

The RDS signal is antipodal [2]. Wozencraft and Jacobs [6, pp. 250] give the bit error rate for antipodal signals with additive white Gaussian noise for a given $\gamma_0$:

$$P[\epsilon] = Q\left(\sqrt{\frac{2E_s}{N_0}}\right) \tag{4}$$

Here $N_0$ is the noise power, which is defined as $\frac{1}{2}N_0=\sigma_n^2$ [6], $\sigma_n^2$ is the variance of the noise, and $E_s$ is the signal energy. $E_s=E_b r_b=\frac{1}{2}f_{\Delta,RDS}^2 T_b r_b$ and $N_0=\eta_{RDS} r_b$, where $E_b$ is the energy per bit, $r_b$ is the bitrate, $f_{\Delta,RDS}$ is the frequency deviation of the RDS signal, and $T_b$ is the bit duration. Now it is found that

$$P[\epsilon] = Q\left(\sqrt{\frac{2E_b}{\eta_{RDS}}}\right) = Q\left(\sqrt{\frac{2f_{\Delta,RDS}^2 T_b}{f_{sc}^2}\gamma_0 B_T}\right) \tag{5}$$

To calculate the mean bit error rate for transmission of data to mobile terminals, Rayleigh fading is assumed in the RDS channel. Then the average bit error probability is:

$$\begin{aligned}\overline{P[\epsilon]} &= \int_0^\infty \frac{1}{\bar{\gamma}_0} e^{-\gamma_0/\bar{\gamma}_0} Q\left(\sqrt{\frac{2f_{\Delta,RDS}^2 T_b}{f_{sc}^2}\gamma_0 B_T}\right) d\gamma_0 \\ &= \int_0^\infty \frac{1}{\bar{\gamma}_0} e^{-\gamma_0/\bar{\gamma}_0} Q\left(\sqrt{c\gamma_0}\right) d\gamma_0 \end{aligned} \tag{6}$$

$$\text{with } c = \frac{2f_{\Delta,RDS}^2 T_b}{f_{sc}^2} B_T$$

If equation (6) is rewritten using the error function erfc(..) instead of Q(..), and some calculations are performed which can be found in Proakis [7, pp. 716 and 717], then the result is:

$$\overline{P[\epsilon]} = \frac{1}{2}\left(1-\sqrt{\frac{c\bar{\gamma}_0}{2+c\bar{\gamma}_0}}\right) \tag{7}$$

In fig. 3 the results of this calculation are given for different values of the average C/I-ratio, using the following parameters: $f_{\Delta,RDS}=7.5$ kHz, $T_b=1/r_b=1/1.1875\times10^3$, $f_{sc}^2=57$ kHz. Here we see that for an increasing bandwidth the mean bit error rate decreases.

In these computations the FM-threshold effect has not been taken into consideration for simplicity.

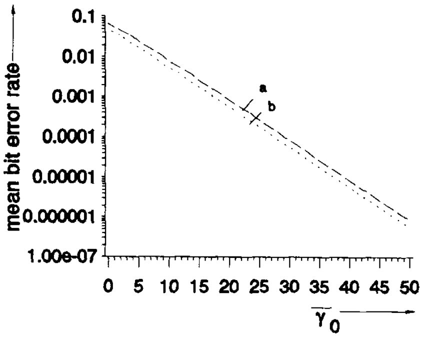


**Figure 3** Bit error ratio of the RDS signal in a Rayleigh fading channel

a) bandwidth = 200 kHz

b) bandwidth = 300 kHz

### B. NEW DATA SYSTEM BASED ON RDS

In the present radio systems the frequency band from 0 to 53 kHz in the baseband is reserved for the stereo radio signal, and only the band above 53 kHz remains for the transmission of RDS data. Now if the pilot tone is suppressed and the mono signal is transmitted in the band from 0 to 15 kHz, as usual, the FM-receiver can only demodulate the mono signal and will ignore the information in the rest of the band. In that case the frequencies above 23 kHz can be used for transmission of an extended RDS signal (fig. 4). The frequency band from 15 to 23 kHz has to remain free to make sure the receiver does not detect a pilot tone.

For FM-broadcasting in the VHF-band the maximum instantaneous frequency is defined in the CCIR Recommendations [8] to be 76 kHz. Now the frequency band from 23 to 76 kHz is available for transmission of data. If this bandwidth is used in full, the bandwidth of the modulated data signal will be 53 kHz. For the original RDS-signal, where the bandwidth is 4.8 kHz, the bitrate is 1.1875 kHz. So if the same spectral efficiency can be used for the data signal with a bandwidth of 53 kHz the bitrate of the new system will be (53/4.8)x1.1875=13.1 kbit/s.

This system could be very interesting for commercial radio stations. These stations could decide to transmit their programs in mono, and supplement their income by renting the data transmission capacity to interested parties.

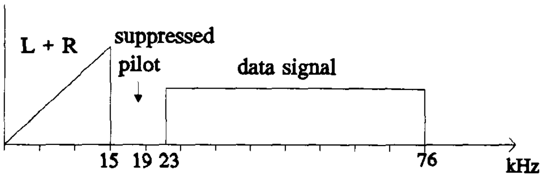


**Figure 4** Baseband spectrum of the mono radio signal multiplexed with a data signal

### C. MEAN BIT ERROR RATE OF THE DATA SIGNAL

For frequency modulation the noise spectrum at the output of the detector is quadratic. In the original RDS system the noise spectrum within the relevant bandwidth is approximately flat. But for the new data system there is a large difference in the noise between the lower and the higher frequencies.

Now the bandlimited RDS signal will be differentiated before it is transmitted and the resulting signal is phase modulated. Then after detection and integration the additive noise has a flat spectrum [4]:

$$G_{data}(f) \approx \frac{\eta}{2S_R} \tag{8}$$

The spectral density of the interference signal is the same as in equation (2).

Now the noise spectrum of the data signal can be approximated as

$$\eta_{data} \triangleq G_{data}(f) \approx \frac{\eta_i}{2S_R} = \frac{1}{2\gamma_0 B_T} \tag{9}$$

And the bit error rate becomes:

$$P[\epsilon] = Q\left(\sqrt{\frac{2E_b}{\eta_{data}}}\right) = Q\left(\sqrt{2\phi_{\Delta,data}^{\ 2}\, T_b \gamma_0 B_T}\right) \tag{10}$$

Here we used $E_b = ½\phi_{\Delta,data}^{\ 2} T_b$. For the Rayleigh fading channel the average bit error probability is written as

$$\begin{aligned}\overline{P[\epsilon]} &= \int_0^\infty \frac{1}{\overline{\gamma_0}} e^{-\gamma_0/\overline{\gamma_0}} Q\left(\sqrt{2\phi_{\Delta,data}^{\ 2}\, T_b \gamma_0 B_T}\right) d\gamma_0 \\ &= \int_0^\infty \frac{1}{\overline{\gamma_0}} e^{-\gamma_0/\overline{\gamma_0}} Q\left(\sqrt{c' \cdot \gamma_0}\right) d\gamma_0\end{aligned} \tag{11}$$

with $c' = 2\phi_{\Delta,data}^{\ 2}\, T_b B_T$

where $\phi_{\Delta,data}$ is the frequency deviation of the phase modulated data signal, and $B_T$ is the transmission bandwidth (200 kHz). Following the same computations as in eqs. (6) and (7) we find the mean bit error probability for the data signal:

$$\overline{P[\epsilon]} = \frac{1}{2}\left(1 - \sqrt{\frac{c\overline{\gamma_0}}{2 + c\overline{\gamma_0}}}\right) \tag{12}$$

In these equations $\phi_{\Delta,data}$ is calculated using the relationship $B_T \approx 2(\phi_\Delta + 1)W$, where W is the bandwidth of the baseband channel [4]. Using the full baseband channel bandwidth W=76 kHz, we find $\phi_\Delta$ = 0.32 radian. If the bandwidth is the same as in the FM stereo multiplex signal (W ≈ 60 kHz) we find $\phi_\Delta$=0.66 radian.

In fig. 5 the mean bit error rate of the data signal is given as a function of the C/I-ratio (carrier to interference ratio), for an FM-channel with a bandwidth $B_T$=200 kHz and for $\phi_\Delta$=0.32 radian, for different bitrates. In this figure it is shown, that for an increasing bitrate the mean bit error rate also increases. In fig. 6 the mean bit error rate is calculated for $\phi_\Delta$=0.66 radian. From figs. 5 and 6 it is obvious that for an increasing phase deviation the mean bit error rate decreases.

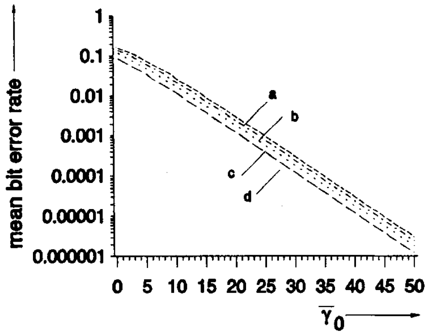


**Figure 5** Mean bit error rate for the phase modulated data signal with $\phi_A$ = 0.32 rad
a) bitrate = 25 kbit/s c) bitrate = 15 kbit/s
b) bitrate = 20 kbit/s d) bitrate = 10 kbit/s

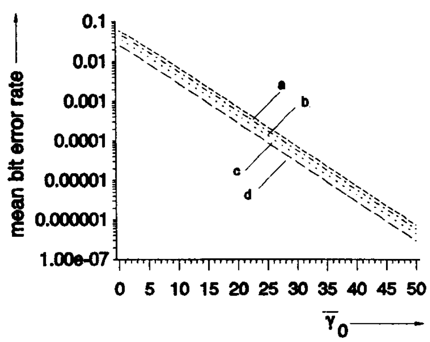


**Figure 6** Mean bit error rate for the phase modulated data signal with $\phi_A$ = 0.66 rad
a) bitrate = 25 kbit/s c) bitrate = 15 kbit/s
b) bitrate = 20 kbit/s d) bitrate = 10 kbit/s

## III. DIGITAL AUDIO BROADCASTING

The Digital Audio Broadcasting (DAB) system is developed by Eureka project 147 for transmission of digital sound. The results were acquired up to mid-1990.

In the frequency spectrum band II, 87.5 to 108 MHz, is reserved for VHF-FM. This service has to be replaced by DAB. In Germany DAB is planned to be introduced as of 1995, and it will take a period of 10-15 years to replace the present VHF-FM system by DAB [9]. For this transition time a parking position has been found in the frequency band of 104 to 108 MHz [10].

### A. DAB SYSTEM PLAN

A general description of a DAB system is given in [11]. First the left and the right analog signals of the stereophonic programme are converted into a digital signals in the transmitting studio . The bitrate of this signal is reduced using source coding. Here MUSICAM (Masking Pattern Universal Subband Integrated Coding And Multiplexing) is used [11], which uses subband coding. Then extra information, such as RDS and traffic control information, is added to the signal. After source coding and addition of extra information the bitrate of the datastream is 200 kbit/s. To enable the use of concealment at the receiver one of the two stereophonic signals is delayed.

The resulting signal is sent to the transmitter, where it is multiplexed with other programmes, which are coded in the same way. In the transmitter the final multiplexed signal is channel coded to add error protection. A convolutional code with an efficiency of ½ (one protection bit per data bit) is used [12]-[14]. Then the signal is modulated using OFDM and finally it is broadcasted. The combination of channel coding and modulation is called COFDM (Coded Orthogonal Frequency Division Multiplex) [15].

In the receiver the delay between the two signals is cancelled. Then concealment can be used, if the signal was lost for 1 to 1000 msec during transmission [11]. The signal is demodulated, decoded using a Viterbi decoder and demultiplexed. Finally, after error correction the resulting digital signal is converted to an analog signal.

The DAB system is designed for transmission of digital coded audio. The IRT (Institüt für Rundfunktechnik), however has decided that the system should also be able to transmit other digital information (for example text). That is the reason why an error protection code with a fixed efficiency is chosen. This means that it is relatively easy to use this system for data transmission: the data must be inserted in the system at a point after the analog to digital conversion of the audio signal and subtracted before the digital to analog conversion in the receiver. Then the rest of the transmission path can be the same as that for transmission of the digital radio signal.

### B. MODULATION USING OFDM

OFDM (Orthogonal Frequency Division Multiplex) is a modulation method which tries to solve the problem of diversity. OFDM was first proposed by Cimini [16] as a method for digital communication over Rayleigh fading mobile radio channels.

A modulation method using ODFM combines the advantages of smallband modulation (less distortion) with the advantages of wideband systems (less disturbance of the signal) [11]. The information of each radio programme is divided over many single carriers each carrying a low bitrate. This way wideband channel, which has a minimum bandwidth of 1.5 to 2 MHz, which is frequency selective is transformed into a large number of non-selective narrowband sub-channels. Then the carriers of a bundle of 6 programmes are frequency multiplexed into one beam with a large bandwidth of 1.5 MHz [15]. To make more efficient use of the

spectrum the frequency spectra of the carriers can be allowed to overlap in such a way that the information content of each signal is mutually orthogonal [17]. Due to this orthogonality there is no inter symbol interference at the sampling time.
Inter symbol interference, caused by the transmission channel, can have the result that the properties of orthogonality in the OFDM signal are lost. This problem can be solved by preceding each signal $\psi_{j,k}$ by a guard interval,which absorbs the inter symbol interference. In [15] the spectral density of an OFDM signal with N=32 is calculated, when a guard interval with a duration $\Delta = T_s/4$ is added. In practice much higher values for N are used and then the spectrum tends asymptotically to a rectangular spectrum. Because this spectrum just fits in the transmission channel the spectral efficiency closely approaches the 2 bps/Hz theoretical limit for 4-PSK [17].

The structure of an OFDM receiver is similar to that of a FDM receiver: for each of the emitted carriers, a filter is needed, which is matched to the signal $g_k(t)$. Due to the orthogonality of the signals $g_k(t)$ the number of operations necessary is considerably reduced.
The OFDM receiver is based on the use of a partial FFT (Fast Fourier Transform) algorithm. This algorithm is called partial FFT because the receiver deals with only one programme out of K (we only want to receive one out of K programmes), and the FFT algorithm does not have to be executed completely. A detailed description of the demodulation scheme is given by Alard and Lassalle [15]. More information about OFDM is given in [11,15] and [18]-[21].

### C. USING OFDM FOR TRANSMISSION OF DATA OVER AN FM-CHANNEL

Unfortunately the OFDM system, as described in this section, needs a minimum bandwidth of 1.5 MHz, due to frequency division multiplexing of the carriers of different programmes. So it is not possible to use this system for transmission of data over an existing FM-channel with a bandwidth of 200 kHz. For transmission of data in an FM-channel all the OFDM-techniques can be used except for frequency division multiplex. The carriers of one coded data stream will have to be spread over one FM-channel with a bandwidth of 200 - 300 kHz, instead of being multiplexed with the carriers of other programmes. A proposal for such a system will be described in more detail in section IV.
Since the carriers of one data stream are now transmitted in one frequency band and not multiplexed with the carriers of other data streams this system is not a wideband system, like OFDM. Therefore we will have to find another way to resolve the problem of channel selectivity, for instance additional error coding.

## IV. NEW SYSTEM BASED ON OFDM

Before we can determine how the present OFDM system can be modified, we have to consider the properties of the OFDM modulation method. The two basic properties of OFDM are:

- orthogonality of the subcarriers.
  This orthogonality allows a better spectral efficiency, because the spectra of the subcarriers can be allowed to overlap.
- frequency division multiplex (FDM) of the subcarriers of K different programmes.
  FDM is introduced in the system to reduce the effects of channel selectivity: the carriers of the programmes are spread over a large part of the spectrum, so a frequency selective fade will affect a few carriers of each programme instead of many carriers of one programme.

The second property, multiplexing the carriers in frequency, makes that OFDM needs such a large bandwidth, because the K multiplexed programmes have to be transmitted in one frequency band. For orthogonality of the subcarriers a large bandwidth is not required.
So if we want to design a system which uses the OFDM-techniques, we cannot use frequency division multiplex. We can use, however, all the other techniques described in section III: the elementary orthogonal signals, the guard interval, FFT for demodulation of the signal and channel coding with convolutional codes and maximum likelihood decoding.

### A. DESCRIPTION OF THE NEW SYSTEM

The new system for data transmission proposed in this section looks much like the DAB-system described in section III. The transmitter and the receiver of this new system are given in fig. 7.

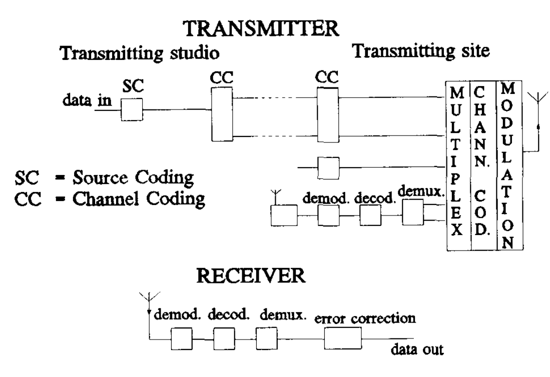


**Figure 7** Data transmission system using techniques from OFDM

In the transmitting studio the digital data can be source coded to remove redundancy. In the DAB system MUSICAM is used for source coding. MUSICAM transforms an analog radio signal into a digital source coded signal. In this system we already have a digital signal available, so we will need another type of source coding.
If the transmitting studio is not at the same location as the transmitting site, the data is channel coded and transmitted to the transmitting site. There the data signal can be multiplexed with other data signals, and then it is channel coded and modulated before it is transmitted. For channel coding convolutional codes can be used, as in the DAB system [12,15].
As said before the modulation method will be different than the modulation method in DAB. In the next section we will propose a modulation method which is based on OFDM.

At the receiver the signal is demodulated and then decoded. As in DAB we could use a Viterbi decoder. After detection the data signal is demultiplexed and then error correction is used.

## B. MODULATION

In this section we look at the modulation method in the new system. For modulation a modified version of OFDM can be used.
The information of the data signal will be divided over a number of orthogonal carriers, which are 4-PSK modulated, like in OFDM. However, the distribution of the carriers in the frequency spectrum will have to be different than for OFDM, because frequency division multiplexing cannot be used. In this system the carriers of one coded data stream will be spread over one FM-channel with a bandwidth of 200 kHz instead of multiplexed with the carriers of other programmes. The resulting frequency spectrum is given in fig. 8.
As we already mentioned in section 3, this system is not a wideband system, like OFDM, and it is necessary to find another way to resolve the problem of channel selectivity.
The separate subchannels of an OFDM channel, however, are frequency nonselective. Therefore in the next section the mean error probability for the 4-PSK modulated subcarriers of the OFDM system will be calculated. Then a brief discussion is given of the influence of frequency selectivity on the modulated data signal, compared to the influence on an OFDM signal.

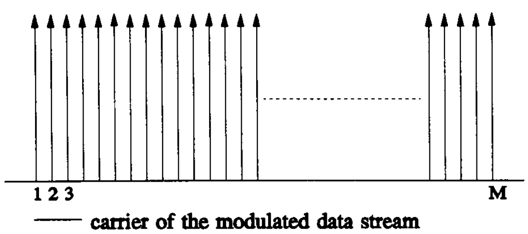


**Figure 8** Distribution of the carriers of one data stream

## C. MEAN BIT ERROR RATE OF A 4-PSK MODULATED SUBCARRIER

In OFDM 1536 carrier waves are spaced through a bandwidth of 1.5 MHz, and each carrier is modulated with 2 kbit/s, so the spectral efficiency is about 2 bps/Hz [11]. The carrier frequencies are separated by $1.5*10^6/1536=977$ Hz.

The error function for 4-PSK in a channel with additive white Gaussian noise is [4,7]

$$P_e = Q(\sqrt{2\gamma_b}) = \frac{1}{2}\mathrm{erfc}(\sqrt{\gamma_b}) \quad \text{with } \gamma_b = E_b/\eta \; ; \; E_b = \frac{1}{2}\phi^2_\Delta T_b \; ; \; \eta = \frac{1}{2\gamma_0 B_T} \tag{13}$$

where $\gamma_b$ is the signal to noise ratio per bit, $E_b$ is the bit energy, $\phi_\Delta$ is the frequency deviation, $T_b$ is the bit duration, $\gamma_0$ is the carrier to interference ratio and $B_T$ is the channel bandwidth. $\eta$ is the positive power density of the noise and is defined in the same way as $\eta_{data}$ in equation (9).
Equation (13) leads to

$$P_e = \frac{1}{2}\mathrm{erfc}(\sqrt{\phi^2_\Delta T_b B_T \gamma_0}) \tag{14}$$

So for each 4-PSK modulated carrier the error rate per bit is given by equation (14). This equation holds both for the frequency multiplexed carriers of the conventional OFDM signal and for each carrier of the new data signal.

A frequency non-selective Rayleigh fading channel is assumed here. In that case the mean bit error probability for a 4-PSK modulated carrier is

$$P_e = \int_0^\infty \frac{1}{\bar{\gamma}_0} e^{-\gamma_0/\bar{\gamma}_0} \frac{1}{2}\mathrm{erfc}(\sqrt{c\gamma_0})\, d\gamma_0 \tag{15}$$

where it is put that

$$c = \phi^2_\Delta T_b B_T \tag{16}$$

According to Proakis [7] the result of this integration is:

$$\bar{P}_e = \frac{1}{2}\left(1 - \sqrt{\frac{\bar{\gamma}_b}{1+\bar{\gamma}_b}}\right) \quad \text{with } \bar{\gamma}_b = c\bar{\gamma}_0 = E_b/\eta = \phi^2_\Delta T_b B_T \gamma_0 \tag{17}$$

In this expression the spectral efficiency is $1/(T_b B_T)=2$ bps/Hz. The phase shifts in 4-PSK are $\pm 3\pi/4$ and $\pm\pi/4$, so the maximum phase deviation $\phi_\Delta$ is $3\pi/4$. The results of these calculations are given in fig. 9 for different values of the carrier to interference ratio $\gamma_0$.

Frequency selectivity of the channel has the effect that subcarriers at different frequencies will be affected differently by frequency selective fades.
A frequency selective fade will distort a number of adjacent subcarriers. In the conventional OFDM system the effects of such a

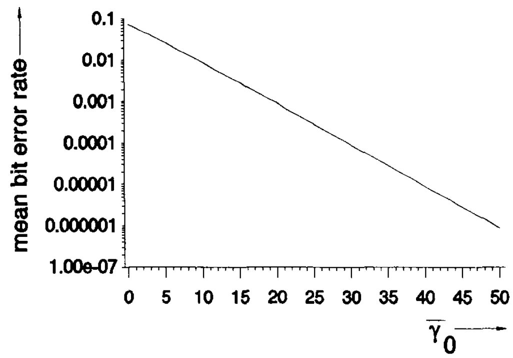


**Figure 9** Mean bit error rate for 4-PSK modulated carrier waves in a frequency nonselective Rayleigh fading channel

fade are less severe than in the new system, because in the conventional system only a few subcarriers of the 6 different programmes will be affected, and the lost bits can be restored using error correction. In the new system a frequency selective fade will affect many subcarriers of one programme and error correction will be much more difficult. This problem could be solved by using additional error correction in the new system.

## V. CONCLUSIONS

In this paper it is shown how FM-channels can be used for transmission of digital data. Modifications of existing systems were investigated.

After these investigations the following conclusions can be drawn:

1. If an **RDS** signal is multiplexed with the mono radio signal instead of the stereo signal, more bandwidth is available for the RDS signal. Now the bandwidth of the RDS signal can be increased, and thereby the bitrate can be increased to approximately 13 kbit/s.

2. Calculation of the mean bit error rate of the RDS signal in a Rayleigh fading channel indicates that the error probability for the new system is slightly higher than for the present RDS system.

3. This new system could be very interesting for local radio stations that could decide to transmit their programmes in mono and earn money by renting data transmission capacity to others.

4. Once the **DAB** system is operational, this system can easily be used for data transmission. It is designed for transmission of digital audio, but it is designed in such a way that it can also be used for transmission of other types of digital data. In the DAB system a bitrate of about 400 kbit/s will be assigned to each channel. (in case of radio transmission: 192 kbit/s for the stereo signal, 8 bit/s for additional information and 200 kbit/s for error correction)

5. Because in **DAB** the modulation method OFDM is used the system needs a minimum bandwidth of 1.5 MHz, so it can't be implemented in the present FM-system. Therefore we can't use this system before it is fully operational.

6. **OFDM** needs such a large bandwidth due to frequency multiplexing of the carriers of 6 different programmes in one beam

7. Due to this multiplexing in **OFDM**, a frequency selective fade will affect only a few carriers of the 6 programmes, instead of many carriers of one programme, and with error correction these carriers can be restored. If we reduce the bandwidth of **OFDM** by transmitting the carriers of one programme in one frequency band, a fade will affect many carriers of this one programme and error correction will be more difficult.

8. Still this option -using all the **OFDM** techniques except frequency multiplexing- is very interesting, because in data transmission it should be possible to solve the problems of frequency selectivity by extra error correction coding, and then we would have a system with a maximum capacity of 400 kbit/s. (note that a part of this capacity should be used for error correction)

9. One teletext page comprises 8 kbit. In table I the bitrates of the different systems discussed in this paper and the teletext system are given. Also the time is given which is needed for transmission of one teletext page for the different systems.

Table I: Bitrates for different data transmission systems and transmission time per page of Teletext for each system

| | bitrate (kbit/s) | transmission time per page (sec) |
|---|---|---|
| Teletext | 16 | 0.5 |
| RDS | 1.1875 | 6.75 |
| New RDS system | 13 | 0.53 |
| OFDM | 400 | 0.02 |

REFERENCES

[1] Recommendations and Reports of the CCIR, Broadcasting Service (sound), Specifications of the Radio Data System, Vol. X-I, 1986, Rec. 643 Annex I.

[2] Radio-Daten-System: So kommt RDS zum Sender, Erzeugen und Übertragen von RDS-Signalen, Funkschau 16, 1987. (in German)

[3] Scomazzon, P.: Comparative study of digital modulations accompanying VHF/FM broadcasts, EBU Review: Technical, August 1991, nr. 248, pp. 170-183.

[4] Carlson, A. B: Communication systems, McGraw-Hill Book Company,1986.

[5] Recommendations and Reports of the CCIR, Broadcasting Service (sound), Planning standards for FM sound broadcasting at VHF, Vol. X-I, 1986, Rec. 412-4.

[6] Wozencraft, J. M. and Jacobs, I. M.: Principles of communications engineering, John Wiley & Sons, Inc., 1965.

[7] Proakis, J. G.: Digital communications, McGraw-Hill book company, 1989.

[8] Recommendations and Reports of the CCIR, Broadcasting Service (sound), Transmission standards for FM sound broadcasting at VHF, Vol. X-I, 1986, Rec. 450-1.

[9] Müller-Römer, F: Digital audio broadcasting (DAB) via terrestrial transmitters and satellite, 6th World Telecommunication Forum, Integration, Interoperation and Interconnection, the way to global services, International Telecommunication Union, Technical Symposium, Geneva, 10-15 October 1991, part 2, Vol. II, pp. 121-125.

[10] Prosch, T. A.: A possible frequency planning method and related model calculations for the sharing of VHF band II (87.5-108 MHz) between FM and DAB (Digital Audio Broadcast) systems, IEEE Transactions on Broadcasting, Vol. 37, pp. 55-63, June 1991.

[11] Plenge, G,: DAB - Ein neues Hörrundfunksystem, Stand der Entwicklung und Wege zu seiner Einführung, Rundfunktechnische Mitteilungen 35 (1991), pg. 45 to 66. (in German)

[12] Cain, J. B. and Clark, G. C.: Punctured convolutional codes of rate n-1/n and simplified maximum likelihood decoding, IEEE Transactions on Information Theory, Vol. IT-25, pp. 97-100, January 1979.

[13] Clark, C. G. and Cain, J. B.: Error correction coding for digital communications, Published by Plenum Press, New York, 1981.

[14] Yasuda, Y. et al.: High rate punctured convolutional codes for soft decision decoding, IEEE Transactions on Communications, Vol. COM-32, pp. 315-318, March 1984.

[15] Alard, M. and Lassalle, R.: Principles of modulation and channel coding for digital broadcasting for mobile receivers, EBU Review: Technical (1987), nr. 224, pp. 168-190.

[16] Cimini, L. J.: Analysis and simulation of a digital mobile channel using orthogonal frequency division multiplexing, IEEE Transactions on Communications, Vol. COM-33, pp. 665-675, July 1985.

[17] Mason, A.: Terrestrial broadcasting to the home of digital television and data in a frequency band shared with analogue television, 6th World Telecommunication Forum, Integration, Interoperation and Interconnection, the way to global services, International Telecommunication Union, Technical Symposium, Geneva, 10-15 October 1991, part 2, Vol. II, pp. 127-130.

[18] Hoeher, P. et al.: Performance of an RCPC-Coded OFDM-based Digital Audio Broadcasting (DAB) system, IEEE Global Telecommunications Conference, Phoenix, Arizona, Dec. 2-5, 1991. pp. 2.1.1-2.1.7.

[19] Ratliff, P. A.: EBU/DAB studies for a new digital sound radio broadcasting system - CD quality for mobiles, Speakers' papers, technical symposium, new horizons in electronic media, International Telecommunication Union, Geneva, 4-7 October 1989.

[20] Casas, E. F. and Leung, C.: OFDM for data communication over mobile radio FM channels - Part I: Analysis and experimental results, IEEE Transactions on Communications, Vol. 39, pp. 783-793, May 1991.

[21] Weinstein, S. B. and Ebert, P. M.: Data transmission by frequency division multiplexing using the discrete Fourier transform, IEEE transactions on Communication Technology, Vol. COM-19, pp. 628-634.

**Marion de Ridder-de Groote** was born in Breda, The Netherlands, in 1969. She received the M.Sc. degree in Electrical engineering from Delft University of Technology, The Netherlands, in 1992. During 1990 she participated in the research on Differential Chain Coding for line drawings. In 1991 she joined The Dutch PTT for three months, where she investigated Wide Area Computer Networks. During 1991-1992 she worked on data transmission over FM radio channels.

**Ramjee Prasad** (M'88-SM'90) was born in Babhnaur (Gaya), Bihar, India on July 1, 1946. He received the B.Sc. (Eng.) from Bihar Institute of Technology, Sindri, India, and the M.Sc. (Eng.) and Ph.D. degrees from Birla Institute of 'Technology (BIT), Ranchi, India in 1968, 1970 and 1979, respectively.

He joined BIT as Senior Research Fellow in 1970 and became Associate Professor in 1980. During 1983-1988 he was with the University of Dar es Salaam (UDSM), Tanzania, where he became Professor in Telecommunications at the Department of Electrical Engineering in 1986. Since February 1988, he has been with the Telecommunications and Traffic Control Systems Group, Delft University of Technology, The Netherlands, where he is actively involved in the area of mobile, indoor and personal radio communications. While he was with BIT, he supervised many research projects in the area of Microwave and Plasma Engineering. At UDSM he was responsible for the collaborative project "Satellite Communications for Rural Zones" with Eindhoven University of Technology, The Netherlands. He has published over 150 technical papers. His current research interest lies in packet communications, adaptive equalizers, spread-spectrum CDMA systems and multi-media communications.

He has served as a member of advisory and programme committees of several IEEE international conferences. He has also presented tutorials on Mobile and Indoor Radio Communications at various universities, technical institutions and IEEE conferences. He is also a member of a working group of European co-operation in the field of scientific and technical research (COST-231) project dealing with "Evolution of Land Mobile Radio (including personal) Communications" as an expert for the Netherlands.

Prof. Prasad is listed in the US Who's Who in the World. He was Organizer and Interim Chairman of IEEE Vehicular Technology/Communications Society Joint Chapter, Benelux Section. Now he is the elected chairman of the joint chapter. He is also founder of the IEEE Symposium on Communications and Vehicular Technology (SCVT) in the Benelux and he was the Symposium Chairman of SCVT'93. He is one of the Editors-in Chief of a new journal on "Wireless Personal Communications" and also a member of the editorial board of other international journals including IEEE Communications Magazine. He is the Technical Program Chairman of PIMRC'94 International Symposium to be held in The Hague, The Netherlands during September 19-23, 1994 and also of the Third Communication Theory Mini-Conference in conjunction with GLOBECOM'94 to be held in San Francisco, California during November 27-30, 1994. He is a Fellow of IEE, a Fellow of the Institution of Electronics & Telecommunication Engineers, a Senior Member of IEEE and a Member of the New York Academy of Sciences.

**Jan H. Bons** was born in Schiedam, the Netherlands, on July 15, 1948. He joined the Delft University of Technology in 1968 and is now member of its scientific staff. After his military service in 1969-1970, and his higher electronics education (H.B.O.), he participated in the researchand development of graphical telecommunications at the Delft University of Technology, with a particular emphasis on telewriting systems. His present work is in the area of new telematic services and systems, particularly electronic mail and teletext. He was managing the Graphimail project, sponsored by The Dutch PTT, aimed at demonstrating telegraphics internationally during the ITU world exhibition TELECOM '87 held in Geneva in October 1987. He is the coauthor of a number of papers on telewriting and is a guest lecturer at the TopTech telecommunication management course in Delft. He is also active in the telematics industry.