REPORT 903-2

DIGITAL TRANSMISSION IN THE LAND MOBILE SERVICE

(Question 40/8)

(1982-1986-1990)

1. SPECIFICATION OF DIGITAL TRANSMISSION SYSTEMS

1.1 Introduction

This section identifies performance specifications for a digital transmission system and shows how these should be related to the system design parameters. The performance specifications are determined from the operational requirements of the user.

Traffic capacity is specified in Erlangs per MHz per unit area. The required system bandwidth is determined by the digital modulation method and frequency reuse factor and access protocol. The frequency reuse factor is related to the minimum carrier to interference ratio that can be tolerated by the digital modulation method.

1.2 Performance specifications

The performance specifications to be extracted from the user operational requirements are traffic capacity, system response time, false message probability and coverage area.

1.2.1 The *traffic capacity* is determined from:

$$T = d \zeta r P_s$$

where:

- d: data rate is related to the digital modulation method and channel bandwidth;
- ζ: channel access control efficiency is related to the signalling and supervision methods;
- r: protocol efficiency is related to the data transmission format;
- P_s: successful message probability is related to the error control techniques.

1.2.2 The *system response time* is related to the successful message probability, channel access control efficiency and data rate.

1.2.3 The *false message probability* is related to the error control techniques.

1.2.4 The *coverage area* is related to the minimum average RF signal level requirement for an acceptable successful message probability. This level will affect the number of radio sites and/or the transmitter power required.

2. Error performance

2.1 Introduction

This section is concerned with the error performance of digital transmission and with error reduction and error control techniques.

2.2 Error patterns

2.2.1 Effects of fading and shadowing

The measured bit error ratio (BER) at VHF for a moving vehicle has been shown to agree with the theoretical prediction [French, 1980] for a fading signal. The theoretical effect of fading and shadowing together is also shown in [French, 1980]. (Shadowing is also known as location variability.) Results at UHF from Japan are shown in [Daikoku et al., 1981].

2.2.2 Dependence on bit rate

[French, 1980] discusses a critical bit rate below which errors are rare and above which the BER is high and nearly constant. Above the critical rate, the bit rate should be high enough to permit error control coding, but not so high that distortion or adjacent channel interference results. In his example the critical bit rate is in the range 75 to 150 bit/s for a vehicle speed of approximately 30 km/h.

[Hata and Miki, 1984] shows the BER performance of MSK with differential detection, measured in an urban environment. When the transmission bit rate is higher than 64 kbit/s, some corrective techniques are required to mitigate the effects of multipath propagation.

In high speed digital transmission, bit and frame synchronizations are of major importance. Trial systems using transmissions at bit rates higher than 1 Mbit/s have been reported [Böhm, 1982].

2.2.3 Error distribution

Typical error distributions at VHF and UHF show that errors caused by fading occur in bursts [French, 1980]. Typically, for bit rates of 1200 bit/s, BER of 10^{-3} and vehicle speeds of approximately 30 km/h, the probability of eight or more errors occurring in a code word of 64 bits is in the order of 10^{-3}.

2.3 Error-free run length distributions

Data obtained from a field trial in the city of Ottawa, Canada [Towaij et al., 1983] on continuous digital transmission at 2400 bit/s showed that, as anticipated, errors occur in bursts. The bit error patterns from the field measurements were analysed in terms of cumulative distributions of error free runs in the three frequency bands 150 MHz, 450 MHz, and 850 MHz. It was, however, demonstrated [Figure 25, Report 903], that the 450 MHz band maintains a consistent behaviour with fluctuations in the S/N from 24 dB to 16 dB while both the 850 and 150 MHz bands resulted in a considerable variation in the error-free run length.

Experimental results for discriminator-MLSE detection of a GTFM signal in the presence of fast Rayleigh fading [Chung, 1987] shows that consecutive errors are restricted to about ten bits, even for a BER of 8%, and that diversity has little influence on these patterns.

2.4 Improvements with diversity

Diversity reception is considered to be an effective technique to mitigate multipath fading.

Figures 1 and 2 [Miki and Hata, 1984] show the effects of post-detection selection diversity improvement on dynamic thermal noise and co-channel interference performance measured in laboratory simulation tests. The experimental results agree closely with theory. Similar performances have been measured in field experiments at 920 MHz in an urban environment [Miki and Hata, 1984].

In a slow Rayleigh fading environment the error probability of a diversity scheme that selects the maximum output of square-law detectors is smaller than that for the scheme that selects the diversity receiver with the maximum S/N ratio [Chyi et al., 1989].

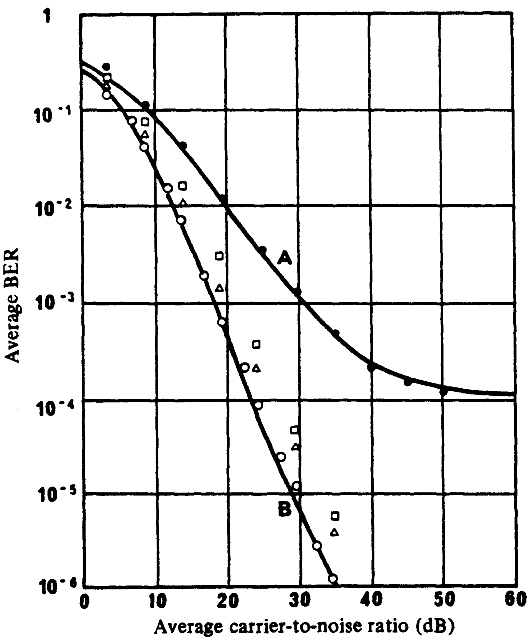

FIGURE 1 – *Thermal noise performance of GMSK transmission with correlated two-branch selection diversity*

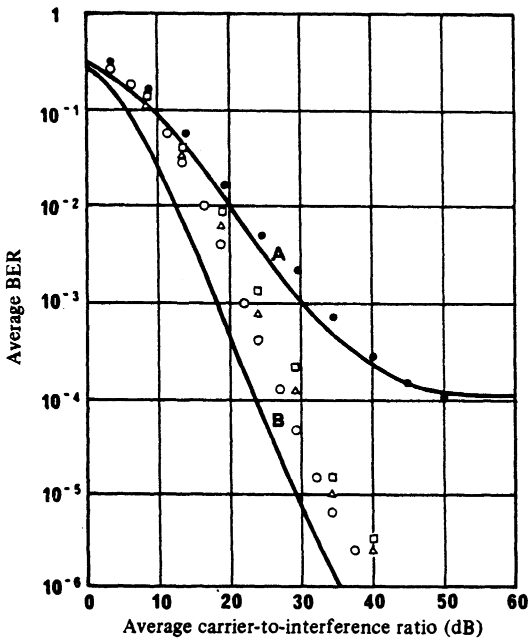

FIGURE 2 – *Co-channel interference performance of GMSK transmission with correlated two-branch selection diversity*

Curves A: no-diversity

B: 2-branch selection diversity

o: $\rho = 0$

△: $\rho = 0.5$

□: $\rho = 0.8$

16 kbit/s – GMSK ($B_bT = 0.25$)

cos $2T$ differential detection

Fading rate 40 Hz

2.5 Improvements with interleaving and coding

Mabey [1978] has shown that cyclic block codes can be used to detect errors in transmission to achieve an arbitrarily low false rate, and the repetition of message can realize a message error rate that is acceptable for relatively short transmissions. However, it has been shown [Freeburg, 1979] that longer transmissions can benefit from the application of error-correcting techniques.

Dorsch [1980] has shown that there is a continuing improvement to be gained by increasing the redundancy of a code and increasing the transmission speed to compensate, up to the maximum practical rate for the channel in question, set by interference criteria. This effect tends to flatten for code rates less than about 1/3.

Also, recent work [Daikoku *et al.*, 1981] indicates that there is a minimum achievable error ratio, mandating the use of error-correcting techniques for messages of any substantial length.

Further improvement can be achieved for both error correction and detection with the introduction of bit interleaving which can also result in reduction in the required overhead.

Error detecting codes may be selected based on the error patterns and the required data block length. Alternatively, the channel may be modelled [Chouinard et al., 1988] and the performance of codes analysed according to [Drukarev et al., 1986]. Soft decision decoding of block codes may yield up to 5 dB improvement [Matsumoto, 1989].

An estimate of the transmission channel state (CSI - Channel State Information) can be suitably used by the decoder, with significant improvement of decoding performance [Hagenauer, 1980].

2.5.1 *Bit interleaving technique*

In the land mobile environment, both random and burst errors occur. Random error correcting codes or single-burst error-correcting codes will not be very effective in combating this problem. Provided the transmitter or receiver is in motion, techniques such as bit interleaving can be used to distribute the errors, thus leading to an effective means for detection and/or correction of errors and at the same time can reduce the overhead required for the code design.

Figure 33 of Report 903-1 (Dubrovnik, 1986), based on field measurements [Towaij et al., 1983] demonstrates the improvement in reduction of the number of errors within a packet of length 31 bits with the introduction of interleaving technique, with an interleaving degree of (32), for digital transmission in the land mobile environment. This behaviour was reflected in the three frequency bands 150 MHz, 450 MHz, and 850 MHz.

2.6 BER degradation due to interference

2.6.1 Co-channel interference

Figures 3 and 4 show measurements of BER against signal level with signal-to-interference ratios, S/I, as a parameter, for both frequency and amplitude modulation with non-fluctuating signals. At low signal levels ($S = -10$ dB(1 μV)) the BER is high due to front end receiver noise. At medium signal levels (e.g. $S = 10$ dB) the BER is strongly dependent on S/I ratio. For example in Fig. 4 with $S/I = 1$ dB the BER $= 10^{-2}$, but with $S/I = 2$ dB the BER drops at least two orders of magnitude. At high signal levels (e.g. $S \geqslant 20$ dB) errors are caused solely by co-channel interference and the error probability [French, 1981] is as follows:

$$P_e = 0.5 \qquad \text{when } S < I$$

$$P_e = 0 \qquad \text{when } S > I \text{ or}$$

$$\therefore P_e = 0.5 \qquad \text{Prob } (S < I)$$

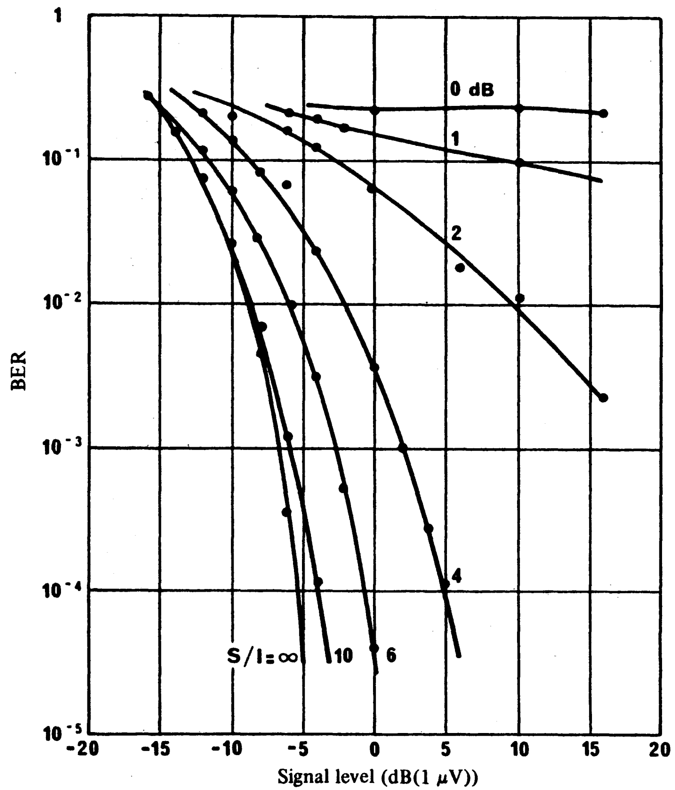

FIGURE 3 – *BER due to co-channel interference, stationary, FM 1200 bit/s FFSK*

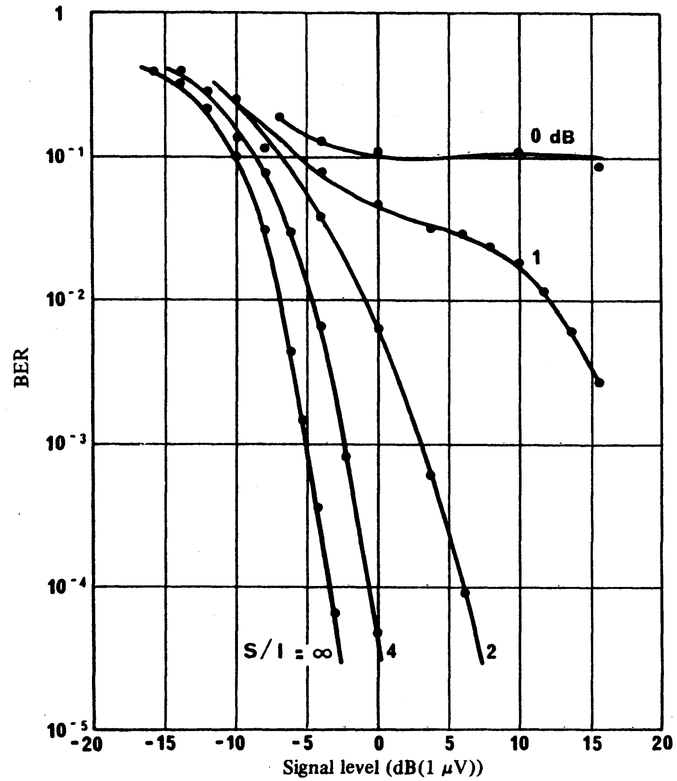

FIGURE 4 – *BER due to co-channel interference, stationary, AM 1200 bit/s FFSK*

[French, 1981] shows the measured BER (on a route where the signal and interference were slightly correlated) compared with the theoretical BER with multipath fading and also with both fading and shadowing, for a moving vehicle. The corresponding error distribution is shown in [French, 1981], where for a probability of 10^{-3} the number of errors in a code word of 64 bits is 8 or more for BER of 10^{-3}, 25 or more for BER of 6×10^{-3} and 32 or more for BER of 30×10^{-3}, in fading conditions with co-channel interference.

2.6.2 Adjacent-channel interference

2.7 The effects of ignition and other man-made noises

2.7.1 Ignition noise

Typical BERs in dense vehicle traffic in the United Kingdom were reported in [French, 1980]. Typically, at a bit rate of 1200 bit/s, a received signal level of +18 dB microvolt pd is required at VHF to maintain BER of 10^{-5}; -4 dB microvolt pd is required at UHF.

For the same received signal level at VHF and UHF the BERs for bit rate of 4800 bits/s are similar [French, 1980]. Typically, +10 dB microvolt pd provides BER in the order of 10^{-3} in both bands. At bit rate of 1200 bit/s, however, there are fewer errors at UHF than at VHF for the same received signal level [French, 1980]. Typically, +0 dB microvolts pd provides BER in the order of 10^{-6} at UHF and only 10^{-3} at VHF.

At the bit rates of 1200 bit/s and less, errors caused by ignition noise alone are normally isolated errors [French, 1980].

3. Digital modulation methods

3.1 Introduction

This section describes digital modulation techniques in applications where analogue speech transmission is not required and the radio equipment can be optimized for digital transmission.

3.2 Requirements

The following characteristics of digital modulation systems are important:

3.2.1 To achieve the required bit error ratio (BER) under fading conditions, good carrier-to-noise ratio (C/N) and carrier-to-interference ratio (C/I) performance is needed.

3.2.2 The technique used must provide high transmission efficiency (in terms of bit/s/Hz) within the constraint of the narrowband assignment.

3.2.3 Use of simplified and miniaturized circuitry is needed to ensure that weight and size are comparable with analogue equipment.

3.2.4 Amplifiers with good power economy should be used but out-of-band radiation must be reduced to a low level.

3.2.5 To minimize the number of errors caused by deep signal fading, rapid bit re-synchronization is required.

3.3 Narrow-band techniques

3.3.1 Channel characteristics - non dispersive

Filter shaped binary phase shift keying (BPSK), quaternary phase shift keying (QPSK) and 16 quadrature amplitude modulation (QAM) are not suitable for non-linear mobile radio channels because the occupied bandwidth requirement is violated. However these modulation methods are applicable to linear mobile radio channels.

Techniques in amplifier design continue to advance, practical linear amplifiers with good power economy are now available [Johnson, 1987]. These amplifiers allow the effective use of linear modulation methods in mobile radio systems. Examples of the bit error ratio (BER) performance of QPSK and 16 QAM under fading conditions are shown in [Adachi, 1989] and [Sampei , 1989].

The signalling schemes suitable for use on these channels can be divided into the two classes of sub-carrier modulation and direct (data) frequency-shift. Sub-carrier modulation is often preferred because of its characteristic of eliminating all low-frequency components in the modulation signal.

3.3.2 Modulation schemes

Frequency shift keying of a suitable sub-carrier is a good choice for low speeds. Gaussian filtered minimum shift keying (GMSK)* and tamed frequency modulation (TFM) of a carrier are suitable for high speed, as they produce a nearly constant envelope and a sufficiently compact frequency spectrum when combined with the low-pass filtering normally present in land-mobile radio equipment. Figure 5 shows the spectra of the modulating signals produced by TFM and GMSK methods.

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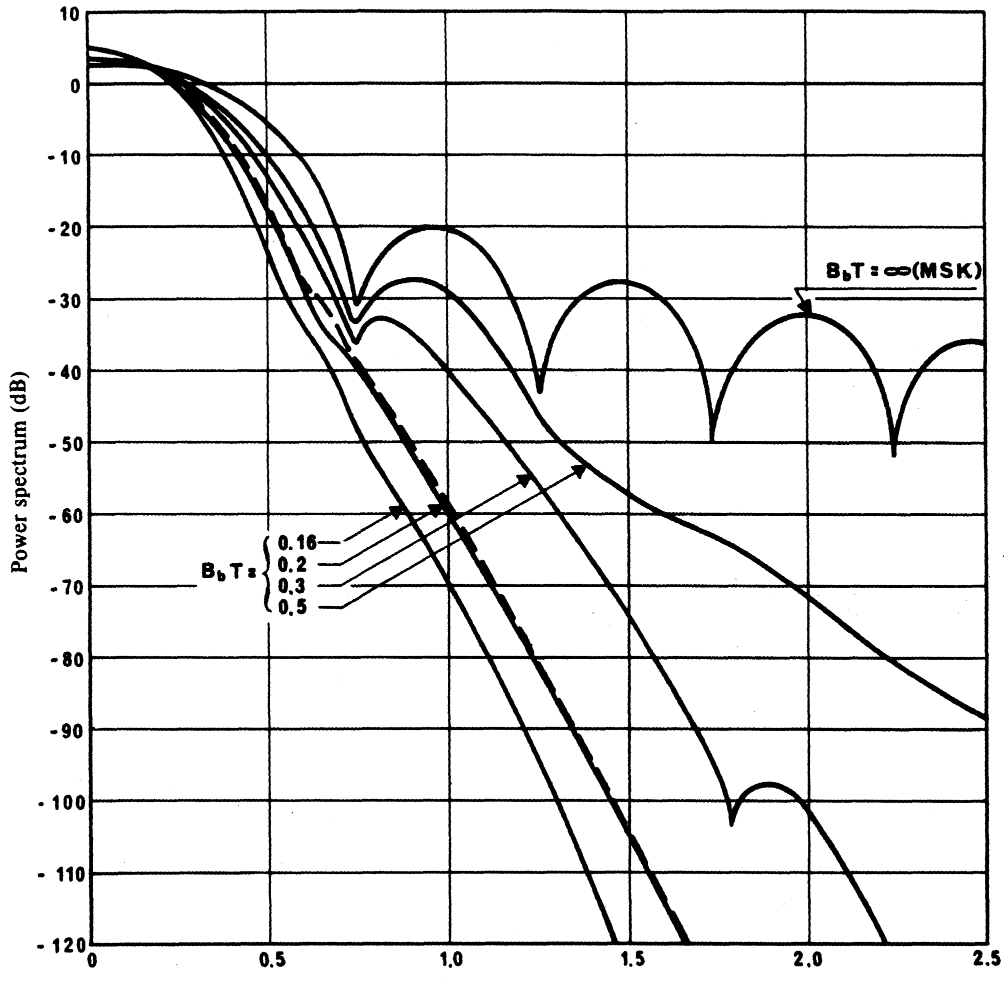
* Also known as Gaussian filtered fast frequency shift keying.



Normalized frequency difference $(f - f_c) / f_b$

FIGURE 5 – *Power spectra of TFM and GMSK*

----: TFM

B_bT: normalized bandwidth of premodulation gaussian bandpass filter (GMSK)

f_c: carrier frequency (Hz)

f_b: bandwidth (Hz)

In general, the narrower the bandwidth of the Gaussian pre-modulation filter, the more compact the output power spectrum may be made. However, bit error ratio performance may be degraded. For this reason, it may be preferred to use direct frequency-shift modulation. The spectrum presented to the modulator in this case is shown in Figure 6, compared with GMSK. Measurements agree with the theory in Fig. 3 of Report 903-1 (Dubrovnik, 1986)

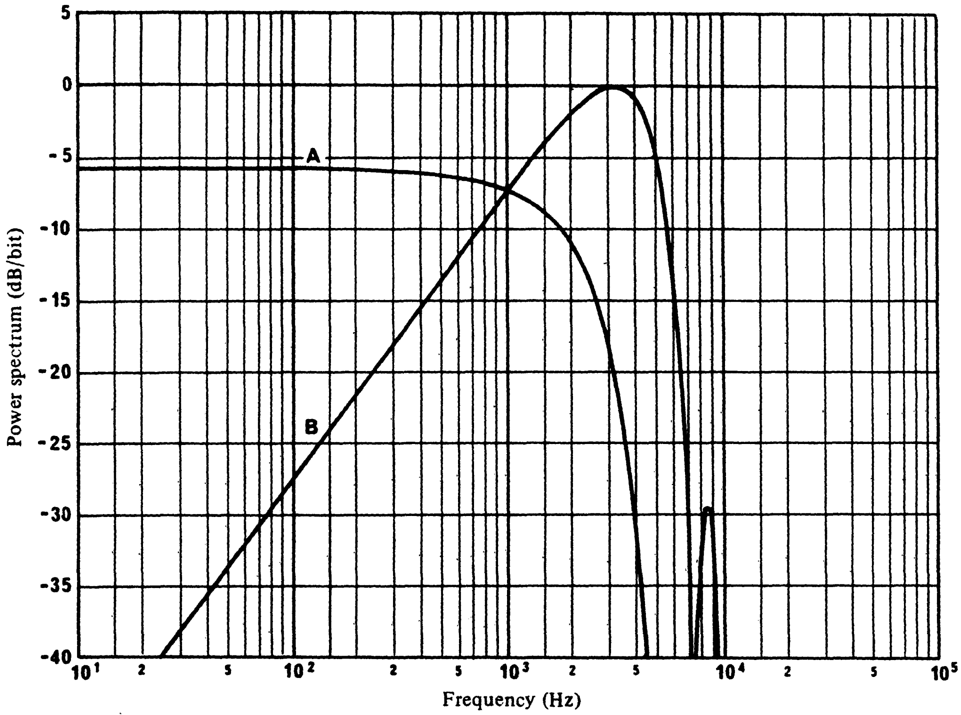


FIGURE 6 – *Power spectra of filtered baseband*

Curves A: baseband spectrum
B: filtered FFSK (GMSK)

Data and GMSK

Data rate: 4800 bit/s

GMSK: 2400 Hz/4800 Hz

B_bT: normalized bandwidth of pre-modulation filter

$B_bT = 1$

Baseband $B_bT = 1/2$

Required E_b/N_0 for the specified BER of 10^{-3} versus B_bT* is theoretically estimated for non-fading conditions as shown in Fig.7, where the GMSK signal is demodulated by orthogonal coherent detection or maximum likelihood detection with differential decoding. The measured results with orthogonal coherent detection are shown in the same figure. The degradations of E_b/N_0 of GMSK ($B_bT = 0.25$) with othogonal coherent and maximum likelihood detections from the ideal binary or quaternary PSK modulation are 1.5 dB and 0.7 dB, respectively. For TFM, the theoretical degradation of a filter-and-sample detector is 1 dB with respect to the ideal binary and quaternary PSK modulation where differential decoding is not adopted [Murota and Hirade, 1981; Muilwijk, 1979].

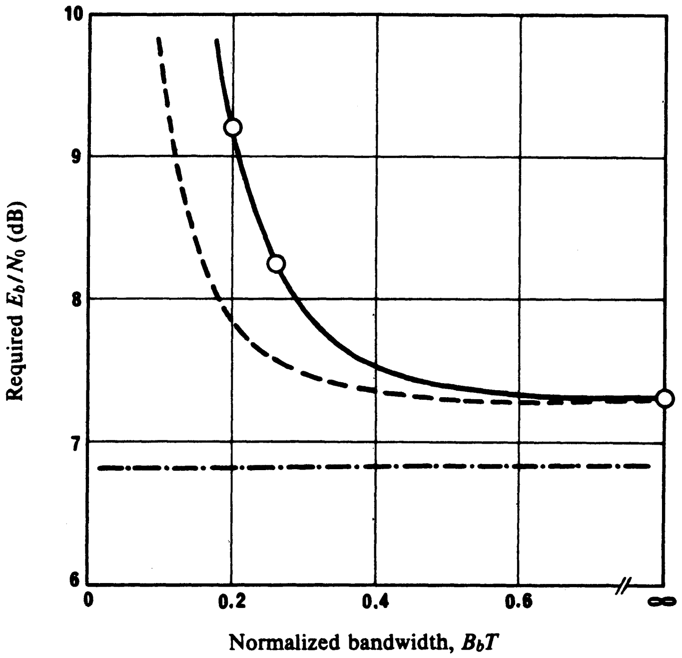


FIGURE 7 – *Required E_b/N_0 versus B_bT*

———— : orthogonal coherent detection with differential decoding
– – – – : maximum-likelihood detection with differential decoding
—·—·— : ideal orthogonal coherent detection
O: measured results for GMSK with orthogonal coherent detection

P_e: bit error ratio (BER) = 10^{-3} non fading
B_bT: normalized bandwidth of premodulation Gaussian low-pass filter
E_b/N_0: signal power per bit noise power density

* B_bT is the normalized 3 dB bandwidth of the filter.

The spectrum space factor F versus B_bT where α relates the received signal power and the distance between base and mobile stations [Murota and Hirade, 1981] is shown in Fig.8 . The minimum value of F is obtained at $B_bT = 0.25$ irrespective of the value of α and the application of diversity. It might be desirable to adopt the GMSK with $B_bT = 0.25$ from the viewpoint of maximizing the spectrum efficiency of digital land mobile radio [Murota *et al.*, 1981]. For TFM the spectrum space factor F is about equal to that for GMSK with $B_bT = 0.25$. The non-linear process of frequency modulation broadens the spectrum substantially as is shown by Fig. 6.

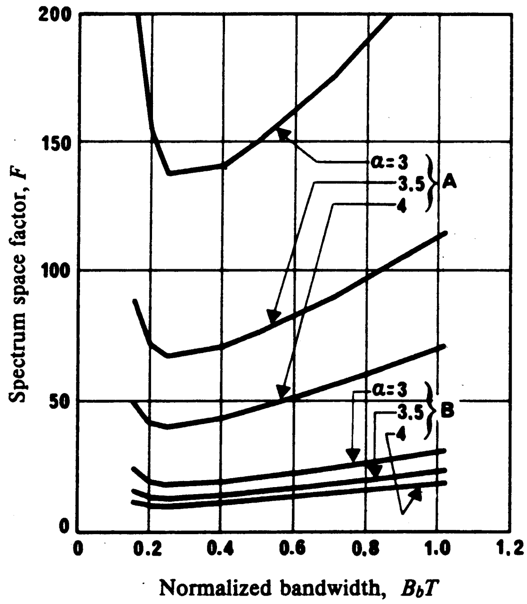


FIGURE 8 – *Spectrum space factor of GMSK*

A: no diversity
B: maximum ratio diversity

In addition to TFM and GMSK, several other digital modulation techniques such as 4-level FM [Akaiwa et al., 1981], PLL-4-PSK [Honma et al., 1980], exist and these are suitable for land mobile radio. Digital phase modulation [Maseng, 1985], 12PM3 [Quacchia et al., 1988], and other types of continuous phase modulation (CPM) [Anderson et al., 1986] ; and transparent tone in band methods [Bateman et al., 1984], exist and are also suitable.

Not only the modulation technique, but also the demodulation technique, determine transmission performance. Expressions for the bit error ratio with differential and discriminator detection of CPM in a fading environment appear in [Svensson et al., 1986], [Elnoubi, 1987 and 1988]; and with differentially coherent detection in [Kaleh, 1989].

3.3.3 Modulation/coding schemes - trellis

Combined modulation and coding, such as trellis coded modulations, can yield improved performance on fading channels.

3.3.4 Channel frequency spacing

Where digital mobile radio systems must co-exist with analogue mobile radio systems, the adjacent channel interference requirements of the analogue system may limit the maximum transmitted bit rate of the digital systems [Constantinou and Towaij, 1981].

The dominant factor in channel spacing is adjacent-channel interference performance; adjacent-channel interference levels for several different modulation schemes and speeds are presented in Table I.

TABLE I – *Measured adjacent-channel interference levels (U/W) (dB) by the method specified by CEPT*

		Channel separation				
		12.5 kHz	20 kHz		25 kHz	
		Deviation	Deviation		Deviation	
		±1.2 kHz	±1.2 kHz	±3.0 kHz	±1.2 kHz	±3.0 kHz
Sub-carrier modulation	600 bit/s FSK	76	> 90	> 90	> 90	> 90
	1200 bit/s GMSK	76	> 90	> 90	> 90	> 90
	2400 bit/s GMSK	72	> 90	> 90	> 90	> 90
	4800 bit/s GMSK	55	> 90	60	> 90	78
Direct FM	4800 bit/s	76	> 90	> 90	> 90	> 90
	9600 bit/s	53 (1)	67 (1)	not available	76 (1)	not available

U/W: Ratio of unwanted to wanted signal level.

(1) ± 2.4 kHz deviation.

Figure 9 shows the adjacent-channel interference performance where both the wanted and unwanted signals are digital. When the normalized frequency difference (the ratio of frequency difference to transmission bit rate) is 1.5, the ratio of unwanted-to-wanted signal level (U/W) is approximately 45 dB.

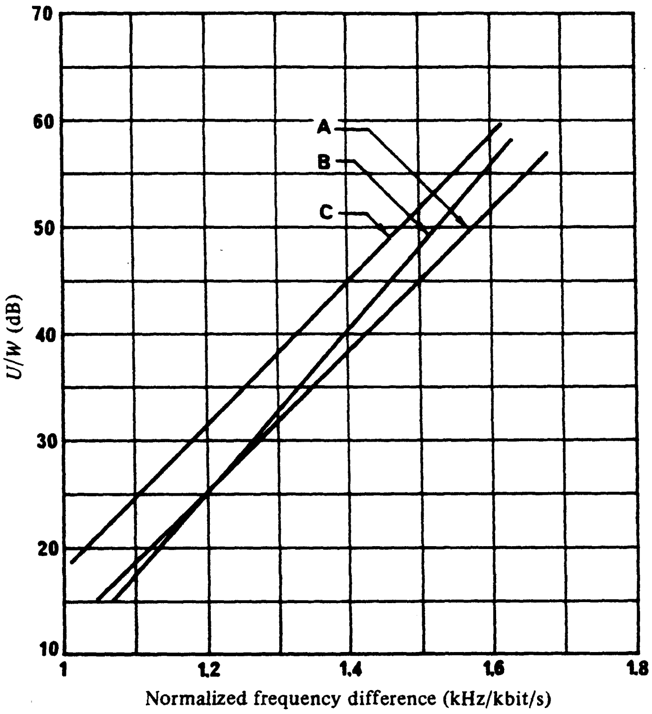


FIGURE 9 – *Adjacent-channel interference performance*

Wanted signal: W-level corresponds to BER = 1×10^{-2}. The signal is modulated with a 9-stage PN sequence

Unwanted signal: U-level corresponds to BER = 1×10^{-2} when desired signal level is 3 dB in excess of W-level. The signal is modulated with a 15-stage PN sequence

Modulation: wanted and unwanted signals are modulated by
A: GMSK;
B: 4-level FM;
C: PLL-4-PSK.

The adjacent-channel interference performance is shown in Fig. 10 where the wanted signal is a digital signal and the unwanted signal is an analogue FM signal. At the normalized frequency difference 1.5, *U/W* is more than 60 dB when the analogue FM signal is modulated by a 1 kHz tone, and approximately 40 dB when the analogue FM signal is modulated by an artificial voice signal.

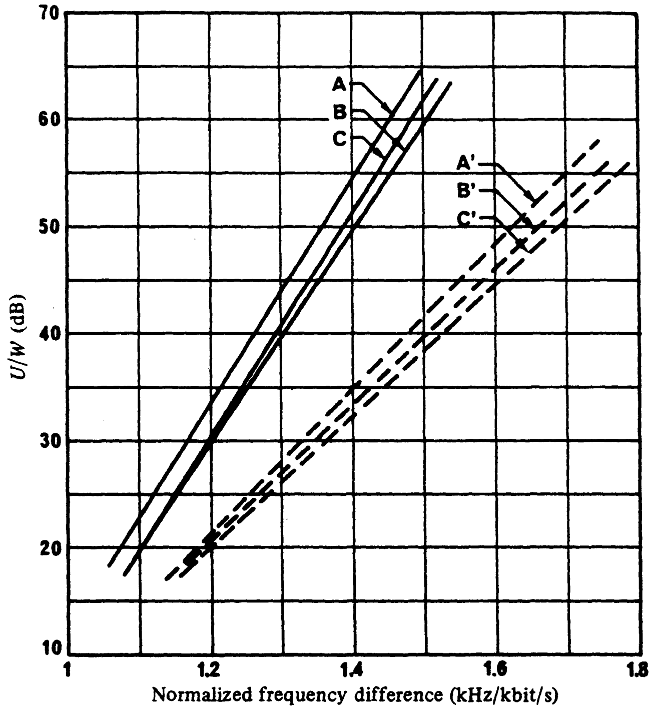


FIGURE 10– *Adjacent-channel interference performance*

Wanted signal: W-level corresponds to BER = 1×10^{-2}. The signal is modulated with a 9-stage PN sequence

Unwanted signal: U-level corresponds to BER = 1×10^{-2} when wanted signal level is 3 dB in excess of W-level. The signal is modulated with:

– a 1 kHz tone at 1.5 kHz frequency deviation (A, B, C);

– an artificial voice signal specified en Recommendation G.227 of the CCITT (A', B', C')

Modulation: wanted signal is modulated by

A, A': GMSK;

B, B' : 4-level FM;

C, C' : PLL -4-PSK;

unwanted signal is modulated by analogue FM.

Figure 11 shows the adjacent-channel interference performance when the wanted signal is an analogue FM signal and the unwanted signal is a digital signal. At the normalized frequency 1.5, *U/W* is approximately 50 dB.

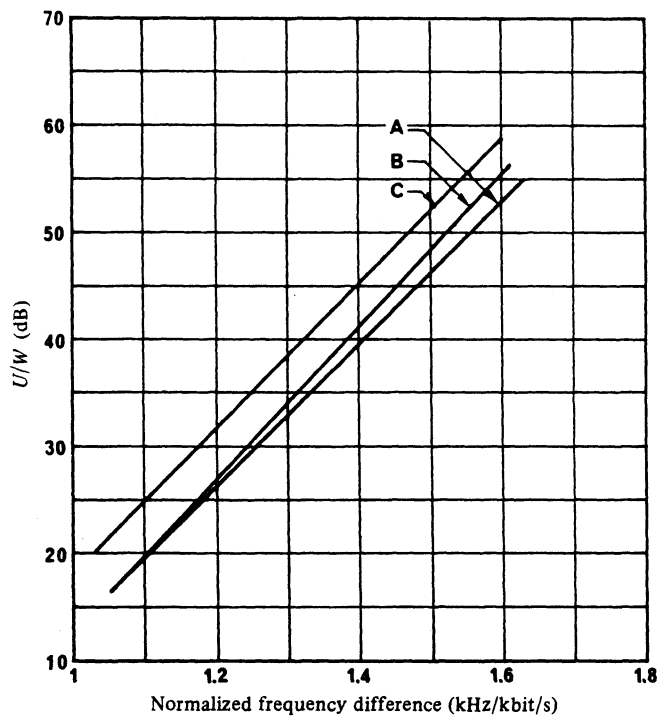


FIGURE 11 – *Adjacent-channel interference performance*

Wanted signal: W-level corresponds to SINAD = 12 dB. The signal is modulated with a 1 kHz tone at 1.5 kHz frequency deviation

Unwanted signal: U-level corresponds to SINAD = 12 dB when wanted signal level is 3 dB in excess of W-level. The signal is modulated with a 15-stage PN sequence

Modulation: wanted signal is modulated by analogue FM, and unwanted signal is modulated by

A: GMSK;

B: 4-level FM;

C: PLL-4-PSK.

These results suggest that channel spacing should be determined based on a value equal to 1.5 times the transmission bit rate, conventional channel spacing, and allowable carrier drift conditions.

In addition, adjacent channel interferences for CPFSK systems [Andrisano et al., 1988] show that suitable values of maximum frequency deviation must be chosen to counteract the combined effects of co-channel and adjacent channel interference. As the frequency offset increases, there is a gradual progression towards favouring the smoothed modulations with lower side lobes [Rhodes et al., 1987]. For small frequency offsets near one-half the bit rate, however, QPSK out performs the other cases, due to its narrower main lobe.

3.4 Wideband techniques

3.4.1 *Spaced frequency correlation function*

The performance of a digital communication system on land mobile radio channels is affected, among other things, by the coherence bandwidth [Bello and Nelin, 1963; 1964] over which random variations on the channel are statistically correlated. The value of correlation that defines the coherence bandwidth B_c is a function of the transmitted symbol shape and the modulation technique employed.

A cumulative distribution function for coherence bandwidth was calculated from spaced frequency correlation functions for data collected during experiments in the 900 MHz mobile radio band in the urban centre of Ottawa, Canada [Bultitude, 1983, 1989]. For these calculations, the coherence bandwidth was defined as that bandwidth at which the envelope of the spaced frequency correlation function dropped below a value of 25%. Since computed correlation functions were asymmetrical with respect to the centre of the measurement bandwidth, the minimum of the positive and negative bandwidth values for each channel measurement was used in the distribution calculations. It was found that the function, when plotted on probability paper was approximatively piecewise linear, with a significant increase in slope for bandwidths greater than 8 MHz Fig. 14 of Report 903-1 (Dubrovnik, 1986). Coherence bandwidths were 0.8 MHz, 2.8 MHz, 7.5 MHz, and 9 MHz with 90, 50, 30 and 10 percentile points of the distribution, respectively. Reduction in correlation across a channel bandwidth is the result of time dispersion. It can be considered that inter-symbol interference on a digital system operating over a channel would be negligible in a city similar to Ottawa for the listed percentages of urban area if transmission bandwidths are maintained below 2.5% of the given bandwidths. Nevertheless, systems may be designed to include equalization, diversity, or other features to counteract or make use of the time dispersion.

For a given signalling pulse waveform, the effects of intersymbol interference on the average and irreducible error probabilities are characterized by the normalized r.m.s. multipath spread [Garber, 1988].

Additional data for microcellular urban mobile radio channels at 910 MHz appears in [Bultitude, 1989]. The relationship between multipath delay characteristics and bit error rate performance depends on the modulation scheme used [Takechi et al., 1988]. The bit error rate with double phase shift keying [Ariyavisitakul et al., 1987] is significantly reduced compared to the error rate of conventional BPSK.

Advanced equalization techniques, like maximum likelihood sequence estimation (MLSE), implemented in adaptive form, can provide a significant performance improvement in a dispersive multipath channel [D'Avella et al., 1989].

4. DATA TRANSMISSION OVER ANALOGUE SPEECH CHANNELS

4.1 Introduction

This section is concerned with data transmission over mobile radio channels originally designed for analogue speech to provide new facilities in dispatch* systems like digital selective calling and status reporting or for end-to-end transmission in systems interconnected to the PSTN.

4.2 Accommodation of telematic services

It is desirable that mobile units which are interconnected with the PSTN should be able to send and receive services such as Videotex, telex and facsimile.

However, further study is needed:

- to reduce the error rates caused by fading, ignition noise and co-channel interference to acceptable levels;
- to establish that the protocols defined in the appropriate CCITT Recommendations can be used effectively in mobile radio systems for both stationary and moving vehicles;
- to ensure that the transmission of data messages should be completed in the minimum channel airtime possible.

4.3 Interfacing data terminals to mobile radio equipment

The principal problem is the proper specification of the interface between the data device and conventional mobile radio equipment primarily designed for speech communications but which with appropriate modifications can become suitable for data transmission.

4.3.1 *Constraints of voice circuits on data speeds*

Bit rates up to 3600 bit/s are presently being used in Canada for transmission over the existing 30 kHz VHF and 25 kHz UHF land mobile channels. One method of transmission is the use of the Miller code [Lindsey and Simon, 1973] over sub-carrier/FM systems passing through the speech processing circuitry (pre-emphasis/de-emphasis). The appropriate equalization circuits are used to minimize the potential inter-symbol interference that will otherwise severely reduce the effective data transmission rates [Constantinou and Towaij, 1981].

One method of transmitting data over voice channels at 1200 bit/s uses a sub-carrier modulation scheme called FFSK, with a logical '1' transmitted by one cycle of a 1200 Hz sinwave and a logical '0' by one and a half cycles of an 1800 Hz sinwave. The scheme is in use in FMS (status message transmission system of the police in the Federal Republic of Germany), in ZVEI (digital selectocall system of the Federal Republic of Germany) and in other systems.

Error performance with this modulation scheme is given in section 6. The generation and demodulation of the data signal is possible with a single microprocessor [Stein and Gibson, 1981].

To realize high bit rates it may be necessary to modify the speech circuits or bypass them and apply the sub-carrier data signal directly to the modulator and recover it directly from the demodulator.

4.3.2 Modulation method and coding

The mobile radio equipment interface should be transparent to the sub-modulation method and coding. There may be benefits from using standards such as CCITT Recommendations that have been developed for land line applications but further study is needed to ensure that these standards can be applied practically to the mobile radio channel. The sub-modulation method selected should be applicable to all expected radio modulation schemes (FM, PM, AM and possibly SSB).

4.3.3 *Control signals*

Since many users share a radio channel, data transmissions must be inhibited when the channel is occupied. Therefore, a carrier sense (COR) signal is needed from the radio receiver to inhibit the data modem.

* For the purpose of this text "dispatch system" has the meaning: a radio system used to control the operation of a fleet of mobiles, such as aircraft, taxis, police, etc.

There can be widely varying time delays (10-200 ms) from application of the transmitter turn-on signal until receipt of the COR signal at the receiver. Data modems must therefore, at present, have selectable delays which introduce inefficiencies. This would be overcome if the maximum time delay were set at a reasonably achievable level (e.g. 10 ms).

4.3.4 *Data modem/radio interface*

The following characteristics are important:

- transmitted signal spectrum at peak deviation;
- data de-emphasis characteristics;
- data rate and bandwidth;
- impedances and signals levels;
- undetected error rate;
- range of transmitter turn-on times;
- period to acquire modem synchronization;
- control signals (transmitter on, squelch, mute).

4.3.5 Decision feedback equalizers

The performance of the bipolar scheme in Recommendation 623 is analysed in [Korn, 1987]. For binary signalling applied directly to an FM modulator, a decision feedback equalizer [Adachi, 1987; 1988] may be required to compensate for the dc content of the binary signal.

4.4 **Integration of analogue speech and data**

Canada has made analytic and simulation studies of the integration of analogue speech and data, by transmitting data packets in the gaps between consecutive voice calls, or alternatively in all the gaps in a voice conversation. In public mobile telephone channels in Montreal, the measured fraction of time during which the mobile talker is silent, can reach about 70% [Cohen and Haccoun, 1980].

Traffic simulation studies [DaSilva *et al.*, 1980; Callendar, 1981] have shown that most of this idle time may be utilized for data packet transmission.

Further studies are needed to investigate voice intelligibility, packet throughput and mean delay, the design of equipment such as speech detectors and fast acquisition circuits, and the problems caused by channel impairments such as fading and co-channel interference.

5. Digitally encoded speech

5.1 **Introduction**

This section is concerned with digital speech transmission over digital channels.

5.2 **Quality of digitized speech systems**

5.2.1 *Voice coding*

In mobile radio communication, low bit-rate voice coding is desirable to achieve efficient spectrum utilization. Although techniques such as ADM (adaptive delta modulation), ADPCM (adaptive differential pulse code modulation), APC-AB (adaptive predictive coding with adaptive bit allocation), PARCOR (partial autocorrelation), and SBC (sub-band coding) may be suitable, it is necessary to study their quality in a fading channel. The quality will also depend upon other factors, such as synchronization methods and error correcting methods (partial or global) – if any – associated with the speech coder.

A coding technique RPE-LTP (Regular Pulse Excited Linear Prediction Coder with Long Term Prediction) has been specially designed in order to achieve robustness to transmission errors and a quality close to that of the PSTN (Natvig and de Brito, 1987).

5.2.2 *Use of digitized speech systems*

In the United States of America there are digitized speech systems successfully operating within the limits of a 25 kHz channel assignment. These systems use a data rate of 12 kbit/s and a frequency deviation of ± 4 kHz.

From measurements made in the United States using rhyme tests these systems appear to have comparable intelligibility with standard FM systems.

It has been found that the sensitivity of receivers using this modulation is 6 to 8 dB less than an analogue receiver accepting normal 16K0F3E emissions.

Figures 12 and 13 show a comparison between the mean opinion scores for analogue FM transmission and digital voice transmission using GMSK coherent detection with a 16 kbit/s ADM Codec in conventional channel spacing of 25 kHz [Kinoshita *et al.*, 1984]. These figures show that digital voice transmission will be applicable for some systems requiring high security, but with the penalty of quality degradation.

The mean opinion scores for digital voice transmission using GMSK discriminator detection [Hirono *et al.*, 1984] with a 16 kbit/s APC-AB Codec and a 2.4 kbit/s Vocoder are also shown in Fig. 12. A 16 kbit/s APC-AB system achieves the quality comparable with an analogue FM system. A 2.4 kbit/s Vocoder system provides approximately the same quality as a 16 kbit/s ADM system, and is suitable for narrow-band communication.

The RPE-LTP Codec operates at 13 kbit/s, and at that bit rate, its quality, in terms of Mean Opinion Scores, is 4 to 4.3 under good transmission conditions. Subjective testing (Natvig, 1988) experiments carried out in a number of countries have shown that, averaged over a great variety of test conditions, including the insertion of random errors at a BER of 10^{-3} and 10^{-2}, the quality of the RPE-LTP was better than an analogue FM reference system (Mean Opinion Score difference greater than 1.5). During the design of a particular system using the RPE-LTP Codec it was found that it was necessary to protect only 70% of the bits produced by the Codec with an appropriate channel coding technique.

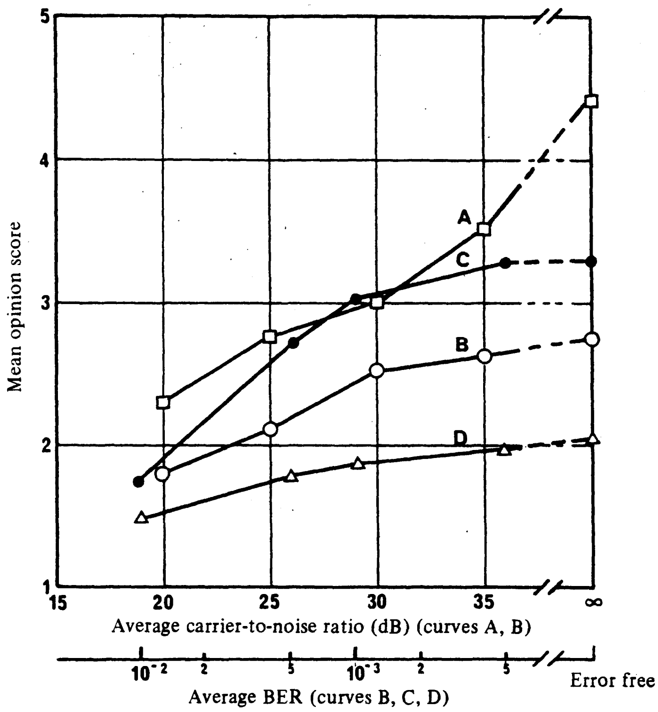


FIGURE 12 – *Comparison of digital and analogue voice transmission; result for thermal noise*

Curves A:	analogue	FM deviation: 3.5 rad/1 kHz
B:	digital	GMSK with coherent detection 16 kbit/s ADM
C:	digital	GMSK with discriminator detection 16 kbit/s APC-AB
D:	digital	GMSK with discriminator detection 2.4 kbit/s Vocoder

Fading rate 20 Hz

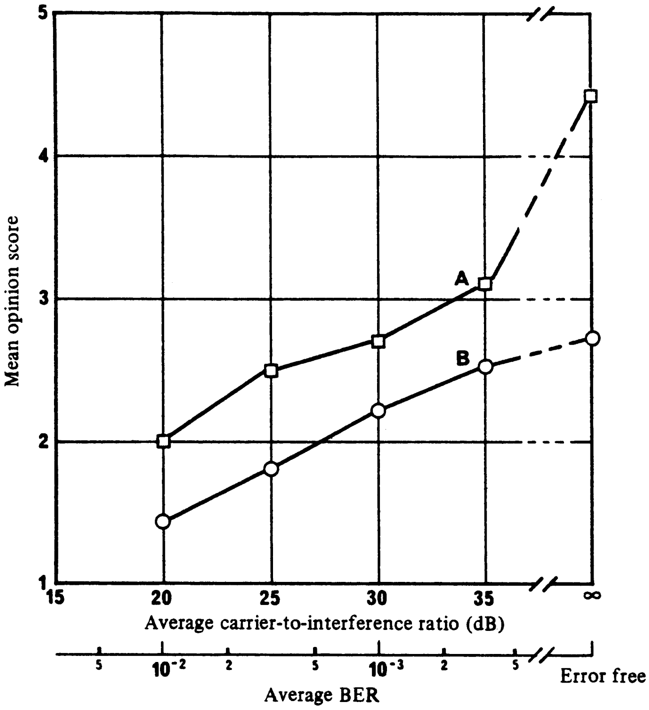


FIGURE 13– *Comparison of digital and analogue voice transmission; result for co-channel interference*

Curves A: analogue FM
deviation: 3.5 rad/1 kHz

B: digital GMSK with coherent detection
16 kbit/s

Fading rate 20 Hz

5.3 Voice privacy

Various levels of privacy can be achieved using commercially available speech privacy techniques. A relative ranking of these different systems in terms of increasing difficulty to intercept is:

- clear speech with verbal code;
- noise masking of analogue speech;
- frequency inversion of analogue speech;
- frequency hopping;
- digitized speech;
- rolling code band splitting of analogue speech;
- linear digital voice scrambler;
- non-linear digital speech.

5.4 Transmitter emission limitations

In the United States of America increasing use of digitized speech signals prompted the adoption of uniform emission standards for digital and analogue land mobile radio in the private mobile services. The principle used in deriving these standards was to restrict digital radio signals to the same emission bandwidths required for analogue speech since both types must occupy and share the same channel space.

Therefore it is necessary to review existing CCIR Recommendations and Reports to ensure that the emission limitations required for operation of digitized voice within the bandwidth occupied by an equivalent analogue signal are properly covered.

5.5 Efficiency of data compared with speech

Typical speech messages are transmitted at a rate of about 150 words/minute [Kelly and Ward, 1973]. In coded character form, this corresponds to a data rate of about 90 bit/s (based on 6 characters/word and 6 bits/character).

Bit rates of presently available data message systems significantly exceed the equivalent speech message rate of 90 bit/s. Such transmissions also normally include the transmitter identification (ID) with each message and this greatly improves the efficiency compared with speech.

Improvement factors (in air time utilization) are expected to vary between 5 and 15 depending upon the detailed system design, propagation effects, retransmission requirements and polling delays [Parness, 1975].

6. SIGNALLING AND SUPERVISION IN PUBLIC MOBILE TELEPHONE SYSTEMS

6.1 Introduction

This section is concerned with various aspects of data transmissions to establish and supervise telephone calls and channels assignments in public mobile telephone systems.

6.2 Transmission channel and modulation techniques

It is important to consider the following parameters, if applicable, in making a choice of modulation technique and data speed:

- available baseband bandwidth;
- occupied bandwidth of the data modulated radio signal;
- tolerance of group delay distortion;
- tolerance to frequency offset;
- tolerance to interference (e.g. impulsive noise, fading).

The telephone channels that link the telephone exchange with the radio base stations affect the data signalling mainly by introducing attenuation distortion, group delay distortion and frequency offset resulting from the use of voice frequency carrier systems.

6.2.1 In the Nordic mobile telephone system (NMT) it was found that the available bandwidth of the transmission channel, using a 25 kHz FM radio channel (16K0F3E), was approximately 2200 Hz (500-2700 Hz). A subcarrier modulation method was selected using a data transmission rate of 1200 bit/s and fast frequency shift keying (FFSK).

For the Nordic mobile telephone system (NMT) measurements were made with results of less than ± 5 Hz for the overall frequency offset.

6.2.2 In the advanced mobile phone service (AMPS) of the USA a 30 kHz radio channel (30K0F3E) is used. A data transmission rate of 10 kbit/s on dedicated channels was chosen with a peak frequency deviation of ± 8 kHz. The system uses direct binary frequency shift keying (FSK) of the carrier with discriminator detection (40K0F9X).

The AMPS transmits data on a fully dedicated digital channel called the "set-up" channel used for paging and access functions and also transmits data on the talking (voice) channel once a call has been established. In this instance the voice channel is interrupted or "blanked" and the data transmission is sent in a "burst" lasting not more than 100 ms. This "blank-and-burst" message is used to effect a channel frequency change in the mobile unit in order to accomplish "hand-off" as the mobile moves from one cell site's coverage to another.

6.2.3 In the Public Land Mobile Radio Telephone System in Japan a 25 kHz FM radio channel (16K0F3E) is used. A 300 bit/s data transmission rate with peak deviation of ± 4.5 kHz was selected to give priority to signalling reliability. The system uses equivalent FM with discriminator detection because modem circuits are used for both voice and data.

6.3 Coding

The choice of coding in public mobile systems must take account of the need to establish telephone calls reliably in the shortest possible time. The power spectrum should also be considered. This usually results in different coding requirements for data transmissions for signalling and supervision of mobile telephone than those used in dispatch radio systems or for telematic services.

6.3.1 For the NMT a convolutional code was selected because it allows arbitrary message lengths, continuous decoding of the messages and is fairly straightforward to implement. Convolution codes are characterized by the maximum length of burst that can be corrected and the error-free zone that is required between two bursts (guard-space).

In the NMT a burst correcting capability of 6 bits was chosen and the guard-space is 19 bits using the Hagelbarger code. This code was felt to be optimal for the NMT regarding the relationship between the burst error correcting capability and the guard-space [Hagelbarger, 1959].

6.3.2 In the AMPS system a biphase (Manchester) bit encoding format was adopted. The peak of the power spectrum produced by this data transmission is well above the voice band. This separation is an advantage in a system transmitting both voice and signalling data on the same channel.

To combat burst errors caused by multipath fading, all ditigal data messages are encoded and repeated several times at the source. The coding used on all radio channels is a shortened (63 : 51) Bose-Chauduri-Hocquenghem code.

In the forward (base-to-mobile) set-up channels, all data messages are interlaced, encoded and repeated five times and a bit-by-bit 3 out of 5 majority vote is taken at the data receiver to determine the best-guess detected message to send to the decoder. In the blank-burst mode over the voice channel, the data messages are repeated 11 times in the forward direction but only 5 times in the reverse direction using bit-by-bit majority vote for both cases. The primary reason for the difference between the 11 and 5 message repeats is that the message from base site to mobile is usually received under poor S/I conditions and is considered a critical function since false interpretation results in a mishandled call [BSTJ, 1979].

6.3.3 In consideration of the changing rate of mobile speed and passage length, the system in Japan uses block codes as error correcting codes for random errors and repeated control signal transmissions for burst errors to combat errors caused by multipath fading. Moreover, diversity effect using a simultaneous multitransmitting technique for set-up channels increases signalling reliability.

Error correcting codes are a shortened (63 : 51) BCH and a shortened (15 : 11) BCH which correct 1 bit error in a frame. Control signal transmissions are repeated twice for paging channels and 4 times for access and voice channels. A Manchester code, which has non-direct current and code redundancy, was selected in consideration of radio channel transmissions and for prevention of erroneous signalling performance due to voice signals.

6.4 Signalling reliability

In the NMT it was found possible to express the requirements for signalling reliability in terms of the RF input signal level. The requirement was established at $\geqslant 0.9$ for input levels $\geqslant 0$ dB(μV) e.m.f. without fading and $\geqslant 10$ dB(μV) e.m.f. with fading.

Data derived from operational experience with 1200 bit/s FSK signalling, a shortened (63 : 45) BCH code with error detection only and a 12 dB reduction in transmitted power towards the mobile when signalling, as compared to the level for voice transmissions [Callendar, 1981] are shown below.

Total successful calls in sample period: 10 533.

TABLE II

Number of retransmissions	0	1	2	3	4	5	6
Number of calls	9619	615	175	64	30	24	6
Percentage	91.3	5.8	1.7	0.6	0.3	0.2	0.05

The handshake protocol for the majority of these calls involved one base to mobile data message. However each individual data message can be repeated, if required, up to six times.

Since 91.3% of these calls required no retransmissions, it was concluded that, in this application with reduced signalling power and no forward error correction, simple ARQ is sufficient for the 8.7% of calls in which retransmissions are required. Correction of even single bit errors in the mobile unit could be expected to significantly reduce the number of retransmissions required.

6.5 **Major system characteristics**

Major system characteristics are contained in Report 742.

7. FORMATS FOR DATA TRANSMISSION

7.1 **Introduction**

This section gives details of some of the data formats that are being used in the land mobile service.

7.2 **United Kingdom preferred format**

The format is preferred for selective calling, status reporting, precoded messages, vehicle location, monitoring and supervisory systems, direct dialling, control in trunked systems and for mobile terminals (printers and displays).

7.2.1 *Format definition*

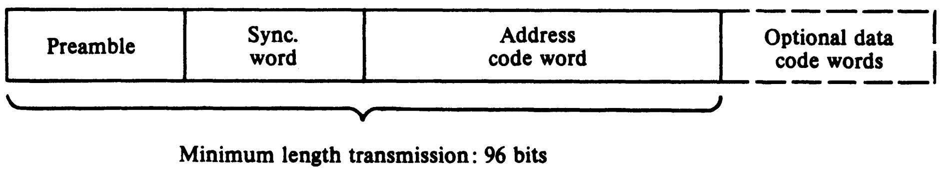


FIGURE 14 – *The format*

7.2.1.1 *Preamble*

16 or more bits "1010 . . . 10" ending with 0.

7.2.1.2 *Synchronization word*

Every message begins with:

Bit No.	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Bit Value	1	1	0	0	0	1	0	0	1	1	0	1	0	1	1	1

(Bit number 1 is transmitted first.)

FIGURE 15 – *Synchronization word*

7.2.1.3 *Code words*

All code words are of 64 bits (including 16 check bits). Short messages consist of a single address code word (which includes some data); longer messages have an address code word followed by data code words.

7.2.1.4 *Address code word*

Bit No.	1	2 8	9 48	49 64
Number of bits	1	7	40	16
	"1"	user's identity	addresses + data	check bits

FIGURE 16 – *Address code word structure*

Bits:
- 1: always "1" to indicate address word
- 2-8: user's identity
- 9-20: addressee's identity (i.e. to) } optional
- 21-32: addressor's identity (i.e. from) } optional
- 33-48: data
- 49-64: check bits (see § 7.2.1.7)

7.2.1.5 *Data code word*

As many as are needed for the message.

Bit No.	1	2 48	49 64
Number of bits	1	47	16
	"0"	data	check bits

FIGURE 17 – *Data code word structure*

Bits:
- 1: always "0" to indicate data
- 2-48: data
- 49-64: check bits (see § 7.2.1.7)

7.2.1.6 *Character sets*

Binary coded decimal (BCD) coding can be used for the addressee and addressor identities. The character sets for messages are BCD for numeric-only messages, and the ISO 7-bit data code for alphanumeric messages. Characters are transmitted in reading order and least significant bit (b_1 in ISO code) first.

7.2.1.7 *Encoding and error checking*

The information bits 1-48 are the coefficients of a polynomial having terms from x^{62} down to x^{15}. This polynomial is divided modulo 2 by the generating polynomial $x^{15} + x^{14} + x^{13} + x^{11} + x^4 + x^2 + 1$. The fifteen check bits, code word bits 49-63, correspond to the coefficients of the terms from x^{14} to x^0 in the remainder polynomial.

The final check bit of the code word (bit 63) is inverted to protect against misframing in the decoder.

One bit is appended to provide an even parity check of the whole 64 bit code word.

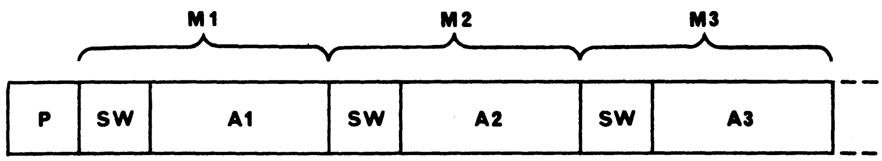
7.2.1.8 *Concatenated messages*

Figure 18 illustrates how several messages may be sent in one transmission.

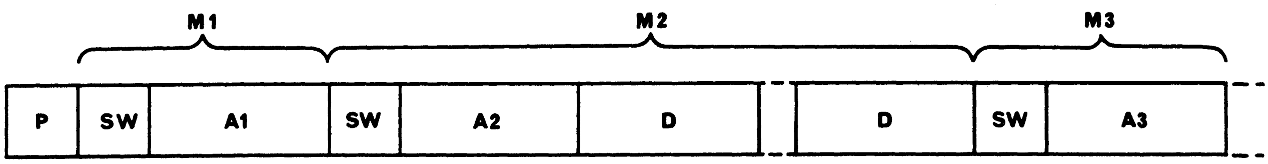
7.2.2 *Format design*

An error detecting code (which has a distance of 5 bits) was chosen rather than an error correcting code because it has an adequate performance and a simple, fast decoder.

The format does not rely on a data operated squelch circuit to prevent false messages by inhibiting decoding at low signal levels. A low false rate is obtained by coding alone.



a) Short messages



b) Mix of short and arbitrary length messages

FIGURE 18 – *Concatenated messages*

P: preamble
SW: synchronization word
A: address code word
D: data code word
M: message

7.2.2.1 *Synchronization word*

The use of a synchronization word is the most efficient method of identifying the start of each message, establishing code word framing and ensuring a low false call rate by inhibiting code word decoding at high bit error ratios, without the need for a signal squelch circuit.

The synchronization word must satisfy the following criteria:

- it must have a good success rate so that the messages which follow are not missed;
- the success rate should be about equal to the success rate in decoding address code words. A suitable synchronization word is then 16 bits;
- the synchronization word should have good correlation properties when it is preceded by preamble so that it is not decoded spuriously during the preamble, and so a preamble of 15 bit with an additional bit is used;
- finally, the synchronization word must provide a high security against false messages.

7.2.2.2 *Error detecting code*

To avoid false messages caused by misframing, the format uses a coset code [Peterson and Weldon, 1972]. Inverting the final bit in the code word is sufficient to ensure that valid code words do not appear for misframing by up to 14 bit positions.

Because a synchronization word may be found falsely at the end of a preceding codeword it is necessary to label data code words with a flag bit to distinguish them from address code words.

7.2.2.3 *Performance with a steady signal level*

The successful message probability P_s and the false message probability P_f have been calculated for a steady signal level in terms of the bit error ratio p, assuming independent errors.

The successful message probability is $P_s = (1 - p)^{80}$ which is 80% at a bit error ratio of $p = 2.8 \times 10^{-3}$.

The false message probability has been calculated as the probability that errors cause a transmitted address code word to be decoded as a different address code word.

$$P_f \simeq P(0,s) \cdot P(\geqslant d,n) \cdot 2^{-r} \qquad (1)$$

$P(0,s)$ is the probability that the s bit synchronization word ($s = 16$) is received error free. For independent errors $P(0,s) = (1-p)^{16}$.

$P(\geqslant d,n) \cdot 2^{-r}$ is the conventional expression for the false rate of a cyclic code [Lucky, Salz and Weldon, 1968] where d = minimum distance of the code ($d = 5$), n = code word length ($n = 63$), and r = number of check bits in a code word ($r = 15$). $P(\geqslant d,n)$ is the probability that an n bit word contains d errors or more.

For independent errors

$$P(\geqslant d,n) = \sum_{i=d}^{n} \binom{n}{i} p^i (1-p)^{n-i}$$

Because the code guarantees detection of all odd numbers of errors, $P(\geqslant d,n)$ was evaluated for even numbers of errors only (i = even integer).

P_f is plotted in Fig.19 which shows that the false message probability is less than 2×10^{-6} per transmitted message.

The values of P_f calculated apply to mobile to base transmissions where the base decoder accepts any valid code word. The false message rate will be lower when some code words remain unused. For base to mobile transmissions the false message probability will be lower because a mobile decoder will only accept messages bearing its own address.

7.2.2.4 *Field measured performance*

Measurements made on a route which had Rayleigh fading and some shadowing (σ = 4 dB) with a vehicle speed of about 50 km/h, without ignition noise present, are given in Fig.20.

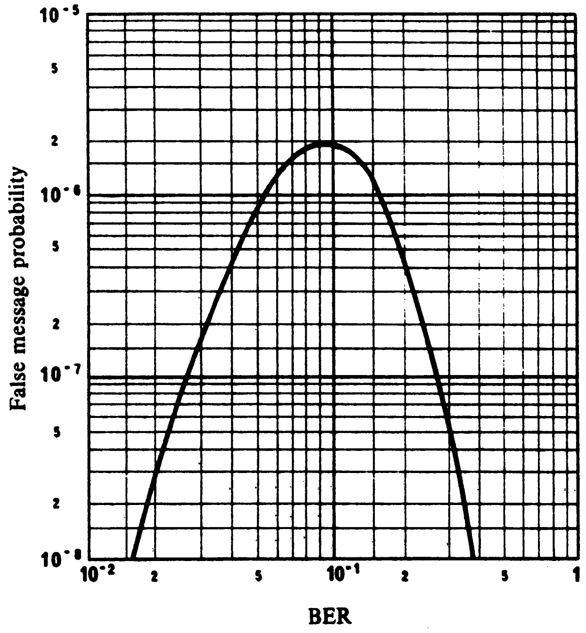


FIGURE 19 – *False message probability with a steady signal*

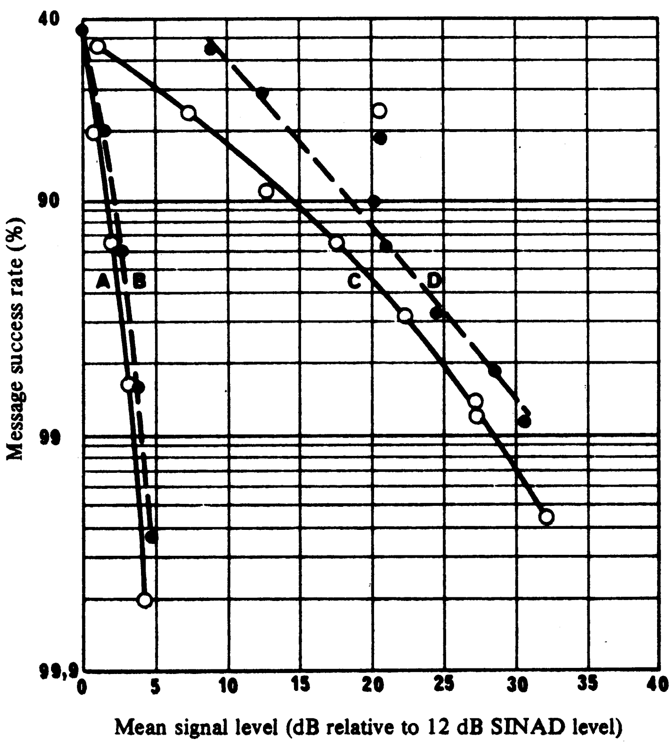


FIGURE 20 - *Field measured performance*

All curves: 1200 bit/s FFSK

A: stationary at 165 MHz FM
B: stationary at 465 MHz FM
C: moving at 50 km/h at 165 MHz FM
D: moving at 50 km/h at 465 MHz FM

7.3 Alternate characteristics preferred in other countries

In France the following synchronization word is preferred:

Bit number	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Value	1	0	1	1	0	1	0	0	0	0	1	1	0	0	1	1

It has the following properties:

– the minimum number of errors for which there may be a false synchronization when receiving the preamble followed by the synchronization word is 7. This number is almost independent of the preamble chosen (all "0", all "1", alternating "0" and "1");

– under these circumstances when using this word, synchronization is achieved if one bit of the synchronization word has been received with one error.

8. Multiple access

This section is concerned with the multiple-access protocols for digital transmission on mobile radio channels and their relationship to the performance specification in section 1. In general, the system response time specification for data messages (not including supervisory, control or emergency messages) may be longer than for voice messages.

8.1 Traffic models

8.1.1 Data only

For most practical systems, data traffic may be modelled as Poisson arrivals (e.g. for dispatch messages) or periodic arrival at regular intervals (e.g. for vehicular monitoring and location). Both traffic models may occur on the same network. The message length distribution (call holding time) may be modelled as fixed or exponential. The traffic load in Erlangs may be different on the inbound (mobile-to-base) channels and the outbound (base-to-mobile) channels.

8.1.2 Data and voice

Voice traffic may be modelled as Poisson arrivals, where the voice message length (call holding time) is on the average longer than that of data messages.

8.2 Multiple access techniques

Multipoint-to-point networks are considered in which many mobile users in a particular geographic area or cell communicate with a single base station in that cell. Thus the mobile users must share access to the inbound channels to communicate with base station.

8.2.1 Fixed assignment

A TDMA or polling type fixed channel assignment scheme is appropriate for a periodic traffic model or channels operating near maximum capacity.

8.2.2 Random assignment

On non-fading channels where all received packets are of the same strength, the maximum throughput of a slotted ALOHA random access scheme is approximately 37% [Gallager, 1985]. On fading channels and where the receiver has a capture effect, the maximum throughput of a slotted ALOHA may be doubled to approximately 65% by employing error correcting codes on the packet address [Habbab, et al., 1989].

8.2.3 Carrier sense multiple access schemes

The throughput of carrier sense multiple access schemes is limited by the time to detect the presence of a packet, during which another packet may begin and thus cause collisions. If this time is zero, then the throughput may be up to 100% of the channel capacity. For practical radio and modem equipment, this time may be from 5 to 50% of the message length; thus the maximum throughput may be reduced significantly.

On Rayleigh fading channels, the throughput of CSMA is further reduced, because packets may not be detected reliably when subject to fading.

8.2.4 Demand assignment

Demand or adaptive schemes may be used to combine the best features of fixed polling schemes for periodic messages and random access exception reporting.

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