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XVIIth PLENARY ASSEMBLY
DÜSSELDORF, 1990



INTERNATIONAL TELECOMMUNICATION UNION

REPORTS OF THE CCIR, 1990

(ALSO DECISIONS)

ANNEX 1 TO VOLUME VIII

LAND MOBILE SERVICE
AMATEUR SERVICE
AMATEUR SATELLITE SERVICE

CCIR INTERNATIONAL RADIO CONSULTATIVE COMMITTEE

Geneva, 1990





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**LAND MOBILE SERVICE
AMATEUR SERVICE
AMATEUR SATELLITE SERVICE**

CCIR INTERNATIONAL RADIO CONSULTATIVE COMMITTEE

92-61-04241-4



Geneva, 1990

ANNEX 1 TO VOLUME VIII

LAND MOBILE SERVICE - AMATEUR SERVICE -
AMATEUR-SATELLITE SERVICE

(Study Group 8)

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SECTION 8A: LAND MOBILE SERVICE AND RELATED SUBJECTS

REPORT 358-5

PROTECTION RATIOS AND MINIMUM FIELD STRENGTHS
REQUIRED IN THE MOBILE SERVICES

(Question 1/8)

(1966-1970-1974-1978-1982-1986)

1. VHF and UHF land and maritime mobile services

1.1 *Protection ratios based on internal noise and distortion in the receiver*

The World Administrative Radio Conference, Geneva, 1979, defined the protection ratio as the minimum value of the wanted-to-unwanted signal ratio, usually expressed in decibels, at the receiver input determined under specified conditions such that a specified reception quality of the wanted signal is achieved at the receiver output (RR No. 164). For further information on the definition see Report 525. This ratio may have different values, according to the type of service desired.

However, in the absence of information submitted to Study Group 8 on subjective measurements made in the VHF and UHF land and maritime mobile services, several administrations submitted the results of laboratory measurements, using appropriate test signals, of the degradation of the signal-to-noise ratio of the wanted test signal, when a co-channel interfering signal is superimposed on the latter. A degradation of the initial signal-to-noise ratio of 20 dB to a signal-to-noise + interference ratio of 14 dB is taken as the criterion. For some systems this grade of service is acceptable.

In the tests described by the various administrations, the frequency deviations are 70% or 60% of the maximum specified frequency deviations, and for amplitude modulation the modulation percentages are 70% or 60%, for both wanted and unwanted signals. From a study of the documents submitted, it may be deduced that the slight differences in measurement conditions and in the characteristics of the receivers used in the different tests, may result in differences in the measured receiver protection ratios, of up to about ± 3 dB.

One administration performed tests to determine the protection ratio for the case where the wanted narrowband G3E signal is interfered by a direct-printing F2B signal (see Recommendation 476) [CCIR, 1978-82]. The e.m.f. of the wanted signal at the receiver input was 2 μ V. In these tests the level of the interfering co-channel F2B signal was so adjusted that the subjective effect on the wanted signal was the same as that of an interfering co-channel narrowband G3E signal attenuated by the protection ratio of 8 dB laid down in Table I for this case. The peak frequency deviations used for the F2B signal were ± 1 , ± 3 and ± 5 kHz respectively. The sub-carrier was 1500 Hz and the frequency shift 170 Hz. 12 dB was found to be a suitable representative value for the protection ratio and is therefore included in Table I.

Although the ability of the receiver to receive the wanted signal is dependent on the passband characteristics of the receiver, the frequency difference between the co-channel wanted and unwanted signals, the frequency deviation, etc., the receiver protection ratios in Table I may be used as the basis for the calculation of system protection ratios for mobile systems for a minimum grade of service. Additional protection should be provided to allow for the effects of multipath propagation, man-made noise, terrain irregularities, and in the case of very closely spaced assignments, adjacent-channel interference (see Report 319).

When using frequency modulation, "capture effect" is enhanced as the frequency deviation of the wanted signal is increased; therefore, a wideband F3E, G3E system requires less protection than a narrowband F3E, G3E system for the same type of interfering source.

If a higher grade of service is required, a higher protection ratio should be adopted, particularly in the case of amplitude-modulated wanted emissions.

1.2 *Man-made noise*

Man-made noise degrades the performance of a mobile system. To maintain a desired grade of service in the presence of man-made noise, it is necessary to increase the level of the field strength of the wanted signal. Motor vehicles have been shown, by measurements [US Advisory Committee, 1967], to be the primary source of man-made noise for frequencies above 30 MHz. Other noise sources are fewer in number and usually radiate from fixed locations.



TABLE I — Typical receiver protection ratios, for use in calculating system protection ratios

Wanted emission (Note 1)	Unwanted emission (Note 1)	Receiver protection ratio (dB)
Wideband F3E, G3E	Wideband F3E, G3E	See Report 319
Narrowband F3E, G3E	Narrowband F3E, G3E	See Report 319
Wideband F3E, G3E	A3E	8
Narrowband F3E, G3E	A3E	10
Narrowband F3E, G3E	Direct printing F2B	12
A3E	Wideband F3E, G3E	8-17 (Note 2)
A3E	Narrowband F3E, G3E	8-17 (Note 2)
A3E	A3E	17

Note 1. — Wideband F3E, G3E systems normally employ frequency deviations with a maximum value in the range ± 12 to ± 15 kHz.

The narrowband F3E, G3E systems considered here normally employ frequency deviations with maximum values of either ± 4 or ± 5 kHz.

The value of the F2B case is with a peak frequency deviation of ± 5 kHz. Frequency deviations of ± 3 and ± 1 kHz do not significantly decrease this value.

Note 2. — The receiver protection ratio may vary within the range shown dependent upon the difference in frequency between the carriers of the wanted and unwanted emissions and the frequency deviation of the unwanted emission. In general, it will tend towards the higher figure as the frequency deviation of the unwanted emission decreases.

For convenience in evaluating the degradation of performance of a base receiver, the following classifications of noise sources are provided:

- high noise locations — traffic density of 100 vehicles/km² at any given instant of time;
- moderate noise locations — traffic density of 10 vehicles/km² at any given instant of time;
- low noise locations — traffic density of 1 vehicle/km² at any given instant of time;
- concentrated noise sources (hot spots): noise radiated from individual sources or closely spaced multiple sources which are usually located within 500 m of the receiving antenna, such as a high concentration of vehicles, manufacturing plants and defective power transmission lines.

Noise data for base stations at high, moderate and low noise locations are presented by a noise amplitude distribution (NAD) (the number of pulses per second equal to or greater than the value shown as ordinate) and are illustrated in Fig. 1. The amplitude (A) (in dB(μ V/MHz)) of noise pulses at a rate of 10 pps (pulse-per-second) is expressed as follows:

$$A = C + 10 \log V - 28 \log f$$

where,

C : constant (tentative value: 106 dB(μ V/MHz))

V : traffic density vehicles/km²

f : channel frequency, MHz.

Noise data for hot spots can also be presented in the form of a noise amplitude distribution. However, due to a wide variety of noise sources, it is not yet practical to provide a classified list.

The constant C is a function of the electrical noise suppression applied to vehicles and may also vary according to the relative proportion of goods and passenger vehicles if the level of suppression is not the same for both categories. A tentative value of 106 dB(μ V/MHz) is shown and this may be revised as more information becomes available.

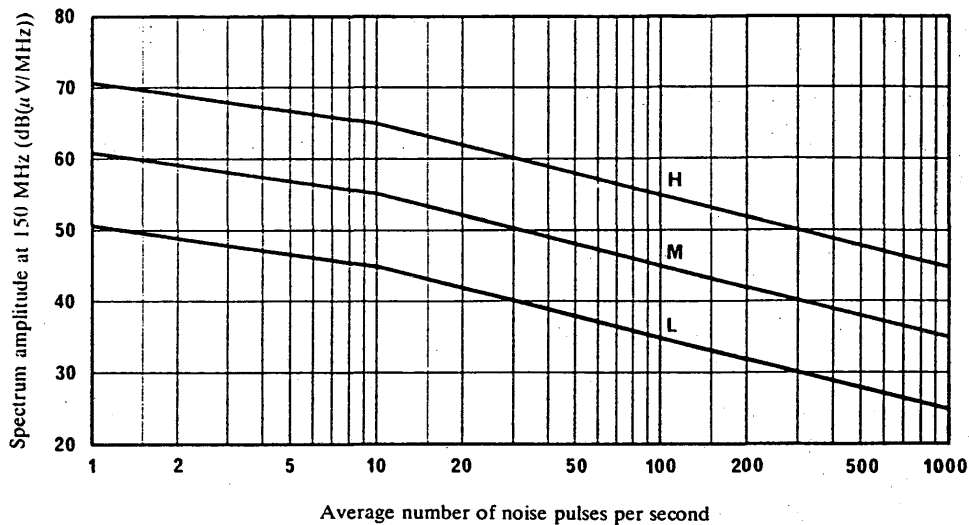


FIGURE 1 – Noise amplitude distribution at base station (150 MHz)

For frequencies other than 150 MHz, raise or lower curves H, M and L in accordance with the formula below.

$$A = C + 10 \log V - 28 \log f$$

where $A = \text{dB } (\mu\text{V/MHz})$ at 10 pps

Curve H: high noise location ($V = 100$)

Curve M: Moderate noise location ($V = 10$)

Curve L: low noise location ($V = 1$)

1.3 Noise Amplitude Distribution (NAD) determination of degradation

1.3.1 Definitions

1.3.1.1 Noise amplitude distribution

A presentation of impulsive noise data in terms of its basic parameters of spectrum amplitude and impulse rate.

1.3.1.2 Spectrum amplitude

The vector sum of the voltages produced by an impulse in a given bandwidth, divided by the bandwidth.

1.3.1.3 Impulse rate

The number of impulses that exceed a given spectrum amplitude in a given period of time.

1.3.1.4 Impulsive-noise tolerance

The spectrum amplitude of impulses at a given pulse-repetition frequency at which the receiver, with an input signal applied at specific levels, produces standard signal-to-noise ratios at the output terminals.

1.3.2 Determination of degradation

Degradation of receivers can be determined as follows:

1.3.2.1 measure the impulsive noise tolerance of the receiving equipment in accordance with applicable IEC standards;

1.3.2.2 measure NAD in accordance with applicable IEC standards;

1.3.2.3 Superimpose the graphs for the receiver impulse noise tolerance and the NAD. An example is shown in Fig. 7.

1.4 Minimum values of field strength to be protected

The minimum values of field strength to be protected in the land mobile service at frequencies above 30 MHz are determined by internal noise generated in the receiver, man-made noise usually in the form of radiation from ignition systems of motor vehicles and the effects of multipath propagation to and from moving vehicles. Some information on the effects of traffic density is now available. In the maritime mobile service, the level of man-made noise depends on the number and nature of high level sources of noise on the ship.

A convenient measure of the threshold of performance for narrowband receivers is a specified value of

$$\frac{S + N + D}{N + D}$$

ratio; the conventionally accepted value being 12 dB (see Recommendation 331).

This defines the minimum usable field strength for any particular installation, in the absence of man-made noise.

The sensitivity of typical receivers is such that an input signal of 0.7 μ V e.m.f. (assuming a receiver input impedance of 50 Ω) would result in a 12 dB

$$\frac{S + N + D}{N + D}$$

ratio at the output. A mobile service is characterized by large variations of field strength as a function of location and time. These variations may be represented by a log-normal distribution for which standard deviations of 8 dB at VHF and 10 dB at UHF are appropriate for terrain irregularities of 50 m (see Recommendation 370). To determine the minimum value of median field strength to be protected, it is necessary to specify the percentage of time for which the minimum usable field strength should be exceeded for different grades of service. For land mobile radiotelephony, a high grade of service would require that the value be exceeded for 99% of the time, but, for a lower (or normal) grade of service, for 90% of the time.

The minimum values of field strength to be protected can be determined subjectively, taking into account man-made noise and multipath propagation. Ignition systems of motor vehicles are usually the most prevalent source of man-made noise. Field strength cancellations due to multipath propagation produce an annoyance somewhat similar to that created by ignition systems. When a mobile unit is in motion, both of these annoyances occur at the same time. Only the effects of receiver noise and man-made noise remain when the mobile unit is stationary. The separation of motor vehicles is generally less with slow-moving or stationary traffic, and under these circumstances, particularly at the lower frequencies, the degradation experienced in a stationary mobile unit is greater than when it is in motion.

Figures 3 and 4 can be used to determine the combined degradation effects of man-made noise and multipath propagation for the case of vehicles in motion. These figures are based on subjective testing under traffic conditions commonly experienced by most mobile vehicles [FCC, 1973]. Specifically, these traffic conditions are the following: in motion while in a low noise area, in motion in traffic surrounded by other vehicles, and stationary surrounded by other stationary or moving vehicles.

The tendency for the curves of Figs. 3 and 4 to merge at the higher frequencies is due to the almost constant multipath degradation effect with frequency and the fact that the degradation effect of man-made noise decreases with frequency.

Degradation is defined as the increase of level necessary in the desired input signal to maintain the receiving signal at the degree of quality obtainable when affected by receiver noise only.

Definitions of signal quality are as follows:

Grade	Interfering effect:
5	Almost nil
4	Noticeable
3	Annoying
2	Very annoying
1	So bad that the presence of speech is barely discernible

Speech understandable, but with increasing effort as the grade decreases

Some information on field strengths can be derived from Recommendation 370. Additional information can be found in the document of the CCIR, [1966-69], and in the article of Okumura *et al.* [1968].

Information on protection ratios and minimum field strengths may also be found in the “Special Agreement between the Administrations of Belgium, the Netherlands, and the Federal Republic of Germany relating to the use of metric and decimetric waves for fixed and mobile services in border areas, Brussels, 1963”, and in the Final Acts of the Special Regional Conference, Geneva, 1960. Similar information may be found in the Agreement between the Telecommunications Administrations of Austria, the Federal Republic of Germany, Italy and Switzerland, Vienna, 1969.

The document of the CCIR [1963-66], deals with the above questions for signal-to-noise ratios of 30 dB and 40 dB at the receiver output.

Until values based on man-made noise and multipath effects are available, the calculated values of minimum and median values of field strength shown in Fig. 2 may be used for hand-portable stations.

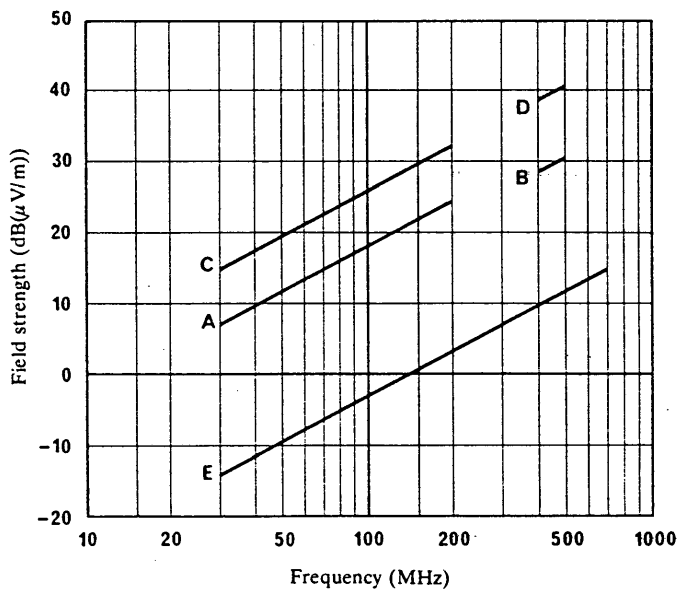


FIGURE 2 – Minimum usable and median field strengths for typical hand-portable stations (based on minimum usable input of 0.7 μV e.m.f., in the absence of man-made noise)

Characteristics assumed: antenna gain { A and C: -9 dB
B and D: -6 dB

A, B: median, normal grade
C, D: median, high grade
E: minimum usable field strength (dipole antenna)

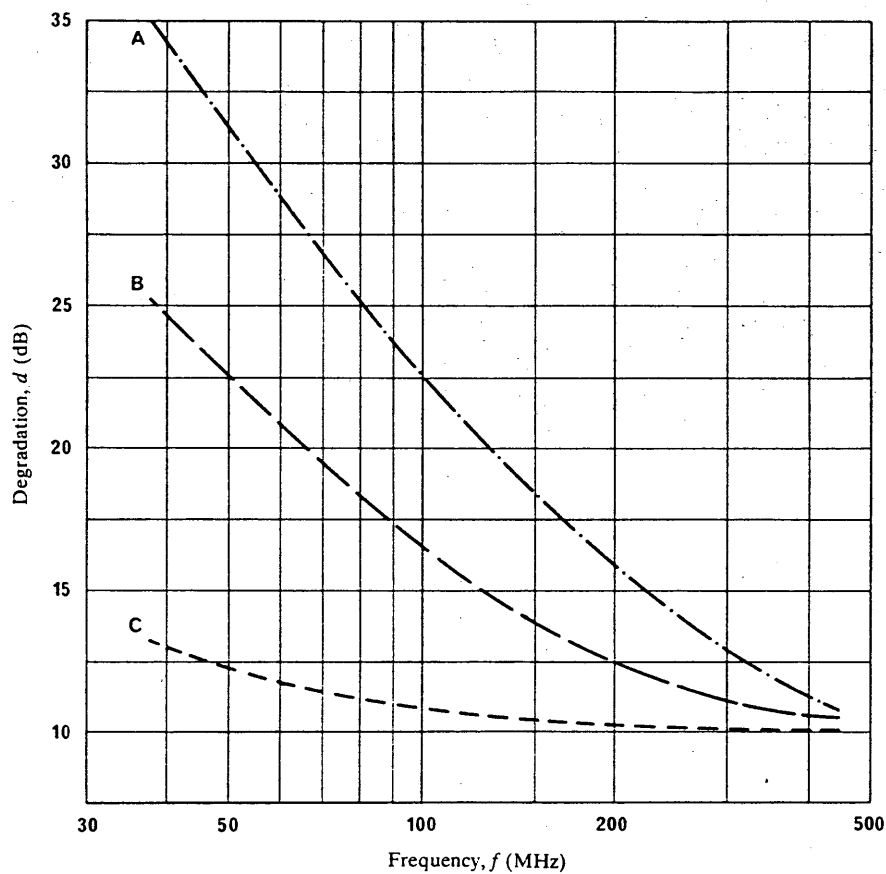


FIGURE 3 – Variation of degradation of mobile reception and minimum values of field strength to be protected for signal quality grade 4 and receiver sensitivity of $0.7 \mu V$ e.m.f.

$$\text{Field strength} = -41 + d + 20 \log f \quad \text{dB}(\mu V/m)$$

- A: mobile vehicle stationary within a high noise area
- B: mobile vehicle in motion within a high noise area
- C: mobile vehicle in motion within a low noise area

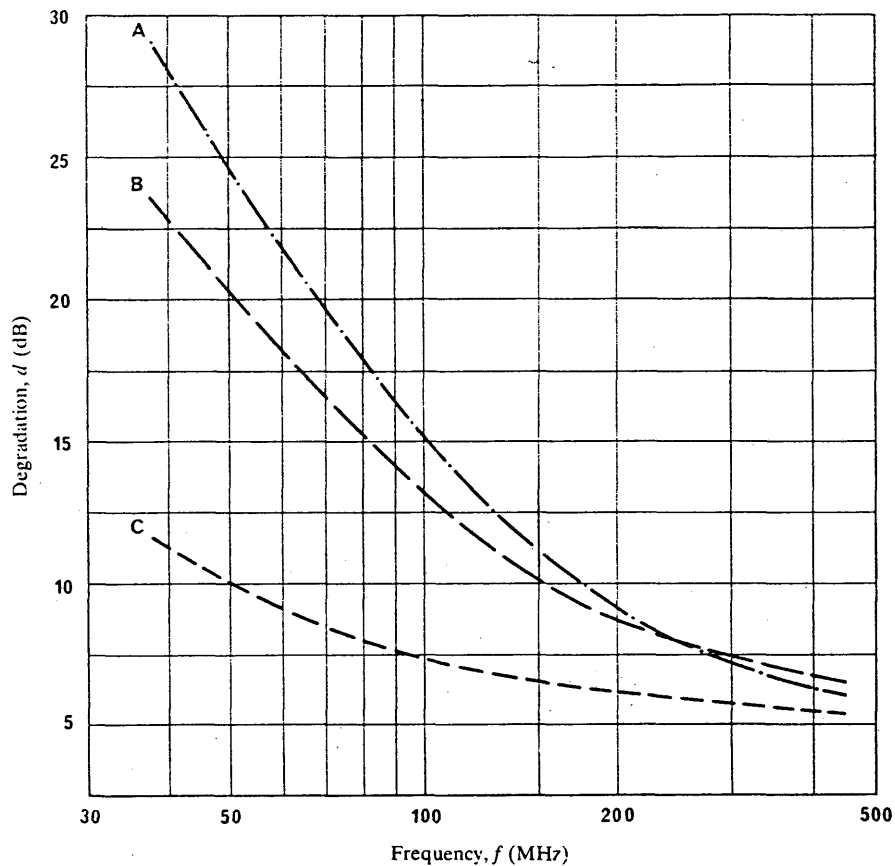


FIGURE 4 - Variation of degradation of mobile reception and minimum values of field strength to be protected for signal quality grade 3 and receiver sensitivity of $0.7 \mu V$ e.m.f.

$$\text{Field strength} = -41 + d + 20 \log f \quad \text{dB}(\mu V/m)$$

- A: mobile vehicle stationary within a high noise area
 B: mobile vehicle in motion within a high noise area
 C: mobile vehicle in motion within a low noise area

Figures 5 and 6 can be used for determining the degradation of base station reception due to ignition noise and multipath propagation.

Curves A and B of Figs. 5 and 6 show the combined degrading effects of multipath propagation and ignition noise for heavy and moderate traffic rates. The speed of the traffic was approximately 80 km per hour. Curves D and E show the degrading effect of ignition noise only. Curve C shows the degrading effects of multipath propagation only.

The data presented here were obtained at a distance of 23.5 m from a heavily travelled thoroughfare. Except for the ignition noise created by the thoroughfare, the base station test site itself was quiet. Curves A and B were obtained by radiating the desired signal from a mobile unit in motion. In this case, degradation is based on median values of voltage at the input terminals of the receiver. The effects of ignition noise, only when the mobile unit is standing still, are shown by curves D and E. In this case the desired signal was obtained from a signal generator. Curve C was obtained by inserting sufficient attenuation at the receiver input terminals in order to eliminate ignition noise pulses. The increased attenuation was compensated for by radiating a stronger signal from the mobile unit.

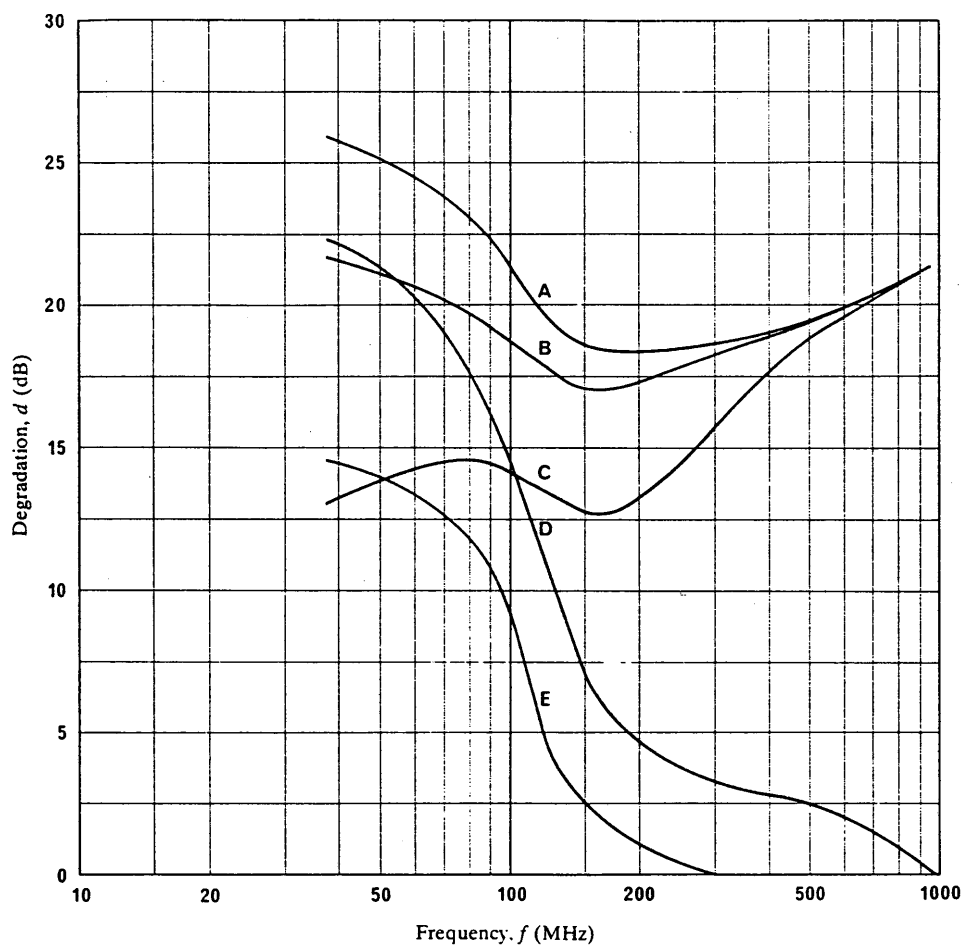


FIGURE 5 – Variation of degradation of base station reception and minimum values of field strength to be protected for signal quality grade 4 and receiver sensitivity of $0.7 \mu V$ e.m.f.

$$\text{Field strength} = -41 + d + 20 \log f \quad \text{dB}(\mu V/m)$$

- A: mobile vehicle moving, traffic rate is 2 vehicles/s
- B: mobile vehicle moving, traffic rate is 1 vehicle/s
- C: mobile vehicle moving, no ignition or ambient noise
- D: mobile vehicle standing still, traffic rate is 2 vehicles/s
- E: mobile vehicle standing still, traffic rate is 1 vehicle/s

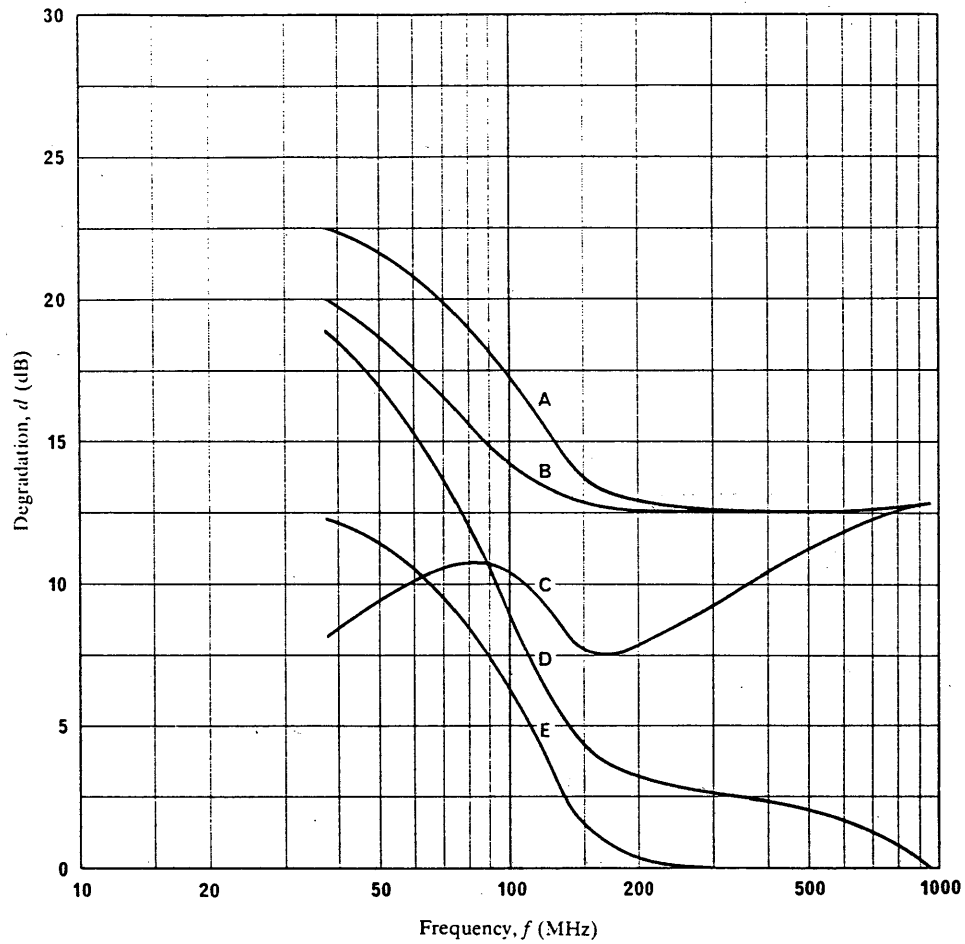


FIGURE 6 – Variation of degradation of base station reception and minimum values of field strength to be protected for signal quality grade 3 and receiver sensitivity of $0.7 \mu V$ e.m.f.

$$\text{Field strength} = -41 + d + 20 \log f \quad \text{dB}(\mu V/m)$$

- A: mobile vehicle moving, traffic rate is 2 vehicles/s
- B: mobile vehicle moving, traffic rate is 1 vehicle/s
- C: mobile vehicle moving, no ignition or ambient noise present
- D: mobile vehicle standing still, traffic rate is 2 vehicles/s
- E: mobile vehicle standing still, traffic rate is 1 vehicle/s

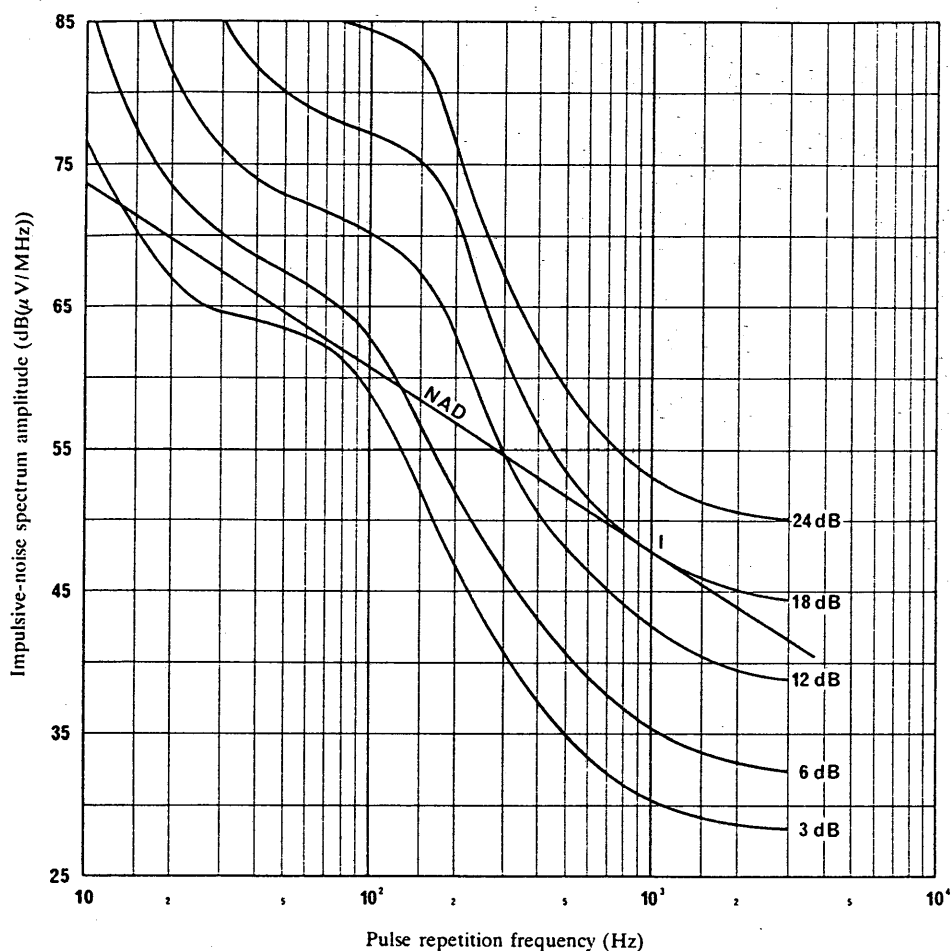


FIGURE 7 – Example showing 18 dB degradation
NAD tangent at 1 to 18 dB curve of the graph for receiver impulse noise tolerance

At approximately 450 MHz the degradation of base station reception due to multipath is greater than that for mobile reception. This is primarily due to the higher level of the ambient aural noise in the mobile unit as compared to that of the base station. The road and vehicle aural noise mask the multipath degradation effect, thereby reducing the value of receiver input signal needed to obtain a given grade.

2. HF maritime mobile service

The question of the protection ratio and the minimum field strength to be protected in the HF maritime mobile service for various classes of emission used by that service needs further study. In so far as HF radiotelephony is concerned the Study Group gave provisional advice to the IFRB in respect of these parameters (see Report 748).

3. Conclusions

Considerable additional work concerning §§ 1 and 2 is necessary to determine more fully the appropriate protection ratios and the values of the field strengths to be protected, and also to determine and record the measurement methods which should be adopted.

REFERENCES

- FCC [August, 1973] Degradation of mobile radio reception at UHF and VHF. Research Division, Rep. No. R-7302.
- OKUMURA, Y., OHMORI, E., KAWANO, T. and FUKUDA, K. [September-October, 1968] Field strength and its variability in VHF and UHF land-mobile radio service. *Rev. Elec. Comm. Labs., NTT*, Vol. 16, 9-10, 825-873.
- US ADVISORY COMMITTEE [1967] Man-made noise; Report from Working Group 3 of the Advisory Committee for the Land Mobile Radio Services, Vol. 2, Part 2, US Govt. Printing Office: 0-281-851, Washington, DC 20402, USA.
- CCIR Documents*
- [1963-66]: XIII/88 (Japan).
- [1966-69]: XIII/146 (Japan); XIII/149 (Japan).
- [1978-82]: 8/3 (Germany (Federal Republic of)).

REPORT 319-7*

**CHARACTERISTICS OF EQUIPMENT AND PRINCIPLES GOVERNING
THE ASSIGNMENT OF FREQUENCY CHANNELS BETWEEN 25 AND 1000 MHz
FOR LAND MOBILE SERVICES**

(Resolution 20)

(1970-1974-1978-1982-1986-1990)

PART A

ASSIGNMENT METHODS

1. Suggested principles

1.1 The following broad principles are suggested for use in the assignment of frequencies in the land mobile service:

- the choice of the most advantageous mode of operation, i.e., single-frequency or two-frequency operation, according to the type of service, bearing in mind the need for coordination between administrations in border areas;
- the gradual adoption, as opportunity occurs, of the same blocks of frequencies for base stations by all administrations, and similarly the same blocks of frequencies for mobile stations, in order to minimize interference between services of different administrations;
- the gradual adoption by all administrations, as opportunity occurs, of the same blocks of frequencies for the same types of service or at least for those services required to provide similar coverage;
- the adoption of compatible frequency plans, with the same channel spacing and the same centre frequencies of the channels and, when suitable, with centre frequencies off-set, e.g. by one half channel, especially in areas where mutual interference might occur between the services of different administrations;
- the use of common channel spacing, preferably 25 kHz (see Note) and the use of equipments which are readily adaptable for a reduction in channel separation without replacement of the whole equipment;
- the allocation of channels in such a way as to minimize the production of interference due to intermodulation products;
- the adoption of optimum sizes and shapes for service areas in relation to frequency economy (Question 37/8 and Report 740 refer to this subject);
- the use of the minimum effective radiated power compatible with the required service range;
- the use of the minimum height of base station antennas compatible with the required service range;
- the siting of co-channel stations with the minimum geographical separation compatible with the protection ratios and minimum field strengths to be protected which are appropriate to the service. Information on this subject is given in Report 358;
- the use by all administrations of common propagation data. References to CCIR documents on this subject are included in Report 358;
- the assignment of the same frequency channel to a number of users in the same area, in such a manner as to permit optimum use of the channel.

* The Director, CCIR, is requested to draw the attention of IEC to Part B, § 3.4. This Report should be brought to the attention of Study Group 1.

1.2 These principles can be applied to full advantage when planning land mobile services, only if all are applied, since they are highly interdependent.

Note. — It is recognized that some administrations use other channel separations. Every opportunity should be taken to achieve the use of common channel separations.

2. Single-frequency and two-frequency operation

It is not usually possible to use all the available frequencies in a given restricted area owing to intermodulation problems, adjacent channel disturbances, receiver desensitization, etc. The problems which arise may be somewhat different with single-frequency and two-frequency operation.

2.1 *Single frequency operation*

- Direct mobile-to-mobile communication independent of base stations is possible.
- Direct base-to-base communications when base stations are within range of each other is possible.
- Mobiles not within range of each other may transmit simultaneously causing interference at the base station.
- Base station can effectively control channel usage.
- Possibility for interference between base stations using the same channel.

2.2 *Two frequency operation-non-repeater mode (at base station)*

- Prevents direct mobile-to-mobile communication.
- Permits full control of channel usage by base station.
- Mobiles may transmit simultaneously causing interference at base station.
- Base-to-base communication not possible.
- Necessary for mobile telephone systems providing full duplex arrangements and interfacing with telephone networks.
- Prevents interference between base stations using the same channel.

2.3 *Two frequency operation-automatic repeater mode (at base station)*

- Mobile-to-mobile communications are automatically rebroadcast (repeated) by the base station thus mobile-to-mobile range is equal to that of base station coverage.
- Every user is aware of every transmission.
- Remote control of repeater easily accomplished from fixed control points by a radio equipment operating on mobile radio frequencies.
- Lends itself to shared use [Mulwijk, 1978].
- Permits use of mobile units in repeater mode to act as relay for portables.
- Permits totally unattended operation of base station.
- Failure of the automatic repeater results in total system failure, i.e. mobile-to-mobile communications is impossible without specially configured mobile units.
- Total channel occupancy may be determined by monitoring the base station frequency only.
- Prevents interference between base stations using the same channel.

2.4 Other factors to be considered when administrations develop two-frequency channelling plans for land mobile services are:

- practical values of transmit/receive frequency separation;
- practical values of maximum channel separation in multi-channel equipment;
- the use of a constant separation between the transmit and receive frequencies over the whole of a band or the sub-bands within a band.

3. Channel spacing considerations

3.1 Spectrum efficiency

During the technical development of the land mobile service a progressive reduction of channel spacing has occurred, thus making available an increased number of channels.

Some administrations, taking into account the need for new channels, especially in limited areas with high-density population, have decided to use frequency modulation with very narrow channel spacing. Despite some disadvantages these administrations consider that they derive substantial benefits from the use of this technique (see Part C, § 1.4).

Nevertheless some conflicting factors have to be taken into account when considering whether it is effective to reduce the spacing between channels by narrowing the bandwidth of the emissions and correspondingly narrowing the bandwidth of the receivers. Therefore, rather than concentrating only on finding ways to increase the number of channels to be derived from a given block of spectrum, the search should be directed to finding the combination of channel spacing and technical characteristics that will result in accommodating the maximum amount of information and/or the maximum number of radio users per MHz in a given geographic area.

Wide-channel spacing affords a large modulation index and hence efficient geographical frequency reuse. The number of available channels, however, is small when the channel spacing is wide. In order to make a valid comparison of different channel spacing in terms of spectrum efficiency, it is necessary to study the relationship between the modulation index and the spectrum utilization factor determined as a consequence of considering available channel numbers and geographical frequency reuse.

Spectrum efficiency in mobile radio communications is usually defined as the product of efficiencies in space, frequency, and time domains [Colavito, 1974; Mulwijk, 1978]. The time factor is determined by how much traffic can be sent through a radio channel in a unit time, and has no relationship to a modulation index.

Co-channel interference performance is the most important characteristic for a cellular system, because the same radio frequency channels are reused in a service area. Figure 1 [Suwa and Hattori, 1985] shows a relationship between carrier-to-interference ratio (C/I) and baseband signal-to-interference ratio (S/I) of FM system under Rayleigh fading condition. The results for various modulation indices which correspond to the peak deviation show that improvement of C/I and S/I performance by increasing modulation index does eventually saturate.

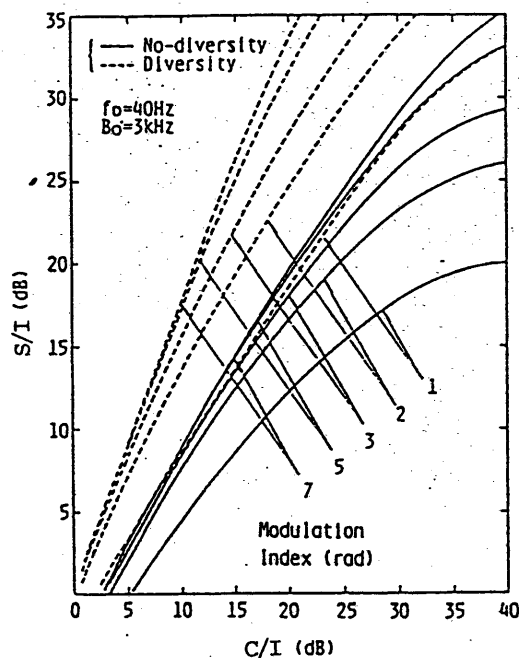


FIGURE 1 - Relation between C/I and S/I of FM system with selection diversity

f_D : maximum Doppler frequency
 B_0 : passband of the baseband filter

Figure 2 [Sakamoto and Hata, 1987] shows a relationship between modulation index and spectrum efficiency. Peak deviation is assumed as $1/0.7$ of the 1 kHz signal modulation level. For example, a 3.5 radian modulation index corresponds to a 5 kHz peak deviation. With regard to efficient spectrum utilization, Fig. 2 shows that 1) it is suitable for the peak deviation to be set around 3 to 4 kHz, 2) the diversity technique is effective in improving the spectrum utilization, and the optimum peak deviation is almost the same as when there is no diversity.

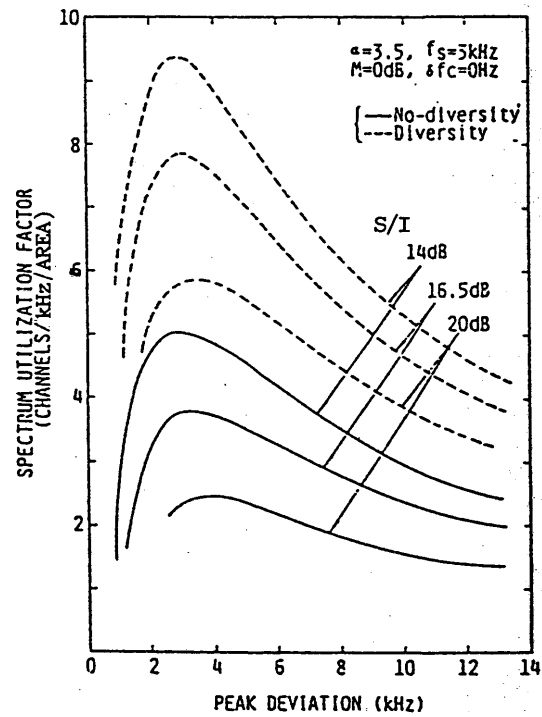


FIGURE 2 - Spectrum utilization factor with S/I parameter

- f_s : maximum baseband signal frequency
 δf_c : carrier drift
 α : propagation constant
 M : C/I margin for shadow-fading

3.2 Experimental quality evaluation

An experimental program has been used [AT&T, 1971] to determine listeners' subjective evaluation of a FM voice channel in the presence of rapid fading, for a variety of **wanted-to-unwanted signal ratio** (W/U) and radio-frequency S/N ratios. The result of this experiment were used to set E_i (local mean (W/U)) and E_n (local mean radio-frequency S/N). The peak frequency deviation of the transmitter was varied, and the effect on listener reaction and thus on E_i and E_n was noted. Listeners evaluated quality in terms of Circuit Merit ratings 5, 4, 3, 2 or 1 (excellent, good, fair, poor or unusable).

Tests were made using peak frequency deviation of ± 12 kHz and ± 6 kHz respectively, the parameters of E_i and E_n being those values which must be exceeded with 90% probability, and a criterion being that at least 75% of the listeners should rate the quality good or better, and at least 90% should rate it fair or better.

The tests showed that:

- For thresholds based on Merit 4 (good) with 75% rating the circuit good or better, the use of ± 12 kHz peak deviation improved the co-channel protection ratio requirement by 8 dB for E_i and by 5 dB for E_n .
- For thresholds based on Merit 3 (fair) with 90% rating the circuit fair or better, the use of ± 12 kHz peak deviation improved the co-channel protection ratio requirement by 6 dB for E_i and by 4 dB for E_n .

These tests were made at 800 MHz in preparation for a cellular system. Further study is necessary to establish the relationship between modulation index and optimum channel spacing, and to determine what values apply for other frequency bands and other service requirements.

3.3 Carrier frequency offset

Additional co-channel protection between cells can be obtained by offsetting the carriers of the channels used in those cells. One example is the use of 16K0F3E emissions on channels offset by 12.5 kHz, using equipment designed for 20 or 25 kHz channel spacing [Brusaferri *et al.*, 1979]. This retains the basic characteristics of the 16K0F3E system while increasing (but not doubling) the number of available channels and the traffic capacity.

In Toronto, Canada, a 150 MHz public mobile telephone system using 15 kHz offset channels to provide automated service has been interleaved with an existing manual service system on the prime channels, spaced 30 kHz. In order to achieve this all base transmitters are co-located and adjusted to give equal effective radiated powers. The following characteristics apply:

- base transmitters have ± 3 ppm frequency tolerance;
- peak frequency deviation is reduced to ± 4 kHz from ± 5 kHz;
- average speech modulation on each transmitter is reduced by 3 dB;
- mobile telephone specifications are unchanged from those required for normal 30 kHz spaced operation;
- all channels use multiple-receive voting techniques to provide good quality audio from almost all locations, even in the presence of adjacent channel mobile transmissions.

4. Transmit/receive frequency separation for high capacity systems

Transmit and receive filters are necessary for full duplex operation of mobile radio telephone systems. In the design of such radio frequency filters, it is necessary to determine types and number of resonators considering such system parameters as transmit/receive frequency separation and allocated transmit and receive bandwidths, and such mobile unit parameters as size, cost, transmitter power, transmitter noise, spurious emission, spurious response, and so on.

However, further study of the transmit/receive frequency separation is required.

5. Use of computers in frequency assignments

Computers provide an economical way of maximizing the utilization of channels by optimizing frequency re-use. It is possible to use for planning detailed topographical data and representative propagation models. The application of computers to frequency assignment is under study in Study Group 1.

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PART B

COMPONENTS AND ENGINEERING TECHNIQUES

1. Integrated engineering at base stations

The generation of spurious emissions and intermodulation products at base stations in the land mobile service can be considerably reduced by using suitably designed filters at the outputs of transmitters and the inputs to receivers, in conjunction with combining networks, so that numbers of transmitters and/or receivers can be coupled to a common antenna. Similar techniques can be used in the UHF band and for larger numbers of circuits.

The use of a single mast and antenna in this way minimizes spurious emissions arising in the external system. Another advantage is that it is possible to optimize the radiation pattern so as to obtain the best coverage of the desired area. Moreover, such a unified antenna and mast can be designed to be much more acceptable aesthetically than separate masts and antennas for each user. This is a consideration which is becoming increasingly important in view of the proliferation of base station antennas on prominent sites and buildings. Another important advantage is that the reduction in spurious emissions and intermodulation products enables better use to be made of the available frequency channels in a given area, as well as improving the quality of individual channels.

The transmit output filters attenuate any spurious emissions and harmonics. The filters may consist of bandpass filters (cavity resonators) and the insertion loss of each filter, for example, can be kept between 1 and 2 dB. The passband of the filter for each transmitter is centred on the transmit frequency and its impedance at that frequency is matched to the transmitter output and to the common cable ("throughline") connecting the transmitters to the combiner. At the frequencies of the other transmitters, the filter presents a high impedance to the "throughline". Whenever possible, the use of dissimilar metals at junctions should be avoided in order to avoid non-linear effects. Care should also be taken to ensure that the physical size and mechanical construction prevents the formation of corona, or brush discharges, which would otherwise cause electrical noise and interference.

On the receive side, two bandpass filters are used to attenuate the transmit frequencies and protect the receivers. This is followed by a low-noise preamplifier, followed by a cascade arrangement of passive hybrid couplers, depending upon the number of outlets required. The insertion loss of the cascade system between the antenna and any receiver can be compensated for by the preamplifier so that the performance of the system is not degraded. In practice, an improvement in signal/noise ratio is obtained compared with coupling the receiver directly to the antenna. For example, an improvement between 1 and 2 dB can be obtained with a VHF preamplifier of 3 dB noise figure and a gain of 18 dB.

If transmitters and receivers share a common antenna, then the combining network should have transmit and receive bandpass filters to provide sufficient additional isolation between the transmitters and receivers. The number of filter sections required depends upon the frequency spacing between transmitters and receivers. The filters in the transmitter arm attenuate any spurious emissions, especially those in the receiver band, and the filters also attenuate harmonics of the transmit frequencies. Care should be taken to avoid the formation of corona.

2. Tone squelch system

One method of providing more efficient use of the spectrum can frequently be achieved by placing more than one user on a single frequency or frequency pair and incorporating a continuous tone controlled squelch system (CTCSS) in the design of the base station and mobile station equipment. Each transmitter when keyed, transmits a tone unique to the particular system. Each receiver in the system must detect not only the presence of the radio frequency signal but also the unique tone before the squelch opens. Thus the user normally hears only those transmissions of his own system and is relieved of listening to the transmissions of others sharing the channel. The tone frequencies may be below 300 Hz or between 300 and 3000 Hz. In both cases they are filtered out of the audio heard by the user.

The selection and coordination of tone frequencies between separate users is necessary to avoid interference. For frequencies used in CTCSS reference should be made to IEC Publication 487-6A.

3. Crystals and oscillators

3.1 Frequency variations with temperature

3.1.1 Some commonly used crystals are listed below:

Designations			Description
(a)	(b)	(c)	
Style D	HC-6/U		Miniature metal can with short pins.
Style J	HC-18/U		Sub-miniature metal can with wires.
Style K	HC-25/U		Sub-miniature metal can with short pins.
Style L	HC-27/U	13	Miniature glass encapsulated with short pins.
Style M	HC-26/U	14	Sub-miniature glass encapsulated with wires.
Style N	HC-29/U	20	Sub-miniature glass encapsulated with short pins.

They exhibit a maximum frequency variation within 20 parts in 10^6 over the range -20°C to $+60^{\circ}\text{C}$, when the power dissipated in the crystal does not exceed about 0.5 mW.

3.1.2 Crystals normally in use exhibit frequency/temperature curves with inversion points usually above the lowest temperature limits required (-20°C) and around the upper temperature limit required ($+60^{\circ}\text{C}$). It is quite difficult to supply crystals to a given tight tolerance for temperature limits much in excess of the inversion points because of the high slope of frequency/temperature curves beyond these points.

3.1.3 With selected crystals fitted in soft-soldered cans, maximum frequency variations of 10 to 12 parts in 10^6 over the range -20°C to $+60^{\circ}\text{C}$ and 5 parts in 10^6 over the range -10°C to $+40^{\circ}\text{C}$ can be achieved.

3.1.4 Glass encapsulated crystals in cold-welded cans may be supplied to similar limits and can be supplied to close tolerances, for example 5 parts in 10^6 .

3.1.5 Smaller frequency variations can be achieved by using crystal ovens. The penalty is higher cost, greater power demand and a degraded mean-time-between-failure (MTBF).

3.1.6 Temperature Compensated Crystal Oscillators (TCXO), which include the maintaining oscillator, can be supplied with maximum frequency variations from a lower limit of about 2 parts in 10^6 to about 6 parts in 10^6 over a range -20°C to $+60^\circ\text{C}$. Adjustment accuracy is better than for a crystal in isolation from its maintaining oscillator and the TCXO may be set to a nominal frequency at a designed temperature.

3.2 Ageing

All crystals and their maintaining oscillators age to some degree. The ageing rate in service is reduced if the crystals are pre-aged during manufacture.

The best commercial performance limits for total variation for the various types of crystal in common use in the land mobile service are as follows:

Soft-soldered can crystals	5 to 20 parts in 10^6 per year
Glass encapsulated crystals	1 to 3 parts in 10^6 per year, after 30 days ageing
Cold-welded can crystals	as for glass encapsulated units.

These figures can be met for ageing of crystals at any temperature in the range -20°C to about $+60^\circ\text{C}$, but ageing rates increase as the temperature increases.

3.3 Effect of maintaining oscillators on frequency variations

Except for TCXO's, all the figures given above are for crystals alone. Most maintaining oscillators contribute appreciably to the frequency variations in practice and these contributions vary widely within a given design as well as between designs. It is estimated that the average contributions of the maintaining oscillator to total frequency variation could be as much as 4 parts in 10^6 over the range -20°C to $+60^\circ\text{C}$, for a change of $\pm 15\%$ in the voltage supplied to the oscillator.

3.4 Frequency synthesizers

In synthesized equipment there is a danger, particularly during synthesizer settling time, of unwanted emissions, which should be avoided by proper design of the equipment.

To prevent the emission of unwanted frequencies (including harmonic and other spurious products) some of the following techniques could be considered:

- in phase locked synthesizers a circuit can be included in the loop to detect the out of lock condition;
- inverse translation of the division ratio to check that the correct channel frequency has been generated;
- correct choice of loop filter and other phase locked loop design parameters;
- generation of the carrier frequency directly by a voltage controlled oscillator in a single phase locked loop;
- adaptive phase locked loop which has a narrow bandwidth when locked, and then reduces spurious responses and noise, but a wider bandwidth (with shorter pull-in-time) when unlocked;
- direct synthesis with high stability oscillators.

The attention of the IEC should be drawn to this problem and they should be invited to prepare suitable methods of measurement.

4. Diversity reception

Diversity reception is considered to be an effective technique to mitigate multipath fading when dual space diversity is used with the mobile unit. Difference in the spacing between two antennas may result in differences in diversity efficiency. The signal from the two antennas can either be combined in some manner, or the signal from one antenna can be selected according to defined criteria. Various criteria may be used, such as:

- highest signal level,
- best signal-to-noise ratio [Jakes, 1974].

The statistical results obtained from measurements using the criterion of highest signal level, on field experiments at 900 MHz in metropolitan areas of Shanghai, are shown in Figure 4 and Figure 5 [Yang, 1986], where d is the distance between the two antennas expressed in terms of the wavelength λ .

In Figure 3, for $d = 0.75$, the mean signal level received with dual diversity is increased by 2.85 dB compared with that received with one antenna.

In Figure 4, for the instantaneous signal received with one antenna, the level is 15 dB or more below the mean signal level for 8.2 percent of the time and 20 dB for 2.9 percent of the time. For dual space diversity, with $d = 0.75$, the level is 15 dB or more below the mean signal level received with one antenna for 2.6 percent of the time and 20 dB for 0.52 percent of the time.

On the basis of these fading statistics, it seems reasonable to highlight the following two points for consideration:

- a) to obtain satisfactory diversity efficiency, the distance between two co-located mobile station antennas should be greater than 0.6 wavelength, i.e. $d > 0.6\lambda$ and it should preferably correspond approximately to odd multiples of $\lambda/4$;
- b) if d is reduced, even to a value as low as $\lambda/4$, reasonable diversity efficiency can still be obtained.

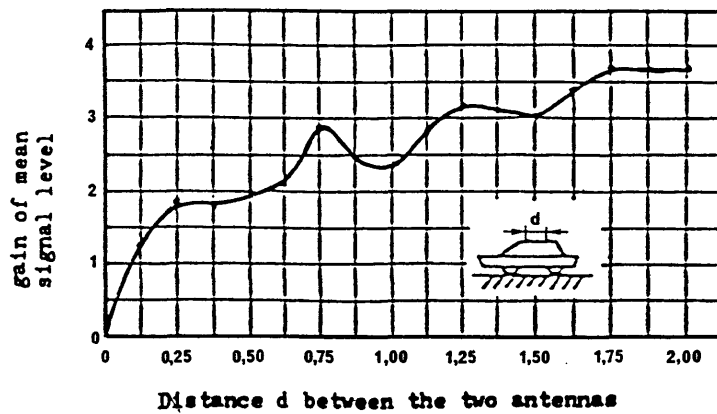


FIGURE 3

Increase of mean signal level versus distance d

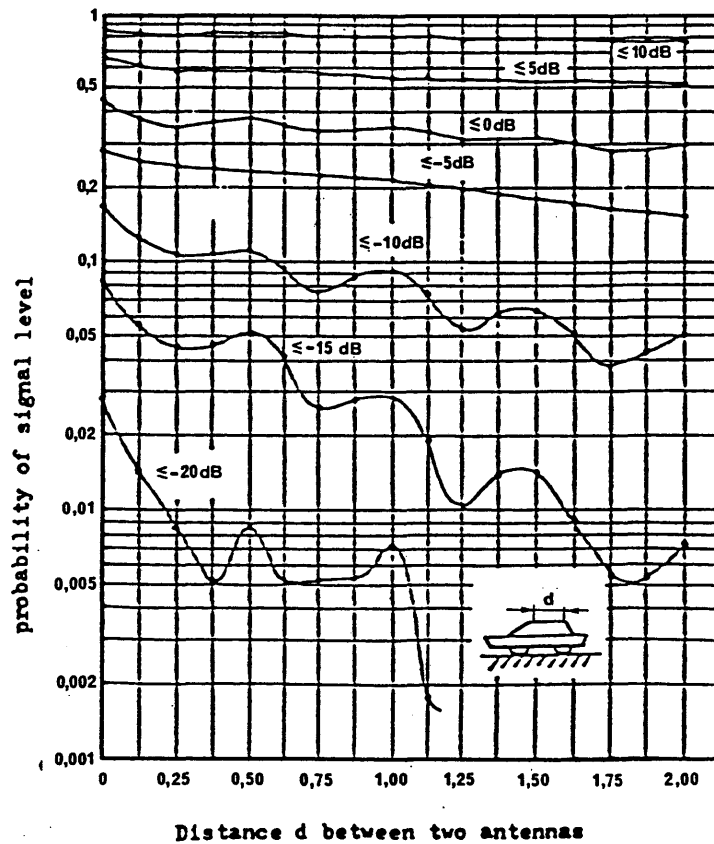


FIGURE 4

Signal level distribution function

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PART C

NATIONAL PRACTICES

1. Summary of contributions

1.1 The information on separations between adjacent channels furnished by administrations, is reproduced in § 2 below. The names of countries are replaced by the appropriate symbols shown in Table I in the Preface to the International Frequency List.

1.2 The Federal Republic of Germany has described methods of the assignment and coordination of frequencies of public and private land mobile services. The planning methods described are, among others, based on the assumption that in an extended network the distance between stations using the same frequencies (co-channel spacing) should be as small as possible for the benefit of a good frequency economy. This can be reached on the one hand by reducing the transmitter range as far as possible and on the other hand by using so-called lattice plans. These lattice plans are set up in such a way that the co-channel spacing is the same for all frequencies and that the number of frequencies for one complete coverage of an extended area is as small as possible. A lattice plan for a public land mobile service using 7 duplex-frequencies for a complete coverage has been developed and a plan for a private land mobile service using 9 simplex-frequencies has been described. Furthermore, two different methods are described by which – when lattice plans are used – the frequency requirements of high traffic areas also can be fulfilled.

The described planning methods have been used in the Federal Republic of Germany with good results for the frequency assignment of about 140 000 land mobile stations.

1.3 Some special information concerning the coordination of frequencies on a multilateral basis is given in the Special Agreement between the Administrations of Belgium, the Netherlands and the Federal Republic of Germany, relating to the use of metric and decimetric waves for fixed and mobile services in border areas (Brussels, 1963).

Agreements also exist between the Administrations of Austria, Germany (Federal Republic of), Italy and Switzerland on the coordination of frequencies between 29.7 MHz and 470 MHz for fixed and land mobile services (Vienna, 1969) and between the Administration of Yugoslavia (Socialist Federal Republic of) and the Administrations of Italy (Vienna, 1969), Austria (Vienna, 1969: revised 1976), Romania (Socialist Republic of) (Belgrade, 1978), the Hungarian People's Republic (Budapest, 1976), Greece (Athens, 1979) and Bulgaria (People's Republic of) (Sofia, 1980). A document from Yugoslavia gives details of the procedures adopted [CCIR, 1978-82a].

1.4 The United Kingdom has drawn attention to the increased number of channels that become available when the spacing is reduced. It suggests that, whenever possible, a spacing of 12.5 kHz should be adopted and that equipment designed for wider channel spacing should be readily adaptable for smaller spacings without total replacement.

In the United Kingdom use is made of computers in frequency assignment [CCIR, 1982-86a]. For an inclusive assignment, a prediction of the field-strength contours representing service and interfering areas is made, and the assignment decided on the basis of overlap criteria with these areas and those of existing stations. For a shared assignment an assessment of the channel loading at the busiest time of the day is made and the assignment decided on the basis of an acceptable total loading.

Iraq reports [Al-Araji and Abdullah, 1982] that by selecting three bands in the audio frequencies delaying them by 0, 4 and 8 ms respectively during transmission and subsequently restoring them at the receiver, the peak impulsive noise level in the audio frequencies can be reduced.

1.5 The USA has provided information concerning disadvantages of decreasing frequency modulation-system deviations from the widely used ± 5 kHz to ± 2.5 kHz; it concludes:

- that increases in impulsive noise interference will be noted;
- that about seven times increase in the number of intermodulation products can be expected.

Other important factors are also listed as follows:

- problems of required increased frequency stability;
- degradation of receiver performance due to adjacent channel transmitter modulation and noise;
- loss in protection ratio;
- loss in frequency-modulation improvement ratio.

1.6 The Federal Republic of Germany has developed a rule which can be used to determine the channels that may be chosen from a number of evenly spaced channels, in such a way that third-order intermodulation products are avoided. According to the rule, channels should be chosen so that the frequency differences between sequential channels, and also the various sums of differences between channels, occur only once. For example, if the evenly spaced channels are numbered 1, 2, 3 etc., then their frequency differences are proportional to the differences between the channel numbers themselves. Thus, assuming that adjacent channels are not to be used, then the chosen channels would be 1, 3, 6, 10, 16, etc., to avoid third-order intermodulation products according to the rule. A proof of the rule is given.

1.7 The USA has studied the interference which is caused by intermodulation products in the land mobile service between 25 and 500 MHz. In areas where there are large concentrations of such services, intermodulation interference is more serious than co-channel interference. Methods are available for predicting the expected level of intermodulation products; these procedures are applied to special transmitters and receivers, but they may be applied generally.

1.8 The People's Republic of Poland has supplied information on the intermodulation response in receivers equipped with crystal filters in the first intermediate-frequency stage. The intermodulation response of such receivers is, in principle, flat over a relatively wide frequency band, in contrast to receivers without crystal filters, in which the slope of this response increases as a function of the degree of detuning the interfering signals from the desired frequency.

In the case of the flat configuration of intermodulation response, there is a greater risk for mobile networks with regard to intermodulation interference caused by the signals widely spaced from the desired frequency. This fact indicates that intermodulation should be measured in a wide frequency band and the results obtained, which constitute the intermodulation response, should be used in planning the allocation of frequencies.

When using receivers with a flat intermodulation response, it is necessary to attempt to achieve the better values of the intermodulation parameter, e.g. by using field effect transistors (FET) or special circuits in the input stages of receivers.

1.9 France has submitted the following list of characteristics and values for possible subsequent insertion in Recommendation 478. Later it may be possible to modify the values quoted in the light of experience.

1.9.1 *Transmitter characteristics*

1.9.1.1 *Transmitter response for modulating frequencies above 3 kHz*

Between 3 kHz and 6 kHz the frequency deviation should not exceed that at 3 kHz. At 6 kHz it should be not more than half the deviation at 1 kHz. For frequencies above 6 kHz and up to the channel spacing, the deviation should decrease with an increase of the modulating frequency and, in addition, the ratio of the initial and final deviations for a doubling of frequency should have a value of 5.

1.9.1.2 *Attenuation of the intermodulation of base station transmitters*

The attenuation of intermodulation, due generally to the non-linearities of the output stage of the transmitter, should be at least 20 dB. Higher values of attenuation might be necessary and may be obtained by means of appropriate protection devices.

1.9.2 *Modulator characteristics*

1.9.2.1 *Limitation*

For a signal at a frequency of 1 kHz, with a level 20 dB greater than that which produces 20% of the maximum permissible deviation, the frequency deviation should be between 70% and 100% of the maximum.

1.9.2.2 *Sensitivity*

For a sound level at the microphone diaphragm of 93 dB relative to 2×10^{-5} Pascal, the deviation should be between 60% and 90% of the maximum permissible deviation.

1.9.2.3 *Audio-frequency response of the transmitter*

For a constant level of the modulating signal, the modulation index (phase modulation) or frequency deviation (frequency modulation) should remain constant, within limits of +1 to -3 dB, when the modulating frequency varies between 300 Hz and 3000 Hz.

1.9.2.4 *Residual modulation*

The residual modulation, in the absence of a modulating signal, should be lower at the output of a linear demodulator by 40 dB relative to the signal corresponding to a deviation of 60% of the maximum permissible deviation.

1.9.2.5 *Harmonic distortion*

The level of harmonic distortion should in no case exceed 10%.

1.9.3 *Receiver characteristics*

1.9.3.1 *Operation of the limiter*

When the radio-frequency varies between 6 dB(μ V) and 100 dB(μ V), the audio-frequency output signal should not vary by more than 3 dB.

1.9.3.2 *Co-channel rejection*

When a wanted signal is applied in the presence of an interfering signal on the same frequency, a reduction in the signal-to-noise ratio at the output from 20 dB to 14 dB should occur when the ratio of interference to signal is not less than -8 dB for 25 kHz channel spacing and not less than -12 dB for 12.5 kHz channel spacing.

1.9.3.3 *Duplex working*

Desensitization of the receiver with simultaneous transmission and reception should not exceed 3 dB.

1.9.3.4 *Output power at audio-frequencies*

The output power at audio-frequencies should not be less than 200 mW in the loudspeaker and 1 mW in the handset earphone.

1.9.3.5 *Audio-frequency response*

For a radio signal with a constant modulation index (phase modulation) or with a constant deviation (frequency modulation), the audio-frequency signal at the output should remain constant to within +1 dB to -3 dB when the modulating frequency varies between 300 Hz and 3000 Hz.

1.9.3.6 *Harmonic distortion*

The level of harmonic distortion should in no case exceed 10%.

1.9.3.7 *Noise and hum*

Noise and hum should not exceed -40 dB with respect to the output level produced by a strong radio signal modulated with a frequency of 1 kHz to a deviation equal to 60% of the maximum permissible deviation.

1.10 Japan has provided information [CCIR, 1978-82b] on tests performed on equipment designed for 12.5 kHz channelling FM systems in the 400 MHz band. The test confirmed that with special attention to protection against adjacent channel interference, a 12.5 kHz channelling FM system in the 400 MHz band, in accordance with characteristics and values of Recommendation 478 is possible. Other characteristics and values are as follows:

1.10.1 *Transmitter characteristics*

The maximum permissible frequency deviation should be 2.5 kHz. For modulating frequencies above 3 kHz, the level should decrease at a rate of 24 dB/octave.

1.10.2 *Receiver local frequency tolerance*

The receiver local frequency tolerance should be 3×10^{-6} .

1.10.3 *Co-channel interference*

The wanted-to-unwanted carrier power ratio (W/U ratio) giving a 12 dB SINAD ratio in the presence of co-channel interference (but with no adjacent channel interference) is shown in Table I. Measurements were made in the laboratory in accordance with IEC Publication 489. The tests showed that under these conditions the co-channel protection ratio required for 12.5 kHz-channelling equipment is only 2.4 dB higher than that required for 25 kHz channelling.

TABLE I

Channel spacing (kHz)	12.5	25
Maximum permissible frequency deviation (kHz)	2.5	5
Wanted-to-unwanted carrier power ratio (dB)	6.8	4.4

1.10.4 Adjacent channel interference

Subjective listener tests showed that adjacent channel interference in the 12.5 kHz channel spacing degraded the speech quality more severely, as compared to the case of 25 kHz channel spacing. Therefore, when introducing the 12.5 kHz channelling system, it is necessary to improve the wanted-to-unwanted power ratio by about 11 dB as shown in the Table II and Fig.5 in order to keep speech quality in 12.5 kHz channel spacing as good as that in 25 kHz channel spacing.

TABLE II - Opinion test conditions of adjacent channel interference

Signal		Modulation level (dB)	Frequency drift (PPM)	
			12.5 kHz	25 kHz
Desired		0	0	0
Undesired (Adjacent channel)	Case 1	0	0	0
	Case 2	10	4	7

Note 1. - The 0 dB modulation level corresponds to the IEC Standard Test Modulation.

Note 2. - Case 2 represents the worst case, i.e. channel spacing reduced by the drift.

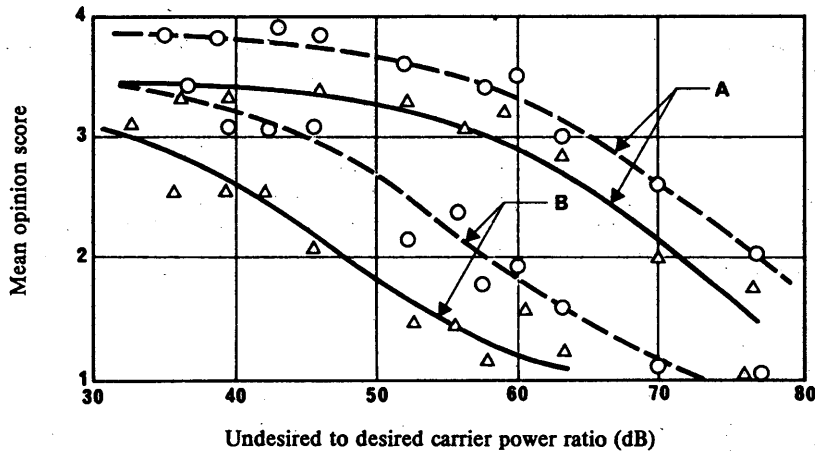


FIGURE 5

A: case 1

B: case 2

△ — △ : 12.5 kHz

○ — ○ : 25 kHz

Desired carrier median power: 9 ~ 13 dBμV

1.10.5 Receiver sensitivity

Figure 6 below shows a comparison of the sensitivity of 12.5 kHz and 25 kHz channel spaced receivers. The reference sensitivities for a 12 dB $(S + N + D)/(N + D)$ (SINAD) ratio are -2.5 and -3 dB(μ V), respectively.

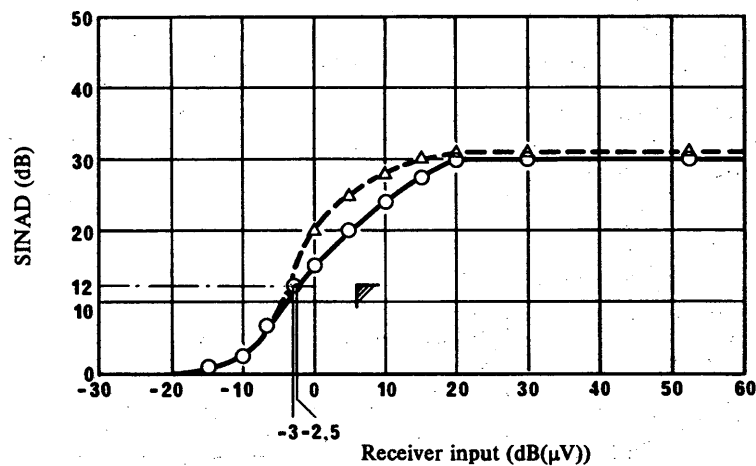


FIGURE 6

- — ○ — ○: 12.5 kHz channel spacing (± 1.5 kHz deviation)
- △ — △ — △: 25 kHz channel spacing (± 3 kHz deviation)

Figure 7 below shows a comparison of opinion tests on the speech quality of 12.5 kHz and 25 kHz channel spaced receivers.

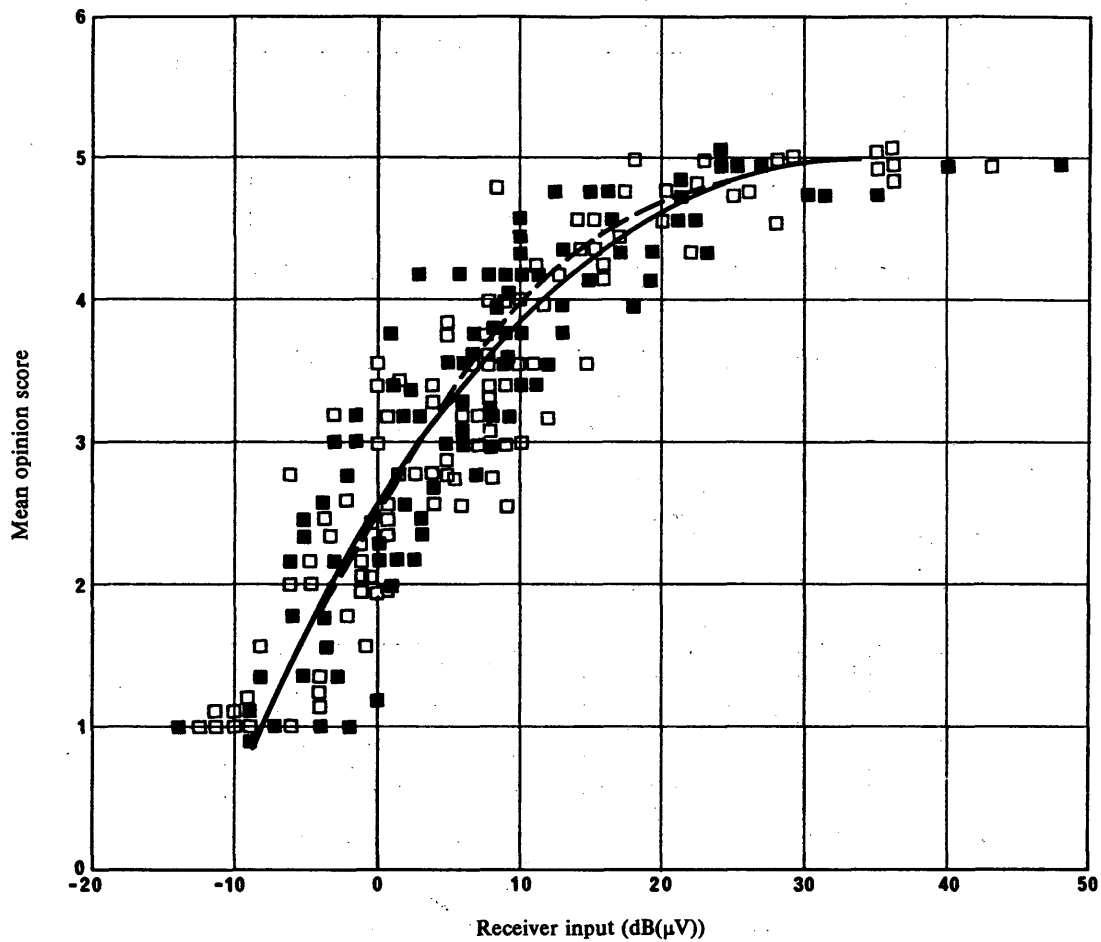


FIGURE 7

□ — □ : 12.5 kHz
■ — ■ : 25 kHz

1.11 Japan has confirmed in the laboratory that a 12.5 kHz channelling FM system in the 900 MHz band is achievable in accordance with the specifications in Recommendation 478. Only the specification of frequency stability in the 900 MHz band should be improved to approximately twice that of the 400 MHz band. Frequency stability and speech quality in the presence of thermal noise and co-channel interference are as follows.

1.11.1 Frequency stability

The frequency drift in the 900 MHz band should be less than 1.5×10^{-6} . It is possible to achieve this frequency stability by adopting the digital TCXO, as shown in Fig.8.

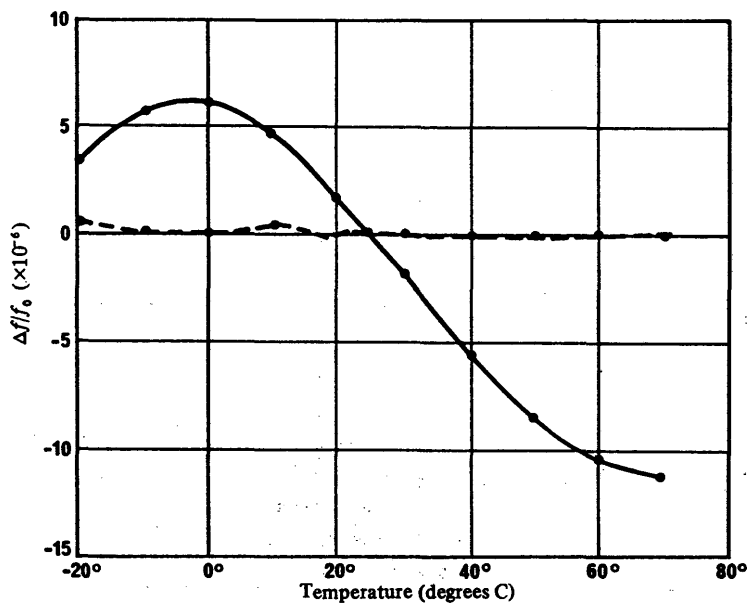


FIGURE 8 – Frequency temperature characteristics of DTCXO

$f_0 = 12.79985$ MHz (AT-cut, fundamental)

—•— uncompensated
 - -•- compensated

1.11.2 Speech quality

As shown in Fig. 9, the same speech quality with 12.5 kHz channel spacing as with 25 kHz channel spacing is obtained at almost the same receiver input voltage, since the C/N giving a MOS of 2.5 with 12.5 kHz channel spacing is only 2.5 dB larger than that with 25 kHz channel spacing. When introducing the 12.5 kHz channelling system, it is necessary to improve the required C/I by almost 3 dB to keep speech quality with 12.5 kHz channel spacing as good as that with 25 kHz channel spacing as shown in Fig.10. These conclusions remain to be confirmed by field tests.

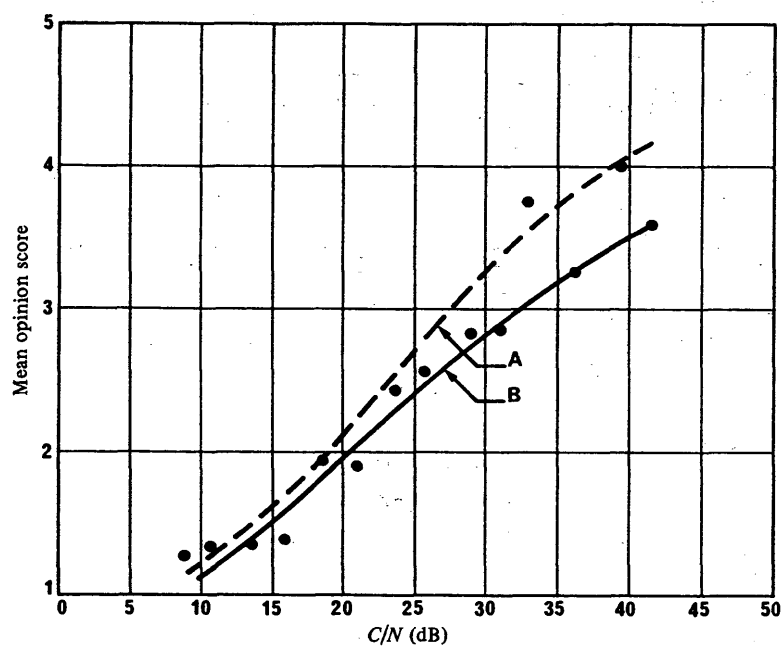


FIGURE 9 – Mean opinion score versus C/N

Curves A: standard modulating level = $3/\sqrt{2}$ rad r.m.s., channel spacing = 25 kHz

B: standard modulating level = $1.5/\sqrt{2}$ rad r.m.s., channel spacing = 12.5 kHz

Fading rate: 34 Hz

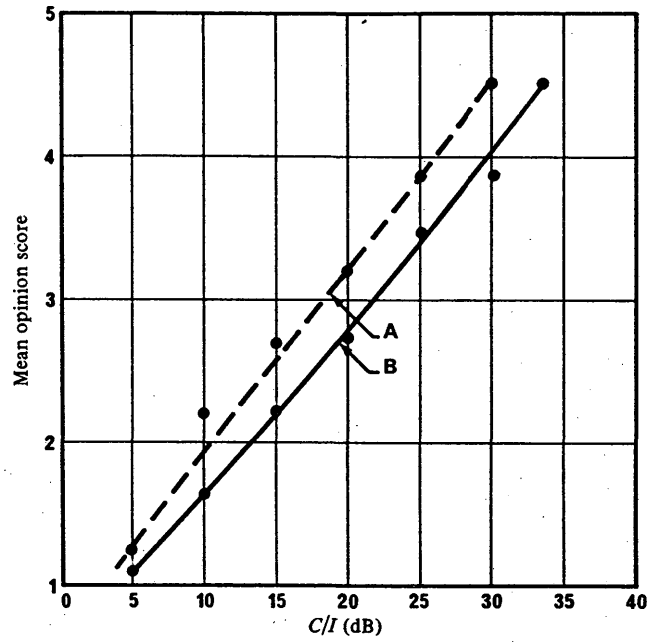


FIGURE 10— Mean opinion score versus C/I

Curves A: standard modulating level = $3/\sqrt{2}$ rad r.m.s., channel spacing = 25 kHz

B: standard modulating level = $1.5/\sqrt{2}$ rad r.m.s., channel spacing = 12.5 kHz

Fading rate: 34 Hz

2. Adjacent channel separations in various frequency bands

2.1 Separations for present and/or future use

TABLE III

Channel separation (kHz)	Frequency ranges (MHz)				
	25-50	50-100	100-200	200-500	500-1000
10	I* D DNK DNK* NOR*	S* SUI* USA NZL			
12.5	F I	F G IRL NZL	F G I IRL NZL E	F NZL AUS J G	J ⁽⁸⁾ S ⁽⁸⁾
15	TCH*	J	USA ⁽¹⁾		
20	CAN D USA ARG AUS	BEL ⁽²⁾ CAN D ARG	BEL ⁽²⁾ D F J ARG	BEL D	
25	AUS CHN DNK E ⁽⁶⁾ F FNL NOR POL DDR S SUI	ROU TCH USSR YUG I	BEL ⁽³⁾ CHN D ⁽⁴⁾ DNK E ⁽⁶⁾ FNL G ⁽⁷⁾ IRL ⁽⁷⁾ NOR POL DDR	ROU S SUI TCH USSR YUG NZL F I	AUS CAN ⁽⁵⁾ CHN D ⁽⁴⁾ DNK E ⁽⁶⁾ F FNL G IRL J NOR
30		AUS	AUS CAN ⁽⁵⁾ USA		USA ⁽⁸⁾ CAN ⁽⁸⁾ AUS ⁽⁸⁾ NZL ⁽⁸⁾
40	USA*				USA ⁽⁸⁾

* When a country symbol is followed by an asterisk, the value given for the separation refers to low-power portable equipment.

⁽¹⁾ An interleaved channel allocation plan is to be used and only alternate channels allocated in any one geographical area. Consequently, the equipment is designed for double the separation shown above.

⁽²⁾ 20 kHz separation is being introduced gradually for new equipment.

⁽³⁾ 25 kHz separation is still accepted in certain parts of the bands reserved for the fixed and mobile services.

⁽⁴⁾ Only in exceptional cases upon multilateral agreement.

⁽⁵⁾ An offset assignment plan is also used. The equipment is designed for the separation shown and advantage taken of geographic separation to assign offset channels at one half the separation shown.

⁽⁶⁾ All assignments are being allowed with this separation since September 1975.

⁽⁷⁾ Certain public services will continue to operate with 25 kHz channel separation.

⁽⁸⁾ Acceptable for cellular mobile telephone systems only.

2.2 Various administrations have interpreted the term "adjacent channel separation" in section 2.1 above in different ways and this has led to some confusion as to where administrations should enter their systems on the table.

An attempt was made at the interim meeting to define "adjacent channel separation", but the resulting discussion revealed that no one definition adequately accounted for all the various national systems. To illustrate the difficulty, adjacent channel separation could be defined, for example, in any one of the following ways:

- minimum frequency separation between channels in the coverage area of a base station;
- the separation between channels as assigned in the national frequency allocation plan;
- the occupied bandwidth of a channel defined by its class of emission;
- frequency separation between adjacent channels which allows for the use of adjacent channels within the same geographical area.

These are only examples, and other and more precise definitions are possible.

Administrations had been invited to input to the Final Meeting 1989 their views on how the information that Table III is intended to contain should be included in the report. Such inputs were to include defining more than one parameter for systems as, for example, is the case of Table I in Part C of Report 742-3 which applies to cellular systems only.

Also administrations had been requested to review the notes associated with Table III to determine if any of them need to be deleted or amended. The attention of administrations was particularly drawn to Notes 2, 6 and 9.

In spite of two input documents to the Final Meeting 1989 no solution to the problems described above could be found during that meeting.

Some administrations expressed the view that Table III is of no value and/or could be misleading, and therefore supported its deletion. Some other administrations, however, could not agree to this and wanted it retained.

Administrations, especially those who would like to see Table III retained, are therefore invited to input to the next Study Group 8 meeting their views on possible solutions.

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REPORT 899-1

**SYSTEMS OF MODULATION WITH HIGH SPECTRUM EFFICIENCY
FOR THE LAND MOBILE SERVICE**

(Question 7/8; Study Programme 7B/8)

(1982-1990)

1. Lincompex system (linked compressor and expander system)**1.1 Introduction**

The Lincompex system has been developed for improving the speech quality of HF radiotelephone service. The characteristics of Lincompex equipment for HF fixed telephone service and HF maritime mobile service, specified by Recommendations 455 and 475, are quite satisfactory for practical HF radiotelephone operations.

The congestion in the VHF band for land mobile services is a serious problem in many countries due to the rapid growth of the land mobile telephone service in limited geographical areas. Though the single-sideband technique is undoubtedly the best way for minimizing the occupied bandwidth, with ordinary single-sideband techniques, it is difficult to meet such inherent propagation conditions of the land mobile service as rapid fading and strong impulsive noise in urban areas.

The Lincompex system has high protection features against these conditions in itself.

It is generally understood that the Lincompex system is possibly applicable to the land mobile telephone service with some improvement of basic characteristics of HF Lincompex system.

1.2 Improved characteristics of Lincompex system for land mobile service

In order to overcome the inherent propagation conditions of the land mobile telephone service, the following improvements had been applied to the equipment used in trials effected in Japan [Kadokawa et al., 1981] :

- In order to reduce the Doppler effect and the influence of frequency error, the sensitivity of frequency modulation in the control channel is expanded to 6 Hz/dB (three times as large as 2 Hz/dB used in the HF band).
- In order to compensate for the deep and rapid fading, attack and recovery times of the compressor and expander are reduced to about one-third, and also the attack time of the fading regulator to about one-tenth compared with HF Lincompex; the lower limit of the fading regulator range is extended by about 15 dB; the level of the control channel is increased up to the same level of that of the speech channel.
- In order to improve the speech quality, a pre-emphasis circuit with 6 dB/octave characteristic is employed.

1.3 Test results

Satisfactory test results were obtained in a field test conducted in Japan. Table I summarizes the results obtained of average field strength corresponding to each grade of evaluation in the field test.

TABLE 1 - The measured results of average field strength necessary for each grade

Grade	Overall rating	Field strength in dB(μ V/m)		
		Lincompex	FM	SSB
5	excellent	41	47	(1)
4	good	25	30	37
3	fair	18	22	24
2	poor	18	16	19
1	unusable	(2)	(2)	(2)

(1) Not found.

(2) Not measured.

It is indicated, in the grading range of 3 to 5, that the Lincompex system tested gave the same grade of speech quality even though the field strength is lower by about 5 dB than that of the FM system. When compared with the SSB system without Lincompex, the evaluators for speech quality felt that the Lincompex system was hardly affected by fading and the protection against city noise was almost the same as that of the FM system.

As for the adjacent-signal selectivity, it is noted that the ratio of the level of unwanted radio signal to the level of wanted radio signal is more than 70 dB at the frequency separation of 6 kHz.

1.4 Future work

In order to improve the technical characteristics and contribute to the reduction of channel separation, further studies should be undertaken.

Administrations are invited to expedite the study of the reduction of channel separation in the VHF land mobile service.

There are however further factors that directly influence the degree to which Lincompex might contribute to reduction of channel separation. In particular, extremely high frequency stability is necessary to achieve the indicated results. Implementation of such stabilities are presently costly and cause substantial increase in the size of portable equipment and some mobile equipment.

2. SSB system (Yugoslavia)

2.1 Introduction

Several thousand VHF, SSB equipments have been used in land mobile applications in Yugoslavia [CCIR, 1974-78] for more than five years.

The performance of SSB was compared with FM by laboratory and also field tests using transceivers which could operate in both modes. The same receiver front-ends and transmitters linear power amplifier were used. The bias of the power amplifier was adjusted separately for the two modes.

2.2 Equipment characteristics

The transceiver characteristics are summarized in Table II.

TABLE II- Transceiver characteristics

Transmitter

	RF output power (W)	Bandwidth	Speech process
FM	2, 6, 20	± 5 kHz deviation	pre-emphasis
SSB	2, 6, 20	2.4 kHz	20 dB-clipper

Receiver

	Sensitivity	IF-bandwidth	Speech process
FM	1 μ V/20 dB SINAD	16 kHz/6 dB	de-emphasis
SSB	1 μ V/20 dB SINAD	2.4 kHz/6 dB	

2.2.1 Frequency stability

A temperature compensated crystal oscillator (TCXO) was used, having the advantages of low power consumption, zero warm up time, and small size.

Frequency variation with temperature: $\pm 1 \times 10^{-6}$ (-30 to $+60$ °C). This stability is sufficient to provide reliable SSB communication up to 100 MHz.

2.3 Test results

2.3.1 Power consumption

In portable equipments, the weight, size and power consumption are most important. Comparisons are based on the same SSB and FM RF output power. In Table III, direct comparison of power consumption between FM and SSB in the same equipment is shown.

TABLE III-Comparison of power consumption of FM and SSB transmitters

Transmission	RF power output	Power consumption (W)		Intermodulation product level (dB)
		Final power amplifier(3)	Transmitter	
FM	20 W_{rms}	36	52	
SSB-I(1)	20 W_{pep}	22	36	-30
SSB-II(2)	20 W_{pep}	14	22	-30

$f = 37$ MHz

(1) SSB-I: Two-tone test conditions (CCIR).

(2) SSB-II: Average speech, clipping ratio of 20 dB.

(3) Final RF power amplifier is separately optimized for SSB and FM.

The comparison of power consumption at different RF output power levels can be seen on Fig. 1.

The diagram shows that the power consumption is more than two times greater in a FM system than in a SSB system of equivalent RF output power.

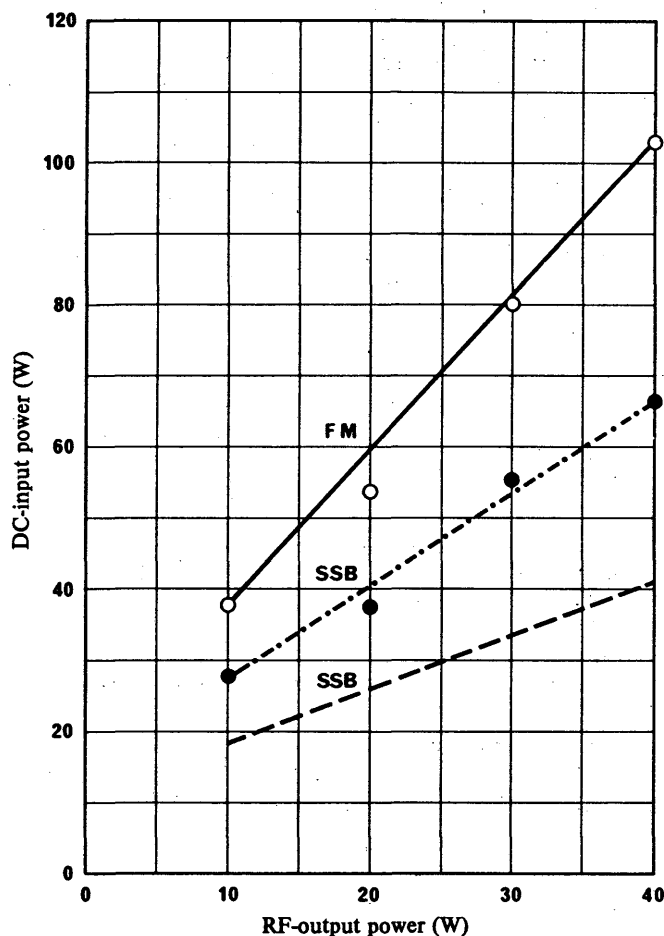


FIGURE 1 – Comparison of power consumptions at different RF power levels

————— FM
 SSB 2-tone
 - - - - - SSB average speech
 Intermodulation product level: -30 dB
 $f = 37 \text{ MHz}$

2.3.2 Maximum range

To compare the efficiency of FM and SSB modulation techniques with respect to intelligibility, a logatom articulation method was used. A complete test comprised 2000 logatoms.

The results of laboratory tests are given in Fig. 2, from which it can be seen that in conditions corresponding to maximum range, the FM transmitter must provide 6 to 10 dB more RF output power than the SSB transmitter.

The extensive field tests carried out show that at maximum range conditions the same acceptable quality of communication was obtained at the level of 2W PEP-SSB and 20W FM radio-frequency output power.

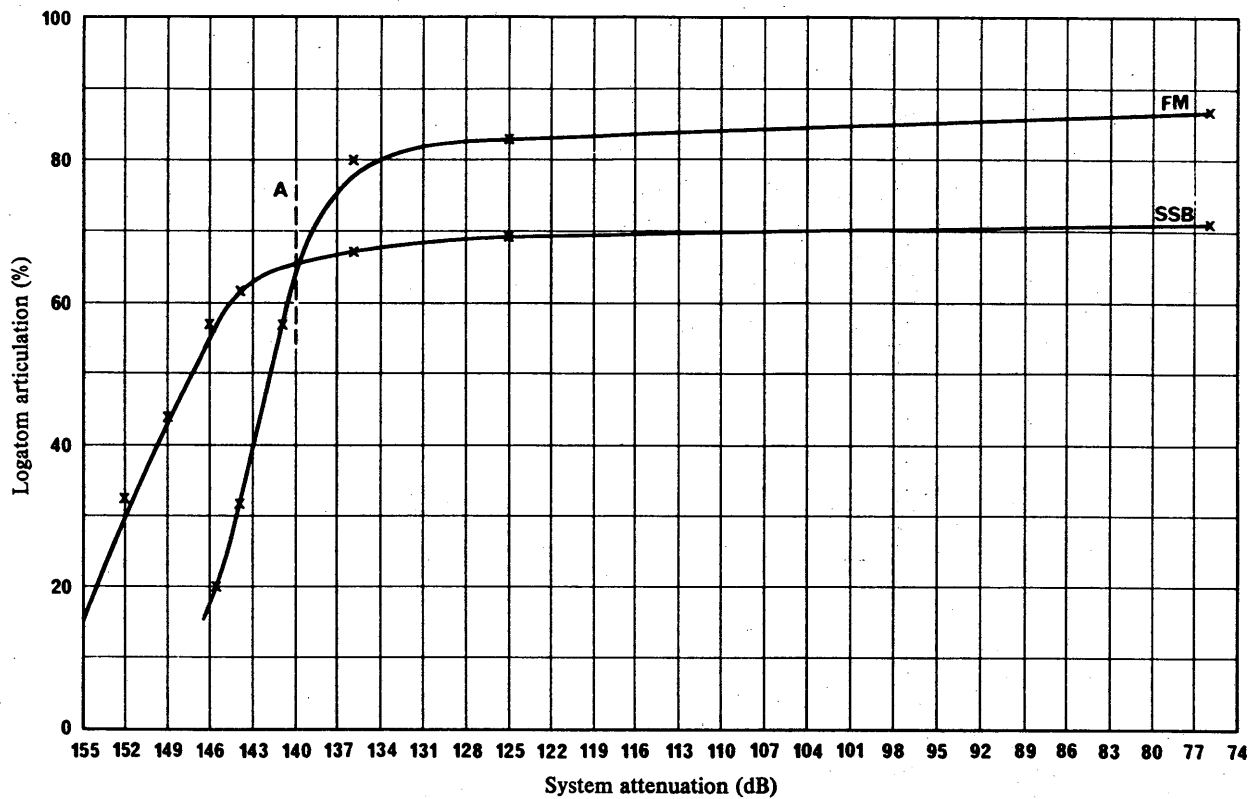


FIGURE 2 - Logatom articulation versus system attenuation

A: FM/SSB receiver sensitivity 1 μ V/20 dB SINAD

FM: RF output power 2 W

SSB: RF output power 2 W (PEP) and 20 dB clipping

2.4 Conclusions

Considerable additional study should be done on:

- adjacent channel selectivity;
- co-channel interference, SSB/SSB and SSB/FM;
- speech processing methods;
- immunity to impulsive noise.

3. SSB system (United Kingdom)

3.1 Introduction

The possibility of using single sideband (SSB) systems to relieve spectrum congestion in the land mobile radio bands below 500 MHz has been investigated in the United Kingdom and a number of laboratory and field tests have been carried out to evaluate the performance of different SSB systems compared to existing AM and FM systems.

3.2 SSB systems

In order to meet the automatic gain control (AGC) requirements of SSB systems operating in a land mobile radio environment, where the received signal is subject to rapid fading, a continuous signal must be transmitted with the speech sideband, to maintain the AGC.

The continuous signal may be provided by one of the following systems:

- a pilot carrier;
- a tone in, or at the edge of, the audio passband;
- a Lincompex control subcarrier;
- a data subcarrier.

Studies in the United Kingdom have concentrated on the first of these systems with pilot signals in the range -10 to -20 dB relative to peak envelope power (PEP).

The necessary RF bandwidth is dependent on the audio bandwidth required and the type of pilot signal employed but in general is not greater than 3000 Hz for land mobile applications.

Feedback AGC, using the pilot signal as a reference may be employed at frequencies up to about 200 MHz. For UHF applications a combination of feedback and feed forward AGC may be employed.

3.3 *Channel spacing and frequency tolerance*

An overall system frequency tolerance of 500 Hz will permit the use of 5 kHz channel spacing.

Economic considerations make it advantageous to employ a high stability base station, with a narrow frequency tolerance, and therefore allow a greater frequency tolerance (approaching ± 500 Hz) for the more numerous mobiles.

3.4 *Impulsive interference*

Experiments have shown that an SSB receiver with an i.f. bandwidth of 3 kHz is not unduly sensitive to impulse noise [Gosling *et al.*, 1979]. Although impulses typical of ignition noise are stretched out to a greater extent by the narrow filter of the SSB receiver the total energy passed by the filter is correspondingly less. Side-by-side comparisons with 25 kHz spaced FM equipment in a high noise environment have shown that the subjective effect of ignition noise is approximately the same [Wells, 1978].

3.5 *Co-channel interference*

Subjective tests have demonstrated that under actual mobile-radio field conditions the effects of co-channel interference in an SSB system with 5 kHz channelling are very similar to those encountered by 25 kHz FM systems [Garner, 1980]. In both cases a signal-to-interference ratio of the order of 16 dB is necessary to achieve a quality of reception corresponding to grade 3 in Table II of § 1.4 of this Report.

The range of subjective gradings observed with various co-channel interference ratios is given in Figs. 3a, 3b and 3c for different wanted signal levels and carrier offsets. These curves are based on subjective tests carried out in the United Kingdom [Garner, 1980] in which the wanted and unwanted signals were of the same modulation type. The wanted signal was on tune and the unwanted signal was at a carrier frequency offset of 75 Hz for SSB and 500 Hz for FM, which was considered to give the worst result in each type of system.

Further subjective tests, carried out by the United Kingdom Home Office, have shown that in almost all conditions 25 kHz FM equipment was found to give better co-channel performance than SSB, 12.5 kHz or 12.5 kHz AM. However, in urban and suburban areas SSB was not shown to be markedly inferior to 25 kHz FM and in all conditions SSB performed at least as well as 12.5 kHz FM or AM.

3.6 *Compatibility with existing control signals*

When a pilot signal is used for automatic frequency control most standard signalling systems can be employed.

3.7 *User evaluation trials*

The results of a user evaluation trial carried out in the United Kingdom under typical private land mobile conditions and using speech [Barnes, 1981] have shown that from a users assessment 5 kHz SSB equipment could be used without any loss in quality or intelligibility when compared with 12.5 kHz equipment. Furthermore, in some high field strength situations the quality of the SSB systems would be better than that of existing 25 kHz FM systems.

However, it was judged that under almost all conditions 25 kHz FM equipment would provide a higher quality service.

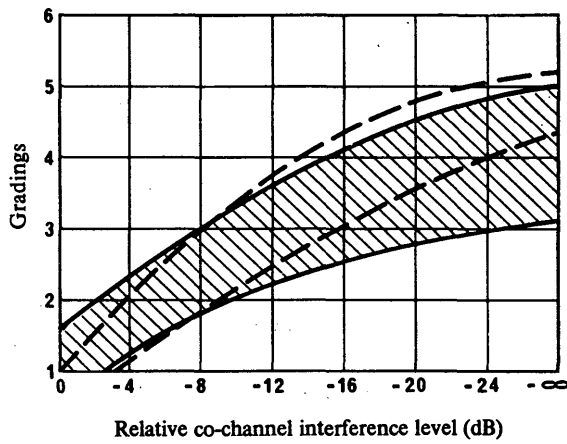


FIGURE 3a

— : SSB
 - - - : FM
 Wanted signal: 10 μ V
 Carrier offset: zero

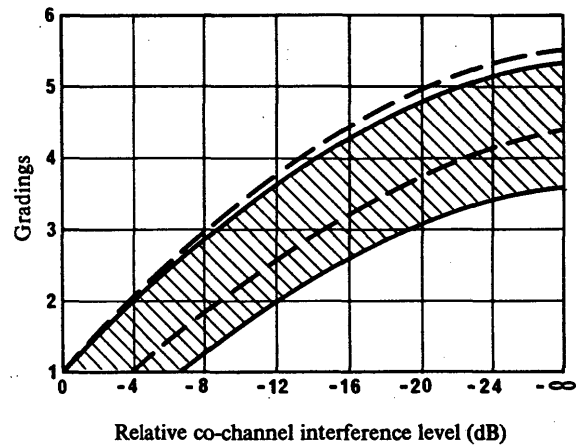


FIGURE 3b

— : SSB
 - - - : FM
 Wanted signal: 3 μ V
 Carrier offset: 75 Hz for SSB
 500 Hz for FM

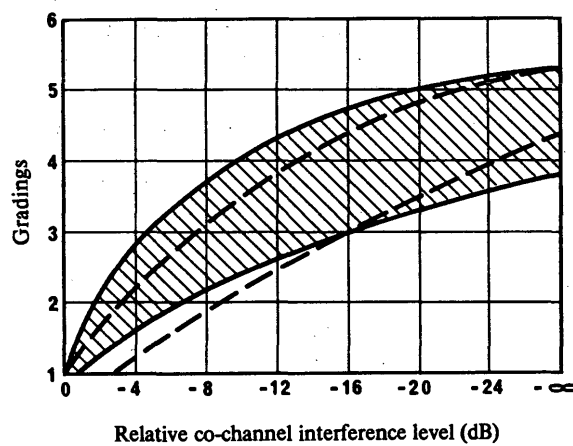


FIGURE 3c

— : SSB
 - - - : FM
 Wanted signal: 10 μ V
 Carrier offset: 75 Hz for SSB
 500 Hz for FM

4. SSB system (United States of America)

4.1 Introduction

Laboratory experiments have been conducted in the United States on the use of Amplitude Companded Single Sideband (ACSB) VHF radios for mobile communications. ACSB radios employ both Amplitude and ACSB Companding and audio pre-emphasis and de-emphasis voice processing. Some comparisons are made with FM, AM and ACSB receivers.

4.2 Tone and voice co-channel tests

The following type of receivers were tested:

- ACSB: For 3K00 R3E emission with 4:1 expander and 12 dB/octave de-emphasis
- 25 kHz FM: For 16K0 F3E emission 6 dB/octave de-emphasis
- 12.5 kHz FM: For 11K0 F3E emission 6 dB/octave de-emphasis
- AM: For 6K00 A3E emission

The reference sensitivity of each receiver was measured in accordance with IEC Publication 489 and adjusted for equal noise figure of 10 dB. The reference sensitivity of the four receivers are as follows:

dB (μV) at receiver input

ACSB	– 6.0
25 kHz FM	– 12.5
12.5 kHz FM	– 15.0
AM	– 3.0

The range of the SINAD performance between -20 dB (μV) and 10 dB μV signal input to each receiver is given in Fig. 4. These results do not necessarily reflect system performance under other than constant tone modulation.

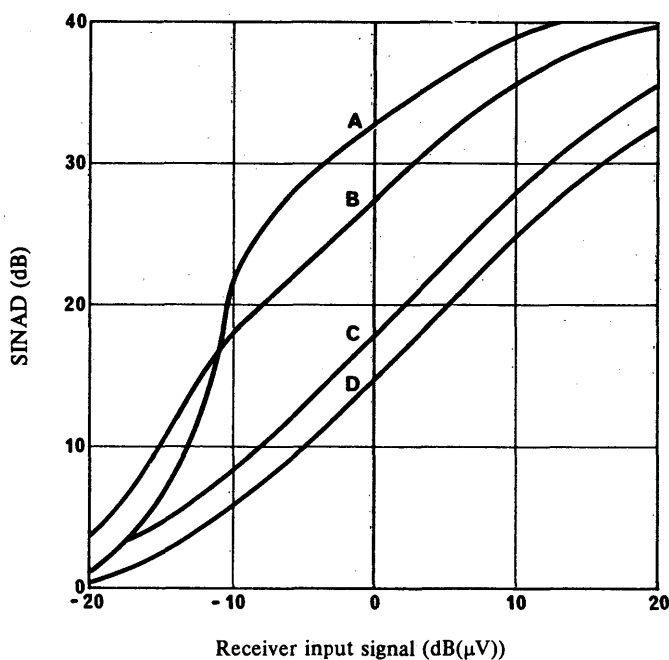


FIGURE 4 – Receiver sensitivity

A: FM – 25 kHz
 B: FM – 12.5 kHz
 C: ACSB
 D: AM

Co-channel interference tests using voice modulation were performed on three of the four receivers.

The protection is based on the ratio (dB) of the mean wanted signal to the mean interfering signal required to prevent an impairment of speech quality. The wanted signal was modulated with extracts from an English language news broadcast and the interfering signal was modulated by a female speaker reading Harvard sentences.

Each signal was passed through a separate multipath simulator and adjusted in level as described above. During the tests the wanted signal mean value was set at a fixed level, and the listener was instructed to set the interfering signal to the highest level and yet not impair speech quality. The results, shown in Fig. 5, were averaged for a team of 5 persons.

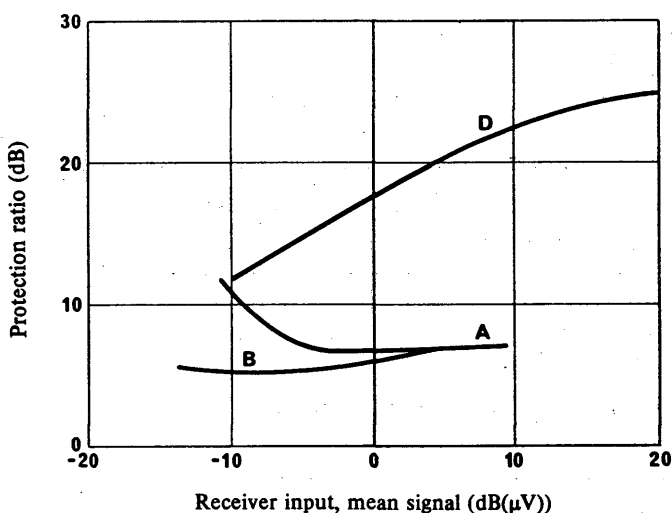


FIGURE 5 - Co-channel protection ratio voice modulation, Rayleigh fading

A: FM - 25 kHz
 B: FM - 12.5 kHz
 C*: ACSB
 D: AM

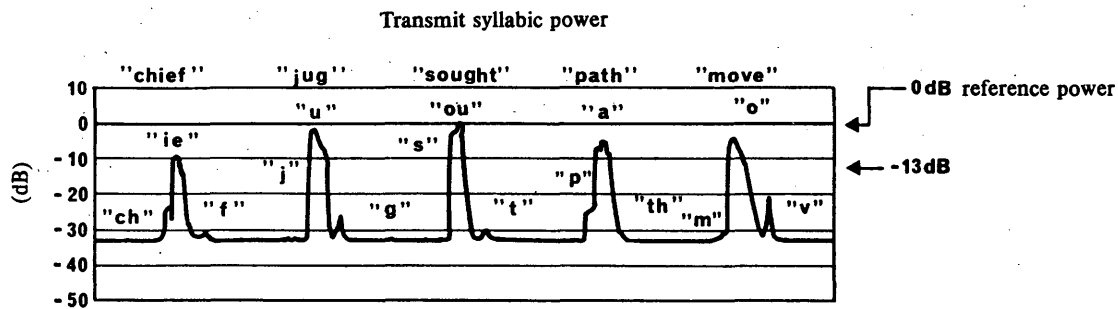
* Complete test results for the ACSB receiver are not yet available, but will be added at a later date.

4.3 Wanted signal voice tests (ACSB receiver)

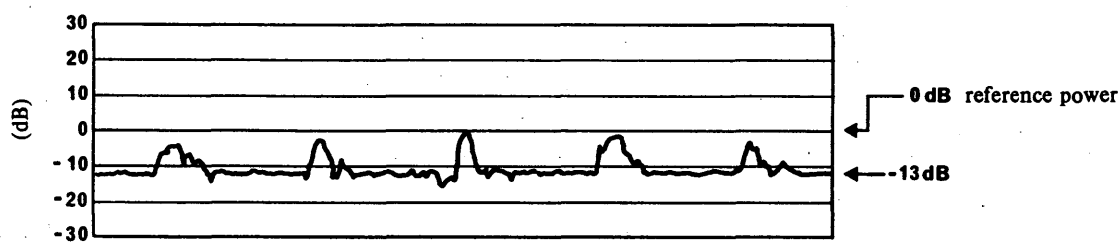
The attached Fig. 6 illustrates how pre-emphasis, de-emphasis and amplitude companding improves the signal-to-noise ratio of certain types of syllables as compared to an SSB transmitted and received signal without the use of audio processing.

Meaningful measurements of ACSB radio performance have been difficult to obtain using standard objective test procedures because these procedures may not account for the use of voice processing. Subjective tests (diagnostic rhyme test) have indicated that ACSB performance is better than implied by objective tests. Such conclusions must be interpreted with care.

The experiments seem promising. Development and field testing continue in the United States.

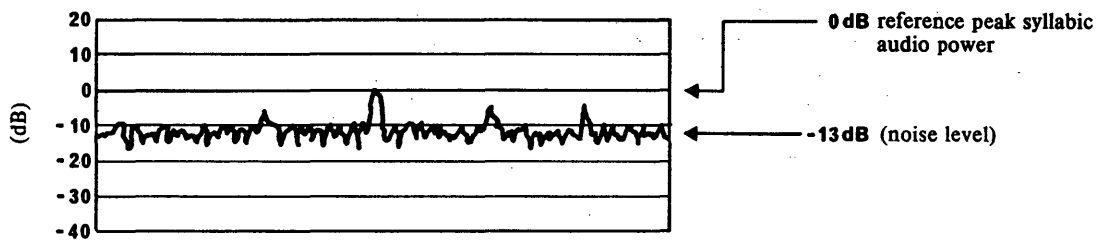


a) SSB

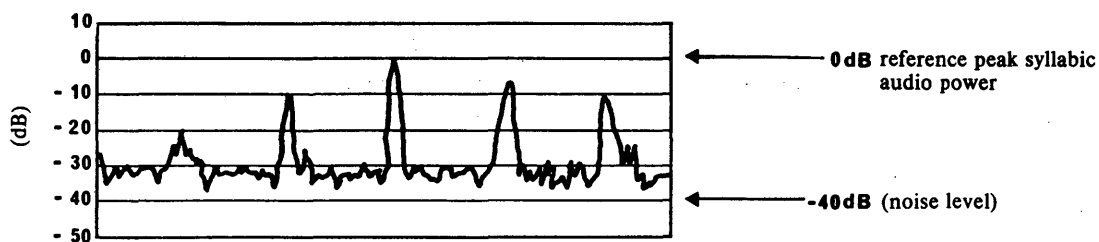


b) ACSB

Receiver audio output signals with noise 13 dB below 0 dB reference power



c) SSB



d) ACSB

FIGURE 6 – Performance of SSB and ACSB radios on English language test words in the absence of multipath fading

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REPORT 1018-1*

**CO-CHANNEL AND ADJACENT-CHANNEL COORDINATION CRITERIA
FOR SIMULTANEOUS USE OF DIFFERENT MODULATION
TECHNIQUES IN THE MOBILE SERVICE**

(Question 72/8 and Study Programme 7B/8)

(1986-1990)

1. Introduction

The United States of America has conducted laboratory and field tests comparing amplitude companded single sideband (ACSSB) to conventional FM for VHF mobile radio. The results of this study are contained in a report issued by the FCC [FCC, 1983]. Some of the information is extracted and included in Figs. 1, 2 and 3. In addition, the National Telecommunications Information Administration [NTIA, 1983] conducted some measurements, and issued a report.

In Canada, _____ a number of laboratory and field tests were carried out on ACSSB systems. These tests demonstrated satisfactory performance with regard to the quality of reception, coverage and immunity to noise. In these tests ACSSB compared favourably with FM [Bischoff and Sieb, 1982; Bonney, 1982].

During the period 1984-1987, _____
_____ subjective and objective laboratory tests of ACSSB systems were carried out by the Communication Research Center, Department of Communications, Canada [Burke and Boucher, 1984]. Intelligibility of voice signals was evaluated for various combinations of wanted and unwanted signals, as a function of frequency separations.

A Canadian experiment was also conducted to identify the possible sharing criteria between frequency hopping (FH) spread spectrum and other mobile services.

* This Report should be brought to the attention of Study Group 1.

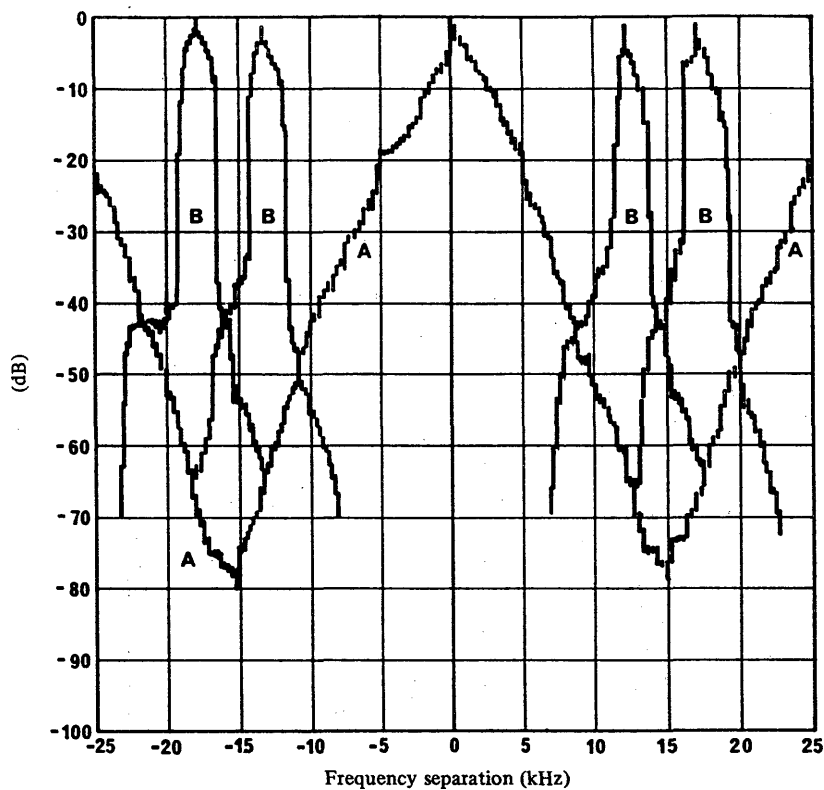


FIGURE 1 – Superimposed spectrum emissions of FM and ACSSB, 30 kHz spaced FM, with 12.5 kHz offset ACSSB drop-ins

Curves A: FM
B: ACSSB

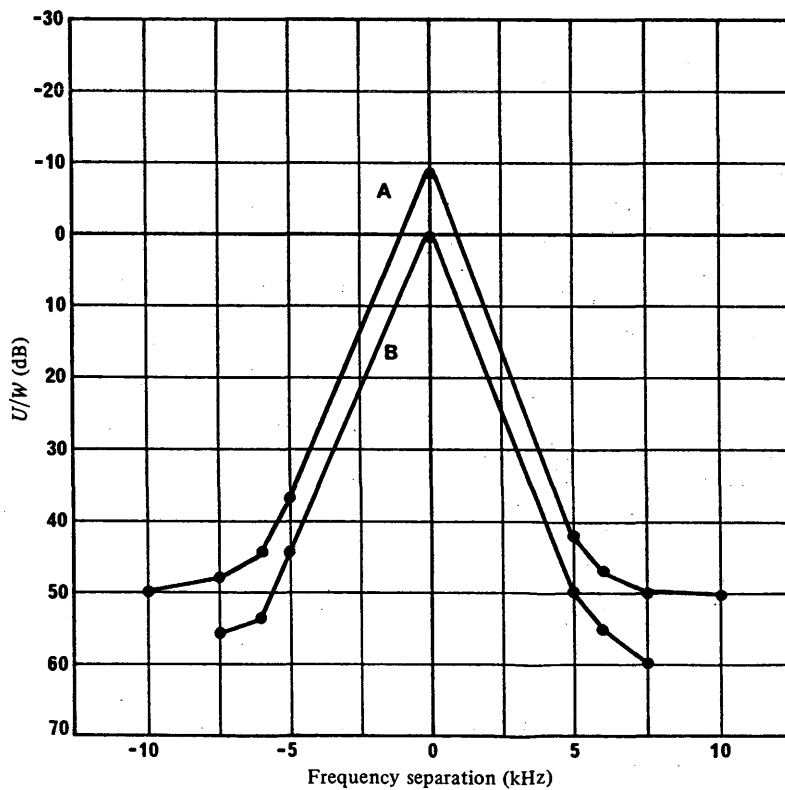


FIGURE 2 – Unwanted/wanted ratio for ACSSB into ACSSB interference

Curves A: just noticeable
B: disruptive

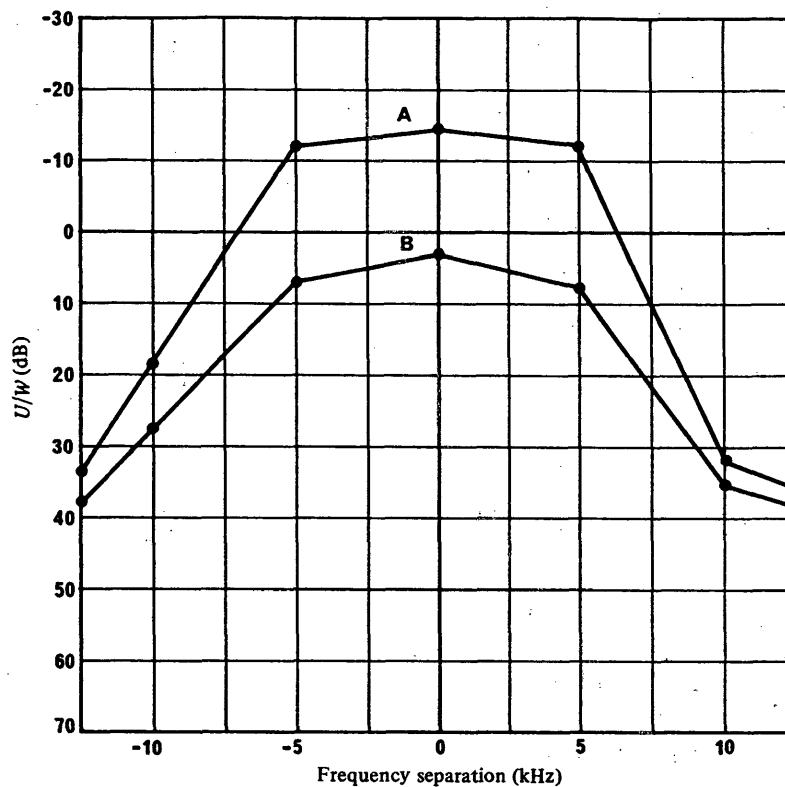


FIGURE 3 – Unwanted/wanted ratio for ACSSB into FM interference

Curves A: just noticeable

B: disruptive

2. Results of the United States' tests on ACSSB versus FM

Figure 1 shows superimposed spectrum emissions of FM and ACSSB systems with 12.5 kHz offset ACSSB channels between existing 30 kHz spaced FM channels. This is referred to as ACSSB drop-in channels. This causes degradation of protection to the wanted FM signal because of the full power of the adjacent ACSSB drop-in signal at the band edge of the FM channel. The intermediate frequency (IF) bandpass of typical FM receivers in use today provides only limited attenuation of signals at 12.5 kHz from centre frequency. Therefore, sufficient geographical separation would be necessary in order to place ACSSB channels (drop-ins) between existing FM channels. A typical FM system operating at 25 or 30 kHz spacing can offer in excess of 70 dB adjacent-channel protection to the wanted FM signal. With ACSSB drop-in channels, the adjacent-channel protection to the wanted FM signal could be as low as 40 dB.

The unwanted to wanted (U/W) ratios for an ACSSB signal from an unwanted ACSSB signal are shown in Fig. 2. The data in Fig. 2 were taken at co-channel, ± 5 kHz, ± 6 kHz, ± 7.5 kHz, ± 10 kHz and ± 12.5 kHz offsets. The data taken beyond 7.5 kHz was limited by the test set-up. The U/W ratio may be greater than 50 dB beyond 7.5 kHz. A positive U/W value indicates that the unwanted signal is stronger than the wanted signal.

The U/W ratios for an FM signal from an unwanted ACSSB signal are shown in Fig. 3. The data in Fig. 3 were taken at co-channel, ± 5 kHz, ± 10 kHz, ± 12.5 kHz offsets. The data shown for both tests represent the average values of what the observers individually obtained. The average difference in data logged between observers was less than 1 dB for "just noticeable" and less than 0.5 dB for "disruptive".

To determine the unwanted to wanted ratios, the wanted channel attenuator in the receiver test area was adjusted for the appropriate test level at the receiver. For each test, the unwanted channel attenuator was adjusted by the observer for "just noticeable" interference, and the difference between attenuators recorded. The unwanted channel attenuator was then adjusted by the observer for "disruptive" interference, and the difference between attenuators recorded. This procedure was repeated for all five wanted levels.

Just noticeable was defined as interference which is detectable but would not be observed if the interfering signal was reduced slightly in level (1 dB).

Disruptive was defined as interference which caused some words to be garbled, requiring occasional repeated message transmissions. Using random sentences with an average length of six words, two or more garbled words in three consecutive sentences was the criterion used to require a repeated transmission.

Any channel plan based on the U/W ratio will need to take account of the frequency stability of the interfering transmitter, and the victim receiver. The selectivity characteristics of the receiver will also affect the U/W ratio. Development and field testing will continue in the United States of America.

3. Results of the Canadian tests on ACSSB versus FM

3.1 Criteria used for interference protection

3.1.1 Interference protection ratio (objective tests)

The "interference protection ratio" is the ratio (in decibels) of the interfering signal relative to the wanted signal level present at the receiver input that would result in degradation in the SINAD at the receiver from 12 dB to 6 dB. The desired signal was modulated with a 1 kHz tone while the interfering signal was modulated with voice.

3.1.2 Disruptive interference (subjective tests)

"Disruptive interference" is that level of interference which will result in some words being garbled, requiring some occasional repeat of message transmission. Both wanted and interfering signals are modulated with voice.

3.1.3 "Disruptive interference ratio" (subjective tests)

The "disruptive interference ratio", with the wanted signal level adjusted to the same level as the 12 dB SINAD, is the ratio of interfering signal relative to the wanted signal that would result in disruptive interference conditions.

3.2 Results for ACSSB to FM and FM to ACSSB interferences

The critical values of relative interfering signal strength to produce objective and disruptive interferences for the ACSSB and FM units are summarized in Table I. Figures 4 and 5 show the results for disruptive and objective interferences, respectively.

TABLE I

Interference	Frequency Separation (kHz)	Objective Interference Protection Ratio (dB) (desired channel modulated with 1 kHz tone)	Disruptive Interference Protection Ratio (dB) (desired channel modulated with voice)
ACSSB to FM	7.5	3	4
	12.5	45	42
	15.0	60	57
	20.0	72	73
FM to ACSSB	7.5	26	28
	12.5	63	57
	15.0	72	70
	20.0	83	80
	25.0	85	87

3.3 Results for ACSSB to ACSSB, and FM to FM interferences

The results of the tests for objective and disruptive interference protection ratios for ACSSB to ACSSB and FM to FM interferences are given in Table 2. Figures 4 and 5 show the protection ratio (dB) for subjective disruptive and objective interferences, respectively.

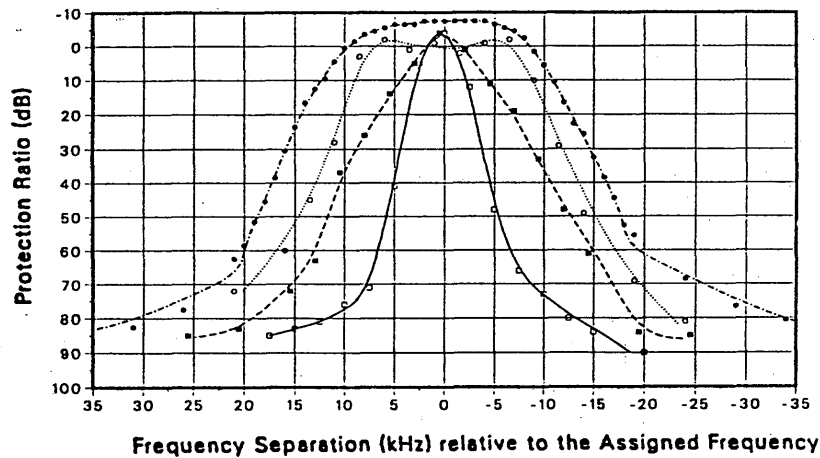


FIGURE 4

Objective 12 to 6 dB SINAD with Voice Interfering

LEGEND

- ACSSB into ACSSB
- FM into ACSSB
- ACSSB into FM
- FM into FM
-

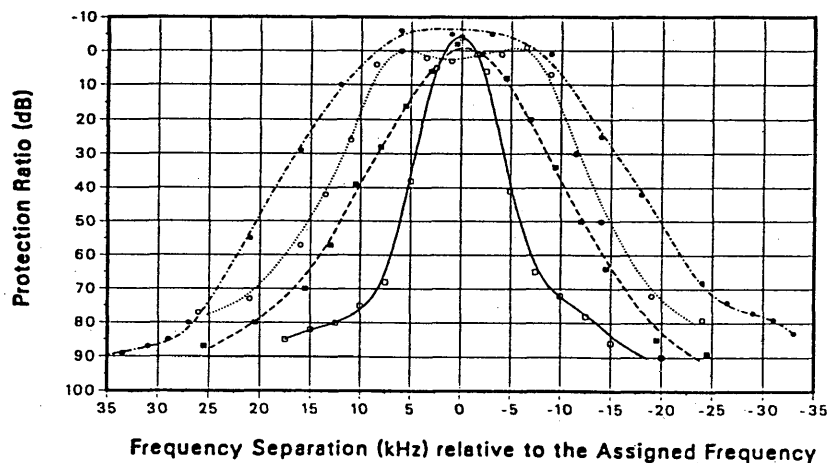


FIGURE 5

Subjective Disruptive Interferences

LEGEND

- ACSSB into ACSSB
- FM into ACSSB
- ACSSB into FM
- FM into FM
-

TABLE II

Interference	Frequency Separation (kHz)	Objective Interference Protection Ratio (dB) (desired channel modulated with 1 kHz tone)	Disruptive Interference Protection Ratio (dB) (desired channel modulated with voice)
FM to FM	11	9.5	10
	15	30.5	29
	20	62.5	55
	25	77.5	73
	30	83.0	87
ACSSB to ACSSB	2.5	5.0	5
	5.0	41.0	38
	7.5	71.0	68
	12.5	81.0	80
	15.0	83.0	82
	17.5	85.0	85

4. Canadian measurements of protection ratios from frequency (FH) hopping spread spectrum

Measurements were conducted to identify co-channel interference criteria applicable to the land mobile service operating in the 30-50 MHz band. Three commercial FM receivers, having channel spacings of 20, 50 and 25 kHz respectively, were used in the measurements. Signal-to-interference (S/I) protection ratios for co-channel operation were measured using the just perceptible (JP) and non-intelligible (NI) interference criteria which are respectively about equivalent to the MINIT and 0.3 AI, as defined in Reports 526 and 525. The results of these measurements are reported in detail in Report 826.

The experimental evidence indicates that large propagation losses are required to protect an FM receiver against the just perceptible level of FH co-channel interference. On the other hand surprisingly small propagation losses are required to protect against the non-intelligible harmful interference level. Moreover, considering that present land mobile users of the 30-50 MHz band use analogue transmission techniques, an interference of 10 ms duration, such as the one used in the Canadian experiment, will not be annoying if it does not occur too often. For a given land mobile channel, interference from one source was present on average 0.4% of the time so that a land mobile system (analogue voice) can tolerate a greater number of FH sources, the maximum number being determined by the desired grade of service.

This experiment did not attempt to evaluate the impact of FH interference on digital systems nor did it study the relationship between the number of FH interference sources and the degradation of a land mobile signal.

It should be noted that the degradation is additive, in the sense that the simultaneous operation of n frequency hopping transmitters will increase the duty cycle of the interference by a factor n . Finally, values given by Recommendation 478 have to be kept in mind (spurious emissions on discrete frequencies are limited to 2.5 μ W or to 70 dB below the carrier power).

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REPORT 1019*

SOURCES OF UNWANTED SIGNALS IN MULTIPLE BASE STATION SITES IN THE LAND MOBILE SERVICE

(Question 7-2/8)

(1986)

1. Introduction

The greatly increased use of land mobile services has resulted in a dramatic increase in the number of base stations on any one site, particularly on those sites strategically placed to serve large built-up areas. This has led to instances of severe interference due to unwanted signals being generated at the site. This Report is not intended to examine every possible type of interference but rather to indicate the more commonly occurring sources. It should be particularly noted that transmitters of other services may be involved.

2. Simple frequency relationships

As land mobile frequency bands are used throughout the VHF/UHF spectrum there may be harmonic relationships between frequencies in the various bands. The equipment cabinet, the power supply cabling and land-line cabling can contribute to the level of these unwanted harmonic signals.

Other interfering signals can be caused by simple mixes either in transmitter output stages or at the antenna mast. As an example, if the signal from a VHF broadcasting transmitter at 93 MHz mixes with a signal of the mobile service at 170.5 MHz, a difference signal of 77.5 MHz can be produced. This can cause a problem if it is a receive frequency of the mobile service.

3. Complex frequency relationships

3.1 *Generation of intermediate frequency and/or its derivatives*

Interference can be caused in a receiver where signals are received from two transmitters whose frequencies are separated by an amount equal to the IF, or a submultiple of the IF, of the receiver.

3.2 *Generation of transmit/receive (Tx/Rx) difference frequency*

This problem arises on sites where there are several base stations having "repeater" or "talk-through" facilities, i.e. the transmitters and receivers are in use simultaneously. If the Tx/Rx spacing is constant (D), an incoming signal from a mobile station will produce in the base station transmitter output stage a difference frequency, D . Any other base station transmitter may now mix with D to produce its own receiver frequency in the same band.

4. Intermodulation products

4.1 *Generated external to the site*

Under this heading, products arise from stations on adjacent sites, and, in particular, the third order product i.e. $2f_1 - f_2$, which is prevalent in large built-up areas. In some instances significant intermodulation products up to and including the seventh order have been noted and in exceptional cases the interference has been traced to the nineteenth order.

4.2 *Intermodulation products generated on-site by non-linear junctions on the mast*

More study is required to verify the mechanisms and levels of such interference, which certainly exists in the land mobile bands. However, at lower radiated powers, the significance of these products is reduced, compared with other forms of non-linearity, e.g. § 4.1 and 4.3.

4.3 *Intermodulation products generated on-site by non-linearity in components of the system*

Junctions between dissimilar metals cause non-linearity, and therefore intermodulation products, when subjected to radio frequency currents, and recent work has highlighted such products up to the eleventh order at VHF caused by connectors, cables and dissimilar junctions in what might be regarded as otherwise innocuous components.

For the long-term development of the land mobile radio industry, it may be necessary to define the non-linearity of passive components in the system.

* This Report should be brought to the attention of Study Group 1.

5. Transmitter noise

Until quite recently, most transmitters on base station sites had valve output stages, which fortuitously were not a major contributor to the noise spectrum compared with the more modern solid-state output stages.

With a valve output stage, the unwanted noise is generally narrow-band, having frequencies which are multiples of the crystal oscillator frequency or a combination derived from the multipliers. However, in the case of solid-state output stages the noise is generally wideband and higher in level.

Figures 1, 2 and 3 give the graphical results of measurements made in the United Kingdom of noise from VHF transmitters with thermionic valve output stages and with solid-state output stages for the VHF "high band" (150-170 MHz) and VHF "low band" (71.5-87.9 MHz).

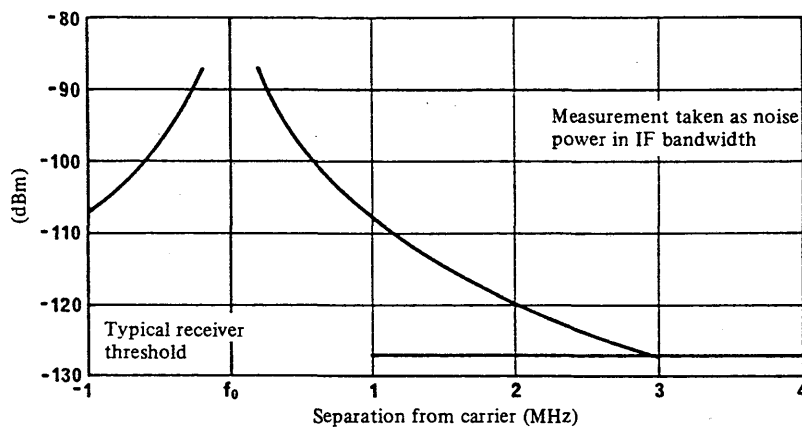


FIGURE 1 – Typical transmitter noise VHF high band (150-170 MHz):
thermionic output stage

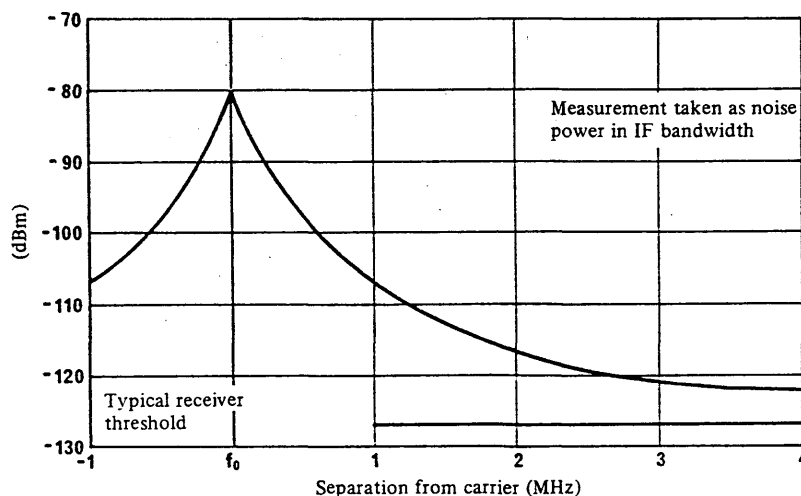


FIGURE 2 – Typical transmitter noise VHF high band (150-170 MHz):
solid-state output stage

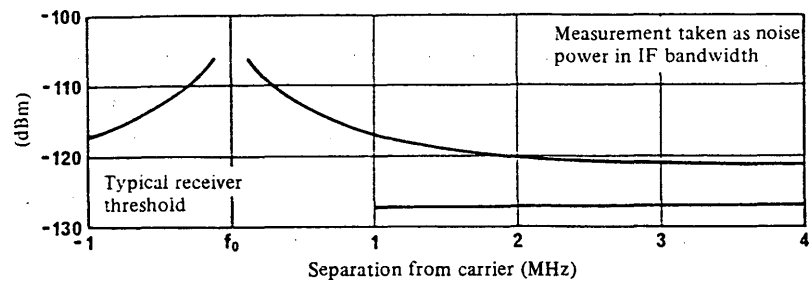


FIGURE 3 – Typical transmitter noise VHF low band (71.5-87.9 MHz):
solid-state output stage

Figure 4 shows practical operational noise curves for a base station without filtering of spurious emissions and noise from the transmitter output (curves A). These results can be compared with those of curves B which show the benefit of filtering in reducing noise and unwanted emissions by some additional 30 dB.

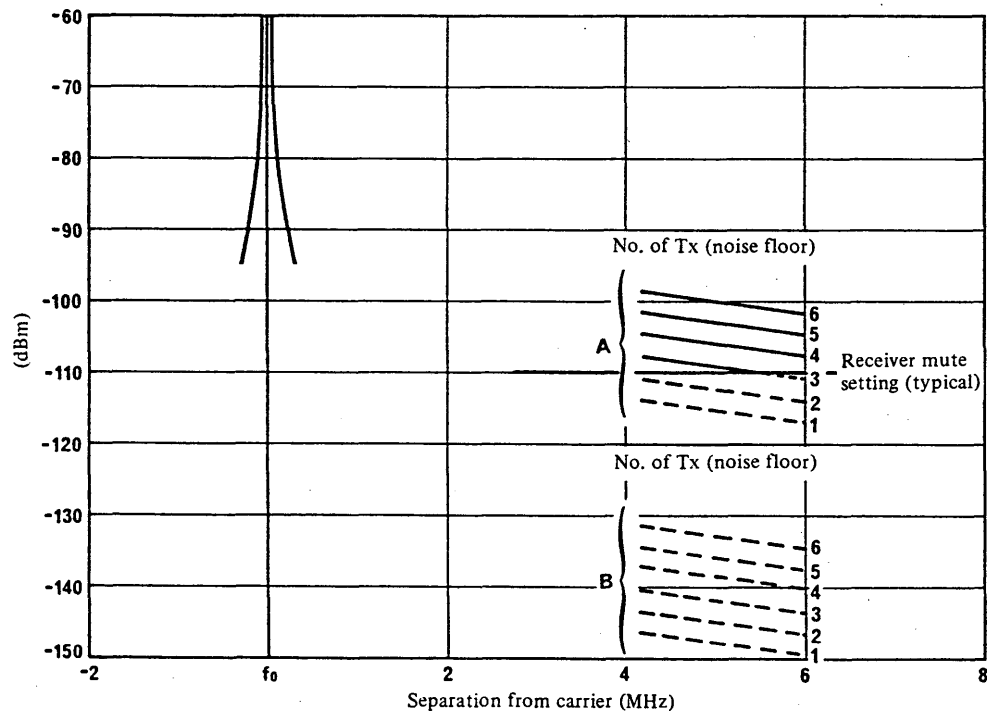


FIGURE 4 – Practical operational noise curves

Curves A: site without transmitter filters
B: site with transmitter filters
Minimum isolation at f_{Rx} = 30 dB
All transmitters have solid-state output stages
Carrier power per transmitter typically +43 dBm

Figure 5 shows a typical transmitter filtering system and Fig. 6 shows the detailed response of the spectrum dividing filter shown in Fig. 5.

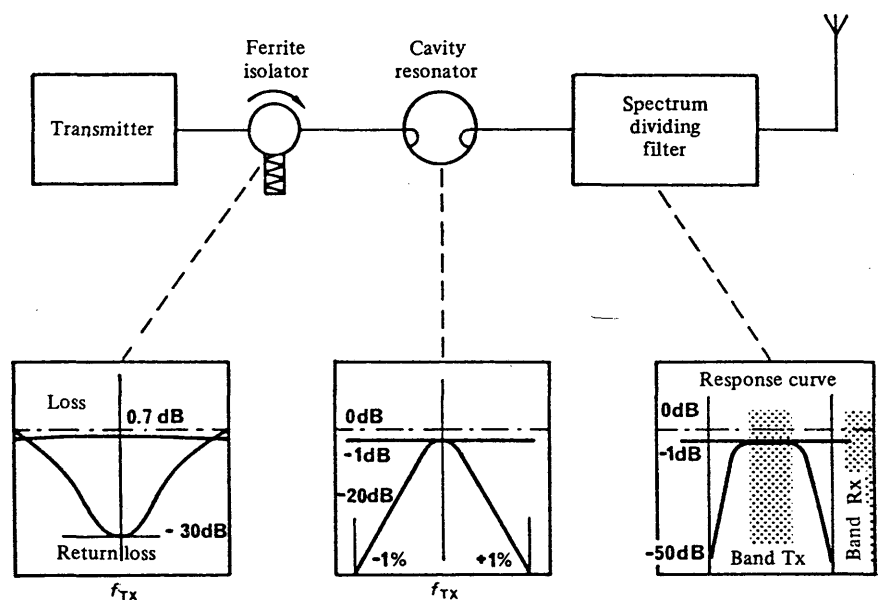


FIGURE 5 – Typical Tx filter system for talk through operation
VHF 150-160 MHz

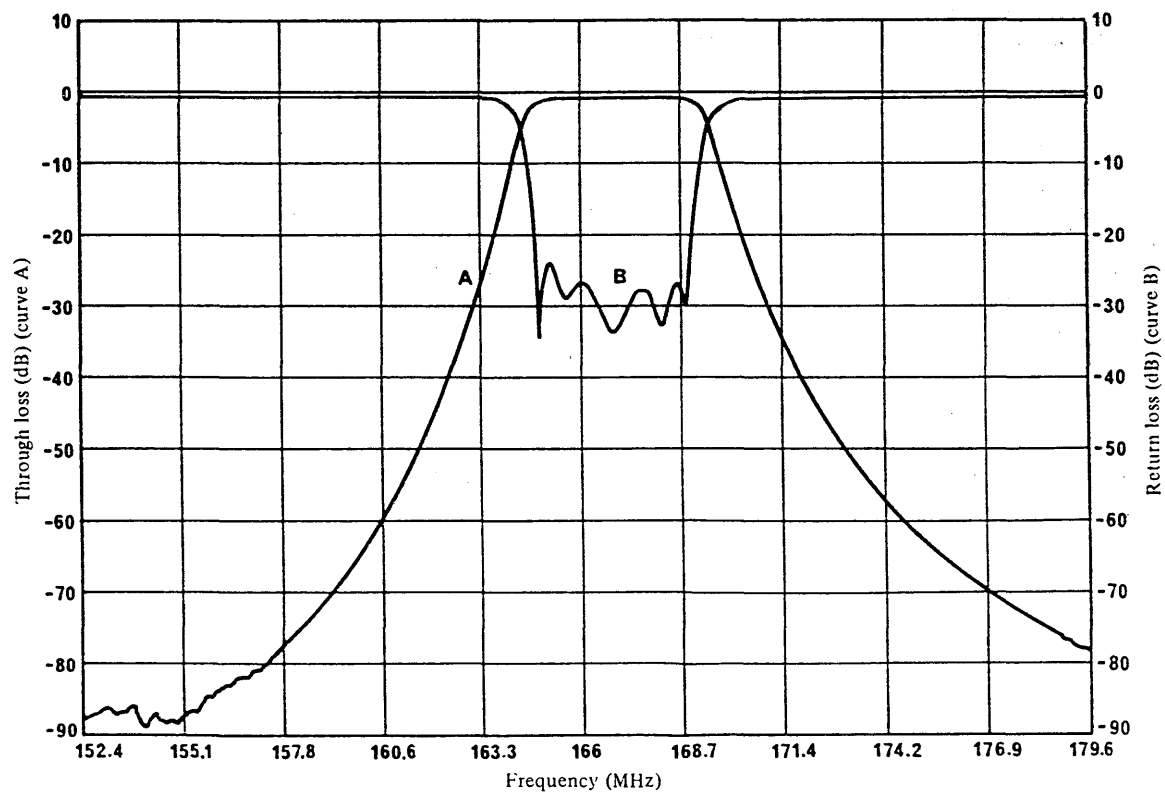


FIGURE 6 – Spectrum dividing filter response curve

A typical calculation of the noise levels to be expected on a site with 15 transmitters using the filtering arrangements outlined in Fig. 5 is given below:

*Calculation of typical transmitter noise levels
(Tx/Rx spacings 4.8 MHz)
VHF high band 150-170 MHz*

Per Tx			
Noise at FRx:	(Typical value)		- 137 dB (relative to carrier)
Now insert additional isolation:			
Add ferrite (- 1 dB)			- 138 dB
Add cavity (- 30 dB)			- 168 dB
Add SD filter (- 35 dB)			
(6 section)			<u>- 203 dB</u>
Now take carrier ref. +42 dBm:	Noise	=	- 161 dBm
Add say 15 × Tx at (+3 dB/Tx)			<u>+45 dB</u>
		=	<u>- 116 dBm</u>
Add say further 5 × Tx at (3 dB/Tx)			<u>+15 dB</u>
Final noise level at fRx		=	<u>- 101 dBm</u>
			(i.e. 2 μV p.d.)*

6. External electrical noise

Apart from ignition noise, there are the well-known sources of radio interference, which continue to proliferate, particularly from industrial users, i.e. RF heating, microwave ovens, X-ray and medical equipments. These normally provide a broad spectrum of noise which tends to vary in frequency.

Screening or suppression of the interfering equipment normally reduces the problem to an acceptable level.

There is however, a new family of sources, namely computers and computer peripherals, which are currently causing problems with broadband noise over the VHF spectrum.

7. Summary

There are instances where the present engineering practices in multiple transmitter sites have allowed the generation of excessive unwanted signals. With the increased use of land mobile radio it is desirable to perfect techniques to reduce interference effects in the future. There is a need for better site engineering in order to establish "quiet" base station sites for trunking networks and cellular radio.

The following should be considered:

- spurious emissions from transmitters;
- filtering of transmitter outputs to reduce spurious emissions and noise at frequencies near the carrier;
- use of directive isolators in transmitter output stages;
- additional filtering to provide protection at adjacent frequency bands;
- non-linear effects at all points in the system.

* μV p.d. is μV potential difference, measured with circuit closed.

REPORT 499-5

RADIO-PAGING SYSTEMS

(Question 12/8)

(1970-1974-1978-1982-1986-1990)

Introduction

This report describes the technical characteristics of radio-paging systems. It includes two parts, A and B and one annex.

Part A focuses more particularly on the transmission aspects, as seen in existing systems.

Part B focuses more particularly on the services and network aspects as presented in more recent studies carried out by several European countries. It considers in more detail the general approach to international roaming and its implications. Alternative solutions to those of Recommendation 539-2 are also presented.

Annex I gives an overview of a particular implementation of the new concepts introduced in Part B: the system studied for pan-European usage.

PART A

TRANSMISSION CHARACTERISTICS OF RADIO-PAGING SYSTEMS

1. Introduction

1.1 The radio-paging systems referred to in this part are described as one-way selective signalling systems without speech facilities and designed as extensions of the telephone network.

1.2 Some principle modes of operation are:

1.2.1 Dialed call to a common terminal, where a telephone number dialled into the telephone network is routed to a common terminal for processing into a radio-paging call;

1.2.2 Dialed call, with audio-frequency secondary end-to-end signalling, to a common terminal, where the common terminal accepts directly dialled digits and then accepts second stage audio-frequency digits to complete the information for processing into a radio-paging call;

1.2.3 Operator-handled calls.

1.2.4 Other modes are presented in Part B of this report;

1.3 False calls and messages should be eliminated so far as practicable, e.g. not more than one incident per user per year.

In the case of multiple-area paging systems, the user should be able to choose the area or areas within which he desires to be paged (see Part B).

The radio signal strength should be as uniform as practicable within the service area but restricted outside it. Special system considerations may be necessary to operate an aircraft paging receiver.

Radio paging could also be used to enhance other mobile services, e.g. paging aircraft in the ground-to-air direction.

2. Control terminal design concepts

For a high-capacity system, the use of a stored programme device seems to be desirable to perform the storing and forwarding functions of the paging calls.

It would be desirable for the control terminal to make a validity check of all calls entering the system.

The control terminal should return the necessary supervisory signals to the telephone switching network.

The control terminal should generate directly or indirectly the transmitter modulating signals.

The number of control terminals needs to be limited to ease the telephone access and switching problems. For access to the control terminal, the dialling codes used over the public telephone network should conform to agreed national and international standards.

As the control terminal will be an integral part of the switched telephone network it should conform to the normal requirements for equipment in that network. The capacity of each control terminal should therefore be either 1000 or 10 000 or 100 000 paging addresses, i.e. discrete subscriber dialling codes. The last-mentioned capacity of 100 000 is considered to be optimum between all the requirements for the majority of applications and leads to an economic design. To obtain greater user capacity in any system, a number of such terminals could be used.

3. Radio-frequency considerations

3.1 The following factors need to be taken into account in the choice of a suitable radio frequency channel(s):

- economics of the system for a given area;
- availability of frequencies;
- propagation considerations and operational requirements;
- environmental noise levels;
- practical limits of receiver sensitivity;
- permitted limits of emitted power levels and antenna heights according to local regulations;
- levels of paging traffic.

3.2 Possible frequency bands

All three ITU Regions have some or all of the following frequency bands allocated to mobile services:

26.1	to	50 MHz
68	to	88 MHz
146	to	174 MHz
450	to	470 MHz
806	to	960 MHz

In the future, it is possible that higher frequencies may be allocated to the mobile service in all three Regions and consequently be available for radio paging, but the utility of such frequencies is not yet proved.

In addition, documents submitted by Sweden illustrate the possible use of the VHF-FM sound broadcasting transmitter network in the frequency band 87.5 to 104 MHz for a wide-area paging system over the coverage area of the broadcast transmission [CCIR, 1978-82].

3.3 *The cost of coverage*

The costs and ease of providing base station antenna gain at 150 MHz and 450 MHz, to overcome increased propagation losses at these frequencies, compared with the lower bands such as 26.1 to 50 MHz (in North America) and 68 to 88 MHz (in Europe), are of the same order as those for the basic unity gain antenna systems for the lower frequencies.

3.4 *Effect of man-made noise*

For receivers with identical gain, which is the current situation with paging receivers designed to operate in the various frequency bands, the receiver noise factor increases with frequency.

The level of man-made noise, which is particularly high in inner city areas and on busy highways, where paging systems find most of their subscribers, is inversely proportional to frequency.

In the category of man-made noise we can also include on-frequency interference. 150 and 450 MHz (the latter being the highest frequency for which any practical paging system has been designed) provide relative freedom from long-distance transmission and thus from interference, which is a major disadvantage of the bands around 50 MHz and below.

3.5 *Radio propagation into buildings*

Measurements results submitted by a number of administrations have indicated that frequencies in the range of 80 to 460 MHz are suitable for personal radio paging in urban areas with high building densities. It is possible that frequencies in the bands allocated around 900 MHz may also be suitable but that higher frequencies are less suitable.

From measurements made in Japan, the following median values of the propagation loss suffered by signals in penetrating buildings (building penetration loss) have been derived. These are shown in Table I below:

TABLE I

Frequency	150 MHz	250 MHz	400 MHz	800 MHz
Building penetration loss (see note)	22 dB	18 dB	18 dB	17 dB (1)

(1) Somewhat less accurate than the other results.

Note. — The loss is given as the ratio between the median value of the field strengths measured over the lower floors of buildings and the median value of the field strengths measured on the street outside.

Similar measurements made in other countries confirm the general trend but the values of building penetration loss vary about those shown. For instance, measurements made in the United Kingdom indicate that building penetration loss at 160 MHz is about 14 dB and about 12 dB at 460 MHz.

Frequencies of about 80 MHz suffer losses similar to those at 150 MHz, but still lower frequencies e.g., 35 MHz, 26 MHz have been shown [Mino *et al.*, 1965; Rice, 1959] to be less suitable for use in urban areas but have some slight advantage over higher frequencies in suburban fringe areas.

For radio-paging systems which are intended to cover large areas with little urban development, the frequency bands around 80 MHz and 160 MHz seem to be most suitable.

3.6 *Techniques applicable to multiple transmitter zones*

To cover a service area effectively, it will usually be necessary to use a number of transmitters. When the coverage from a single transmitter is small, a single radio-frequency channel should be used so as to avoid the need for multi-channel receivers. In these circumstances, the separate transmitters may operate sequentially or simultaneously. In the latter case, the technique of off-setting carrier frequencies, by an amount appropriate to the coding system employed, is often used. It is also necessary to compensate for the differences in delay to the modulating signals arising from the characteristics of the individual land-lines to the transmitters. One way to do this is to carry out the synchronization of the code bits via the radio paging channel. Information is required about the bit rates which this synchronization method would permit.

It is preferable that the frequency off-set of the transmitter carrier frequencies in a binary digital radio-paging system be at least twice the signal fundamental frequency.

It is also preferable that delay differences between the modulations of the transmitters in a binary digital paging system should be less than a quarter of the duration of a bit if direct FSK, non-return to zero modulation is used. For sub-carrier systems the corresponding limit should be less than 1/8 of a cycle of sub-carrier frequency.

Studies are required to determine optimum methods for the transmission of signals over land lines and for simultaneous operation of a number of transmitters.

3.7 *Receiver design*

Built-in antennas can be designed for 150 MHz with reasonable efficiency. A typical radio-paging receiver antenna using a small ferrite rod exhibits a loss factor of about 16 dB relative to a half-wave dipole.

The feasibility of large scale integration of circuits (LSI) seems now to be realized. Component costs for a receiver using these techniques are low.

The majority of wide-area systems which have been established have employed some form of angle modulation.

Repeated transmission of calls can be used to improve the paging success rate of tone alert pagers. If p is the probability of receiving a single call, then $1 - (1 - p)^n$ is the probability of receiving a call transmitted n times, provided the calls are uncorrelated. Correlation under Rayleigh fading conditions can be largely removed by spacing the call more than 1 s apart. Longer delays between subsequent transmission (≈ 20 s) are required to improve the success rate under shadowing conditions.

Receivers with numeric or alpha-numeric message display can only take advantage of call repetitions if the supplementary messages are used to detect and correct errors.

4. *Signalling format*

The signalling format should be standardized. The choice of the appropriate coding techniques should take into account the required capacity of code combinations, speed of transmission, call success rate, and lowest practicable false calling rate. The code should be designed to allow for transmission of various types of messages. Recommendation 584 provides details of a recommended code and format.

Cyclic block codes such as the Bose-Chaudhuri-Hoquenghem (BCH) codes, permit the reliability of signalling to be improved and the probability of false calling to be much reduced, because of their distance and their inherent error detection and correction capability.

It is desirable that the standardized code can easily share a channel with other codes.

Message repetition is one possible way of increasing the successful call probability.

For the measurement of the signalling reliability of equipment, it is understood that the IEC is working on this subject. Results from field tests are also desirable.

5. System capacity

The number of users to be catered for still has to be determined.

In a document submitted by France, it was estimated that the available overall capacity at a national level should be at least 20 per 1000 inhabitants.

The capacity of any system is affected by at least the following:

- the number and the characteristics of the radio channels used;
- the number of times each channel is re-used within the system;
- the actual paging location requirements of the individual users;
- the peak information (address and message) requirement in any location(s);
- tolerable paging delay;
- data transmission rate;
- code efficiency;
- method of using the total code capacity throughout the system (this may also affect the system's capabilities for "roaming");
- any inefficiency introduced by battery saving provisions.

In addition to the above, there may also be telephone system input restrictions.

6. Compatibility between international and national radio-paging systems

It was recognized that a high degree of compatibility would be necessary between the national and international radio-paging systems. However, this does not preclude the establishment of radio-paging systems in factories, buildings, etc., using different standards.

On an international basis, between systems that are technically compatible, the international user should be able to move to another country, and the service provided in his home service area be extended to him in remote service areas of another nation.

One method of providing service between systems which are technically incompatible, would be to exchange the subscriber's paging receiver, and to use an agreed method to transfer the access data between national telephone systems.

7. Conclusion

In respect of codes and formats needs have been largely satisfied by Recommendation 584. The various existing domestic and international systems may soon be followed by new systems. Some administrations have an urgent need for a standard for their future systems and have made new studies in order to permit easy implementation of cross-border systems, sharing of users between various system providers, and to give good guidance to providers of paging services. The studies necessary to define the requirements for international radio-paging systems are not complete and should be continued. Results obtained to date are presented in Part B, and Annex I provides an example of a particular system design.

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PART B

TECHNICAL AND OPERATIONAL CHARACTERISTICS OF PAGING
SYSTEMS SUITABLE FOR INTERNATIONAL ROAMING1. Introduction

Part B describes the technical characteristics of radio-paging systems on which international agreement is desirable, including consideration of harmonization if more than one solution (for example, to cater for a range of transmission rates) is needed. The most important of these are the capability to carry text and data, and the possibility to use a single receiver in all locations allowing for roaming within the total coverage area.

Annex I provides an alternative configuration to those described in Recommendation 539 and Report 900 which do not address system concepts. Thus, this Annex does not comply fully with Recommendation 539 which lists generic features suitable for international paging, e.g. multiple transmitter zones, rate and type of data modulation, selectivity, signalling code and format.

2. Architecture

Any wide area or local paging system must have at least two well defined interfaces:

- the radio interface between the base station transmitter and the paging receivers;
- the interface between the paging system controller and the access networks.

In addition other internal interfaces may also be defined.

3. Services and facilities

Since in many cases wide area paging systems will be introduced when other national systems are already in operation, they should be able to support efficiently the main services characteristics of the existing systems such as tone only, numerical and alphanumerical pagings. In addition to these basic services a number of supplementary services may be provided, however, the networks should be capable of supporting a minimum set of services to ensure full compatibility with the basic version receiver.

The main services and facilities can be summarized as follows:

- international roaming;
- choice of call destination;
- temporary barring of incoming traffic;
- constitution of closed users' groups (the possibility of mobile and fixed subscribers to establish a group with only internal communication possibility; the mobile or fixed subscribers can be members of more than one closed users' group);

- protection against messages lost;
- establishment of priority levels;
- diversion of traffic to other receivers;
- security facilities;
- charging facilities;
- out of range indication;
- deferred delivery.

Regarding the paging categories, they can be summarized as follows:

- tone only;
- numeric;
- alphanumeric.

The type of message will be:

- individual calls;
- group calls (using common or multiple radio identity code).

4. Radio coverage

As far as the radio coverage is concerned, the system should have the possibility to guarantee the continuous coverage of a given territory (e.g. a whole country).

A paging area is defined as the area served by a single transmitter or set of transmitters which send the same information. It is possible to foresee that the system area may be divided into several paging areas, overlapping or not. Messages may be transmitted in a limited number of paging areas and a dedicated procedure will enable roaming subscribers to temporarily reroute the calls to one or more alternative areas where appropriate.

5. Radio network structure

The radio network structure needs to be defined in such a way as to permit maximum flexibility for national implementation and efficient spectrum utilization (specifically in border areas), whilst retaining essential common characteristics.

6. Network aspects

The main principle of a wide area system is to keep the access to the system as easy as possible. Access to the system could be given through telephone or data networks, either private or public.

ANNEX I

TECHNICAL AND OPERATIONAL CHARACTERISTICS OF ERMES

1. Introduction

A number of European countries are developing a new pan-European radio paging system named ERMES (European Radio MESSage System) capable of offering service both to national subscribers and to those roaming outside their home country, in the whole CEPT area.

The complete draft standard, according to the present working programme will be ready in June 1990 and the introduction of ERMES is foreseen in early 1993.

This annex contains a general description of ERMES.

2. General description

ERMES architecture is shown in Figure 1. In this figure the following main interfaces are shown:

- I1: Radio interface, based on the following characteristics:
 - frequency band: 169.4 - 169.8 MHz;
 - 25 kHz channel bandwidth;
 - modulation method: 4PAM/FM;
 - symbol rate: 3.125 kbaud (6.25 kbit/s bit rate);
 - transmission protocol as described in Figure 2;
 - frequency agile receiver (16 channels).
- I2: Interface between the Paging Area Controller (PAC) and the base station;
- I3: Interface between the PAC and Paging Network Controller (PNC);
- I4: Interface between different PNCs, that ensures the roaming between countries and/or operators;
- I5: Interface between the PNC and access networks;
- I6: User access interface.

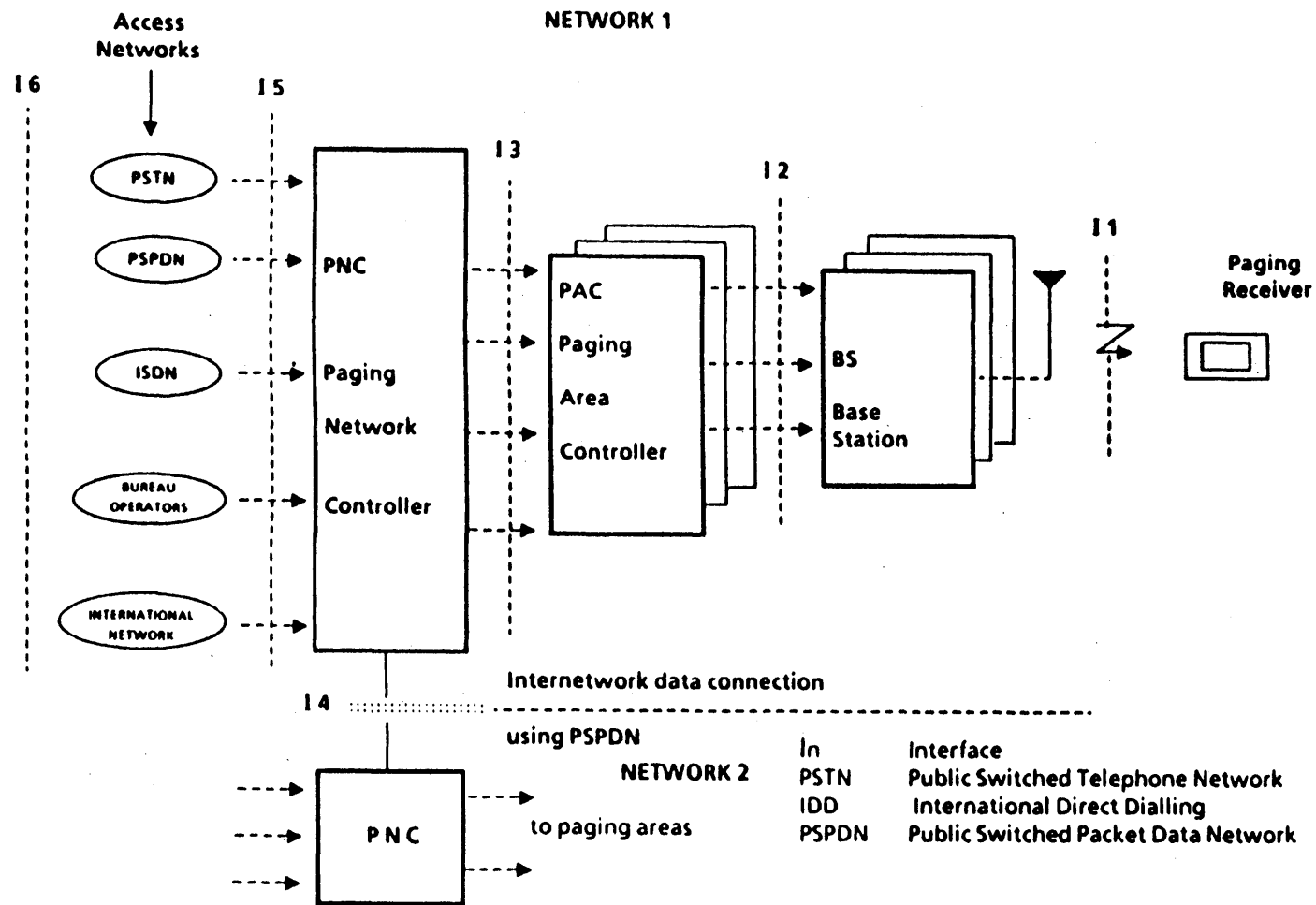
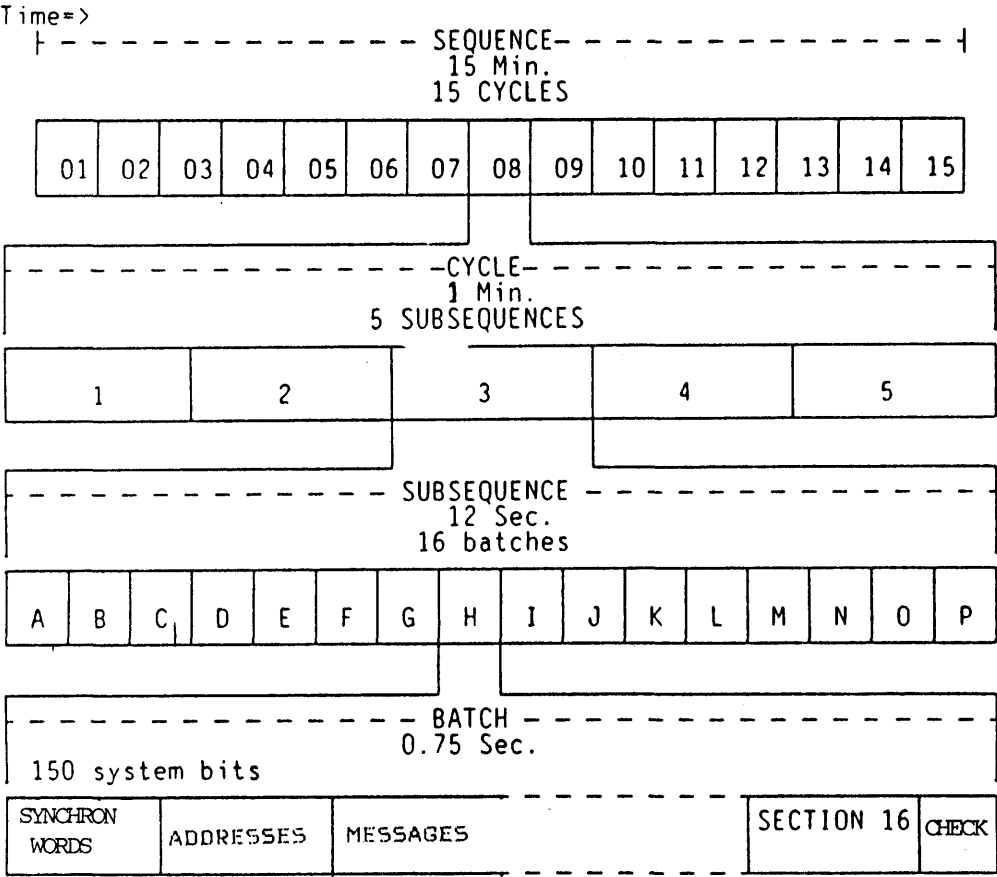


Fig. 1 ERMES architecture



Structure in the radio protocol on one frequency

BATCH NO.	0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	0	0	0	0	0	0	0	1	1	1	1			
	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	1	2	3	4	5	6	7	8	9	0	1	2	3
SUBSEQUENCE 12 Sec.																													
CH.	■ = On time for a scanning type A pager.																												
01	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M
02	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L
03	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K
04	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J
05	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I
06	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H
07	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G
08	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F
09	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E
10	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D
11	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C
12	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B
13	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■
14	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
15	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N	O
16	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	■	B	C	D	E	F	G	H	I	J	K	L	M	N

Synchronization between channels

Fig. 2 ERMES transmission protocol

3. Services and facilities

ERMES provides the basic services as follows:

- tone only (8 different alerts possible per radio identity code) (RIC);
- numeric (20 characters);
- alphanumeric (400 characters);
- transparent data (in the form of an arbitrary bit data stream) for several applications such as graphics, telecommands, coded voice, etc.

In addition to these basic services a great number of supplementary services will be provided, some of which might be offered by individual countries, such as:

- international roaming;
- choice of call destination (under control of the mobile subscriber or of the calling party);
- temporary barring of incoming traffic;
- constitution of closed user groups (the possibility of mobile and fixed subscribers to establish a group with only internal communication possibility; the mobile or fixed subscribers can be members of more than one closed user group);
- protection against messages loss (repetition, numbering, storing and retrieval, etc.);
- establishment of priority levels (urgent calls, normal calls, non-critical time-related calls);
- diversion of traffic to other receivers;
- security facilities (encryption, legitimization code, verification for access);
- charging facilities (standard rate, reverse charge, charging information);
- out-of-range indication;
- deferred delivery.

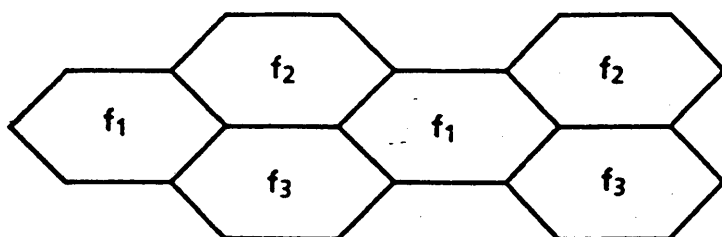
Regarding the paging categories they can be summarized as follows:

The types of messages will be:

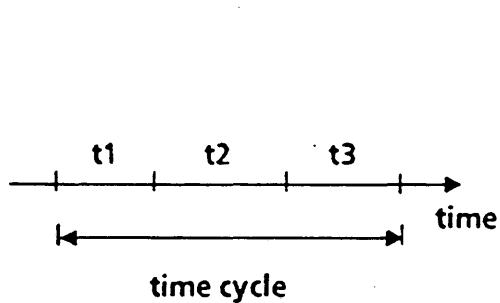
- individual calls;
- group calls (using common or multiple RIC);
- radiodistribution, understood as the possibility of a calling party to broadcast information for a selective group of receivers.

4. Radio network structure

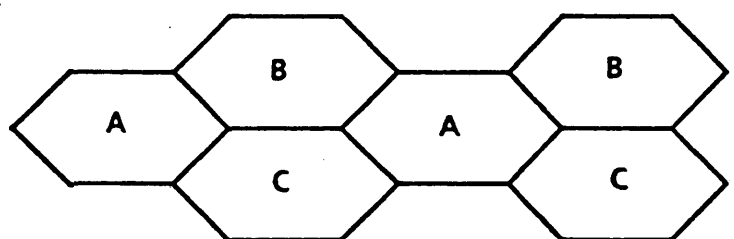
The ERMES radio network structure is based on time divided and/or frequency divided network operation and it is up to network operators in each country to decide the mode of operation. Figure 3 shows the three modes of operation viable in the system.



FREQUENCY DIVIDED NETWORK



TIME DIVIDED NETWORK



FREQUENCY AND TIME DIVIDED NETWORK

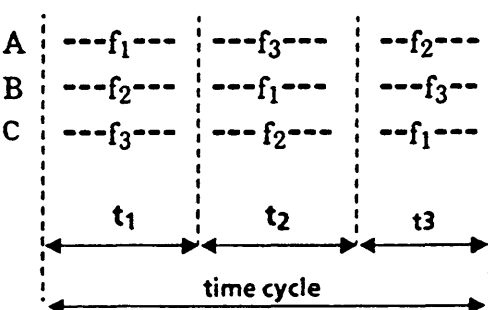


Fig. 3 Possible configurations of ERMES radio structure

5. Network aspects

The main principle of ERMES is to keep access to the system as easy as possible. Since access to the system should be through telephone or data networks, the following methods are possible:

- Direct: which consist of two main phases: first, input the address code (ADC); second, input the message;
- End to End: access with three main phases: input the service number, input the ADC, input the message.

Direct access applies only to the access via telephone network.

ADC is used for identification of the mobile subscriber and it consists of three parts:

- Country number: having the same meaning of that used in the PSTN;
- Network identification: which identifies the network operator within the country;
- Subscriber identification.

Regarding the different phases of call processing, the PNC has three functional entities:

- PNC-I for the input of the ADC messages;
- PNC-H checking the validity of input (ADC);
- PNC-T sending the message to the appropriate paging area controller.

In any case the responsibility of the overall call processing is ensured by PNC-H.

REFERENCES

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P.O. Box 152, Sophia Antipolis 06561 Valbonne France

REPORT 900-2

RADIO-PAGING SYSTEMS

Standardization of code and format

(Question 12/8, Study Programme 12A/8)

(1982-1986-1990)

1. Introduction

An urgent need was expressed by some administrations for a standard code and format which could be used for systems due to enter service in the immediate future. Recommendation 584 is intended to provide advice and help to such administrations but this Report supplements the information.

2. Information on existing codes

2.1 The Swedish radio-paging system is an addition to the radio data system (RDS), recommended for the transmission of information to facilitate station or programme selection in FM receivers (see Recommendation 643). The system has been used for radio paging in Sweden since 1978. It uses a 57 kHz subcarrier on VHF/FM broadcasting transmitters. The modulation method and the code and format are specially designed to be compatible with the broadcast monophonic and stereophonic programmes. A paging address comprises six decimal digits. After that may follow an additional message, either numeric with 10 or 18 decimal digits, or alphanumeric with unlimited length [CCIR, 1978-82a, 1982-86a]. Details can be also found in Annex II.

2.2 Documents submitted by the United States of America describe a binary digital code and format with a single address capacity of up to 400 000 with non-coded battery saving and 4 000 000 with coded preamble. The code uses a Golay (23 : 12) cyclic code with two codewords representing an address. Messages are encoded using a BCH (15 : 7) code. The code and format provide queueing and numeric/alphanumeric message flexibility and ability to operate in a mixed mode of transmission with other formats. The details of this code and format are contained in Annex I [CCIR, 1978-82b; 1982-86b].

2.3 Japan uses a BCH (31 : 16) codeword with a Hamming distance of 7. The format gives approximately 65 000 addresses, 15 groups for battery economy and a total cycle length of 4185 bits. Each group includes 8 address codewords headed by a 31-bit synchronizing and group indicating signal [CCIR, 1978-82c].

2.4 The United Kingdom employs a BCH 32 : 21 plus even parity codeword with a Hamming distance of 6. The code format caters for over 8 million addresses and can be expanded. It also caters for any type of data message (hexadecimal and CCITT Alphabet No. 5 are already standardized). It is designed to share a channel with other codes and to permit mixed simultaneous and sequential multi-transmitter operation at the normal (512 bit/s) data transmission rate [CCIR, 1978-82d]. This code, sometimes referred to as POCSAG, has been adopted as CCIR Radio-Paging Code No. 1 (RPC1) (Recommendation 584).

2.5 Radio-Paging Code No.1 is also used for National paging services in Australia, Finland, New Zealand and the People's Republic of China, as well as in United States where numerous other codes and formats are currently in use.

2.6 France has implemented a public paging system in 1987 using Radio-Paging Code No. 1 and allowing, in particular, the transmission of alphanumeric messages. These messages, of up to 80 characters, are sent using "Minitel" home terminals (several millions of such terminals being available at PSTN subscriber premises). This service, called "Alphapage", also caters for the transmission of alerting tones, and numeric messages of up to 15 digits can also be sent out using DTMF telephone sets (CCITT Recommendation Q.23).

3. Factors affecting standardization

Report 499 in § 4 identifies the need to standardize the signalling format and points out that the choice of the appropriate coding technique should take into account the required capacity of code combinations, the speed of transmission, call success rate and lowest practicable false calling rate. In addition the code should be designed to allow for transmission of various types of messages (for example, single address, multiple address, address plus numeric message, address plus alphanumeric message) and should easily share a channel with other codes.

Other factors which will need to be considered are interrelated. For example the number of subscribers to be served, the number of addresses which they require, the traffic generated, including that due to the transmission of messages and the signalling rate that is technically feasible. In some cases this last may be determined by the linking land line network characteristics, particularly when simultaneous radio transmission is employed, as well as by the radio system characteristics.

The choice of a code and format cannot be made without considering the system in which they are to be used. Certain codes and formats are capable of operating in a variety of systems but this does not apply to all.

4. Considerations of different codes and formats

4.1 In the choice of a standard code and format the following system requirements are considered important and should be taken into account:

- the number of subscribers to be served;
- the number of addresses assigned to each subscriber;
- the calling rate expected including that from any included message facility;
- the zoning arrangements;
- the data transmission rates possible over the linking network and radio channel(s), taking account of the propagation factors of the radio frequencies to be used;
- the type of service, e.g. vehicular or personal, urban or rural;
- the acceptable queueing delay.

4.2 From considerations of the above, codes may be compared by their characteristics in respect of:

- Code address capacity;
- Number of bits per address;
- Code efficiency, e.g. number of information bits/total number of bits, per codeword;
- Codeword Hamming distance;
- Error detecting capability;
- Error correcting capability;
- Message capability and length;
- Battery saving capability;
- The ability to share a channel with other codes;
- The capability of meeting the needs of participating administrations with systems varying in respect of size, transmission mode, e.g. simultaneous and/or sequential, etc.

4.3 The various characteristics of coding systems described in the documents summarized in § 2 appear in Table I below.

5. Address capacity

Report 499 includes an estimate made by France that the available overall capacity on a national scale for radio-paging receivers should be at least 20 per 1000 inhabitants.

Assuming that particular national systems and international paging use the same code and format on a common frequency then the address capacity required may be large. On the other hand, if different formats and/or frequencies are chosen then the address capacity requirements could be smaller.

Further studies on the capacity required are needed urgently.

6. Additional information concerning Radio-Paging Code No. 1 (Recommendation 584)

Recommendation 584 provides details of Radio-Paging Code No. 1 but the following additional information may be useful in connection with the requirements for receivers to operate in systems using this code. [CCIR, 1978-82e].

6.1 *Loss of synchronization*

If a receiver loses synchronization, or if it commences receiving after the preamble has been completed, it is desirable that it can achieve synchronization on receipt of a number of valid batches.

6.2 *Decimal representation of pager identities*

A decimal representation of the pager identity might be useful. If so, it is suggested that it should be the decimal equivalent of the 21 bit identity.

6.3 *Message reception, display and alerting*

6.3.1 *End of message*

It is desirable that the pager ceases decoding a message when either an idle or address codeword is received or when two successive information codewords are indecipherable, even if they immediately follow a message indicating pager address.

6.3.2 *Minimum message storage capacity*

Some form of storage will be necessary for pagers which do not provide a printed output. It is suggested that the minimum storage capacity in pagers designed for the "numeric-only" format should be 20 characters and for the "alpha-numeric" format, 40 characters.

TABLE I – Characteristics of different codes and formats

Characteristics	Sweden	United States	RPC1	Japan
<i>Codeword related characteristics:</i> Codeword type	Kasami 26 : 16 Truncated cyclic code	Golay 23 : 12 (address) BCH 15 : 7 (message)	BCH 32 : 21 plus 1 bit even parity	BCH 31 : 16
Hamming distance	3 (Note 1)	7 (address) 5 (message)	6	7
Codeword error detection capability per word (Note 5)	Burst: 10 errors Random: 2 errors	Burst: 11 errors } address Random: 6 errors } message	Burst: 11 errors Random: 5 errors	Burst: 15 errors Random: 6 errors
Codeword error correction capability per word (Note 5)	Burst: 5 errors Random: 1 error	Burst: 5 errors } address Random: 3 errors } message	Burst: 5 errors Random: 2 errors	Burst: 7 errors Random: 3 errors
Codeword efficiency (No. of information/ total codeword bits) (Note 9)	16/26 = 0.62	12/23 = 0.52 (address) 7/15 = 0.47 (message)	21/32 = 0.66	16/31 = 0.52
<i>System related characteristics:</i> Code address capacity	1 × 10 ⁴	409 600 non-coded preamble Over 4 000 000 coded preamble (Note 3)	Over 8 × 10 ⁴ (Note 4)	65 000
Number of bits per address	52	60.5	32 (+ 3 implied) + 2 overhead + preamble	34.9 (Note 7)
System bit rate (bit/s)	1187.5	300/600 (address) (Note 2) 600 (message)	512	200
System message capability and length	Decimal, 18 digits; alphanumeric, unlimited	Dual address, 4 functions, numeric and alphanumeric: length indefinite when asynchronous	4 addresses per identity: unlimited message length, hexadecimal or alphanumeric	Dual address
Battery saving feature and type	Call grouping. Other information transmitted	Yes	No transmission. Other codes in transmission (Note 6)	Call grouping into 15 groups
Ability to share a channel with other codes	Yes	Yes	Yes	No
Suitability of format to simultaneous and/or sequential transmission	Simultaneous: yes Sequential: not relevant to system as implemented in Sweden	Yes	Yes	Yes
Radio channel in use	FM Broadcasting 87.5-104 MHz	VHF and UHF mobile 150 MHz, 450 MHz, and 800 MHz	VHF mobile 150 MHz (Note 10)	VHF mobile 250 MHz
In service since	1978	1973 (address) 1982 (message)	(Notes 8 and 10)	1978

Notes relative to Table I

Note 1. — Out of the 2600 possible 3-bit error patterns, 7 will convert any valid codeword to another valid codeword. There are known to be also some 4- and 5-bit error patterns which have the same effect.

Note 2. — The transmission system must be able to transmit the 1/2 bit duration signals at double rate, i.e. 600 bit/s.

Note 3. — With known means of expansion.

Note 4. — Although as implemented the code and format provide for more than 8×10^6 addresses, compatible methods of expansion are known.

Note 5. — The higher the error detection the greater is the protection against false calls and false messages. Conversely greater error correction will tend to increase the call success rate at the expense of protection against false calls.

The actual values of error correction and detection depend not only upon the codeword structure but also upon each particular design of receiver and upon the code and format.

Note 6. —

Transmission condition	Battery saving method
No transmission	Sampling to establish the presence of preamble
Other codes	Discrimination by chosen bit rate
Code and format	Grouping calls

Note 7. — This takes account of the 31 overhead bits shared between 8 calls in a group.

Note 8. — In service in the United Kingdom since 1980; subsequently brought into service in Australia, Finland and New Zealand, and in experimental service in the People's Republic of China.

Note 9. — Not equal to overall efficiency, which is affected by format.

Note 10. — Radio-paging services in the United States in use on 150, 450 and 900 MHz bands; also public radio-paging services in use in the People's Republic of China on 160 MHz in 1985.

6.3.3 Alerting tones

6.3.3.1 The following repeated alerting tone patterns as shown in Table II are suggested:

TABLE II

Function Bit Combination	Alerting Tone
00	1 beep
01	2 beeps
10	3 beeps
11	4 beeps

6.3.3.2 When digital message pagers are used the following repeated alerting tone patterns as shown in Table III are suggested:

TABLE III

Function Bit Combination	Following Message Type	Alerting Tones
00	Numeric-only	1 beep
01	No message	2 beeps
10	No message	3 beeps
11	Alphanumeric	1 beep

Thus the recipient will expect only a single beep irrespective of the type of message.

7. Comparison of codes

7.1 Comparison by simulation

7.1.1 In Italy both the radio paging Code No. 1 (RPC1) and Golay code are used for the public radio-paging system. In order to control environmental parameters, e.g. propagation, that affect any measurement programme, it was decided to compare the RPC1 and Golay codes by laboratory simulation and to verify this simulation by some field measurements. A computer-aided comparison between the two codes was carried out [CCIR, 1982-86c]. Performances for both address and message reception (numeric and alphanumeric) were evaluated for two radio channel models (Gaussian and Rayleigh fading).

The results indicated that, for message success rate per paging attempt, the Golay code has a better performance with respect to 1 bit error correction per code word of RPC1 and almost the same performance with respect to the foreseen RPC1 2 bit error correction algorithm under these channel conditions. The results for 1 bit error correction are summarized in Table IV in which the performances are presented in terms of $S/N(\text{RPC1}) - S/N(\text{Golay})$ (dB) in order to obtain the same (95%) success level per paging attempt.

TABLE IV – Code performance comparison (dB) (see text)

Type of signal	Channel		
	Gaussian	Rayleigh at 5 km/h	Rayleigh at 40 km/h
Alert	0.6	1	3
10 character numeric message	1	4	8.5
80 character alphanumeric message	1	8	11

For the criteria of a false message conditioned by a successful call, results are shown in Figs. 1 and 2. They indicate that with the 2 bit error correction algorithm of RPC1 the probability of obtaining a false message is highest, that the probability for Golay is less and that for RPC1 with 1 bit error correction is least.

7.1.2 The United States believes that the additional field tests it has carried out confirm these laboratory simulations [CCIR, 1982-86d].

7.2 *Comments on the comparison*

7.2.1 In order to obtain the equivalent results by means of field measurements it would be necessary to mount a large measurement programme, especially to distinguish results that are close to one another.

7.2.2 A bad impression of the service might be given to a subscriber who receives a false message following a successful alert; and thus the curves in Figs. 1 and 2 are of particular interest. The translation of the data from these curves into the terms suggested in Report 499, taking into account also Fig. 3, needs study. Significant pre-knowledge of the likely message content often exists, and this may ameliorate this user impression.

7.2.3 In order to identify the consequences on system planning criteria, study is needed to determine how to weight the simulation results by factors such as time spent in marginal situations.

Field tests carried out in the United Kingdom on an RPC1 alphanumeric pager utilizing a 1 bit error correction algorithm gave the sensitivity decrease between an alert and an 80 character message for a success rate of 90% as approximately;

$$3.2 + 0.055 W \quad (\text{dB})$$

where W was the percentage of time spent *walking* at least 5 km/h inside a single room in each of two buildings and the streets beside one of them [CCIR, 1982-86e].

7.2.4 Message reception with new decoding methods (hard and soft decision) for both RPC1 and Golay has not yet been fully explored [CCIR, 1982-86f].

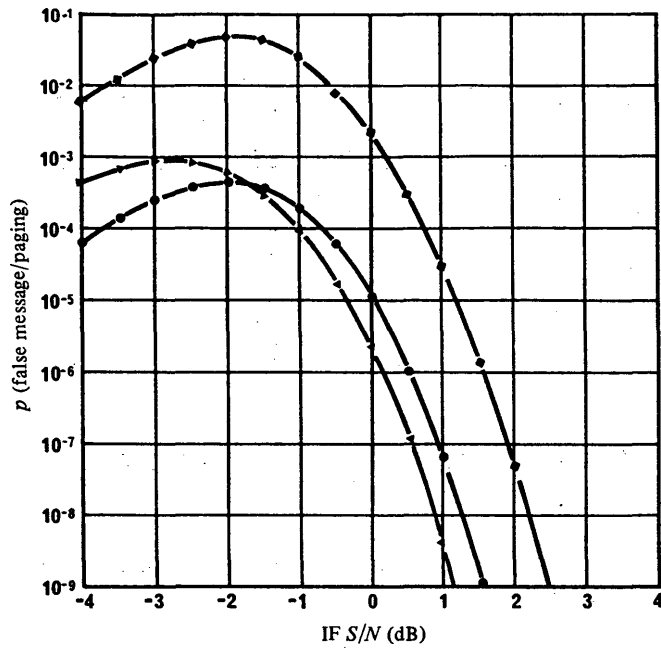


FIGURE 1 – False numeric message rate (10 figures), conditioned to the probability of associated address reception, for a pager moving in a Gaussian channel

—●— POCSAG (1 bit correction)
 —◆— POCSAG (2 bit correction)
 —▶— GSC

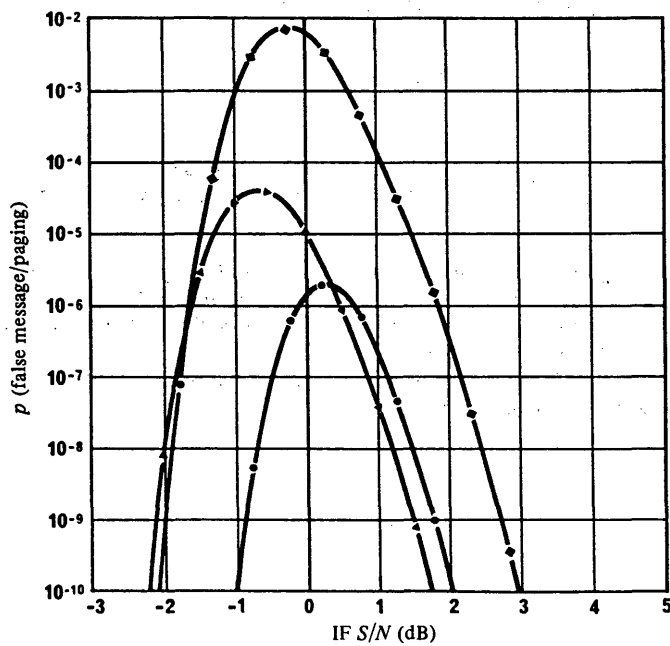


FIGURE 2 – False alphanumeric message rate (80 characters), conditioned to the probability of associated address reception, for a pager moving in a Gaussian channel

—●— POCSAG (1 bit correction)
 —◆— POCSAG (2 bit correction)
 —▶— GSC

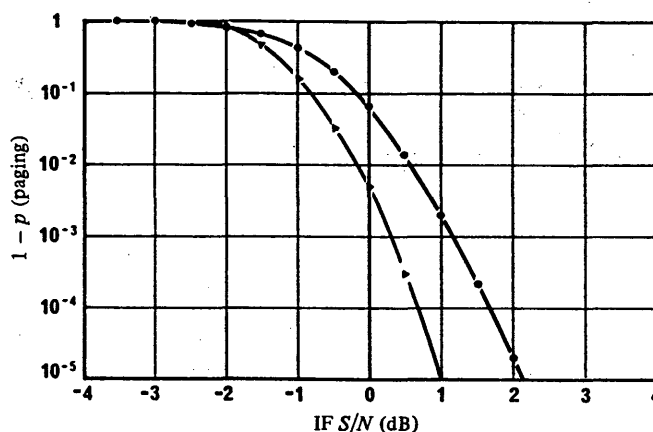


FIGURE 3 – Paging failure rate, for a pager moving in a Gaussian channel

—●— POCSAG
 -▲- GSC

8. Conclusions

In order to reach agreement on a universal standard signalling format it is highly important to carefully identify the common bases for comparing codes and formats, taking into account the system requirements given in § 4.

However, agreement on a standard code and format will only be truly effective in international paging when agreement is also reached on radio channel assignments and modulation techniques, and on the method(s) for transferring calls between the systems in the participating countries.

REFERENCES

CCIR Documents

[1978-82]: a. 8/63 (Sweden); b. 8/82 (USA); c. 8/210 (Japan); d. 8/112 (United Kingdom); e. 8/397 (United Kingdom).

[1982-86]: a. 8/417 (Sweden); b. 8/112 and 8/113 (USA); c. 8/446 (Italy); d. 8/375 (USA); e. 8/447 (United Kingdom); f. 8/295 (United Kingdom).

Note. — Statement by the United Kingdom concerning the POCSAG Radio-Paging Code: The members of the Post Office Code Standardisation Advisory Group (POCSAG) have stated that they would refrain from making any patent claims for the code itself (as distinct from any means of transmitting or receiving it). So far as they are concerned, the code is free of charge for use by any administration or other service provider, manufacturer or user.

For information, solely for its own purposes British Telecom has had a reasonably thorough patent search carried out without discovering any patent which the code might infringe.

ANNEX I

GOLAY SEQUENTIAL CODE

1. Introduction

Golay paging codes are those codes that use as their basis a binary code discovered by Marcel Golay. The basic code has 23 bits, 12 of which are information bits. This code is used in a binary paging format generally characterized by 300 baud NRZ data. The original Golay code, in service since 1973, has an uncoded preamble and supports up to 400 000 individual pager addresses; it is briefly described in § 2.5. Subsequently, the Golay paging format was expanded to include data and a coded preamble, and it is entitled the Golay sequential code (GSC). The GSC code supports up to 4 million addresses and is compatible with the older Golay paging code under certain address restrictions. Some forms of GSC support voice operation also but are not discussed in this Annex.

2. Signalling format

The GSC code format allows paging calls to be transmitted individually or in batches. Transmission begins with a preamble followed by a start code and a single address or a batch of addresses as illustrated in Fig. 4. The polarity of the preamble identifies the transmission mode (single call or batch). The preamble also serves to divide the population of pagers within the system into groups for improved battery life, as well as to uniquely identify GSC transmissions. The start code delimits the end of the preamble and supplies timing information for batch mode decoding, and the address identifies the individual pager.

In the implementation of the code, a positive frequency shift represents a binary "1", and a negative frequency shift represents a binary "0".

2.1 Batch operation

The batch transmission format, as shown in Fig. 4, begins with an inverted preamble followed by the start code and up to 16 pager addresses or data blocks. The arriving paging requests should be grouped as a function of preamble code and transmitted on a time or traffic basis as required for the system.

Unfilled batches should be filled out with comma (see § 2.3). A new preamble batch, however, may be initiated after 11 addresses of the preceding batch have been transmitted or 3.85 s has elapsed since the start of the preceding batch. In message paging, a message (not an address) may be started at the 17th address block or before and may be continued until completion.

2.2 Extended batch

As defined herein, a preamble and start code must be transmitted with each additional 16, or equivalent, addresses sent. This differs from some Golay implementations which allow batch extension through the use of start code repeat only.

2.3 Preamble, start code, and address structure

The preamble, start code, and address codes are constructed from words selected from the Golay (23 : 12) cyclic code and transmitted at 300 bit/s.

A 1, 0 bit reversal pattern or comma transmitted at 600 bit/s acts as a separator between preamble, start code, and address. One comma bit is half the length of an address bit and identical in length to a message or data bit.

2.3.1 Preamble structure

The preamble as shown in Fig. 4 consists of 28 bits of comma followed by 18 repeats of the selected preamble word (ten preambles divide the population of pagers into battery saver groups). The starting polarity of the comma must be the same as the first bit of preamble. As previously noted, an inversion or complement of the preamble and its associated comma bits identifies the batch mode of transmission. The normal polarity of the preamble words is tabulated in Table V (least significant bit (LSB) to the right), and the preamble words are transmitted LSB first. The decimal representation of the information binary bit pattern is tabulated in column 2 of Table V.

TABLE V – Preamble words

Preamble number	Decimal	Binary bit pattern	
		Parity	Information
0	2030	1000000011	01111101110
1	1628	0000111110	01100101100
2	3198	1100001001	11000111110
3	647	0111111000	00101000011
4	191	0000111010	00001011111
5	3315	0000011001	11001110011
6	1949	0001111000	01111001101
7	2540	0100000111	10011101100
8	1560	0111111001	01100001000
9	2335	11100010001	10010001111

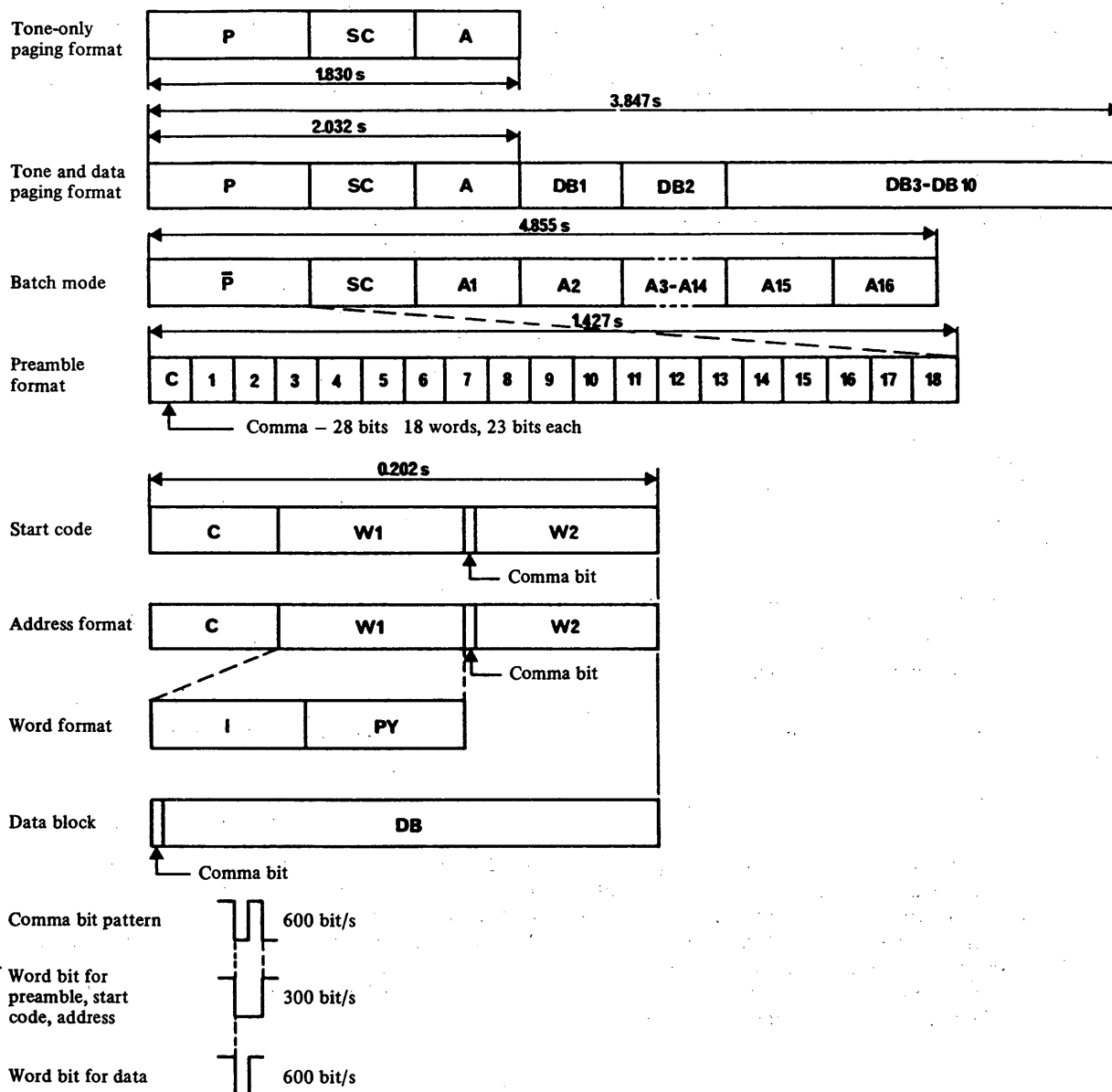


FIGURE 4 – Golay sequential code format

P : preamble
 SC : start code
 A : address
 DB : data block (8 (15:7) BCH code words)
 C : comma – 28 bits
 W1 : start code or address word 1-23 bits
 W2 : start code or address word 2-23 bits
 I : information – 12 bits
 PY : parity – 11 bits

2.3.2 Start code

The start code and address codes use a two-word format consisting of 28 bits of comma followed by two (23 : 12) code words separated by one comma bit. The starting polarity of the 28 bit comma must be the same as the first bit of the first word and the polarity of the single comma bit between words must be opposite to that of the second word. The start code is fixed for the system, and its Word 2 is the complement of its Word 1. The start code Word 1 is:

Start code	Decimal	Binary bit pattern	
		Parity	Information
Word 1	713	0100000011	001011001001

The start code format permits the pager's decoding circuitry to detect either the start code Word 1 or Word 2 and identify which pattern was detected, thus maximizing start code detection sensitivity.

2.3.3 Address code

The address format is identical to the start code format with regard to the number of bits and the rules for comma. The address Word 2 code set consists of all (23 : 12) code words except for all 0's, all 1's, and all cyclic rotations of the start code. Thus, there are approximately 4000 Word 2's. There are 100 first words (50 words and their complements) used in the GSC system. Fifty words are determined by table look-up (Table VI), and the remaining 50 are achieved by complementing the tabulated words. The Word 1 and Word 2 code words for each pager address may be derived from the pager code (see § 3).

TABLE VI – Address Words 1's

Word No.	Decimal	Word No.	Decimal	Word No.	Decimal	Word No.	Decimal	Word No.	Decimal
00	721	10	2692	20	2285	30	375	40	1575
01	2731	11	696	21	2608	31	1232	41	3463
02	2952	12	1667	22	899	32	2824	42	3152
03	1387	13	3800	23	3684	33	1840	43	2572
04	1578	14	3552	24	3129	34	408	44	1252
05	1708	15	3424	25	2124	35	3127	45	2592
06	2650	16	1384	26	1287	36	3387	46	1552
07	1747	17	3595	27	2616	37	882	47	835
08	2580	18	876	28	1647	38	3468	48	1440
09	1376	19	3124	29	3216	39	3267	49	160

2.3.4 Multi-function address capability

The two-word format of the GSC address code contains a built-in multi-function capability. There are four possible Word 1/Word 2 combinations that can be easily detected (see § 3.3). These combinations are used to designate the type of page (tone-only or data). Each pager in the GSC system is assigned at least one four-function address.

2.4 Code word generation 23 : 12 Golay

To generate the binary bit patterns for the (23 : 12) Golay code, the decimal representation of the code word is converted to binary. This binary representation is rewritten LSB to the left. These 12 information bits (there may be leading or trailing 0's) now correspond to the coefficients of a polynomial having terms from x^{22} down to x^{11} . This polynomial is divided, modulo 2, by the generator polynomial $x^{11} + x^9 + x^7 + x^6 + x^5 + x + 1$. The parity bits correspond to the coefficients of the terms from x^{10} to x^0 in the remainder polynomial found at the completion of this division. The complete block, consisting of the information bits followed by the parity bits, corresponds to the coefficients of a polynomial which is integrally divisible in modulo 2 fashion by the generator polynomial.

2.5 Non-battery saver operation

The older Golay format was based on decoding of the address words independently of any preamble or start code information. This non-battery saver operation is the simplest transmission mode and achieves the highest throughput; however, this form of operation reduces pager battery life and lacks address expansion.

In non-battery saver operation, the coded preamble and start code are not used; only the pager addresses need to be transmitted for tone only paging. However, it is suggested that a preamble consisting of a 1, 1, 0, 0 pattern (75 Hz square wave) be transmitted for at least 1.25 s after transmitter turn-on. Following the simple 75 Hz preamble, any number of consecutive paging calls may be transmitted. Any of the standard preambles can be used in place of the 75 Hz preamble.

When older Golay pagers and GSC pagers are mixed in a system, the address (Word 1, Word 2 combination) independent of the preamble must not be repeated, as is allowed in GSC only systems. Data pagers are also allowed in non-battery saver systems.

2.6 Data paging

A data paging consists of a pager address followed by one or more data blocks of alphanumeric characters. A data block is identical in length to an address block and may be freely substituted for addresses in the batch operating mode. The single call mode can also be used by following the pager address with the data message. Data information is transmitted at 600 bit/s using a (15 : 7) BCH code.

2.6.1 Data block definition

A data block consists of 8 BCH (15 : 7) code words. The data block structure is defined in Fig. 5; each block consists of 56 information bits and 64 parity bits. A single comma bit opposite in polarity to the first data bit transmitted is added at the start of each message block to make the data block identical in length to an address block.

Transmission on the channel starts with the comma bit followed by the least significant bit (LSB) of Word 1, then the LSB of Word 2 ... LSB of Word 8 then jumps back to the second bit of Word 1, etc. Assembling the data block code words by rows and then transmitting on the channel by columns protects against burst errors.

The method for embedding the information into the data block is shown in Fig. 5. One bit of the data block is used as a continue bit, and 7 bits are used for a block check character to minimize undetected message errors. The continue bit has been assigned the following meaning: a "1" indicates additional data blocks to follows, and a "0" indicates the end of message.

For ease in implementing the block check character, the 7 information words of 7 bits each of each (15 : 7) code word are added together (arithmetically), and the least significant 7 bits of the sum form the block check character.

Partially filled message blocks should be filled out with NULL characters.

2.6.2 Symbol assignment

The character sets for use in the numeric and alphanumeric systems are defined in Table VII. In the numeric system, information is assigned on the basis of 4 bits per symbol, and in an alphanumeric system, 6 bits per symbol are used. The numeric set includes the numeric digits (0-9), as well as all necessary control and punctuation characters. The shifted numeric set allows certain alphanumeric designators to be inserted within the numeric data. The alphanumeric set is a slight variation of the ASCII or ISO international 7 bit code with bit number 6 set equal to 1 and not transmitted. Modifications are made to this basic set only to allow insertion of the needed control characters. It is expected that the pagers themselves will display the lower case alphas as upper case alphas.

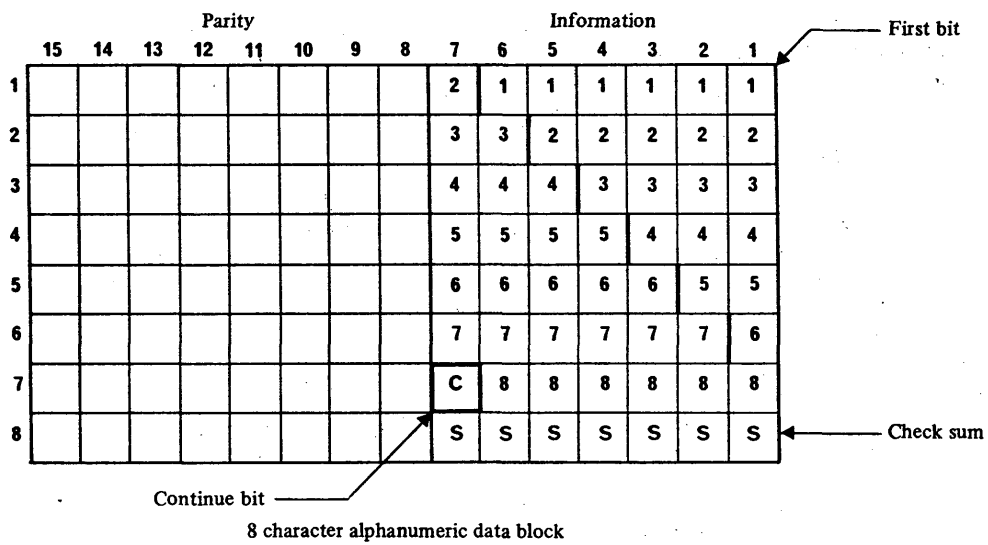
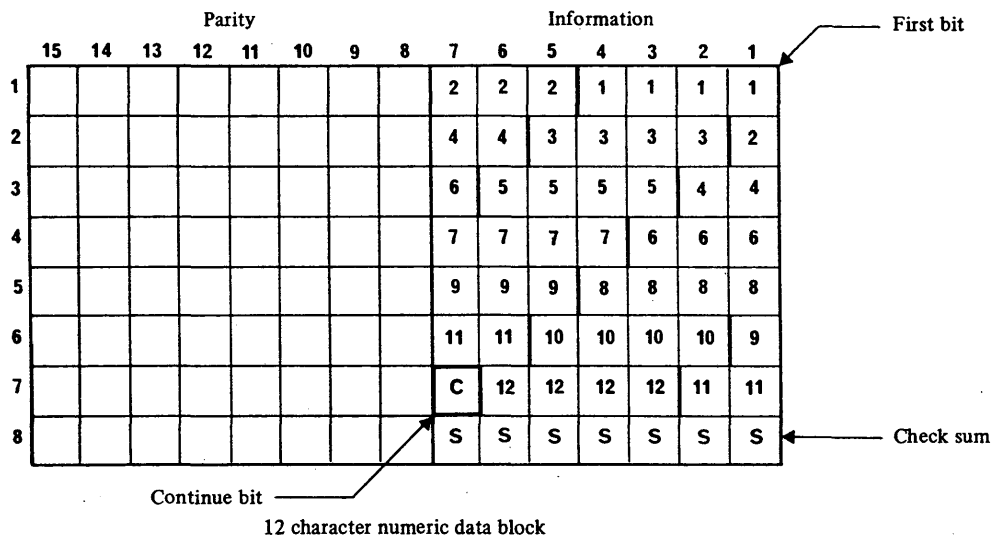


FIGURE 5 - Data block structure

TABLE VII – Numeric and alphanumeric character assignments

Alphanumeric set						Unshifted (basic) numeric set	Shifted numeric set
Bits (1)	7	0	0	1	1	No applicable	No applicable
4 3 2 1	5	0	1	0	1	No applicable	No applicable
0000		Space	0	\	p	0	A
0001		!	1	a	q	1	B
0010		"	2	b	r	2	C
0011		#	3	c	s	3	D
0100		\$	4	d	t	4	E
0101		%	5	e	u	5	Space
0110		&	6	f	v	6	F
0111		'	7	g	w	7	G
1000		(	8	h	x	8	H
1001		)	9	i	y	9	J
1010		*	:	j	z	Null	Null
1011		+	;	k	{	U	L
1100		,	<	l	CR/LF	Space	N
1101		-	=	m	}	-	P
1110		.	>	n	Null/EOM	≡	R
1111		/	?	o	Spare	Shift or (E) (1)	Spare

(1) Bit No. 6 never sent.
(2) (E) is displayed by pagers not employing the shifted numeric set.

2.6.3 Number of data blocks per message

It is suggested that numeric messages be limited to 2 data blocks (24 characters) and that alphanumeric messages be limited to 80 characters or 10 data blocks. A data message may follow the last address in a 16 address batch transmission and may be continued until completion.

2.6.4 Code word generation (15 : 7) BCH code

The generator polynomial for the (15 : 7) code is $x^8 + x^7 + x^6 + x^4 + 1$. The parity information for this code is calculated in the same fashion as was done for the (23 : 12) code. The information bit pattern is divided by the generator polynomial, modulo 2 and the parity bits are the remainder from this division.

3. Pager code structure

The pager code is a six-digit decimal number followed by a series of function digits from which the address Word 1, Word 2 and preamble information can be derived. The pager code digits are defined as follows:

I	G1 G0	A2 A1 A0	f
Preamble index	Group	Address	Function digits

The range of assigned codes is $00001 \leq G1\ G0\ A2\ A1\ A0 \leq 99999$. The sixth digit (I) is used to expand the number of address codes from 100 000 to 1 000 000. The range of the preamble index is $0 \leq I \leq 9$. If the letter N is used in place of the preamble index number, non-battery saver operation is specified. The function digits indicate which address functions are active for any particular pager and what type of page is represented by each function. Up to four function digits may be assigned to each pager code. In the basic large system code plan, at least one pager code is assigned per pager in a sequential fashion starting with 000001.

3.1 Algorithm for obtaining the preamble, Word 1 and Word 2 information from the pager code

In order to transmit a page to any pager, the preamble, Word 1 and Word 2 information must be derived from the pager code. To obtain this information, follow the flow diagram outlined in Fig. 6. For example, the preamble, Word 1, and Word 2 information for the pager code 954853 is as follows:

- the preamble number = 3 (see Table V);
- the Word 1 number = 4 (see Table VI);
- the Word 2 decimal = 1753. Convert 1753 to binary and calculate the parity information obtaining the following:

Decimal	Parity	Information
1753	00010101101	011011011001

The actual polarity of the transmitted bit patterns is determined by the function digit discussed in § 1.3 and whether the single call or batch transmission mode is to be used.

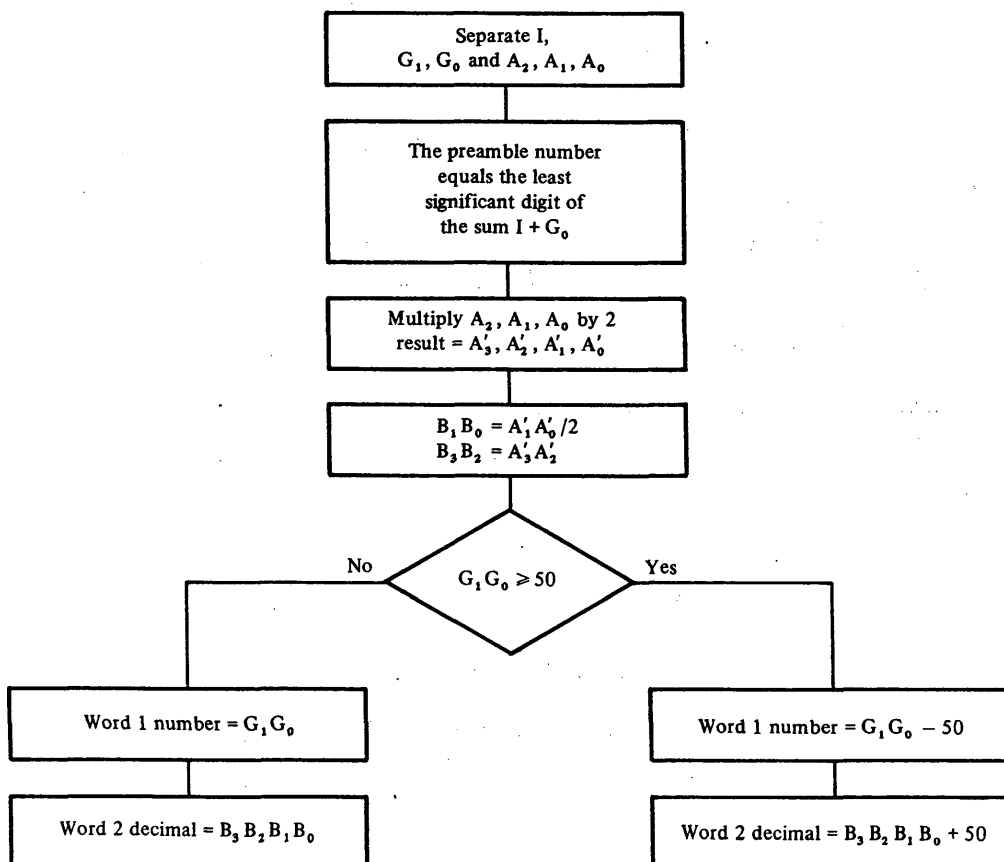


FIGURE 6 – Algorithm for obtaining preamble, Word 1 and Word 2 information from the pager code

3.2 *Pager code restrictions*

Since the start code is included within the range of address Word 2's, some assignable pager codes must be eliminated to prevent a false start code detection. These illegal codes are listed in Table VIII.

TABLE VIII – *Illegal pager codes*

If G_1 G_0 equals	Then do not allow A_2 A_1 A_0 to equal
00-49	000, 025, 051, 103, 206, 340, 363, 412, 445, 530, 642, 726, 782, 810, 825, 877
50-99	000, 292, 425, 584, 631, 841, 851

3.3 *Function digit assignment*

The GSC code allows four functions to be assigned to each independent pager address. In turn, each address function may be assigned independently to 1 of 2 page types (tone only or data). The f digit suffix is a means for making this assignment. Table IX relates address function (Word 1 and 2 polarity) with page type and function digit.

TABLE IX – *Function assignment (non-voice)*

Address function	Binary word format	Function suffix (f)	
		If page type is data, f =	If page type is tone only, f =
1	W_1 W_2	5	9
2	W_1 $\overline{W_2}$	6	0
3	$\overline{W_1}$ W_2	7	3
4	$\overline{W_1}$ $\overline{W_2}$	8	4

ANNEX II

RADIO PAGING USING THE RADIO DATA SYSTEM (RDS) ON FM BROADCAST TRANSMITTERS

1. Introduction

This Annex describes the additional characteristics of the radio data system (RDS) as defined in Recommendation 643 when it is used to combine FM broadcasting with radio paging.

2. Modulation and baseband coding [EBU, 1984; Swedish Telecommunications Administration, 1976]

The modulation of the data channel and the baseband coding including message format are in accordance with Recommendation 643.

3. Additional characteristics for paging [EBU, 1984; Swedish Telecommunications Administration, 1976]

3.1 General

3.1.1 Group type 4A, clock-time and date (CT), is transmitted at the start of every minute.

3.1.2 Group type 1A, programme-item number (PIN), is transmitted at least once per second. The five last bits of its block 2 (spare bits) are used as follows:

- bits B4-B2 :3-bit transmitter network group designation;
- bits B1-B0 :battery saving interval synchronization and identification.

3.1.3 Group type 7A is used to convey the paging information.

3.2 Transmitter network group designation

The first three bits of the five spare bits of block 2 of group type 1A are used to designate the transmitter network to a group of pager group codes. Pagers not belonging to the designated group codes must not lock to the transmitter.

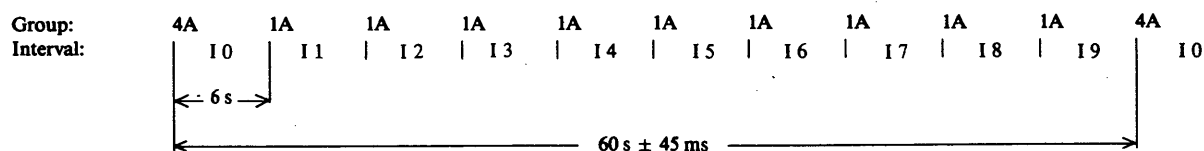
The group designations are as follows:

B4	B3	B2	Group codes	Number of group codes
0	0	0	No paging on channel	
0	0	1	00-99	100
0	1	0	00-39	40
0	1	1	40-99	60
1	0	0	40-69	30
1	0	1	70-99	30
1	1	0	00-19	20
1	1	1	20-39	20

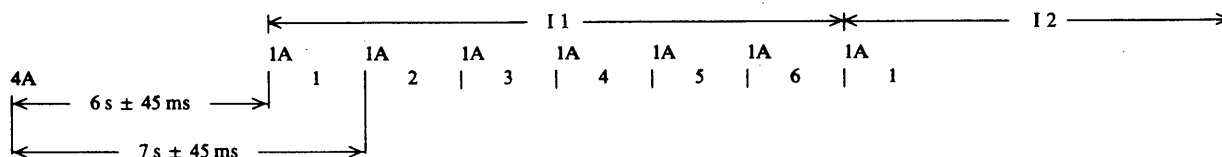
The transmitter network group designation makes it possible to distribute the paging calls over one to four networks, e.g. several networks during day-time and a single network during night-time. The number of group codes in each network are shown below for the different number of networks in operation.

Number of transmitter networks	Number of group codes respectively
1	100
2	40/60
3	40/30/30
4	20/20/30/30

3.3 *Transmission sequence (battery saving)*



Timing within intervals:



1A number within interval	1	2	3	4	5	6	1
Bit B1	1	1	0	0	0	0	1
Bit B0	0	J3	J2	J1	J0	1	0

For battery saving purposes, each minute is divided into 10 intervals of equal length (I 0 ... I 9). Each paging receiver belongs to the interval corresponding to the last digit of its individual code (digit 0 belongs to I 0 and so on). Paging calls are placed within the interval corresponding to the last digit or within the two intervals following that interval.

To enable the receivers to synchronize to the correct interval, the last two bits, B1 and B0, of the spare bits of block 2 of group type 1A are used. The start of an interval is indicated by the transmission of two 1A groups with B1 = 1 (in interval I 0 the first 1A group is replaced by 4A). The first 1A (or 4A for I 0) group is transmitted at the start interval and the other 1 s later. Accuracy is ± 45 ms in relation to the medium time of the preceding 4A group. Within an interval at least 3 more 1A groups are transmitted (bit B1 = 0). Bit B0 of 1A groups number 2, 3, 4 and 5 is used to sequentially transmit the four bits J3 J2 J1 J0 of the BCD-coded interval number 0 ... 9. Excessive 1A groups within an interval have their bit B0 = 1.

The receiver may enter battery saving mode after start of its interval:

- if at least 10 groups differing from group type 7A have been received, or
- if a paging call, belonging to an interval different from the receivers' own and the two preceding intervals, has been received, or
- after the start of the third interval after its own interval.

The receiver shall be considered to have lost its interval synchronization:

- if there is a paging call within the receivers' own interval to a receiver not belonging to the interval or the two preceding intervals, or
- if an error-free reception of the interval marking (J3 J2 J1 J0) is not the one expected.

(Checking of J3 J2 J1 J0 is not necessary each time the receiver leaves battery saving mode.)

3.4 *Locking to a channel*

3.4.1 The receiver searches for one of the offset words A ... D. When this is found, it searches for the next expected offset word at a distance of $n \times 26$ bits, $n = 1 \dots 6$. When two offset words have been found, the receiver is synchronized to both block and group.

After block and group synchronization, the receiver must find the correct country code (within the PI-code) and group designation of the transmitter network.

3.4.2 When scanning the frequency band, block and group synchronization must occur within 1 s and correct country code and group designation must be found within 2 s after block and group synchronization. Otherwise the receiver must leave the channel.

3.4.3 When locking to the channel after battery saving mode, block and group synchronization and the reception of correct country code and transmitter group designation must occur within 15 s. Otherwise the receiver shall leave the channel.

3.4.4 For quick scanning, the information about alternative frequencies in group type 0A may be used.

3.5 *Loss of synchronization*

3.5.1 Clockslip may be detected by using the fact that the program identification (PI) code is very seldom altered. By calculating the syndrome for this block and the block shifted plus/minus one bit, it is possible to see whether clockslip has occurred. If the information becomes correct after a one bit shift, it is considered that a clockslip has occurred, all received data is shifted accordingly and the receiver is correctly synchronized.

3.5.2 When 43 out of the last received 45 blocks have a syndrome different from zero (for the respective offset words), the channel locking is lost and the receiver shall scan the band for a better channel.

3.5.3 If the group code of the receiver is no longer in accordance with the transmitter group designation code, the receiver shall leave the channel and scan the band for a new channel.

3.6 Group type 7A message format

3.6.1 General

Group type 7A:

Block 1	Block 2	Block 3	Block 4
PI ***	TP PTY ***	Paging ***	Paging ***
Offset A	Offset B	Offset C	Offset D

Block 2 bit map:

0	1	1	1	:	0	TP		PTY		AB : T3	T2	T1	T0	///	*Checkword*
														///	
7	A	Paging A/B				Text segment address code									

Block 1 comprises the PI code found as the first block of every RDS group type. Blocks 3 and 4 are used for paging information.

In block 2 the five last bits are used to control the paging information. Bit AB, paging A/B, is used as a flag which changes its value between different paging calls thus indicating the start of a new call. Bits T3-T0 are used as a 4-bit text segment address code and to indicate the type of additional message that follows:

T3	T2	T1	T0	
0	0	0	0	no additional message,
0	0	1	X	10 digit numeric message
0	1	X	X	18digit numeric message
1	X	X	X	alphanumeric message

X indicates state 0 or 1.

3.6.2 Paging without additional message

Group type 7A:

Block 1	Block 2	Block 3	Block 4
PI ***	7A TP PTY ***	Y1Y2 Z1Z2 ***	Z3Z4 n.u. ***

Block 2 bit map:

0	1	1	1	:	0	TP		PTY		AB : 0	0	0	0	///	*Checkword*
														///	
7	A	Paging A/B				Text segment address code									

Text segment address code: 0 0 0 0

Y1Y2 denote the group code

Z1-Z4 denote the individual code within the group

Yn and Zn denote BCD-coded digits 0 ... 9

n.u. 8 last bits of block 4 not used.

3.6.3 Paging with additional numeric message

The additional numeric message is transmitted in 1 or 2 7A groups following the first 7A group of the call. Other group types may be transmitted in between:

Other group types	7A group 1	Other group types	7A group 2	Other group types	7A group 3

Third 7A group only transmitted in case of an 18 digit message.

7A group 1:

PI ***	7A TP PTY ***	Y1Y2 Z1Z2 ***	Z3Z4 A1A2 ***

7A group 2:

PI ***	7A TP PTY ***	A3A4 A5A6 ***	A7A8A9A10 ***

7A group 3 (only with 18 digit message):

PI ***	7A TP PTY ***	A11A12A13A14 ***	A15A16A17A18***

Block 2 bit map:

0	1	1	1	:	0	TP		PTY		AB : T3	T2	T1	T0	///	*Checkword*	///
						7	A	Paging A/B				Text segment address code				

The text segment address code is used to indicate the contents of blocks 3 and 4 in respective groups:

T3	T2	T1	T0	Contents of blocks 3 and 4
				10 digit message:
0	0	1	0	Group and individual code Y1Y2 Z1-Z4 plus message digits A1-A2.
0	0	1	1	Message digits A3-A10.
				18 digit message:
0	1	0	0	Group and individual code Y1Y2 Z1-Z4 plus message digits A1-A2.
0	1	0	1	Message digits A3-A10.
0	1	1	0	Message digits A11-A18.

Y1Y2 denote the group code

Z1-Z4 denote the individual code within the group

Yn and Zn denote BCD-coded digits 0 ... 9.

A1-A18 denote the numeric message.

An denotes a binary coded hexadecimal digit 0 ... A.

Hexadecimal A is used to indicate a blank position in the message.

— A new call is marked by the altering of the "paging A/B" flag.

3.6.4 Paging with additional alphanumeric message

The additional message is transmitted in consecutive 7A groups. Other group types may be transmitted in between:

Other group types	7A group 1	Other group types	7A group 2	Other group types	7A group 3	... etc.

7A group 1:

PI ***	7A TP PTY ***	Y1Y2 Z1Z2 ***	Z3Z4	n.u. ***

Following 7A groups:

PI ***	7A TP PTY ***	Cm-Cm+1 ***	Cm+2-Cm+3 ***

Each of the groups contains 4 characters coded in 8 bits each.

Block 2 bit map:

0	1	1	1	:	0	TP	PTY	AB : T3	T2	T1	T0	*Checkword*
7						A	Paging A/B			Text segment address code		

The text segment address code is used to indicate the contents of blocks 3 and 4 in respective groups:

T3	T2	T1	T0	Contents of blocks 3 and 4
1	0	0	0	Group and individual code Y1Y2 Z1-Z4.
1	0	0	1	Message characters Cn-Cn+3.
1	0	1	0	Message characters Cn+4-Cn+7.
1	0	1	1	Message characters Cn+8-Cn+11.
1	1	0	0	Message characters Cn+12-Cn+15.
1	1	0	1	Message characters Cn+16-Cn+19.
1	1	1	0	Message characters Cn+20-Cn+23.
1	1	1	1	End of alphanumeric message.

Text segment address code is repeated cyclically 1001 ... 1110 for every 24 characters of the message transmitted (n is increased by 24 for each cycle).

End of message is indicated by the transmission of text segment address code 1111 or a new call (indicated by the altering of the "paging A/B" flag).

Maximum length of message is 80 characters.

Y1Y2 denote the group code

Z1-Z4 denote the individual code within the group

Yn and Zn denote BCD-coded digits 0 ... 9.

Cn denotes a message character coded in 8 bits.

n.u. 8 last bits of block 4 of Group 1 not used.

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REPORT 739-1 *

INTERFERENCE DUE TO INTERMODULATION PRODUCTS IN THE
LAND MOBILE SERVICE BETWEEN 25 AND 1000 MHz

(Study Programme 7C/8)

(1978-1986)

1. Introduction

Intermodulation causes a degradation to radio services when:

- unwanted emissions are generated in transmitters;
 - unwanted emissions are generated in non-linear elements external to the transmitters;
- or
- in-band intermodulation products are generated in the radio-frequency stages of receivers.

These cases occur with varying probability and varying severity. They may be reduced by equipment design or careful choice of channels, but solutions of the latter type to one case of intermodulation may increase another.

2. Transmitters

The last active stage of a transmitter is usually an amplifier. The current in this stage will be repeatedly swept from zero amplitude to a maximum and the impedance of the output active device is liable to contain a small amount of non-linearity.

If any other signal from another emission is also present at the output of this stage, the non-linearity will give rise to a number of products having frequencies with specific frequency relationships to the frequency of both the wanted and unwanted signals. These products are called intermodulation products, and their frequencies may be expressed as

$$f_i = C_1 \cdot f_1 + C_2 \cdot f_2 + \dots + C_n \cdot f_n \quad (1)$$

where the sum $|C_1| + |C_2| + \dots + |C_n|$ is the order of the product.

The odd-order intermodulation products may be relatively close in frequency to the wanted signal frequency and thus coupled via the output circuit to the antenna with minimal attenuation.

In order to be able to calculate the effects of these products, it is necessary to establish certain terms.

2.1 Coupling loss, A_c

The coupling loss, A_c , in dB, is the ratio of the power emitted from one transmitter to the power level of that emission at the output of another transmitter which may produce the unwanted intermodulation product.

Typical values for the coupling loss on a common site are of the order of 30 dB.

2.2 Intermodulation conversion loss, A_I

The intermodulation conversion loss A_I , in dB, is the ratio of power levels of the interfering signal from an external source and the intermodulation product, both measured at the output of the transmitter.

Without any special precautions, typical values for semi-conductor transmitters are to be found in the range of 5 to 20 dB and for valve transmitters, in the range of 10 to 30 dB, in respect of the 3rd order product ($2f_1 - f_2$).

The overall loss between a transmitter providing the unwanted emission giving rise to the intermodulation product and a receiver operating at the frequency of the product is:

$$A = A_c + A_I + A_p \quad (2)$$

where A_p , in dB, is the propagation loss of the intermodulation product between the relevant transmitter output and the receiver input.

* This Report should be brought to the attention of Study Group 1.

Note that the power level of the transmitter in which the intermodulation is produced is not included in the formula but this level may have an effect on the value of the intermodulation conversion loss A_I .

Example

Signal frequency of transmitter producing intermodulation product:	f_1
Signal frequency of transmitter whose emission is coupled into transmitter (f_2):	f_2
Power level of transmitter (f_2):	+ 10 dBW
Assumed coupling loss A_c :	30 dB
Assumed conversion loss A_I :	15 dB
Assumed receiver threshold signal level:	-150 dBW
Overall path loss is equal to 10 dBW - (-150 dBW) = 160 dB.	
If $A_c + A_I = 45$ dB, then the required value of A_p is 115 dB.	

Figure 1 gives an example of propagation path losses at 100 MHz and, under free space conditions, a very large distance is required between the "product producing" transmitter and the receiver. If the receiver is a mobile station, this distance is considerably reduced. It may be concluded therefore that 2-frequency operation provides better conditions for the reduction of the effects of inter-transmitter intermodulation if the base receive frequency band is remote from the transmit frequency band.

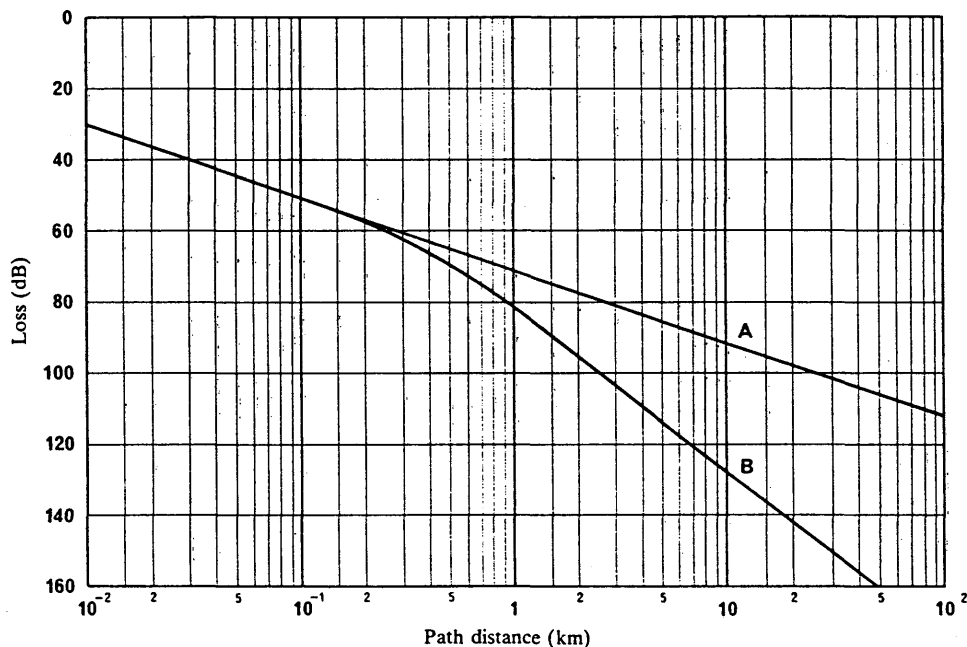


FIGURE 1 - Short range path loss at 100 MHz ($\frac{\lambda}{2}$ dipoles assumed)

Curves A: free space

B: Recommendation 370-3; $h_1 = 37.5$ m, $h_2 = 2$ m

The intermodulation caused by two or more mobile transmitters will be worse when the mobiles are closest together and when the desired signal originates at a mobile at the edge of the service area, an event which is associated with some (perhaps small) probability. The mobile being interfered with will be received at its base as a signal of widely varying level (due to fades and shadows) which will be independent of the IM interference. These wide and independent variations can allow the IM to reach harmful values for periods of time, even when its average value is much less than that of the signal.

3. External non-linear elements

On most sites, external non-linear elements will be at junctions in masts, feeders, and other antennas which are closely coupled to the radiating elements of nearby transmitters.

It would be useful to determine conversion losses for masts etc., of various qualities in terms of the isotropic loss between transmitters and the masts, etc. It would then be possible to establish specific values as good engineering practice.

4. Receivers

An intermodulation response is a response at the output of a receiver from an in-band signal generated in the RF stages of the receiver. This in-band signal is generated by the presence of two (or more) high-level signals in a non-linear section of the RF stages. As with transmitters, the two (or more) unwanted signals must have specific frequencies such that the intermodulation product lies within the frequency band accepted by the receiver.

This receiver characteristic is normally recorded as a single measurement with the level of the unwanted signals equal and is given as a single ratio which is:

the ratio of the level of these two equal signals

to

the apparent level of the intermodulation product at the input to the receiver.

It is possible, however, to cause a similar product level when the unwanted signals are not equal.

Figure 2 gives examples (3 theoretical and 1 measured) of the overall third order intermodulation characteristic of receivers. It shows that intermodulation may easily be a problem when one of the unwanted signals is not excessively high. Such curves can be used to calculate other intermodulation product levels when the unwanted signals do not have values equal to those plotted.

For a product with a frequency relationship of the form $(2f_1 - f_2)$, the level will be proportional to the level of the signal at frequency f_2 , but will vary as the square of the level of the signal f_1 ; i.e. the product will have an amplitude of the form $k \cdot V_1^2 \cdot V_2$, where V_1 , V_2 are the amplitudes of the signals at frequencies f_1 and f_2 respectively.

When a mobile receiver is used in a multi-channel system it will be subject to an intermodulation response due to many equally spaced high level signals. The following relationship has been suggested by the People's Republic of China to relate the maximum permissible signal level with the intermodulation response rejection ratio of the receiver [CCIR, 1982-86a]:

$$E_s + 3E_M \geq 3E_{I_{max}} + B + k(n,p)$$

where:

E_s : wanted signal level (dB) above sensitivity;

$E_{I_{max}}$: maximum interference signal level (dB) above sensitivity;

E_M : receiver's third-order intermodulation rejection ratio (dB) (for two signals);

B : RF protection ratio (dB);

$k(n,p)$: a constant dependent on the number of channels n and channel sequence p .

The derivation of this formula and the calculation of $k(n,p)$ are given in Annex I.

5. Reduction of intermodulation product levels in transmitters

5.1 Intermodulation conversion loss

It is obvious that a reduction of the non-linearity, particularly of the odd-numbered orders, will improve the overall performance and increase the value of the intermodulation conversion loss A_I .

From the example in § 2, it is evident that a considerable improvement is necessary before the relevant path loss reduces to manageable values.

5.2 Coupling loss

The coupling loss can obviously be increased by increasing the distance between the relevant transmitters but it may not always be possible to do so effectively at a particular site.

Ferrite isolators could be used in the output circuits of the transmitter in which the product is generated but present production units do not provide much more than 25 dB additional loss and the use of multiple units is inhibited by the inherent non-linearity of the isolators themselves. To suppress undesirable products, filters may be required after such isolators. These isolators are equally effective irrespective of the frequency spacing between f_1 and f_2 .

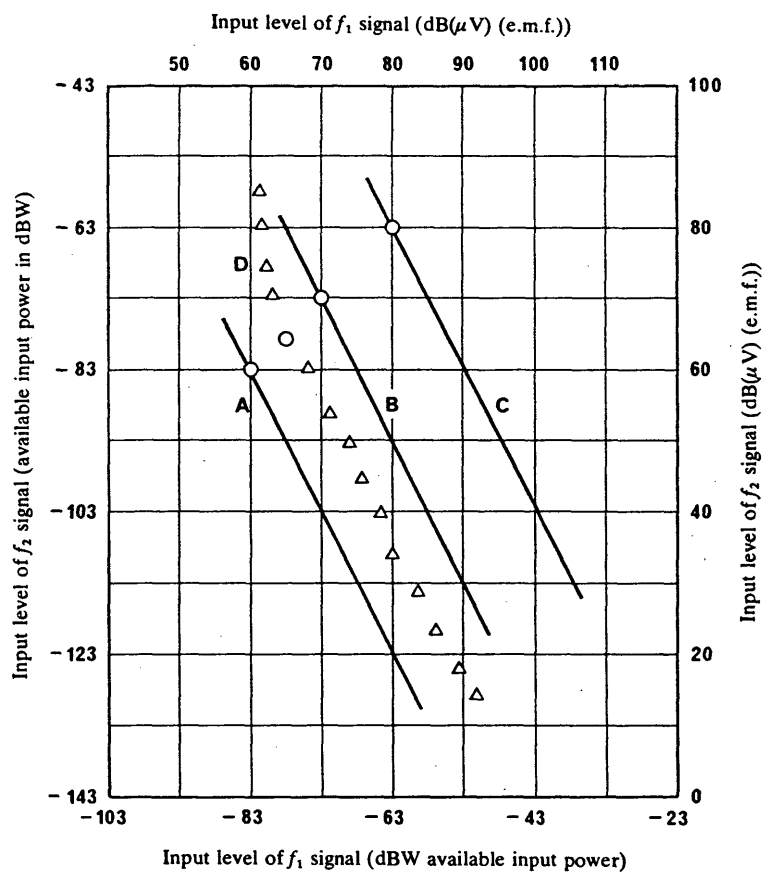


FIGURE 2 – Receiver intermodulation characteristic

Levels of unwanted input signals which together produce a constant product level.

Curves A, B and C: derived characteristics based on a single recorded value of the receiver's third order intermodulation characteristic, i.e. for $(2f_1 - f_2)$.

Curves A: based on a single value, with both input levels at a level of 60 dB(μ V) (e.m.f. to 50 ohms).

B: based on a single value, with both input levels at a level of 70 dB(μ V) (e.m.f. to 50 ohms).

C: based on a single value, with both input levels at a level of 80 dB(μ V) (e.m.f. to 50 ohms).

D: measured values for a receiver for which the specified criterion is achieved with equal input signal levels of 65.5 dB(μ V) (e.m.f. to 50 ohms).

Cavity filters can also be used and examples of their theoretical responses are given in Fig. 3. They may be used in cascade or in more complex series-parallel combinations but in all cases, their performance is dependent on the frequency spacing between f_1 and f_2 . They have the advantage that they will also attenuate the product level at the input to the antenna or transmission line and thus increase A_f .

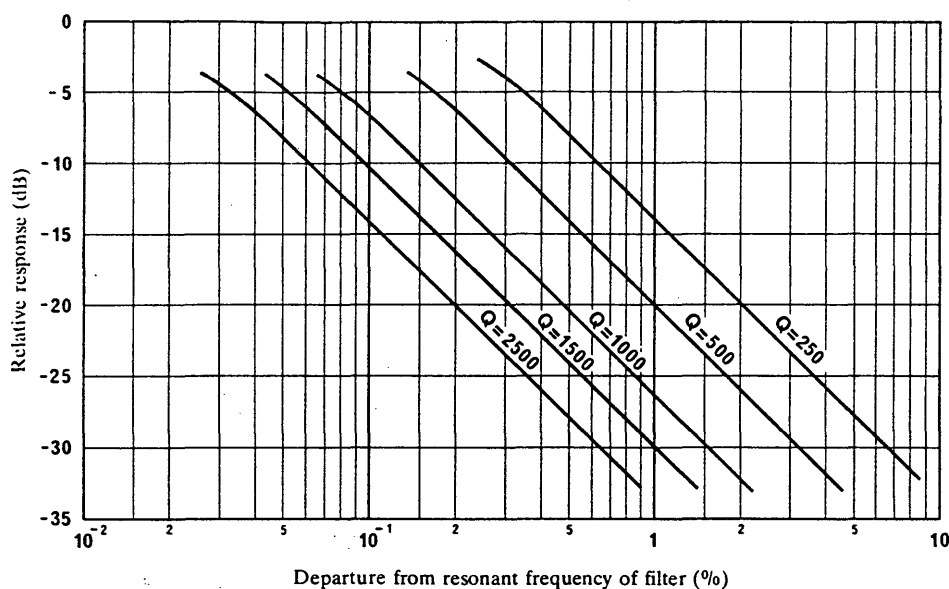


FIGURE 3 – Theoretical response of cavity band-pass filters

For values of loaded Q of 250-2500.

Note. — The unloaded Q should be at least 5 times the loaded Q and preferably 10 times.

An economic and efficient filter is the coaxial cavity resonator, either in its pure quarter-wavelength form or with varying degrees of modification to reduce the overall length and improve the value of the loaded Q . The resonator should be robust, simple to tune, highly efficient in terms of transmission loss, and provide a high degree of isolation at the required frequencies. Resonators for use with transmitters should have a low temperature coefficient and good thermal conductivity, so that their performance is not affected by changes in ambient temperature or through being heated by transmission losses. Temperature compensation can be employed to maintain the length of the centre conductors. Physical robustness is necessary to avoid changes in technical parameters from being caused by mechanical shock or deformation. The physical and mechanical design should also prevent the formation of electrical discharges or corona. Adjustable telescopic centre conductor assemblies permit a variation of resonant frequency of, typically, $\pm 15\%$ of the centre frequency.

Reliable and economical resonators can be manufactured from high-conductivity aluminium for the larger units, and silver-plated copper or brass for smaller units. Practical limitations of mechanical engineering govern the upper limits of Q obtainable with a cavity resonator. As the diameter is increased, the value of the unloaded Q is increased, but the sensitivity of tuning and the temperature coefficient become more critical. Practical and satisfactory resonators with a power handling capacity of up to 250 W can, however, be made for the band 150-170 MHz, for example, having an unloaded Q as large as 18 000, with a diameter of 0.58 m, and length 0.63 m, giving 35 dB discrimination at a frequency 1% removed from the resonant frequency.

It is not usual to employ cavity resonators for values of Q_0 below about 1000, since there are more satisfactory techniques, e.g. helical resonators, which can be coupled together to form smaller but relatively efficient filter units. Tables I and II give the choice of types of filter and their relative costs.

TABLE I – Relative sizes and costs of resonators (150–174 MHz)

1	2	3	4	5	6
Reference	Q_0	Q_L	Attenuation at 1% F_0 (dB)	Diameter (m)	Relative cost of practical resonators
A	920	100	7	0.03	1.0
B	2300	250	14	0.07	1.7
C	4600	500	20	0.14	2.8
D	6900	750	24	0.21	3.3
E	9200	1000	26	0.29	3.9
F	11700	1250	28	0.37	4.6
G	13800	1500	30	0.46	5.3
H	16100	1750	32	0.53	6.8
I	18400	2000	35	0.58	7.1

TABLE II – Relative costs of practical resonators for other frequencies

Resonant frequency (MHz)	Cavity height (m)	Unloaded Q					
		920	2300	4600	6900	9200	13800
50–60	1.55	*	*	8.7	12.0	14.7	+
60–80	1.15	*	*	5.5	7.3	10.6	14.9
95–110	0.85	*	3.3	4.1	5.2	6.4	10.7
120–150	0.68	*	2.6	3.3	4.2	5.0	8.9
150–174	0.63	1.0	1.7	2.8	3.3	3.9	5.3
160–180	0.52	0.9	1.5	2.4	2.9	3.4	4.6
400–500	0.24	0.8	1.0	1.5	2.0	2.2	3.0

Note. – Items not tabulated are identified as follows:

* Helical resonator superior

+ Single cavity large and somewhat uneconomic.

Compared with the total cost of the radio equipment at a base station, cavity resonator filters are an economical and efficient means of reducing spurious emissions and preventing or minimizing interference.

5.3 Identification of the source of an intermodulation product

The frequency of the third order intermodulation resulting from the interaction of two transmitters may be expressed as either $2f_1 - f_2$ or $2f_2 - f_1$.

If the product is $2f_1 - f_2$, the mixing is occurring within or close to the transmitter operating on f_1 .

Conversely, if the product is $2f_2 - f_1$, the mixing is occurring within or close to the transmitter operating on f_2 .

In the case of FM or PM emissions, the deviation caused by modulation is doubled when a second harmonic is generated. So if the modulation on one of the intermodulation products appears to be excessive, this modulation is probably transferred from the f_1 signal of a $2f_1 - f_2$ mixing.

6. Reduction of intermodulation products in receivers

As with transmitters, a reduction in the non-linearity of a receiver will improve the performance.

Attenuation at the input of the receiver may be used to reduce the level of an intermodulation product. The levels of these products are related to the levels of the signals that produce them, in such a way that the attenuation (in dB) of each " n^{th} " order product will, in most cases, be n times the attenuation (in dB) of the wanted signal.

For example, a 3 dB attenuator will reduce a third order product by 9 dB while reducing the wanted signal by 3 dB. This may also be used as a test device to prove that the intermodulation product is being generated in the receiver.

Cavity filters can be used, either as rejection filters to f_1 and/or f_2 , or as band-pass filters to the wanted signal. Again the effectiveness of these filters depends on the frequency spacings involved.

7. Reduction of intermodulation interference by frequency arrangements

The frequencies to be used can be arranged so that no receiver on the product frequency is required to operate in an area where the unwanted signals may produce an intermodulation product of sufficient level to disturb the service. If this level is at the maximum sensitivity level of the receiver; it will mean receivers cannot be used for distances up to 2 km from the sites of the base station operating at f_1 and f_2 . This applies even when the f_1 and f_2 stations are separated by several kilometres and thus implies that the base station on the product channel must be sited outside the service area of stations operating on f_1 and f_2 . This leads to very poor use of the frequency spectrum.

In systems that operate a number of frequency channels, most cases of harmful base transmitter and mobile receiver intermodulation within the system can be alleviated by the choice of even channel sets at the base stations. This means that the channels of each base station are evenly distributed at a constant frequency separation. In a service area the intermodulation products within the band used will in that case coincide with channels of the set, and the ratio of the desired signal to the intermodulation product in a mobile receiver is independent of the distance and propagation characteristics.

8. Reduction of intermodulation interference by other arrangements

If continuous tone signalling is used, the receiver will operate only in the presence of this signalling tone and it is then necessary only to ensure that the wanted signal on the product channel exceeds the level of an unwanted product of f_1 and f_2 by an amount in excess of the required protection ratio. This can be best assured by siting the product channel base transmitter at the same, or near to, the site of stations operating on f_1 and f_2 . Under these conditions, the need for filters or other devices in the transmitter or receiver is reduced.

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CCIR Documents

[1982-86]: a. 8/3 China (People's Republic of).

ANNEX I

RECEIVER INTERMODULATION RESPONSE IN A MULTI-CHANNEL SYSTEM

1. Number of third-order intermodulation products in a multi-channel system

When a system consists of n channels with equal intervals and n is an even number ($n \geq 4$), the number of third-order intermodulation products S_p falling into each channel is shown in Fig. 4 [Morinaga, 1972] including types $2A - B$ and $A + B - C$, designated as type III-1 and type III-2 respectively. From channel 1 to channel n , each channel has $\left(\frac{n}{2} - 1\right)$ type III-1 products and the rest are type III-2 products. The type III-2 products are 6 dB above type III-1's. Since there are three unwanted signals involved, it is valid only when $n \geq 4$.

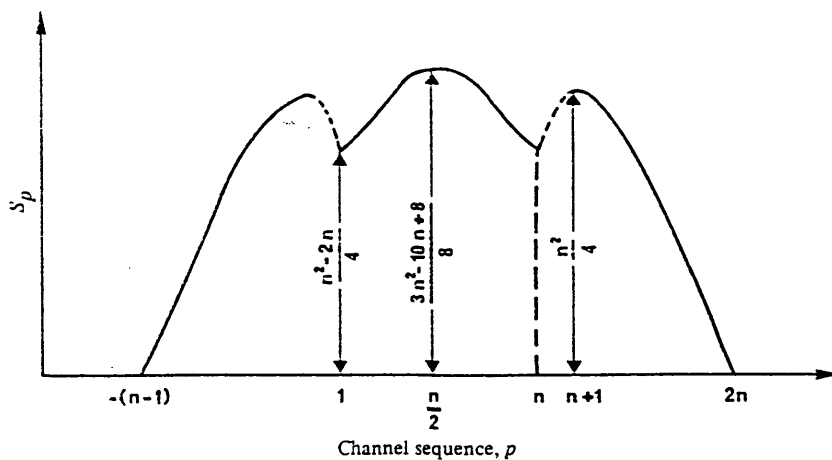


FIGURE 4

Due to the fact that $S_{p \max}$ with an odd number of n is still equal to or less than that with an even number of n , n can be taken as an even number for the following analysis.

2. Intermodulation products in a multi-channel system

As the third-order intermodulation products are random in phase, the intermodulation product level in each channel will be $k(n,p)$ dB higher than the value of sensitivity.

For mid-channels:

$$\begin{aligned} k(n,p)_{\max} &= 20 \log \sqrt{\left[\frac{3n^2 - 10n + 8}{8} - \left(\frac{n}{2} - 1 \right) \right] \times 2^2 + \left(\frac{n}{2} - 1 \right) \times 1^2} \\ &= 10 \log \frac{1}{2} (3n - 7)(n - 2) \end{aligned}$$

For channels on the edge:

$$\begin{aligned} k(n,p)_{\min} &= 20 \log \sqrt{\left[\frac{n^2 - 2n}{4} - \left(\frac{n}{2} - 1 \right) \right] \times 2^2 + \left(\frac{n}{2} - 1 \right) \times 1^2} \\ &= 10 \log \frac{1}{2} (2n - 3)(n - 2) \end{aligned}$$

3. Influence on systems with adjacent channels

Due to the presence of the third-order intermodulation, the frequency spectrum is extended three times, so a system with n channels would influence the bandwidth for $3n$ channels.

Similarly:

$$\begin{aligned} k'(n,p)_{\max} &= 20 \log \sqrt{\left(\frac{n^2}{4} - \frac{n}{2} \right) \times 2^2 + \frac{n}{2} \times 1^2} \\ &= 10 \log \frac{1}{2} n (2n - 3) \end{aligned}$$

$$k'(n,p)_{\min} = 0$$

4. If $E_{I\max} > E_M$ (i.e. the maximum interference signal level exceeds the intermodulation rejection ratio), the third-order intermodulation products level (in dB) will increase by $3(E_{I\max} - E_M)$. So the level of the intermodulation product (E_I) will be:

$$E_I = k(n,p) + 3(E_{I\max} - E_M)$$

For satisfactory system operation:

$$E_s - E_I \geq B$$

So:

$$E_s + 3 E_M \geq 3 E_{I\max} + B + k(n,p)$$

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MORINAGA, T. [March, 1972] Mobile Communication – Theory and Design. Electronic Communication Academy of Japan. 85-91.

REPORT 740-2*

GENERAL ASPECTS OF CELLULAR SYSTEMS

(Question 37/8)

(1978-1982-1986)

1. Introduction

Systems are in use for the land mobile service which utilize the frequency spectrum much more efficiently than conventional systems [Schulte and Cornell, 1960; Frenkiel, 1970; Staras and Schiff, 1970]. The technique which such systems have in common is the division of the desired service area into a number of sub-areas (cells), with one or more base stations providing radio coverage to each cell.

Appropriate frequency planning permits frequencies to be used several times in a coverage area. The cellular technique can also be combined with the trunking concept (see Report 741), thereby improving spectrum utilization efficiency.

2. General system characteristics

When a mobile unit either originates or is prepared to receive a call, the cell which can best serve the unit is selected (e.g. through an algorithm resident in the logic of the mobile unit) and it communicates through a base station covering that cell. With the proper spacing of co-channel cells, each frequency may be used simultaneously in several cells in the service area, thereby multiplying the capacity obtainable in conventional systems.

* This Report should be brought to the attention of Study Group 1.

The general features of a cellular system are:

- accommodation of more than one base station in a service area and thus the expansion of the service area of the system beyond the coverage that a single site provides;
- “hand-off” or transfer of the responsibility for radio coverage of a mobile or hand-held unit from one site to another, i.e. from one radio frequency to another;
- re-use of the same RF assignment simultaneously by more than one base station and for more than one message; this is the cornerstone of the *spectral utilization efficiency* of cellular systems;
- growth; that is, the system must be able to start with a few large cells and gradually grow until many small cells are created at the points of highest traffic density.

These features are essential for a complete cellular system. However, specific equipment arrangements may create systems that are “cellular compatible” (that is, inter-operable with true cellular systems) even though they do not possess all of these attributes. For reasons of economy, the base station equipment used in a small city may not permit the simultaneous existence of two cell sizes, while more sophisticated equipment used in a large city where maximum capacity is required will be fully compliant; the same mobile units will work in either area.

3. Channel re-use considerations

3.1 General

The rejection of co-channel interference depends solely on the carrier-to-interference (C/I) ratio and not on absolute amplitude. Since median C and median I are inversely proportional to a power of the distance from the source, the required co-channel cell spacing (called D) can be specified as some multiple of the cell radius (called R), depending on the median C/I desired.

To a good first approximation:

$$C = k/R^n, \quad I = k/D^n$$

thus:

$$C/I = \left(\frac{D}{R}\right)^n \quad \text{or} \quad D = R \left(\frac{C}{I}\right)^{1/n}$$

Reducing D and R in proportion would allow each frequency to be re-used simultaneously in more cells in the service area. However, it could also increase the mutual interference between co-channel stations beyond usable limits, unless other techniques, such as diversity [Lundquist and Peritsky, 1971] and power control [US Advisory Committee, 1967] are used. The re-use distance must be large enough to give an acceptably low probability of co-channel interference as a result of the propagation conditions [Okumura, 1968], of multipath fading (which has a Rayleigh density function) and log-normal shadowing (slow-fading) experienced in the land mobile service.

The re-use ratio D/R is related to number of channel sets n by:

$$D/R = \sqrt{3n}$$

A typical value of D/R is 4.6/1, corresponding to $n = 7$. For a regular grid of base sites packed in a hexangular grid, only certain values of n are possible: 3, 4, 7, 9, 12, 13 etc.; these correspond to evaluating:

$$n = i^2 + j^2 + ij$$

for non-negative integer values of i and j .

The system designer must decide what speech quality is required and choose the C/I objective. For example, experience in simulations, subjective tests, field trials, and early operational systems at 850 MHz indicates that a 90th percentile equal to or greater than 17 dB median C/I in an urban propagation environment will achieve a user acceptance goal of “good” or higher by the majority of telephone users [see Report 319, § 3.1.1 and 3.1.2]; systems for dispatch or private usage can be designed for a lower objective. During early growth phases when cells are relatively large and horizon effects limit interference, a 7-channel set re-use pattern is usually adequate, using conventional omnidirectional antennas; this assumes that no more than one or two interfering sites provide signals above the threshold level at any specific point in a base station’s coverage area.

As growth to a more densely packed system takes place, either a larger channel re-use pattern (e.g. 12 sets) can be used, or a change-over to directional antennas for transmit and/or receive with a "sector" channel assignment pattern can be implemented. Arrangements using both 60° and 120° sectors have been designed; for a given n value, use of a sector assignment plan with directional antennas increases C/I by up to 5 dB (120° sectors) or 8 dB (60° sectors). In any event, the system planner should be satisfied that the *a priori* objective is exceeded in both the mobile-to-base and the base-to-mobile directions. The mobile-to-base direction is more complicated because of the random location of the potential interferers, because of the mix of vehicular and portable transmitters with different powers, and because of the building penetration loss that the portable unit can encounter.

To accommodate varying user density over the area of service, the cell plan should allow for different size cells to coexist in a system at the same growth stage. This generally means a doubling of the number of channel sets into which the allocation must be divided. Administrations should allow for this if the largest city or populated area is to grow to the divided-cell stage.

If the cellular channel plan avoids adjacent channels at the base station sites, the frequency deviation can be increased to enable closer re-use, resulting from an improved protection ratio.

Private dispatch systems which can handle traffic with a blocked-calls-queued strategy can be efficient with a smaller allocation than can mobile telephone systems.

3.2 *Co-channel interference experienced by stationary units*

Stationary units will not experience multipath fading as a loss in speech quality, resulting from rapid movement through the fading pattern, as mobile units do; rather, they will experience a very large number of points where the wanted signal has faded and the channel is unusable because of co-channel interference. In addition, shadowing will result in areas in which this problem is more severe due to a reduced local mean signal level.

The probability of co-channel interference, $P(s_1 \leq p_{s_2})$ for this case of simultaneous fading and shadowing with an uncorrelated wanted signal and a single interferer, has been calculated [French, 1979] as shown in Fig. 1.

Re-use distance ratios calculated using Fig. 1 and an inverse fourth power propagation law to give particular values of interference probability at the cell edge are shown in Table I, for protection ratios, P , of, for example 8 and 12 dB and shadowing standard deviations, σ , of 6 and 12 dB. Both fading and shadowing were included in the calculation.

Other values of protection ratio that may arise in particular systems should be used with Fig. 1 and the inverse power propagation law to find the corresponding re-use distances. Note that hand-off techniques may provide some diversity against fading, which would reduce the probability of call cut-off.

3.3 *Other comments*

3.3.1 It should be noted that in situations where it is intended to transmit a single message to a large number of vehicles distributed over several cells, the efficiency of frequency utilization is reduced for the duration of that message. Further studies are needed to determine the effect on spectrum efficiency of combining private and public systems.

3.3.2 Among the methods used to exchange channel control signals (channel assignment, dialling, etc.) are:

- the exchange of control signals over channels which can also be used as communication channels;
- the provision of channels dedicated either to control or communication but not both.

Both methods subtract from the total traffic capacity of the system. The choice between the above methods depends primarily on the amount of signalling traffic. In high capacity systems, the channels may be more efficiently utilized if the control channels are dedicated and are capable of being divided into the sub-sets of paging channels and access channels.

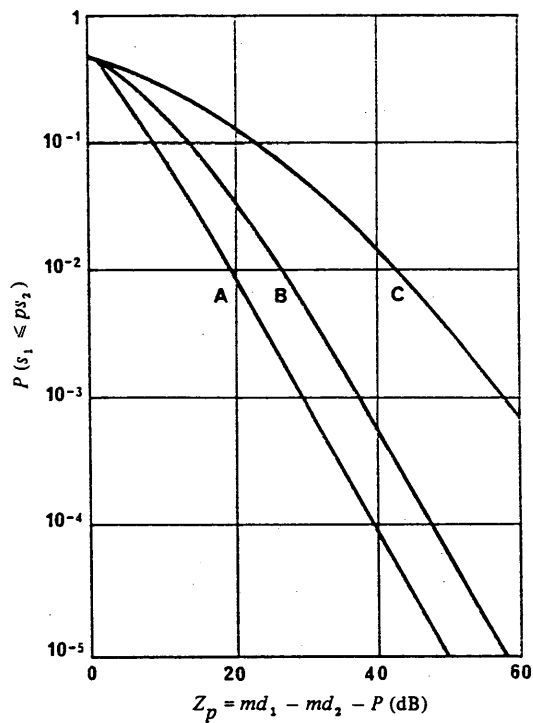


FIGURE 1 – Probability of co-channel interference
(with Rayleigh fading and log-normal shadowing)

Curves A: $\sigma = 0$ dB
B: $\sigma = 6$ dB
C: $\sigma = 12$ dB

s_1 : wanted signal level (V)
 s_2 : interfering signal (V)
 p : protection ratio of modulation system
 $P = 20 \log_{10} p$
 md_1 : area mean level of wanted signal (dB)
 md_2 : area mean level of interfering signal (dB)
 σ : standard deviation of the shadowing
 $Z_p = md_1 - md_2 - P$

TABLE I – Re-use distance ratio (D/R)

$P(s_1 \leq ps_2)$	$\sigma = 6$ dB		$\sigma = 12$ dB	
	$P = 8$ dB	$P = 12$ dB	$P = 8$ dB	$P = 12$ dB
0.5	2.6	3	2.6	3
0.1	4.7	5.6	7.3	8.9
0.03	6.6	8	13	16

3.3.3 If the method which provides control channels for control use only is chosen, control channel frequency assignment may be independent of communication channel cell size.

Two control channel frequency assignment methods are:

- assignment of a separate control channel for each communication cell;
- assignment of single control channel to cover a group of adjacent communication cells.

The choice between the above methods is made in accordance with call processing reliability and control traffic in each system.

3.3.4 Control channels are subject to similar C/I constraints as communication channels. To ensure that a system is not limited by its signalling capacity, the system designer should set aside an adequate number of signalling (control) channels. Preferably, this quantity should be dynamically changeable as the system grows.

4. Diversity

One or more diversity techniques to combat the effects of shadowing and multipath fading may be used, but are not absolutely necessary. The effect of such diversity is to increase significantly the efficiency of spectrum utilization [Lundquist and Peritsky, 1971]. The effects on system cost remain to be determined. While reception diversity at the base station is not a requirement of either a conventional or a cellular system, it can serve:

- to balance the C/I and C/N ratios in the mobile-base and base-mobile directions; and
- to improve the baseband S/N ratio for a given C/I or C/N ratio.

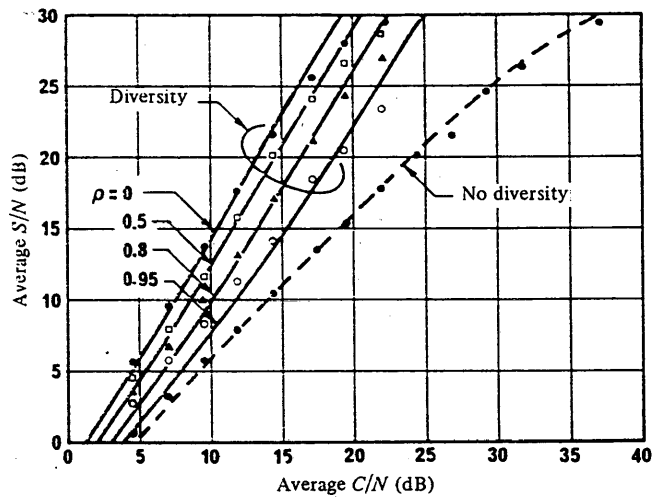
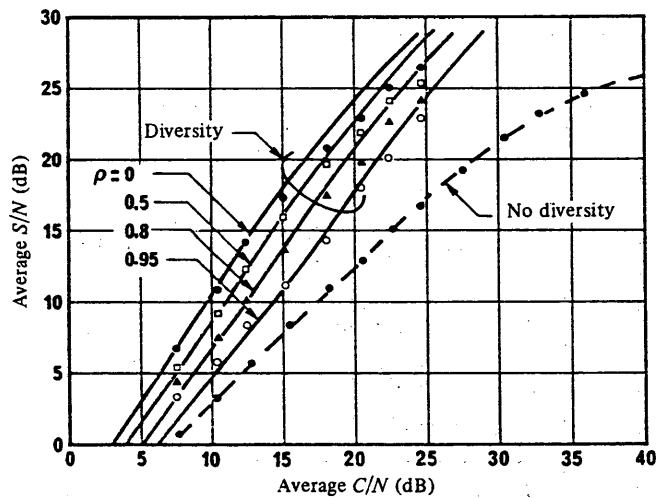
These factors are especially important for service to hand-held portables. At present, space diversity is the preferred method, currently realisable versions of frequency-diversity and time-diversity are not spectrally efficient, although research is continuing in techniques such as spread-spectrum and time-division re-transmission. The coherent combining methods (maximal ratio combining and equal-gain combining) achieve the best improvement compared with other methods. However, the equipment structure is complicated because of the requirement for a control function to ensure that the received signals are aligned in phase. In addition, the cost is high. On the other hand, the antenna switched method has a simple structure but voice quality signal suffers from switching noise and switching delay. A selection combining method with two diversity branches is more practical than two-branch coherent combining, although the former is a little inferior to the latter with respect to voice and data transmission. Switching noise generated in selecting the strongest signal level branch is reduced, and the voice signal quality is improved, by a post-detection selection combining technique. This technique is based on the selection of the baseband demodulated output of the diversity branch with the maximum reception level.

4.1 *Improved characteristics for thermal noise*

An example of experimental results [Suwa *et al.*, 1984] of thermal noise reduction in a voice signal using post-detection selection combining is shown in Fig. 2. The relative C/N is defined as the difference in average C/N required to obtain a given baseband S/N of 8 dB with diversity compared to the average C/N required to obtain that S/N without diversity. The relative C/N is more than 6 dB for an envelope correlation coefficient $\rho = 0$ and greater than 5 dB for $\rho = 0.5$.

4.2 *Improved characteristics for co-channel interference*

An example of experimental results [Suwa and Hattori, 1985] of co-channel interference reduction in voice signal using post-detection selection combining is shown in Fig. 3. The relative C/I is defined as the difference in average C/I required to obtain a given baseband S/I of 14 dB with diversity compared to the average C/I required to obtain that S/I without diversity. The relative C/I is more than 6 dB for an envelope correlation coefficient of $\rho = 0$ and greater than 5 dB for $\rho = 0.5$.

(a) $B = 16$ kHz(b) $B = 8$ kHzFIGURE 2 - C/N versus S/N performance

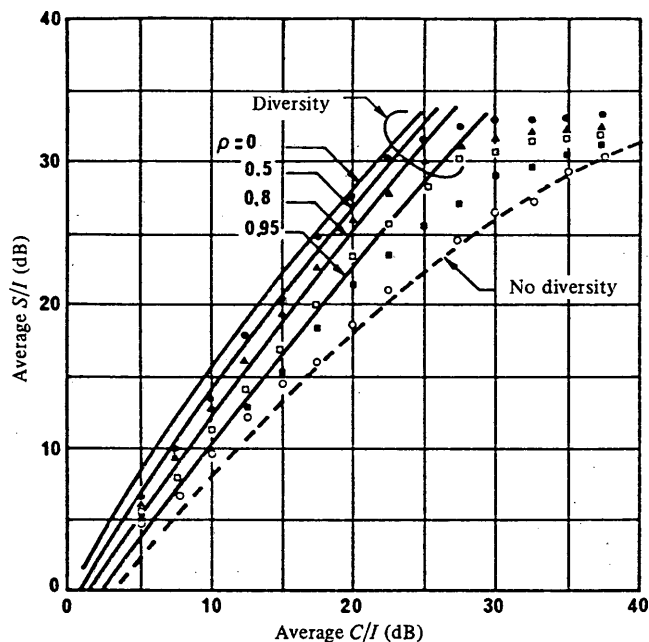
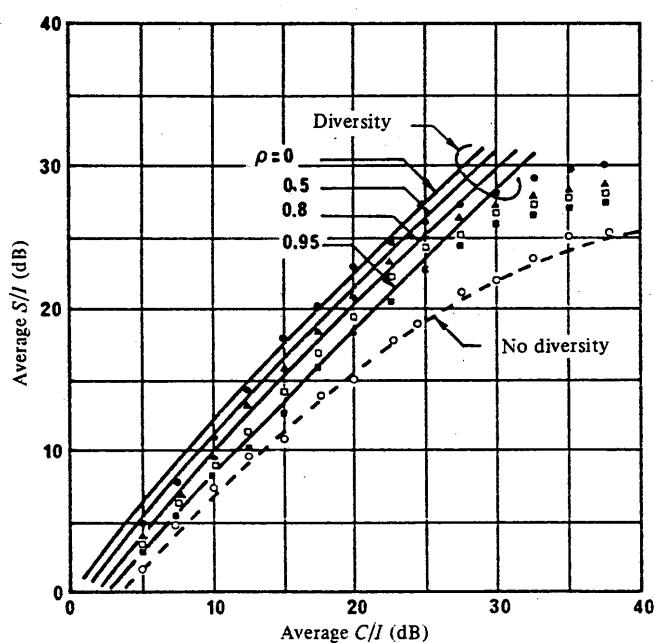
Condition: - carrier frequency: 900 MHz band
 - IF filter bandwidth B : 16 and 8 kHz
 - fading rate: 34 Hz
 - standard modulation level: $3.5/\sqrt{2}$ rad r.m.s. - $B = 16$ kHz
 $1.75/\sqrt{2}$ rad r.m.s. - $B = 8$ kHz

Compressor reference level is set equal to standard modulation level

C/N is measured in 16 kHz and 8 kHz IF filter bandwidth

S/N is measured in 0.3-3 kHz baseband filter bandwidth

A 2:1 syllabic compandor is applied

a) $B = 16$ kHzb) $B = 8$ kHzFIGURE 3 – C/I versus S/I performance

Condition: – carrier frequency : 900 MHz band
 – IF filter bandwidth B : 16 and 8 kHz
 – fading rate : 34 Hz
 – standard modulation level: $3.5/\sqrt{2}$ rad r.m.s. – $B = 16$ kHz
 $1.75/\sqrt{2}$ rad r.m.s. – $B = 8$ kHz

Compressor reference level is set equal to standard modulation level

C/I is measured in 16 kHz and 8 kHz IF filter bandwidth

S/I is measured in 0.3-3 kHz baseband filter bandwidth

A 2:1 syllabic compandor is applied

5. Coping with vehicle movement

In a cellular system, mobile detection, location and registration as well as inter-cell switching are necessary when a mobile station moves across a cell boundary to:

- limit channel switching traffic;
- maintain adequate transmission quality;
- prevent the increase of co-channel interference;
- minimize the impairment caused by channel switching.

When the signal from a mobile unit engaged in a cell becomes inadequate, the system may be equipped to become aware of this fact and to effect a transfer (hand-off) from the original base station to a more appropriate one; conversely, the system may only locate the mobile unit and assign a base station at the beginning of a call, maintaining that assignment for the duration of the call. If a mobile unit is allowed to continue to use a base station when it moves out of that station's cell, the signal it encounters will have greater variability and lower average value. Similarly, since it can now move closer to the co-channel base stations serving other mobile units on the same frequency, the interference it encounters, and the interference it causes, will increase. Therefore, for any given quality of transmission, co-channel base stations would have to, in general, be spaced further apart if "hand-off" is not employed than they would have to be if it is; this adversely affects spectrum efficiency. For these reasons, hand-off is considered an essential feature of all cellular systems. One appropriate method to identify the radio cell from which a moving vehicle can best be served is the field-strength level of the carrier received from the mobile unit.

Field experience indicates that the voice quality of a cellular system is no better than the system's hand-off algorithm permits it to be. Too tight a control on signal level requires more hand-offs than necessary; this

- detracts from the capacity of the controller; and
- can cause frequent distraction to the users.

Too loose control, on the other hand, permits signal statistics to remain at unnecessarily low levels for long periods, can lead to high dropped-call probabilities, and can impair co-channel calls.

Mobile location registration may preferably be retained in the home exchange memory. Each base station broadcasts its unique identification code via the control channel. The mobile stations in the area keep this code in memory. When a new identification code is received by the mobile station, it can autonomously identify itself to the new serving area via a base station in that area. This information can be communicated to the home exchange (see Recommendation 624).

6. Intermodulation interference considerations

In a high capacity mobile telephone system using cells, it is necessary that intermodulation interference be minimized. However, because of the orderly spatial organization and the hand-off capability of cellular systems and as a result of new state-of-the-art transmitter/antenna coupling and combining arrangements used in cellular systems, intra-system intermodulation interference is considered to be controllable.

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REPORT 1156

DIGITAL CELLULAR PUBLIC LAND MOBILE TELECOMMUNICATION SYSTEMS (DCPLMTS)

(Study Programme 39A/8)

(1990)

1. Introduction

Digital cellular public land mobile telecommunication systems (DCPLMTS) are defined to be land mobile systems for voice and data services using digital cellular radio technology for the connection to the public switched telephone network (PSTN), integrated services digital network (ISDN) and public data networks (PDNs).

Part A of this report deals with the general principles of DCPLMTS and, in particular, addresses the rationale for digital cellular systems design. This part also provides a basic comparison with existing analogue cellular systems and discusses some architectural features of these systems.

The major characteristics of some DCPLMTS currently under development are presented in Part B of this report, together with a brief description of the current status of these systems and any unique features of their design.

PART A

GENERAL PRINCIPLES OF DCPLMTS

1. Introduction

Several DCPLMTS are currently under development. A Pan-European system called GSM is now under validation. The specification for a North American system is scheduled for completion by year end 1989. A Japanese system is also at an advanced stage of design. These systems are being designed for international operation and are expected to be widely implemented during the 1990s. In comparison, the on-going studies (Decision 69) necessary to specify future public land mobile telecommunication systems (FPLMTS) are not yet completed. It is expected that the FPLMTS will complement the DCPLMTS in the future. Thus it is timely to develop a report describing the key features of the emerging digital cellular public land mobile telecommunication systems.

2. General objectives

The general objectives of DCPLMTS are to provide:

- systems with high spectrum utilization efficiency, thereby accommodating more users within the limited spectrum resource than existing analogue cellular public land mobile telecommunication system (PLMTS);
- users with a wide range of services and facilities, both voice and non-voice, that are compatible with, and access, those offered by the public fixed networks (PSTN, ISDN, PDN, etc);
- services and facilities exclusive to mobile systems, including facilities for automatic roaming, locating and updating mobile users;
- users with a variety of mobile stations consistent with their requirements, ranging from vehicle mounted to hand-held stations with voice and non-voice interfaces;
- services of high quality and integrity at an economic cost;
- mobile equipment and infrastructure at the reduced cost, weight, size and power drain offered by the adoption of digital processing and VLSI technology.

3. Cellular mobile systems

3.1 Cellular systems background

Cellular systems divide the desired service area into a number of sub-areas (cells) with one or more base stations providing radio coverage to each cell. Appropriate frequency planning permits frequencies to be used several times in a coverage area.

When a mobile station either originates or is prepared to receive a call, the cell which can best serve the station is selected. With proper spacing of co-channels, each frequency may be used simultaneously in several cells in the service area and thereby multiplying the capacity obtainable in non-cellular radio systems.

The general features of a cellular system are:

- accommodation of more than one base station in a service area and thus the expansion of the service area of the system beyond the coverage that a single site could provide;
- "roaming" or automatic registration of mobile stations in appropriate location areas;
- "handover" or transfer of the responsibility for radio contact with a mobile station from one base station to another. This may also involve a transfer from one frequency or time slot to another;

- reuse of the same RF resource simultaneously by more than one base station and for more than one message. This is the basis of the spectral utilization efficiency of cellular systems;
- growth; that is, the system must be able to start with a few large cells and accommodate traffic growth by creating smaller cells at the points of highest traffic density.

These features are the basis for cellular systems. A more complete description of the general and specific aspects of cellular systems can be found in Reports 740 and 742.

3.2 Digital cellular systems

The cellular systems described in the previous section have been very successful. This success has led to an increasing number of users and to an increasing demand for new and improved services. Since the introduction of the cellular systems, there have also been changes in the telecommunications network. The most notable of these are the services introduced by ISDN. These forces have led to the development of new, digital PLMTS systems.

3.2.1 Digital technology

The digital technology is introduced into the PLMTS in six major areas:

- digital radio modulation/demodulation;
- time division multiple access (TDMA);
- digital speech coding;
- channel coding and digital signal processing;
- digital control and data channels;
- privacy and authentication.

3.2.1.1 Digital radio modulation/demodulation

The digital radio modulation/demodulation makes efficient use of the spectrum and enables operation at lower carrier to interference (C/I) ratios than is possible with analogue systems. This enables the DCPLMTS to accommodate a wide range of cell sizes tailored to suit the local traffic conditions. The use of digital techniques also allows implementation of much of the circuitry using very large scale integration (VLSI) processes and thus reduces costs (particularly of mobile stations). The new modulation techniques, which enable operation at lower C/I ratios, also permit the use of improved frequency re-use patterns and cell sectoring to further increase system efficiency.

3.2.1.2 TDMA

The introduction of TDMA results in much improved system signalling, operation and cost.

A mobile station can exchange system control signals with the base station without interruption of speech (or data) transmission. This facilitates the introduction of new network and user services. The mobile station can also

check the signal level from nearby cells by momentarily switching to a new time-slot and radio channel. This enables the mobile station to assist with handover operations and thereby improve the continuity of service in response to motion or signal fading conditions. The availability of signal strength information at both the base and mobile stations, together with suitable algorithms in the station controllers, allows further spectrum efficiency through the use of dynamic channel assignment and power control.

The cost of the base stations using TDMA can be reduced as radio equipment is shared by several traffic channels. A reduced number of transceivers will lead to a reduction of multiplexer complexity. Outside the major metropolitan areas, the required traffic capacity for a base station may, in many cases, be served by one (or two) transceivers. The saving in the number of transceivers results in a significantly reduced cost.

A further advantage of TDMA is increased system flexibility. Different voice and non-voice services may be assigned a number of time-slots appropriate to the service. For example, as more efficient speech codecs are perfected, increased capacity may be achieved by the assignment of a reduced number of time slots for voice traffic. TDMA also facilitates the introduction of digital data and signalling services as well as the possible later introduction of such further capacity improvements as digital speech interpolation (DSI).

3.2.1.3 Digital speech coding

The advanced digital coding of speech (at rates less than 16 kbit/s) contributes to system efficiency. This coding also provides potential for capacity growth through the use of "half-rate" codecs (when they are perfected). The digital coding of the speech also allows powerful error correction and detection techniques to be used. This contributes significantly to improved speech quality and to satisfactory operation at low C/I ratios.

3.2.1.4 Channel coding and digital signal processing

In cellular systems the use of adequate digital error control and signal processing techniques can contribute to system spectral efficiency. The error control allows operation at low C/I ratios and in high noise radio environments. Digital signal processing techniques also facilitate the use of adaptive channel equalization processes to compensate for dispersion in the radio channel. Digital signal processing also permits the use of diversity, frequency hopping, and interleaving techniques to improve operation in a signal fading environment.

3.2.1.5 Digital control and data channels

The digital control and data channels are the key to the flexibility of the new systems and to the introduction of new services. Digital data may be transmitted at high rates to the mobile stations with improved error control and protection. The digital control channel provides an improved facility for the introduction of network services such as message services and simultaneous voice and data communications. The digital control channels also facilitate the introduction of ISDN services into the PLMTS.

3.2.1.6 Privacy and authentication

The combination of the digital coding of the speech and the digital control channels facilitates provision of effective privacy and authentication. The digital speech may be easily protected through the use of digital privacy

coding algorithms. The control channel facilitates the proper distribution of privacy keys. Through the control channel, other system resources (such as the home location register (HLR) and the authentication centre (AUC)) can provide the positive authentication of the mobile users. Authentication assures the accuracy of billing information and facilitates the roaming of users throughout a wide geographic area and between networks. The positive authentication of the mobile user by the authentication centre can also facilitate the introduction of new services which would otherwise not be practical. (This could include, for example, access to account records, or stored message retrieval.)

3.2.2 Practical concerns

The introduction of digital technology is not without some penalties however.

It must be noted that the TDMA technique with interleaving and low bit rate digital speech coding will introduce significant delay into the speech path. Careful concern for echo control will be necessary when the digital mobile systems are connected to the PSTN/ISDN.

The digital system must also be concerned with time synchronization in a dispersive radio channel. This will be of particular concern if the digital radio signals have a wide bandwidth.

The TDMA technique requires higher peak to average power ratio. This has to be taken into account when estimating battery life and possible radiation effects.

4. Comparison with existing designs

The DCPLMTS build upon some of the technology and system design concepts already in use by analogue cellular systems. However, several new features are consequential upon the use of digital techniques and the time division multiplexing method.

4.1 Commonalities

- The RF band used (around 800 - 1 000 MHz) has remained the same as for many analogue cellular systems, leaving the basic propagation statistics (path loss, delay spread, Rayleigh distributed envelope, etc.) unchanged;
- the basic cellular frequency reuse opportunities remain similar, since the standard deviation of path loss is unchanged, as are the outage design goals;
- the network architecture, as dictated by a number of factors, is essentially unchanged. Other constant network attributes are: numbering plan, roaming techniques, and charging methods.

4.2 Differences

- The digital modulation yields potential design economies and lower C/I requirements relative to earlier techniques;
- the digital modulation also offers enhanced ISDN compatibility;

- the time division multiplexing (TDM) offers further design flexibilities, including mobile-assisted handover and commonality between bearer information and control information formats;
- the channel coding adds another dimension of design control;
- for optimum performance of a digital channel, adaptive equalization is required;
- the TDM, interleaving and digital speech coding introduce appreciable (50-100 ms) delay, which necessitates echo control as long as 2-wire land-line circuits are encountered;
- the digital speech coding changes the way the system sounds when impairments are encountered, relative to a companded analogue system. The peak quality of speech in the absence of impairments is an intrinsic characteristic of each system;
- in a digital system, enhanced privacy is relatively simple to implement.

5. Services

5.1 General

The development of the DCPLMTS is driven by the increasing world-wide demand by mobile subscribers for voice and non-voice services. The current emphasis is more towards the provision of enhanced capacity for voice service, but there is a growing demand for non-voice services. These cannot be conveniently or efficiently supported on existing analogue systems. The DCPLMTS will offer a much greater variety of enhanced services.

In parallel with the need for enhanced capacity and services, there is a growing demand for improved quality of service. The DCPLMTS, through the use of such techniques as advanced modulation and coding, can offer improved voice and non-voice service quality.

5.2 Service types

The basic telecommunication services offered by the DCPLMTS can be divided into two types:

- bearer services which give the user the capacity needed to transmit appropriate signals between certain access points;
- teleservices which provide the user with the full capacity, including terminal equipment functions, to communicate with other users.

Supplementary services are also available in association with the basic services.

The services supported by the DCPLMTS in each of these categories are related to those offered by the ISDN, but are confined to lower bit rate channels (typically less than 16 kbit/s) by the limitations of the radio channel. All the DCPLMTS support some services in each category, but the range offered varies between systems.

5.3 Bearer services

Typical bearer services offered include:

- synchronous, asynchronous and packet data at rates up to a maximum of 9.6 kbit/s;
- unrestricted digital capability at specific bit rates (generally less than 16 kbit/s).

In general, connection of voice band modems to the speech path of mobile stations is not supported. Equivalent service to that offered by the use of voice band modems on the PSTN or ISDN can be provided via the bearer services listed above.

5.4 Teleservices

All the DCPLMTS support telephony and facsimile teleservices. Some extend the teleservice offerings to include Videotex, Teletex, etc.

5.5 Supplementary services

The range of supplementary services supported by the DCPLMTS varies depending on the system and also the particular implementation.

6. Architecture common to all digital systems

6.1 Base station layout

The geographical distribution of base stations is organized around two types of structure:

- regular cell structures using omnidirectional antennas and,
- sector cell structures using directional antennas.

Each of these techniques is already used in analogue cellular systems. More details on cellular base station layout can be found in Report 740.

6.2 Channel design

Two basic categories of channel are defined for DCPLMTS:

- traffic channels (TCH) which are used for voice and data transmission (i.e. bearer services and teleservices);
- control channels (CCH) which are used for signalling and control purposes, including handover.

The CCH are further divided into three broad types:

- common control channels (CCCH) which are used for paging, random access, etc;
- broadcast control channels (BCCH) which are used for broadcast messages, and/or synchronization and frequency correction;

- associated control channels (ACCH) which can be divided into slow ACCH (SACCH) and fast ACCH (FACCH) and provide control and signalling functions for individual users.

Some systems may also define other types of control channel for particular applications (e.g. stand-alone dedicated control channels).

The basic terminologies for some of these control channels can be found in the CCITT Q.1000-Series of Recommendations.

6.3 Network architecture and assignment of functions

Figure 1 shows the basic system architecture for a DCPLMTS, including the major functional components. The communication protocols are specified according to the 7-layer OSI model, while the interfaces between mobile switching centres (MSCs) and the interfaces to the ISDN, PSTN and PDN are all specified according to CCITT Recommendations. The numbering plan also follows CCITT Recommendations.

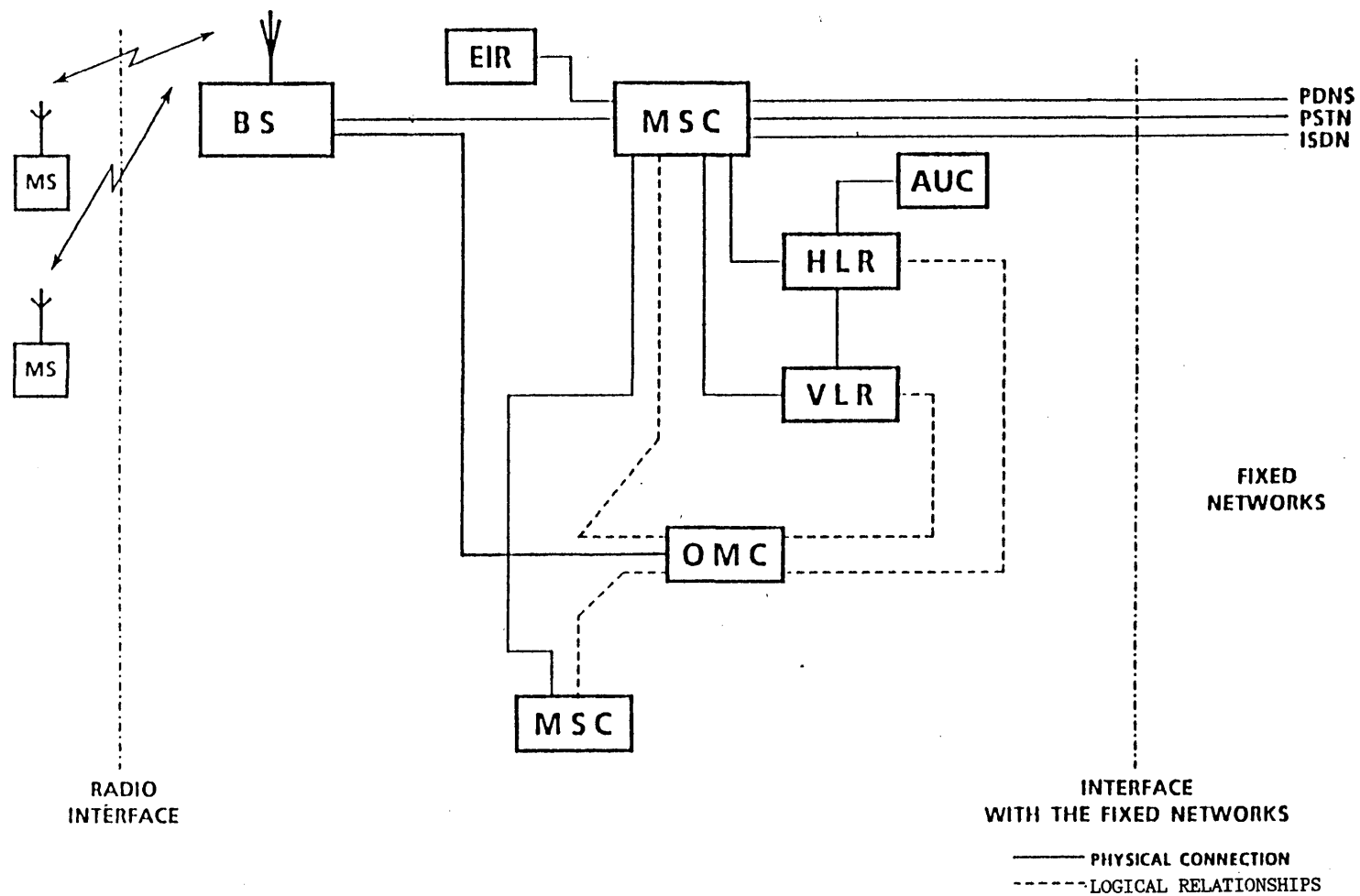


FIGURE 1 - Network architecture

MS : MOBILE STATION
 BS : BASE STATION
 MSC : MOBILE SERVICES SWITCHING CENTRE
 HLR : HOME LOCATION REGISTER
 VLR : VISITOR LOCATION REGISTER
 OMC : OPERATION AND MAINTENANCE CENTRE
 EIR : EQUIPMENT IDENTITY REGISTER
 AUC : AUTHENTICATION CENTRE

PART B

DCPLMTS SYSTEMS BEING INSTALLED OR PLANNED

1. Introduction

High capacity DCPLMTS are being developed in Europe, North America and Japan. Each of these systems has the basic objectives and characteristics outlined in Part A of this report. However, each is being developed with a slightly different focus and with different constraints. This has produced three different system specifications. Each of these systems is introduced below and more detailed descriptions can be found in Annexes I through III.

1.1 GSM

The development and specification of this Pan-European system in the 900 MHz band has been carried out through cooperation among more than 16 European countries. The system will be operated by one or more operators in each country from the beginning of the nineties.

The international aspects of the GSM system were taken into account from the very beginning, making GSM a true international public land mobile network (PLMN). For example, GSM supports roaming between countries and full access to the international PSTN, ISDN and PDNs.

1.2 North America

In North America, a digital cellular standard has been developed. This standard has been designed to meet the requirements to introduce new digital services and to provide additional traffic capacity while maintaining compatibility with the existing advanced mobile phone service (AMPS) analogue standard. The new digital standard will be used in North America, and a similar design can be used by over forty countries that have adopted, installed or specified AMPS or its derivatives (TACS).

The dual mode digital and analogue North America Cellular Standard for PLMTS has evolved from the existing AMPS standard to provide both voice and data services with a ten-times traffic capacity improvement as the goal. The North America Digital Standard supports international roaming and full access to the PSTN, ISDN and PDN.

1.3 Japan

Due to the rapid increase in the number of mobile subscribers in Japan, the capacity of the existing analogue cellular system is approaching saturation. To accommodate more subscribers and provide a greater variety of services, digital cellular technology offers promising opportunities. The Japanese Digital Cellular standard provides new digitized traffic and control channels. The standard is applicable to both the 800/900 MHz and 1.5 GHz bands.

2. Explanation of table

Annexes I, II and III give details of digital cellular systems emanating from Europe, North America and Japan respectively. Table I presents the core parameters for these three systems. In each case complete specifications are, or will be, available from the relevant authorities as indicated in the annexes.

TABLE I

Core parameters

FEATURE	GSM	North America	Japan
Class of emission			
- traffic channels	271KF7W	40KOG7WDT	tbd.
- control channels	271KF7W	40KOG1D	tbd.
Transmit frequency bands (MHz).			
- base stations	935 - 960	869 - 894	810 - 830 (1.5GHz tbd.)
- mobile stations	890 - 915	824 - 849	940 - 960 (1.5GHz tbd.)
Duplex Separation (MHz)	45	45	130 48 (1.5GHz)
RF Carrier spacing (kHz)	200	30	25 interleaved 50
Total number of RF duplex channels	124	832	tbd
Maximum base station erp (W)			
- peak RF carrier	300	300	tbd
- traffic channel average	37.5	100	tbd
Nominal mobile station transmit power (W)	20 - 2.5	9 - 3	tbd
peak - average	8 - 1.0	4.8 - 1.6	
	5 - 0.625	1.8 - 0.6	
	2 - 0.25	tbd - tbd	
Cell radius (km)			
- min	0.5	0.5	0.5
- max	35 (up to 120)	20	20
Access method	TDMA	TDMA	TDMA
-traffic channels/RF carrier			
-initial	8	3	3
-design capability	16	6	6

Note.- tbd. - to be defined

Table I (continued)

Modulation	GMSK (BT = 0.3)	$\pi/4$ diff. encoded QPSK (roll off=0.25)	$\pi/4$ diff. encoded QPSK (roll off=0.5)
-transmission rate (kbit/s)	270.833	48.6	37 - 42
Traffic channel structure			
- full rate speech codec			
- bit rate (kbit/s)	13.0	8	6.5 - 9.6
-error protection	9.8kbit/s FEC + speech processing	5 kbit/s FEC	~3 kbit/s FEC
-coding algorithm	RPE-LTP	CELP	tbd
- half rate speech codec			
-initial	tbd	tbd	tbd
-future	yes	yes	yes
-data			
-initial net rate (kbit/s)	up to 9.6	2.4, 4.8, 9.6	1.2, 2.4, 4.8
-other rates (kbit/s)	up to 12	tbd	8 and higher
Channel coding	Rate one half convolutional code with interleaving plus error detection	Rate one half convolutional code	tbd
Control channel structure			
-common control channel	yes (3)	shared with AMPS	yes
-associated control channel	Fast & slow	Fast & slow	Fast & slow
-broadcast control channel	yes (3)	yes	yes
Delay spread equalisation capability (us)	20	60	tbd

Note.- tbd. - to be defined
diff. - differentially

Table I (continued)

Handover			
- mobile assisted	yes	yes	yes
- inter system capability with existing analogue system	no	between digital and AMPS	no
International roaming capability	yes >16 countries	yes	yes
Design capability for multiple system operators in same area	yes	yes	yes

ANNEX I

GENERAL DESCRIPTION OF THE GSM SYSTEM

1. Introduction

The characteristics of the GSM system that are common to most of the digital cellular systems can be found in Part A of this report.

Therefore this annex highlights only original aspects of the GSM system, and, in fact, only parts of them.

The driving force of the GSM has been its international lay out based on a common availability of virtually "clear" frequency bands. This situation offered a unique opportunity of optimizing the usage of new technologies and therefore spectrum efficiency with a rather limited number of constraints. A very advanced radio design was therefore possible.

Full detailed information on the specifications of the GSM system is given in the reference list.

2. Services

In the process of drafting the GSM standard, the details of the implementation of each particular service together with the required interworking mechanisms have been specified in order to offer full access to the services while roaming, and to minimize the complexity of the mobile station.

2.1 Bearer services

The bearer services offered by the GSM PLMN include transparent and non-transparent data services for circuit as well as packet mode, up to a net data rate of 12 kbits/s.

2.2 Teleservices

Among the main teleservices supported by the GSM are:

- speech i.e. telephony and emergency calls;
- short message service;
- data message handling system access;
- Videotex;
- facsimile.

2.3 Supplementary services

The supplementary services offered by GSM operators can be divided into four main groups:

- call forwarding;
- call completion;

- advice of charge;
- call restriction.

2.4 Security aspects

Further to the provision of a wide range of services, the GSM system has also been designed to ensure a high level of security. Therefore security features are provided to protect the access to the services and the privacy of user-related information. The following security features are implemented in the GSM system:

- subscriber identity confidentiality: it ensures that the mobile subscriber identity (IMSI) cannot be disclosed;
- subscriber identity authentication: it verifies that the subscriber identity sent by the mobile is the one claimed (not duplicated or impersonated);
- user data confidentiality: it ensures that the user data, including speech, transferred on the radiopath cannot be disclosed by unauthorized bodies;
- signalling information element confidentiality: it is the property that a given piece of signalling information (subscriber and equipment identities, directory numbers, etc.) exchanged on the radiopath cannot be used by unauthorized individuals or entities.

The IMSI is the information which uniquely identifies the subscriber, and that has to be present and valid to allow the operation of the mobile station.

Each mobile station has a unique identity that shall be implemented by the manufacturer: the international mobile equipment identity (IMEI).

The security functions for authentication of the subscriber related information, and all processes involving the authentication key are contained in a removable piece of the mobile station called the subscriber identity module (SIM).

3. Overview of the system

The GSM system has been standardized by administrations, operators and manufacturers of over 16 European countries in order to provide full service access to international roamers. The standard of the GSM is described in terms of interfaces and functional entities.

Two interfaces are mandatory: the radio interface (Um) and the interface "A" between the mobile services switching centre (MSC) and the base station system (BSS). A further interface "A bis" within the BSS system is being specified but its implementation is not mandatory.

The functional architecture is given in Figure 2. It shows:

- the MSC, the home location register (HLR), and the visitor location register (VLR), where the networking and switching functions are performed;

- the BSS which includes the base station controller (BSC) and the base station transceivers;
- the operations and maintenance centre (OMC);
- the mobile station (MS).

The MAP is the mobile application part of CCITT Signalling System No. 7 which has been specified to allow the routing of calls to MS which have roamed to different MSC areas or to different networks.

The MSC, HLR, and VLR execute interworking with partner networks, call control, and encryption of signalling and user speech and data. These functions also include authentication of the mobile user, location updating as roaming occurs, and paging of the mobile to indicate incoming calls.

The BSS performs the radio channel management functions which include administration of the radio channel configurations, allocations of radio channels and link supervision, scheduling of messages on broadcast channels, choice of frequency hopping sequences whenever needed, and power control.

4. Technical radio characteristics

These characteristics are specified in GSM Recommendation Series 05 and 06.

4.1 RF equipment requirements

In accordance with GSM Recommendation 05.05 (see reference list).

4.2 Carrier spacing

200 kHz carrier spacing yields at least 18 dB adjacent RF channel selectivity within the system. Second adjacent RF channel at 400 kHz spacing yields at least 50 dB selectivity within the system. The corresponding third adjacent RF channel selectivity yields at least 58 dB.

Frequency hopping is a possible feature.

4.3 Class of emission

271KF7W according to Radio Regulations RR4, i.e. Gaussian minimum shift keying GMSK(BT = 0.3) with modulation rate 270.83 kbit/s per carrier using time division multiple access (TDMA) scheme for 8 basic physical channels.

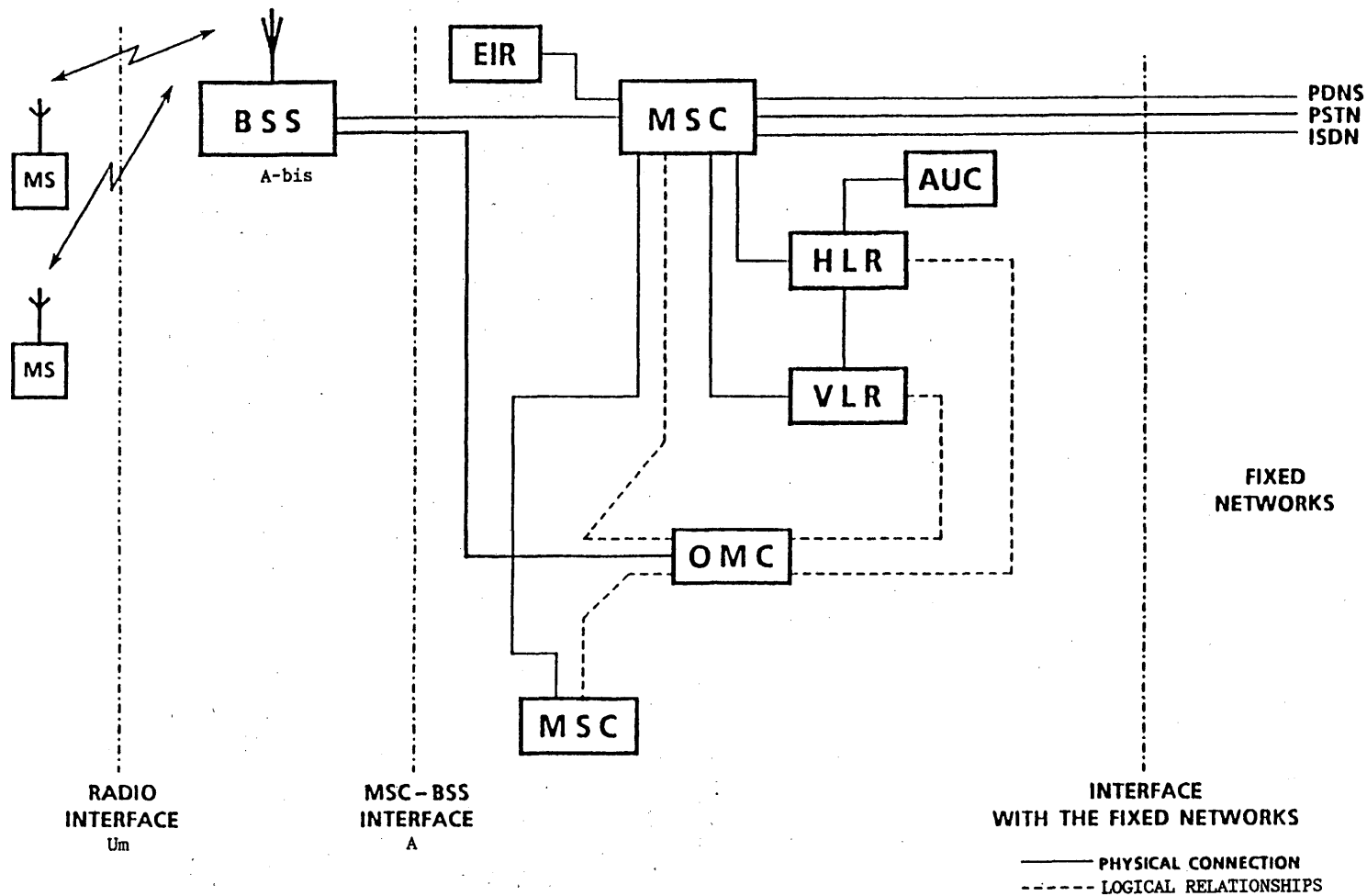


FIGURE 2 : - GSM 01.02 System architecture

MS : MOBILE STATION
 BSS : BASE STATION SYSTEM
 MSC : MOBILE SERVICES SWITCHING CENTRE
 HLR : HOME LOCATION REGISTER
 VLR : VISITOR LOCATION REGISTER
 OMC : OPERATION AND MAINTENANCE CENTRE
 EIR : EQUIPMENT IDENTITY REGISTER
 AUC : AUTHENTICATION CENTRE

4.4 Cell structure and carrier reuse

It is possible to use large cells (up to 35 km base - mobile distance) in rural areas as well as small cells (down to 1 km diameter) in urban areas.

Extended cell operation ranging up to 120 km base-mobile distance is also possible.

In high traffic density peak areas (e.g. city centres) it is possible to build a sector cell structure using directional antennas with a channel concentration at the traffic peak area.

Co-channel protection ratio down to $C/I = 9$ dB is acceptable by the system and yields a possible reuse corresponding to a 9-cell cluster. (3 cell reuse patterns with three sectors per cell.)

The sensitivity, similar to that of existing analogue systems, allows an average power in transmitters about 9 dB lower than in current analogue systems, given the same requirements for maximum cell sizes and the same RF device choices.

4.5 Time-slots and TDMA frames

A burst containing 148 bits, corresponding to 114 coded bits, is sent within a time-slot duration of 0.577 ms. A set of 8 time-slots is used to build up a TDMA frame containing 8 basic physical channels. On each physical channel logical channels are mapped i.e. traffic channels and control channels.

The useful information is distributed in the time-slots in a manner allowing for recovery from total erasure of some time-slots.

Two multiframe structures are defined: one consisting of 26 TDMA frames (recurrence interval of 120 ms) for traffic channels and their associated control channels, and one for the other control channels comprising 51 TDMA frames (recurrence interval of 236 ms).

4.6 Traffic channels

4.6.1 Full and half rate traffic channels

The system is able to support both full and half rate traffic channels, corresponding respectively to the gross bit rates of 22.8 and 11.4 kbit/s. The half rate channel is obtained by the use of only half of the time-slots used by the full rate channel. A carrier therefore provides up to 8 full rate or 16 half rate traffic channels (or a combination of both) with their respective associated control channels.

4.6.2 Speech traffic channels

The full rate speech codec, and the associated error correction and detection mechanisms have been defined (see Report 903). Speech frames of 20 ms, comprising each 260 bits, provide a net bit rate of 13 kbit/s. The coding method, "Regular pulse excited linear prediction coding with long term prediction (RPE-LTP)", has been designed to achieve robustness to transmission errors, and to offer a quality close to that of PSTN, while using a limited bit rate.

Error correction (consisting of a 1/2 rate convolutional code) and interleaving schemes, to selectively protect the most important bits within the speech frame (70% of the bits) have been specified. Furthermore an error detection mechanism has been included, associated with extrapolation techniques which have been described and/or recommended, in order to minimize the impairment of the speech quality resulting of speech frames not correctly received. The usage of speech activity detectors has also been specified in the GSM system. Details can be found in [de Brito and Natvig, 1988] for the speech coding aspects and in [Maloberti, 1989] for the transmission aspects.

4.6.3 Data traffic channels

Transparent and non-transparent data services of up to 9.6 kbit/s are supported by different rate adaptations, channel coding and interleaving schemes, on full rate and/or half rate channels.

Unrestricted digital bearer services with a net bit rate of 12 kbit/s are also supported.

4.6.4 Discontinuous transmission

All traffic channels may use, when possible, discontinuous transmission (i.e. transmitter is silent when no relevant information is to be transmitted). In the case of speech this is possible due to the specification of speech activity detectors.

This feature, combined with frequency hopping which introduces interferer diversity, is expected to increase the system capacity. It will also prolong battery life in handheld-portable stations.

4.7 Control channels

Three categories of control channels are defined: broadcast, common and dedicated.

4.7.1 Broadcast channels

Broadcast channels are divided into frequency correction, synchronisation and broadcast control channels.

4.7.2 Common control channels

Common control channels are divided into paging, random access and access grant channels.

4.7.3 Dedicated control channels

Dedicated control channels are divided into slow and fast associated control channels, as well as stand-alone dedicated control channels with their associated control channels. Under this category also a cell broadcast channel is defined to carry short message service cell broadcast.

Short message, service mobile terminated and mobile originated point-to-point, is supported by the stand-alone dedicated control channel or the slow associated control channels.

5. Operational characteristics

5.1 Cell selection

Whilst in idle mode the mobile station is camped on a cell from which it can reliably decode downlink data, and with which it has a high probability of communicating on the uplink.

The condition to perform cell selection is based on path loss criteria. If these criteria are not met or the mobile station fails to decode paging blocks or fails to make access on the uplink, the mobile station starts to re-select.

5.2 Location updating (roaming)

Roaming is performed in accordance with Recommendation 624.

The mobile station evaluates the received signal and initiates the location updating procedure when necessary.

Roaming is possible between Mobile Service Switching Centres (MSC) and between countries.

5.3 Communication protocols

The communication protocols are layered according to the OSI model and are specified in the GSM Recommendations.

The network layer is divided into three sub-layers: call control, mobile management and radio resource management.

The link layer is based on LapD protocols and makes use of the control channels. Messages between link layer peer entities are source coded into 23 octets, i.e. 184 bits.

5.4 Call setup

5.4.1 Mobile originated call setup

The procedure starts on the random access channel to set up a radio resource. Then authentication is done on the mobile management sub-layer. After ciphering and assignment has been confirmed, the call setup is confirmed on the call control sub-layer.

5.4.2 Mobile terminated call setup

After paging from the network the same procedure follows as in 5.4.1 above.

5.5 Handover

Handover is required to maintain a call in progress as a mobile passes from one cell coverage area to another and may also be employed to meet network management requirements, i.e. relief of congestion (network directed handover).

The handover is done either from a channel on one cell to another channel on another cell, or between channels of the same cell.

The handover strategy employed by the network for radio link control determines the handover decision that will be made based on the measurement results reported by the mobile and base stations and various parameters set for each cell. The exact handover strategies are determined by the network operator.

A procedure is implemented in the mobile station by which it monitors the downlink signal level and quality from its serving cell and the downlink signal level and colour code of surrounding cells.

A procedure is implemented in the base station by which it monitors the uplink signal level and quality from each mobile station being served by the cell.

These radio link measurements are also used for RF power control.

Handover is possible between location areas and between different MSC:s belonging to the same PLMN.

5.6 Radio link failure

The criteria for determining radio link failure are specified in order to ensure that calls which fail either from loss of radio coverage or unacceptable interference are satisfactorily handled by the network. Radio link failure results in either call re-establishment or release of the call in progress.

The criterion for determining radio link failure in the mobile station is based on the success rate of decoding messages on the downlink slow associated control channel.

5.7 Signalling between base station and MSC

The signalling follows a layered approach similar to ISDN in accordance with GSM Recommendations.

5.8 ISDN, PDN and PSTN interfaces

In accordance with CCITT Recommendation Q.700- and Q.1000-series.

5.9 Numbering plan

In accordance with CCITT Recommendations E.164, E.212 and E.213.

5.10 Signalling between MSC:s

CCITT signalling system No. 7 (Recommendations E.214, Q.700-series and GSM 09.02 or CCITT Q.1051).

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ANNEX II

GENERAL DESCRIPTION OF THE NORTH AMERICAN DIGITAL CELLULAR
PUBLIC LAND MOBILE TELECOMMUNICATIONS SYSTEM1. General description1.1 Objectives

The North America digital cellular system for public land mobile telecommunications system (PLMTS) is designed to provide digital voice and data services and to meet a significantly growing capacity requirement. The standard is suitable for new systems and is also compatible with the existing advanced mobile phone service (AMPS) system. RF carrier spacing is compatible with the AMPS spacing and thus both types can co-exist in the same radio environment.

1.2 Evolution strategy

Because of the compatibility of the RF signals, the system provides operators with a smooth evolutionary path for the introduction of digital services and additional traffic capacity to their existing PLMTS. The digital standard can be incorporated into existing networks to allow both digital and analogue traffic. Users with dual mode terminals can receive service from operators who have added a digital capability, and from those operators who have only analogue facilities. Operators need not install digital equipment until their traffic growth warrants additional facilities. Because the digital and analogue traffic share the same band and control channel, the dual mode terminals can economically share many radio and control components. Section 2 of this document outlines the technical features of the system. As the new standard is compatible with the existing AMPS system, only the digital features are highlighted here. For a further description of the AMPS refer to Report 742..

1.3 Capacity improvement

The system uses a linear modulation technique and low bit rate speech coding to achieve high spectral efficiency. This combination enables each 30 kHz radio frequency carrier to handle three traffic channels. Six traffic channels can be accommodated in the future with half-rate codecs. The linear modulation technique also operates at a lower carrier-to-interference ratio than the present analogue system. Capacity can be further increased by the use of improved frequency reuse patterns, cell sectoring and planning techniques. Together, these techniques provide greatly increased traffic capacity as Table II indicates.

The following assumptions were used in the preparation of the table:

- a) a total of 25 MHz is available to the operator,
- b) a 2% blocking probability is acceptable,
- c) a lower C/I threshold is used for the digital system, and
- d) the same ratio of traffic to signalling channels is used for both the analogue and digital systems.

TABLE II

Capacity improvements in the North American digital cellular system over the present analogue system

Evolution stage				Number of TCH	Erlang per Cell	Improve. Factor Relative to (7x3)*	Improve. Factor Relative to (4x6)*
Access Technique	TCH Per RFC	Cell Reuse Pattern	Sector Per Cell				
FDMA	1	7	3	395	37	1.0	
FDMA	1	4	6	395	60	1.6	1.0
TDMA	3	7	3	1185	138	3.7	
TDMA	3	4	6	1185	236	6.4	3.9
TDMA	6	7	3	2370	302	8.7	
TDMA	6	4	6	2370	522	14.1	8.7
TDMA	6	4	3	2370	550	14.9	9.2
TDMA	6	3	3	2370	746	21.3	12.4

TCH - Traffic channels, RFC - Radio frequency carrier

* (mxn) m-cell reuse pattern with n-sectors

2. Technical outline

2.1 RF interface requirements

— Modulation: $\pi/4$ differentially encoded QPSK.

The information is differentially encoded; symbols are transmitted as changes in phase rather than absolute phases.

Gray code is used in the symbol mapping; two di-bit symbols corresponding to adjacent signal phases differ only in a single bit.

2.2 Typical cell radius: 0.5 - 20 km.

It is possible to have three and six sector cells using directional antennas with both digital and analog channels for rural and urban areas.

2.3 Channel coding

- Rate one-half convolutional code,
- two levels of error protection,

- cyclic redundancy code (CRC) to protect the most important bits for speech.

2.4 Timeslots and TDMA frames:

Timeslots are arranged in the following order: transmit-receive-idle. The idle timeslots may be used for measuring and channel monitoring purposes.

The numbering scheme is one through six. Currently, six synchronization words are used to allow up to six users to share one TDMA channel.

Provisions are made to allow other conditions of transmission under base station control to support more users and different services on one TDMA channel.

2.5 Traffic Channel Structure:

- Speech: supports full rate and half rate speech codecs.

For full rate speech codec, 13 kbit/s is allocated to speech coding and forward correction. It is possible to support different speech codecs of different rates at the base station.

- Data: provides data services of 2.4, 4.8, 9.6 kbit/s [1] and [2]. Higher rates also will be specified.

2.6 Control Channel Structure:

- Common control channel:

AMPS common control channel [3] will be shared by both digital and analog mobile stations.

Additional messages for mobile stations in digital mode will be handled in the digital traffic channels to support higher dual mode cellular system capacity.

Digital control channel also will be specified.

- Fast and slow control channels: Dedicated in each channel for each user.

The supervisory and other control can be handled by fast and slow associated control channels.

The digital voice color code will be a separate field to provide the function equivalent to the analog Supervisory Audio Tone (SAT) to detect interference.

2.7 Handover

- Intersystem and intrasystem handovers are specified.
- Handovers between digital and analogue channels are possible.
- Mobile Assisted Handover:

It provides the ability for the mobile stations in the digital mode to measure and report both the received signal strength and channel quality over the current digital connection as well as the received signal strength on other channels as requested by the base station.

2.8 System Architecture:

— Communication protocol:

The network communication protocol reference model is designed according to the OSI model.

— Interfaces [4]:

The interfaces between system functional blocks are modeled according to CCITT Q.1000-series.

2.9 Intersystem Cellular Networking:

— Intersystem handover [5]:

The minimum message set provides intersystem location requests, measurements and response. The trunk selection and call connection sequences are specified IS-41-2.

— Intersystem automatic roaming:

Functions and messages of the Home Location Register (HLR) and Visitor Location Register (VLR) are specified based on the CCITT Recommendation.

The data base access procedures for roaming validation, special feature treatment and call delivery also are defined.

— Intersystem data communications:

Using the protocol of CCITT Recommendation X.25 or Message Transfer Part used with CCITT Signalling System No 7 protocol [5].

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[4] "Definition of Interface Architecture: U.S. Second Generation Cellular System", EIA/TIA TR45.3.2/88.08-1.

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ANNEX III

GENERAL DESCRIPTION OF THE JAPANESE DIGITAL CELLULAR
PUBLIC LAND MOBILE TELECOMMUNICATION SYSTEM1. Introduction

The Japanese Digital Cellular PLMTS is specified to provide various services and to accommodate a great number of subscribers.

The system is applicable to both the 800/900 MHz and the 1.5 GHz bands and supports data, facsimile and ISDN services. To realize efficient frequency utilization, the RF carrier spacing is 25 kHz in accordance with the existing analogue standard.

2. Main features

2.1 RF interface requirements:

- Channel spacing : 25 kHz interleaved channel spacing, 50 kHz channel spacing.
- Modulation : $\pi/4$ differentially encoded QPSK (Roll-off factor;0.5).
- Access method : TDMA;
 - 3 timeslots/25 kHz (for full rate),
 - 6 timeslots/25 kHz (for half rate).
- Transmission bit rate: ranging from 37 to 42 kbit/s will be specified.

2.2 Cell structure and carrier reuse:

- Typical cell radius : 0.5 - 20 km.
- Sector cell structure using directional antennas.

2.3 Timeslots:

- 3 for full rate, 6 for half rate.

2.4 Traffic channels:

- Speech:supports full rate and half rate speech codecs.
 - Full rate speech codecs ranging from 6.5 to 9.6 kbit/s are being tested.
 - Ranging from 9.6 to 12 kbit/s are allocated to speech coding and forward error correction.
 - Half rate speech codec will also be specified.
- Data and other services:
 - Data services of 1.2, 2.4, 4.8 kbit/s.
Higher rates will also be specified.
 - Facsimile.
 - ISDN Sub-rate (8 kbit/s).

2.5 Control channels:

- Broadcast control channels (BCCH) : Control channels for broadcast.
- Common control channels (CCCH) : Control channels for signaling, such as paging.
- Associated control channels (ACCH): Slow ACCH and Fast ACCH.

2.6 Cell selection:

- While in idle mode, the mobile station monitors the downlink signal level and color code from its serving cell and surrounding cells.

2.7 Handover:

- Intersystem and intrasystem handovers are specified.
- Mobile assisted handover.
 - It provides the ability for the mobile stations to measure and report both the received signal strength and channel quality over the current connection as well as the received signal strength on other channels, as requested by the base station.

2.8 Roaming:

- In accordance with Recommendation 624.
- The mobile station evaluates the received signal and coding and initiates the location updating procedure when necessary.
- Roaming is possible between MSCs and between systems.

2.9 System architecture:

- Communication protocol: The network communication protocol reference model is designed according to the OSI model.
- Interfaces : The interfaces between system function blocks are designed according to CCITT Recommendations of the Q.1000-series.

2.10 Networking:

- ISDN and PSTN interfaces: In accordance with CCITT Recommendations Q.700- and Q.1000-series.
 - Numbering plan : In accordance with CCITT Recommendations E.164, E.212, and E.213.
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REPORT 1020

ADAPTATION OF SYSTEM SPECIFICATION TO EASE THE
PRACTICAL IMPLEMENTATION OF RADIO EQUIPMENT

(Question 68/8)

(1986)

1. Introduction

Question 68/8 addresses the possibilities of adapting the technical characteristics of cellular land mobile radio systems (see Report 740) to ease their practical implementation without compromising the system performance. One of the most important system characteristics is good spectrum utilization.

Large systems, based on small-cell structures and allocated an exclusive frequency band, will be of growing importance, at present mainly for mobile telephone use but eventually also for dispatch and data services. A future mobile telephone system might also be integrated with the planned ISDN network.

In advanced cellular systems, mutual interference within the system can be reduced through suitable network arrangements, thus permitting reduced system selectivity (spurious, intermodulation, blocking) with marginal impact on the overall system performance. New technologies such as digital speech transmission and time divided usage of wide-band radio channels (TDMA, packet transmission and time duplex) could be used in new system configurations with improved performance in comparison with present analogue narrow-band systems. Potential improvements are in spectrum economy, transmission quality, new services and ease of implementation.

Small portable terminals will probably be of rapidly growing importance if cost and size can be made attractive for a mass market. It is therefore very desirable that the system specification and configuration ease the implementation of small, inexpensive terminals. The combination of digital transmission, reduced system selectivity and wide-band radio channels could make this possible.

Most of the material in this Report has been presented in more detail in four contributions to the Nordic Seminar on Digital Land Mobile Radiocommunication in Finland, February, 1985 organized by the Telecommunication Administrations of Denmark, Finland, Norway and Sweden [Öhrvik, 1985; Stjernvall, 1985; Ekemark *et al.*, 1985; Uddenfeldt, 1985; Uddenfeldt and Stjernvall, 1985].

Uddenfeldt [1985] discusses the relations between system characteristics and ease of implementation. A more detailed discussion can be found in lecture notes from a post-graduate course held at Lund University, Sweden, in 1984 [Öhrvik, 1984].

2. Comparison between the frequency economy of narrow-band digital speech transmission and present analogue mobile telephone systems

Several research groups have indicated that speech coders of reasonable complexity and power consumption (for portable telephones) could give close to normal telephony speech quality with a total data rate of around 16 kbit/s, including channel coding to accept up to 1% bit error ratio with small degradation in speech quality. Additional channel coding against burst errors due to fading dips, which increases the data rate to around 27 kbit/s, gives a substantial reduction of the required $C/I_{\text{co-channel}}$ during Rayleigh fading. 27 kbit/s can be transmitted over a 25 kHz radio channel with reduced requirements on adjacent-channel attenuation. The reduction in the required C/I corresponds to a substantial reduction in the required frequency re-use distance compared to present mobile telephone systems. As is summarized in Table I [Uddenfeldt, 1985] around three times improvement in spectrum utilization is obtained.

TABLE I — *Spectrum efficiency for companded FM system and two digital narrow-band FDMA systems. Sectorized cells with 3 cells per site are used*

	Analogue system	Digital system	Coded digital system
Description	Comp. FM	RELPG/GMSK	RELPG/GMSK
Speech coder rate (kbit/s)	—	16	16
Transmission rate (kbit/s)	—	16	27
Channel spacing (kHz)	25	15	25
Minimum C/I in fading (dB)	18	20	13
Frequency re-use:			
— No. of frequency groups	21	27	9
— No. of sites	7	9	3
Spectrum efficiency:			
— No. of channels per MHz and cell	1.9	2.4	4.4
— Capacity per cell for 10 MHz system (erlang)	12.4	17.2	35.1

This is of course no general result. Further improvements in analogue mobile telephone systems have not been considered, but on the other hand, large further improvements in technologies for digital speech transmission over fading radio channels can be expected.

3. Frequency economy implications of introducing wide-band radio channels

One major complication in going over to wide-band radio channels is time dispersion. The final result on the system level might be either a degradation or an improvement in frequency utilization through the influence of time dispersion (or frequency selective fading) on the required $C/I_{\text{co-channel}}$.

Under typical propagation conditions, the modulation bandwidth is considerably larger than the coherence bandwidth. Suitable signal processing could take advantage of this condition by introducing frequency diversity to reduce the required fading margin for multipath fading. The negative consequence might be that the effect on receiver sensitivity (required C/I) from time dispersion (inter-symbolic interference) could not be fully eliminated through suitable signal processing (i.e. adaptive channel equalization).

Much further study is needed, both of the detailed wide-band characteristics of the mobile radio channel and of suitable signal processing to take advantage of or combat time dispersion. The effects will also depend on the total system configuration, i.e. system data rate, other diversity arrangements, channel coding and basic type of radio modem. Frequency diversity could also be introduced through coordinated frequency hopping between the channels allocated to a cell.

An advantage of different combinations of TDMA and packet transmission is increased system flexibility, especially if different non-voice services become an important part of future systems. An obvious disadvantage of present FDMA systems is allocation of unnecessarily large channel capacity for narrow-band data transmission and for certain types of signalling. (However, an advantage of a wide-band channel system is the possibility of accommodating bursts of high speed data, i.e. 16 kbit/s ISDN messages. Also digital speech interpolation (DSI) like arrangements might be incorporated.)

The introduction of TDMA/packet results in much improved system signalling. A radio terminal can exchange system signalling with the base station without interruption of speech and data transmission, using separate time slots. The terminal can also check the signal level from nearby cells (momentarily switching both to a new time slot and radio channel). The frequency economy could be improved both through macro diversity between cells to reduce the required fading margin for shadowing and through rapid dynamic channel re-allocation when C/I becomes marginal. Such a system can be designed for a lower average $C/I_{\text{co-channel}}$, and the co-channel cells can be packed closer together, resulting in improved spectrum utilization.

On the other hand, the introduction of TDMA means a basic penalty in frequency economy due to overheads for guard bands between time slots and for burst synchronization.

4. Adaptation of the system specification to ease the practical implementation

4.1 *Reduced system selectivity*

Analysis of the interference situation in a cellular system with suitable control of terminal transmit power and close to optimal hand-over procedures, indicates much reduced mutual interference in the direction, terminals to base, compared to the situation where small independent systems, with overlapping geographic coverage but different base station sites, share the same frequency band. The system selectivity requirements on terminal transmitters and base station receivers could probably be reduced to around 40 dB without any noticeable increase in the $(C/I)_{\text{co-channel}}$ required for acceptable quality.

The interference situation would be roughly the same in the opposite direction if no dynamic power control is applied to the base station transmitters. A simple way to implement high dynamic range (blocking performance) of the terminal receivers might be to introduce automatic gain control.

A possible complication might be the introduction of power control at the base station transmitters in order to reduce the average level of co-channel signals, thus reducing the probability for harmful co-channel interference. One consequence would be requirements for more stringent specifications on transmitter intermodulation.

Reduced requirements on system selectivity would have a considerable influence on equipment cost and size through reduced requirements on dynamic range, noise performance and filter selectivity of different transmitter and receiver sub-systems. Present extreme requirements on spurious suppression have led to complicated receivers, using several stages of mixing and filtering. Reduced selectivity requirements might give important simplifications such as balanced circuits to suppress spurious emissions and direct conversion between base band and transmit frequency.

4.2 *Increased channel bandwidth*

The introduction of wide-band time-division radio channels will have a considerable impact on the practical implementation.

The cost of the concentrated base station sites will be reduced as each radio channel equipment will be shared by several speech or data channels. A reduced number of more widely separated radio channels will lead to a considerable reduction of the multiplexer complexity. Outside of the major metropolitan areas, the required traffic capacity of a base station site could be served by one wide-band radio channel, eliminating the need for multiplexers.

Considering base station complexity, the optimum system data rate would be a compromise between the needs of the major population centres and the more sparsely populated areas, and also a compromise between the early phase of system growth and the final saturated system.

A more important impact will probably be on the terminal's cost and size, as the terminal's cost is typically the major part of the total cost of a cellular system. Introduction of wide-band radio channels leads to a reduction of the requirements for extreme frequency stability, narrow radio-frequency filters and VCOs with good noise and microphony performance. Also the frequency synthesizer design is simplified, especially if there is a system requirement for fast switching between radio channels.

Reduced system selectivity and increased channel spacing impact mainly on the analogue and very high frequency sub-systems. This is of special importance as remaining digital sub-systems, including digital signal processing, can take full advantage of further developments in custom digital VLSI. Eventually, this might lead to the situation that those sub-systems that could not be realized in digital VLSI, will become the dominating cost and size factors of the terminal electronics.

However, introduction of wide-band radio channels also introduces system limitations and design problems. Too wide radio channels would not be practical.

- In many parts of the world and for many services, digital cellular systems for telephony or for dispatch must be introduced gradually into frequency bands presently organized for, and partly occupied by, narrow-band FDMA channels. A possible compromise might be to use intermediate bandwidth radio channels, i.e. combine FDMA and TDMA with around 300 kHz channel spacing, and to use a combination of frequency and time duplex (frequency duplex to adapt to the present frequency allocation principles and time duplex to eliminate the duplex filter from portable terminals).
- Time synchronization between different cells would probably be required.
- TDMA, and especially packet arrangements with retransmission of lost packages will introduce delays. Delays will also be introduced by the speech coder and channel coder. Echo cancelling will probably be necessary also in local and regional mobile telephone networks.
- Due to the use of a burst transmission with low duty cycle, the peak power will be much higher than the average terminal transmit power. The possibilities for considerably reducing the fading margin for multipath and even part of the shadow fading, as has been suggested above, will however reduce the requirements on average transmit power. (That also helps to reduce the battery size of portable terminals.) Using intermediate bandwidth radio channels ("narrow-band TDMA") the net result could be only moderately higher peak power than the CW power of present terminals.
- Going over to more wide-band radio channels which would probably require more complex circuits to utilize or combat the time dispersion, could result in excessive power consumption of the radio modems, especially considering portable terminals. Unfortunately, further improvements in the packing density of digital VLSI will probably not result in equal progress in reduced power consumption per elementary function. Even with CMOS, the power consumption will be considerable, if very high speed logic must be used. As portable telephones would be a major part of future cellular systems for important markets, the consequences on power consumption should be thoroughly considered before deciding on the width of the radio channels in a digital mobile telephone system.

5. Conclusions

This Report has tried to indicate that new technologies both on the system and practical implementation side could make it possible to combine improved system performance, especially frequency economy, with reduced system cost and reduced size of portable terminals. Key areas to consider the optimum system selectivity and channel spacing. A very tentative suggestion made in the Report is that a "narrow-band TDMA" system using digital speech transmission and a channel spacing around 300 kHz is an interesting possibility that merits further study. An important area that should be studied is the detailed characteristics of the wide-band mobile radio channels, including appropriate mathematical models and channel simulators.

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REPORT 741-3

**MULTI-CHANNEL LAND MOBILE SYSTEMS FOR DISPATCH TRAFFIC
(WITH OR WITHOUT PSTN* INTERCONNECTION)**

(Question 37/8)

(1978+1982-1986-1990)

Introduction

Radio spectrum allocated for dispatch traffic is becoming increasingly congested and already many users (networks) have to share a channel with several other users. This practice employs the channel more effectively but gives the participants a reduced "grade of service" in the sense that they may often have to wait for some time for a channel to become free and they suffer from reduced privacy. Not only improved channel utilization but also better grade of service and good privacy conditions are important factors to design dispatch systems.

Part A of this Report deals with the general aspects of multi-channel land mobile systems for dispatch traffic, such as system configuration, characteristics of dispatch traffic, grade of service, traffic handling capability, performance of trunked systems, signalling and so on.

Part B of this Report introduces some systems being installed or planned by some administrations.

For this purpose of this Report, "dispatch system" has the meaning: A radio system used to control the operation of a fleet of mobiles, such as air craft, taxis, police, etc., and "trunked system" has the meaning: A multi-channel system with automatic channel selection, particularly referring to dispatch systems.

However, this Report does not consider the detail of interconnection of trunked dispatch systems with public or private switched telephone networks. Such interconnection requires further study.

* Public switched telephone network.

PART A

GENERAL ASPECTS RELATING TO SPECTRUM
CONSERVATION AND SYSTEM DESIGN

1. System configuration

A typical arrangement of equipment, covering a single radio zone, is shown in Fig. 1 and consists of the following four principal items:

- a number of control posts (CP)(s) (i.e. dispatchers or control stations), consisting of a CP(s), each connected to a switching centre, either by a dedicated line or a radio path using channel equipment (CE);
- a switching centre (SC), (several switching centres may be interconnected);
- a number of base stations (BS) each containing a number n of CE(s).
- a number of groups of mobile stations, each consisting of a_m Mobile Stations (MS) and each being served by its own CP.

To enable a CP to call one of its MSs, and vice versa, each CP and each MS is considered to have access to one of the radio channels by means of an access procedure which is still to be defined.

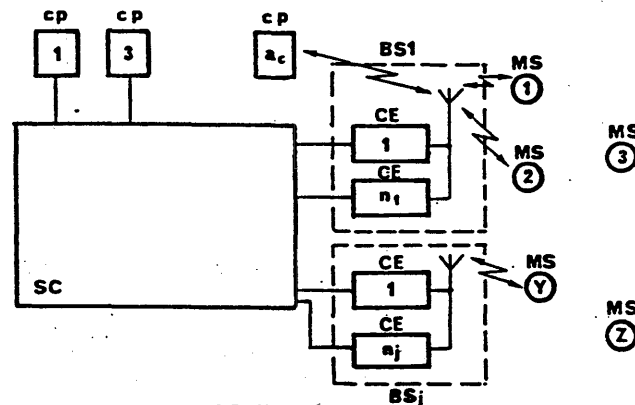


FIGURE 1

Arrangement of equipment in a trunked system

cp: control post (i.e. dispatcher or control station)

CE: channel equipment

MS: mobile station

a_c : number of control posts

SC: switching centre (several switching centres may be interconnected)

BS: base station (some administrations define the base station as one channel equipment)

$n = \sum_{k=1}^j n_k$: total number of channel equipment

2. Characteristics of dispatch traffic in private mobile radio systems

The increase in channel loading which can be achieved by trunking depends on the kind of traffic carried. Mobile radio dispatch traffic differs from public mobile telephony traffic in a number of ways:

- the mean holding time of calls is shorter (15 seconds may be typical);
- many users operate a vehicle fleet system where a single operator controls a number of mobiles;
- the size of fleets varies greatly.

2.1 Fleet operation

The number of users in the system is, in effect, equal to the number of CP operators, not the number of mobiles, because if the operator is busy nobody in the fleet can place a new call. Thus all the traffic of the fleet passes through the operator and a mobile radio system therefore is characterized by a small number of users (operators), some of which may offer large loads and some small. Such a system is termed "unbalanced" (see also § 5).

Fleet operation means that there are actually two kinds of queue. The CP operator only ever has to wait because the channel is occupied by another fleet (the "sharing" delay) but an MS may have to wait not only for other fleets but also because his own operator is busy with a call to another MS (the "fleet" delay). It can be argued that the fleet delay can be ignored because:

- it is experienced only by the MSs, each of which uses the system rather sparingly compared with the CP;
- the delay can be reduced only if the fleet owner is prepared to employ extra operators, a factor which is outside the control of the systems designer;
- the psychological effect of waiting while one's colleagues talk to the CP is not the same as that of waiting while strangers block the channel.

2.2 Random fluctuations

The random nature of mobile radio dispatch traffic means that the busy hour traffic level will fluctuate. The fluctuations can be surprisingly large and last for significant periods. Upward fluctuations in the traffic level will result in a reduced grade of service which will be unacceptable to the users if they occur too often or last too long. It may be desirable to protect against this with an 'overload' criterion of the form:

- the ordinary busy hour grade of service should not degrade by more than, e.g., a factor of 2 for periods longer than half an hour occurring no more than once every 20 days.

2.3 Day to day traffic variations

Telephony experience [Hayward and Wilkinson, 1970] shows that there will be variations in the mean busy hour traffic from day to day in addition to the random fluctuations. With small numbers of users the variations can be expected to be large: the traffic will vary not only because of changes in the activity of the whole community, but also because of sudden changes in the demand of just a few of the users. The day to day variations add to the random fluctuations and so, in practice, the safety margin required for protection against overload is likely to be significantly larger than consideration of the random fluctuations alone would indicate. At present the extent of these variations has not been studied.

2.4 Operating procedures

Operating procedures which may cause excessive simultaneous call requests from many mobile stations can overload the access protocol or the switching centre of a trunked system. For example, transactions in which a control post seeks to allocate work to mobile stations by means of a fleet call requesting offers or "bids" for the work. System design may make special provision for such procedures (see also Annex I to Part A, section 3.3).

3. Grade of service and traffic handling capability

The increase in channel loading that can be achieved with a trunked system also depends on the grade of service required.

3.1 Reference system

The merits of the trunked system with respect to traffic handling capacity and radio channel efficiency can be compared with those of a single channel reference system employing automatic sharing, having the same amount of traffic per MS and the same number of MSs per CP and offering the same grade of service.

It is assumed that a CP is connected to a SC by a dedicated line, and n is less than a_c .

3.2 Traffic assumptions

In addition to the usual assumptions for estimating the volume of the traffic in ordinary telephone networks, the following assumptions may be made:

- communication is possible, in two directions in a simplex mode, between a CP and each of its MSs, but not more than one at a time;
- any other communication, for example between MSs of the same or different networks or between CPs of different networks, is excluded;
- all calls are queued and processed in the order of arrival;
- any degradation due to multipath fading, interference and other deficiencies inherent in radio transmission is disregarded.

3.3 Delay criteria

Because of the relatively short mean holding times encountered in mobile radio networks for private use, the character of the messages and the general pattern of conduct of the user of such networks, a waiting system, also known as delay or queueing system, is desirable.

For the purpose of traffic calculations a delay criterion, both under normal traffic and overload conditions, may then be used as a measure of the grade of service in accordance with current telephone practices.

The following criteria are appropriate:

- under *normal traffic conditions* the probability of excess delay (P), that is, the probability that a given waiting time is exceeded, shall be limited. If, for example, this waiting time is one mean holding time, then $P = P_r$. It may also be desirable to guard against occasional very long delays;
- under *overload conditions* the probability of excess delay shall not be more than, for example, twice the probability under normal traffic conditions (e.g. P_r), in the case that the traffic per MS increases by a given percentage, for example 10%. An alternative way of specifying the overload criteria is given in § 2.2.

3.4 Probability of excess delay in the single-channel reference system

The distribution of waiting time in the theoretical model of one telephone line serving a finite number of traffic offering sources is known and has been published. It can be shown that the model of a single mobile radio network consisting of one CP and a finite number of MSs is equivalent to the telephone model mentioned above, with the CP corresponding to the line and the MSs corresponding to the traffic sources.

However, the single-channel reference system consists of several such networks, each comprising one CP and several MSs. As the CPs are linked by the fact that they have access to only one radio channel, it was necessary to also consider the CPs as traffic sources. Consequently, a single radio channel serving simultaneously several CPs and their MSs will give rise to two rows of queueing calls which are linked to each other.

An exact solution of the distribution of waiting times has not been obtained but an approximate solution by means of computer calculations shows that for a single-channel reference system consisting of 10 CPs, each serving 10 MSs with a traffic load of 0.005 erlang per MS, the probability, P_r , of the waiting time exceeding the mean holding time (e.g. 15s) is equal to about 0.3.

3.5 Channel efficiency and overload capability of a trunked system

From the results of similar, preliminary calculations performed on a trunked system with the same probability of excess delay, $P_r \approx 0.3$, as the single-channel reference system, the following simple empirical relationship between the number of CPs (a_c) and the number of radio-channels n (where $n > 1$), is derived:

$$a_c = \frac{n - 0.6}{a_m \alpha_m}$$

where

a_m : the number of MSs per CP

α_m : the traffic per MS in the busy hour, expressed in erlang.

When the number of radio channels is increased, the admissible number of CPs increases in accordance with the above formula. The channel efficiency expressed in terms of traffic load per channel also increases. This last relationship is shown in Fig. 2 under normal traffic conditions, for a probability of excess delay of about 0.3.

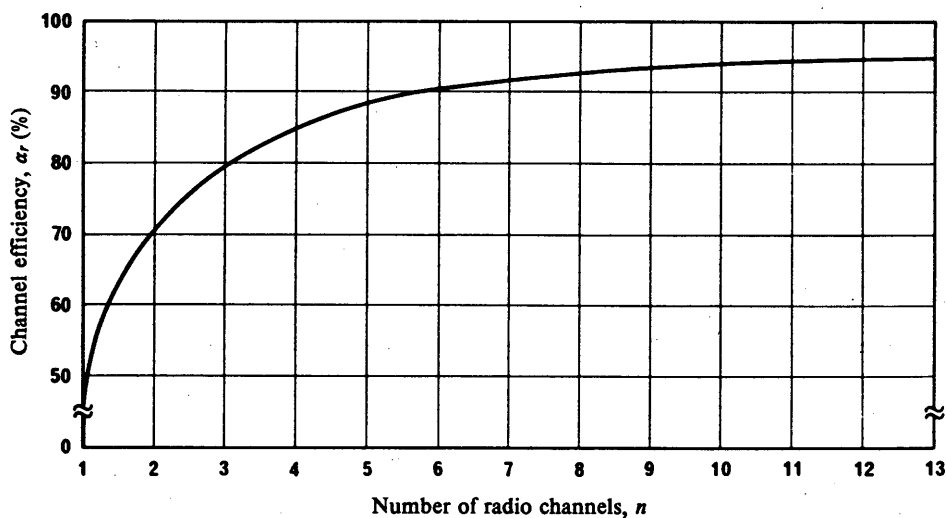


FIGURE 2 - Radio channel efficiency in a trunked system
(for $a_m = 10$, $\alpha_m = 0.005$ E and $P_r = 0.285$)

a_m : traffic per MS

a_m : number of MSs per CP

P_r : probability of excess delay (delay equal to the mean holding time) under normal traffic conditions

However, with an increasing number of radio channels, the system becomes more sensitive to overload due to the increased traffic load per channel. Fig. 3 shows the permissible overload as a function of the probability of excess delay for different numbers of channels. From the overload criterion given in § 3.3, it may be concluded that the maximum number of radio channels in the case considered here amounts to 4 or 5, the overload criterion being slightly exceeded for $n = 5$.

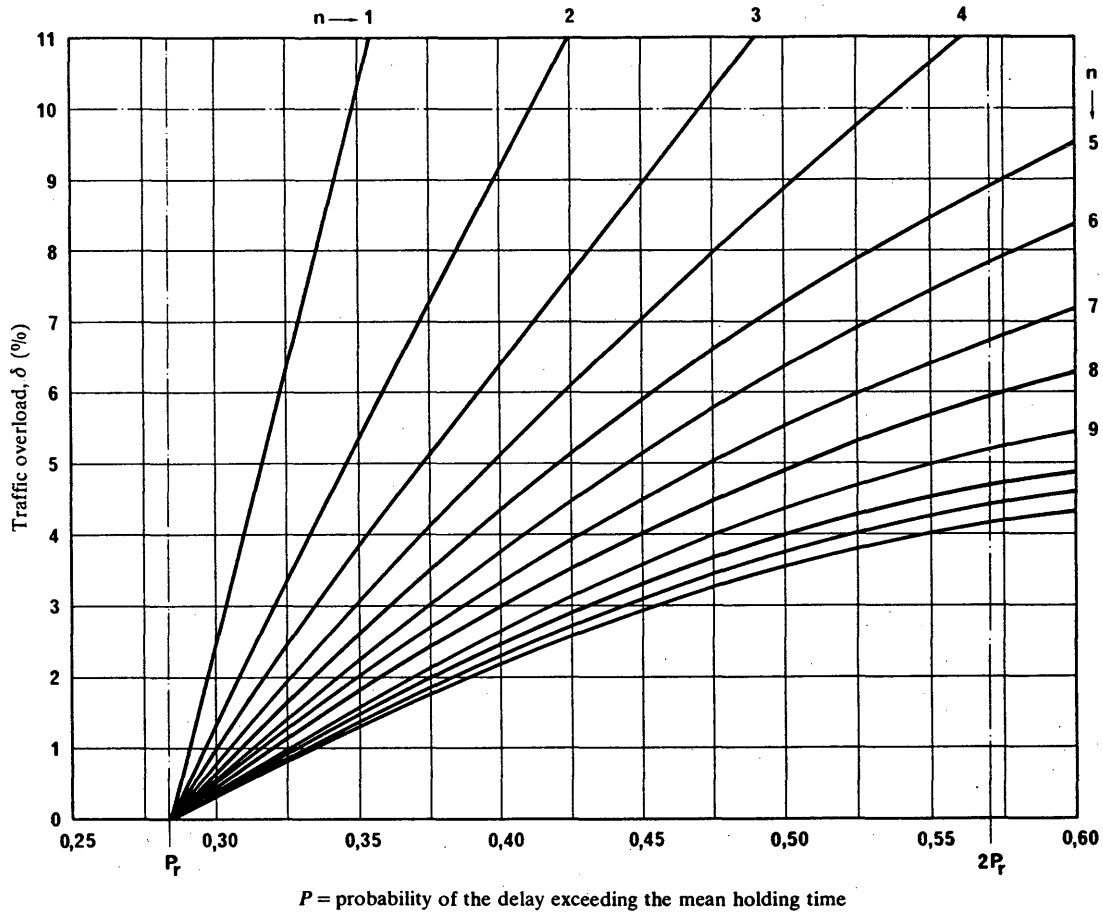


FIGURE 3 – Traffic overload characteristics in a trunked system
(for $a_m = 10$, $\alpha_m = 0.005$ E and $P_r = 0.285$)

- n : number of radio channels
- a_m : traffic per MS
- a_m : number of MSs per CP
- P_r : probability of excess delay (delay equal to the mean holding time) under normal traffic conditions

3.6 Discussion of the results

Assuming five radio channels and 10 MSs per CP, each MS producing 0.005 E, it follows from the formula in § 3.5 that the system is capable of handling 88 CPs and 880 MSs with a radio channel efficiency (see Fig. 2) of 88%.

One single-channel system with 10 CPs employing automatic sharing and serving the same number of MSs per CP with the same amount of traffic per MS, is capable of handling 100 MSs with the same grade of service, but with a channel efficiency of only 50%. The effective gain in spectrum utilization obtained by trunking thus amounts to $100(88 - 50)/50 = 76\%$.

When the system is compared with a manual system using a single radio channel and providing a similar grade of service, the effective gain only can be estimated; it will be of the order of 100%.

Figure 3 shows that if a smaller value of the permissible traffic overload and/or a larger value of the probability of excess delay under overload conditions is chosen, it is possible to increase the number of channels and, consequently, have a better spectrum utilization.

A further improvement is possible if, in contrast to what has been assumed in the present calculations, only those users are selected to participate in a particular trunked system, whose busy periods are not all coinciding.

It should be noted that the gain in spectrum utilization also depends on the inputs for the traffic data and further assumptions made, such as those mentioned in § 4 to 6.

4. Measured data on performance of trunked systems

In an effort to evaluate the performance of trunked systems, the Department of Communications, Canada, collected monitoring data for a ten-channel trunked radio system, with a single dedicated signalling channel, operating in the 800 MHz band and 12 conventional (non-trunked) channels operating in the 400 MHz band. For this trunked system, channels are occupied for the duration of a transmission only, while for the conventional systems the channel is occupied for the duration of the message.

Based on the average peak-hour traffic conditions, transmission occupancy and average probability of waiting were compared and are given in Table I.

TABLE I

	Average probability of waiting (%)	Average peak hour transmission occupancy (%)
Trunked system	13	64
Conventional system	53	33

Table I shows that the use of trunking resulted in considerable reduction in the waiting time and increase in the channel loading compared with conventional systems with similar types of user.

5. Unbalanced systems

As discussed in § 2.1, fleet operation means that the number of users on a mobile radio system will often be small although the number of mobiles may be large. Also, the load offered by different users will often vary considerably [Davis and Mitchell, 1979]. A system in which large and small users are mixed is termed "unbalanced" and in such a system the large users can receive a much better grade of service than the small users [Davis and Mitchell, 1978], a large user offers a large amount of traffic in comparison to a small user. This leads to the problem of whose grade of service should be used to determine the permissible loading of the system.

As an example, consideration is given to the channel efficiency which is possible in an unbalanced system, when a given grade of service criterion is applied to:

- the calls of the largest users only and,
- the calls of the smallest users only.

The grade of service criterion chosen for this example is that approximately 15% of calls may suffer a wait of more than one mean holding time. Only the time spent waiting for a channel is considered; the fleet delay is ignored.

The results for 1, 3 and 5 channel systems are shown in Fig. 4 [Davis and Mitchell, 1979]. When the grade of service criterion is applied only to the largest users then the channel efficiency can be significantly greater than it would be in a balanced but otherwise similar system. This is illustrated by curve A in Fig. 4. The probability of delay for the small users will then be worse than the specified grade of service criterion, but because they offer fewer calls the number of times they actually suffer a significant delay will still be small and perhaps acceptable. Curve A shows that unbalance reduces the amount there is to be gained from trunking in this situation because it allows the loading of a single channel to be increased.

If the grade of service criterion is applied to the small users in an unbalanced system the channel efficiency will, in general, be similar to the balanced situation as shown by curve B.

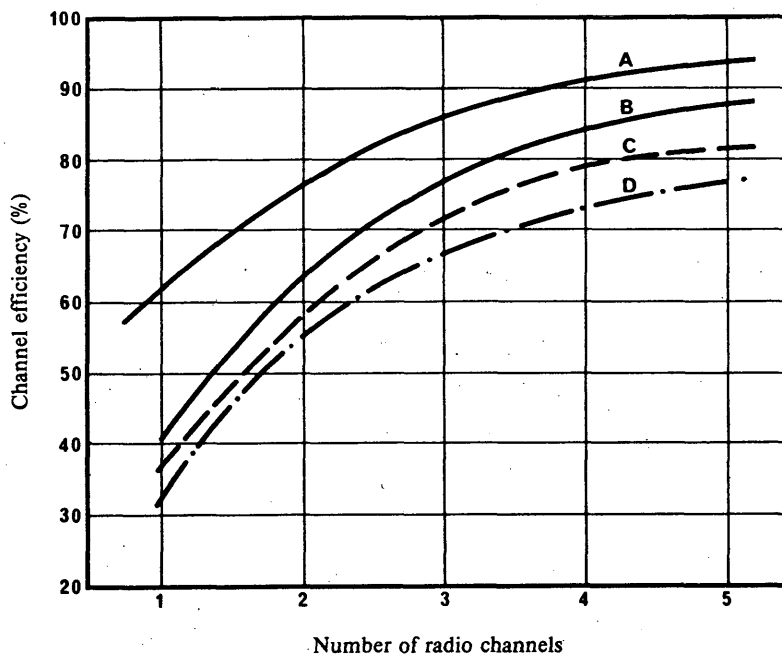


FIGURE 4 – Radio channel efficiency in balanced and unbalanced systems as a function of the number of channels (such that the cumulative probability of delay greater than the mean holding time equals: 0.165)

- Curves A: results when the grade of service criterion is applied only to the large users (CPs) in an unbalanced system
 B: results when the grade of service criterion is applied only to the small users in an unbalanced system
 C: is for a balanced system in which all users offer the same amount of traffic
 D: results from conventional telephone traffic theory

Note. – It is assumed that in the unbalanced case there are 1 large and 11 small equal users/channel (traffic ratio approximately: 11:1). In the balanced case there are 12 users/channel.

6. Signalling

In trunked systems with queueing of incoming call requests the signalling system used for handling these calls is crucial to the efficient operation of the system. The main design considerations are:

- the method for allocating the signalling channel;
- the method of access to the signalling channel;
- the method of contention control;
- the channel allocation time.

6.1 *Method for allocating the signalling channel*

The radio channel used for gaining access to the switching centre may be dedicated solely to signalling, but it is also possible to use any channel for that purpose. If, in this last case, one of the speech channels is temporarily idle, it may be marked as the signalling channel and be used as such during a limited period of time. In that case, the mobile attempting to place a call must detect this channel, for example by scanning the radio channels.

In both cases, dedicated signalling channel and non-dedicated signalling channel systems, the mobiles placing a call must all use the same channel and this may give rise to the problem of contention, that is, the problem of giving each contender a fair chance of gaining access to the channel in accordance with an established rule.

6.2 *Methods of access to the signalling channel*

There are two basic methods of accommodating in time the calls in the available signalling channel:

- an ordered method of access, for example a "polling" system in which each mobile is assigned a particular time slot which enables him to signal if he has a message to send,
- the random access method, for example some form of "ALOHA" [Kleinrock, 1976], in which the mobiles receive a standard starting signal inviting them to signal if they want to do so.

With polling, the signalling load depends on the number of users who have to be polled, rather than on the traffic offered. This is less efficient when there is a large number of users, each of whom offers traffic only occasionally. Moreover, any change in the number of mobiles requires some rearrangement of the polling routine.

The ALOHA type of systems may be much more efficient, provided the probability of "clashing", i.e. corruption of simultaneously transmitted calls, is sufficiently small.

6.3 *Contention control*

With the ALOHA type of systems every signalling message contains bits which are used for error detection allowing the control equipment to determine whether the message was corrupted by clashing with a simultaneous transmission from another MS. If the signal is successful an acknowledgement is sent; if it is not, the MSs concerned repeat their messages but with a randomly chosen delay between the first and second attempts. The process continues until the messages get through or a predetermined time elapses.

However, depending on the intensity of signalling messages and the availability of a signalling channel, there is a possibility of the system becoming unstable, i.e. the number of call requests pending (which are subsequently repeated and add to the new requests) becomes so large that the signalling channel's throughput is finally reduced to zero so that traffic handling ceases completely [Kleinrock and Lam, 1975].

Annex I to Part A reviews some alternative ALOHA systems and methods for contention control to improve stability.

6.4 *Channel allocation time*

For efficient operation the channel allocation time, i.e. the time that elapses between a channel becoming free and its being re-allocated, must be short compared with the average time between channels becoming free.

The channel allocation time depends on the signalling bit rate, the signalling format, the method of access and the system of contention control.

7. **Other important design considerations**

7.1 *Off-air call set up*

In order to minimize the amount of "air-time" wasted while a call is being set up, all calls should be set up "off-air" as much as possible. This will be particularly important where the call is being made through the public telephone network where set-up times may be appreciable compared with typical holding times.

7.2 *End of conversation message*

A reliable end of conversation message should be used so that channels can be made available again as soon as the previous call is completed. Additionally, some form of time supervision of the channel may be required.

8. Conclusions

Although trunking cannot solve all the problems relating to the frequency scarcity in the land mobile service, nevertheless it seems to be a valuable tool for frequency utilization.

As the improvement in channel utilization is very dependent upon the number of users, the way traffic is shared among them and the grade of service required, the improvement in channel utilization can be expected to vary considerably from one system to another.

Taking such limitations into consideration, it should be noted that improved spectrum efficiency and/or grade of service is obtained by trunked systems, regardless of type of modulation, channel bandwidth and the like.

However, in order to convert to a trunked system, the user has to incur additional cost due to a multi-channel radio set with more sophisticated signalling circuitry, and more complex base station operation.

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ANNEX I TO PART A

METHODS OF CONTENTION CONTROL

1. Introduction

This Annex reviews some alternative ALOHA systems and methods of contention control to improve stability, distinguishing between systems with and without feed-back.

2. Systems without feed-back

Carrier sense systems [Kleinrock and Tobagi, 1975] seem to be less suitable as they cannot be used for two-frequency simplex operation and also labour under the so-called hidden terminal problem [Tobagi and Kleinrock, 1975].

By employing "slotted ALOHA", in which the signalling packets are transmitted in successive time slots, the clashing probability is reduced by a factor of about two compared with the conventional ALOHA-system without time slots [Abrahamson, 1977].



Introduction of an additional frame structure such that a mobile may signal in one of the slots of a frame, only if it had the signalling packet ready at the beginning of the frame ("framed slotted ALOHA" [Okada *et al.*, 1977]), leads to a better throughput of the signalling channel than "slotted ALOHA" [Capetanakis, 1979; Schoute, 1980].

In both cases, stability is improved if capturing occurs, for example with FM-transmissions where a strong call request signal may be received correctly even though it occupies the same time slot as the weaker signals.

Although under normal traffic conditions "framed slotted ALOHA" employing, for example, 3 frames of constant length with 4 slots each, is unlikely to become unstable when capturing occurs, some form of dynamic (or feed-back) control may be desirable for the reasons set out in § 3 below.

3. Systems with dynamic (feed-back) control

Among the reasons justifying dynamic control, the following may be mentioned:

3.1 It is generally assumed that the number of new call requests arriving is a random variable following a Poisson distribution. This would be correct if the arrival process is a Poisson process *and* the time between signalling periods is constant. If, as is normally the case, the time between signalling periods is not constant, the assumption is no longer valid. The non-Poisson distribution results in an increased probability of the system becoming unstable, particularly under conditions of increased signalling intensity, such as those mentioned in § 3.2 and 3.3 below.

3.2 An additional burden is placed on the signalling system if some event takes place (e.g. an accident, traffic queue and the like) near the boundary of the radio zone. In this case, an increased amount of weak call requests may be expected during a prolonged time, resulting in a reduced probability of capturing.

3.3 Initial bursts of call request may occur if for some reason (e.g. technical malfunctioning of the system or excessive holding times of a temporary nature) there is no possibility to signal during several minutes in systems with a non-dedicated signalling channel, and mobiles are eager to transmit a request packet.

Dynamic control procedures have been studied by various authors [Lam and Kleinrock, 1975; Fayolle *et al.*, 1977]; a quantitative analysis of instability when there is no feed-back in the case of "framed slotted ALOHA" may be found in [Schoute, 1980].

An example of dynamic control of random access, especially for use in the maritime mobile satellite service, is the binary search procedure, described in Annexes I and II of Report 596, which, when used in conjunction with "framed slotted ALOHA", can give relatively high throughput [Capetanakis, 1979].

Another example, specifically adapted to private land mobile radio, is slotted ALOHA with dynamic frame length control [Schoute, 1980].

With this system the number of slots in a frame is not constant, but is automatically adapted to the expected demand, thus eliminating the cause of instability. The algorithm used in the system is handled by the switching centre and controls the number of available time slots in a frame on account of observations made on the utilisation of the time slots in a previous frame. Each time an ALOHA start signal is transmitted, the mobiles are informed about the available number of slots. Each mobile with a request pending then selects at random one of those slots.

The process of updating the number of slots in the new frame is based on:

- the number of successful, garbled and empty slots in the previous frame;
- the expected number of new calls since the beginning of that frame;
- some data which are computed off-line from stored information concerning the properties of the modulation system, the regional radio propagation conditions and the geographical distribution of mobiles.

Updating of the stored information to cope with gradually changing conditions is possible by means of an automatic learning process based on long term observations, according to appropriate software to be incorporated in the switching centre.

Under certain practical assumptions the number of time slots in a frame is equal to one for low traffic intensities and may increase up to, for example, 15 during the busy hour.

The variable number of time slots also results in a signalling time and a waiting time, which, for low traffic intensities, are both equal to about half of that required for the slotted ALOHA system with fixed frame length mentioned in § 2 above.

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PART B

SYSTEMS BEING INSTALLED OR PLANNED
IN THE NEAR FUTURE

The systems referred to in Part B do not necessarily meet all design indications mentioned in Part A.

The basic specifications for the systems described are shown in Table I.

1. Example of a trunked mobile radio dispatch system in the United States

In the United States, trunked mobile radio systems are being installed for dispatch traffic using between 5 and 20 channels in the 900 MHz band. One channel on each system is a dedicated signalling channel where all requests for service and instructions to mobile stations users are made. The other channels are for two-way voice communication.

The dedicated signalling channel is organized in a time slotted configuration for contention control and uses a 78 bit word format requiring approximately 23 ms to be transmitted. A 21 bit word format at a 150 baud rate on the voice channel provides subaudible connect and disconnect signalling to mobiles. Error detecting and code correcting techniques are used to protect the integrity of information.

Requests for service are queued and mobile stations are served on a first-come first-served basis. Generally users speak for periods of only 2 to 3 seconds, with pauses of a few seconds in between. The system signalling speed allows the processor to recognize these pauses in conversation and reclaim the idle talk channel for reassignment to other groups. When the original user group is ready to continue its message, it will be assigned a new voice channel (or its request will be placed in a priority recent system user queue if no talk channels are available).

The use of this trunked system for extended area coverage, portable units and/or satellite receivers and direct mobile to mobile/portable communications is still being evaluated.

2. Basic characteristics for a trunked dispatch system in Sweden

In Sweden a nation-wide trunked dispatch system called MOBITEK has been introduced [STA, 1986]. A common infrastructure will serve more than 40 000 mobile stations and 4000 control posts throughout the country, using 150-200 base station sites in a cellular structure and some 1000 channel equipments. Automatic roaming between base stations is included.

In MOBITEK data or text messages as well as speech can be exchanged between mobile stations and their control posts or between mobile stations.

Data and text is forwarded using packets with variable length on a store-and-forward basis. MOBITEK is mainly a layered communication protocol suitable for any data rate on VHF or UHF. For the radio path a protocol comprising error correction coding and selective ARQ is used. For the fixed line part (base station - exchange - control post) CCITT Recommendation X.21 *bis* and high level data link control (HDLC) are used.

The method of contention control is mainly that described in § 3 of Annex I to Part A (slotted ALOHA with dynamic frame-length control) with some added features. The maximum packet length, that can be transmitted from a mobile unit without channel access request and the slot length, can be dynamically and independently changed by the base station. The base station also transmits a silence (busy) signal immediately it detects a transmission addressed to it (compare to CSMA). The contention control protocol also has the ability to permit only parts of the mobile fleet to transmit (to compare the binary search procedure).

The radio channels on a base station can be used for different purposes in a dynamic manner. One of the channels is used as a master control channel sharing the frequency with the other base stations on a time division basis. The rest of the channels can be used for speech or data traffic and as additional random access channels depending on the traffic load. Information on how the channels are to be used by the mobile units is transmitted from time to time by the base station.

To prevent excessive delays, restrictions on speech call holding times are used. Connections to PSTN and public data networks are included.

3. A trunked mobile radio dispatch system developed in the Netherlands and the United Kingdom

This trunked dispatch system developed in the Netherlands and the United Kingdom operates in accordance with the queuing principle and employs a switching centre of the Stored-Program Control type. Communication is possible in a semi-duplex mode between a mobile and its control post and also, in a two-frequency simplex mode, between mobiles of the same fleet. A version of the system has now been installed by several operators in the United Kingdom [DTI, 1987; Somerville and Tridgell, 1987]. Other variants of the system are also being installed in Germany (Federal Republic of) and Finland.

Binary signalling is employed with a message format containing the appropriate sync., data and error control bits. The method of contention control is that described in § 3 of the Annex to Part A and is known as slotted ALOHA with dynamic frame-length control.

The system is suitable for one or more radio channels and can be expanded up to the limit of the switch matrix by adding new channel equipment and updating the ROM tables in the central processor and mobiles. Adding mobiles only requires the table of valid users in the central processor memory to be updated. The same is true for the control posts up to the limit of the switch matrix.

The system employs channel scanning when the number of channels is small, but to further improve system efficiency when the number of channels increases, a transition can be made to a dedicated signalling channel mode of operation with only minor software modifications. The signalling channel can also be shared between a number of base stations on a time division basis.

The system provides for ancillary functions such as group calls, talk-through, call forwarding, emergency calls and a range of data message facilities. It also permits the information handling required for multi-zone operation, e.g. roaming, and network management.

To evaluate the effect of trunked operation for different system configurations under various conditions with respect to channel loading, propagation etc., traffic simulation tests have been made by employing a computerized Traffic Simulator and Analyser [CCIR, 1978-82a] which provides the signalling and simulates the traffic of an arbitrary number of virtual control posts and mobile stations, and analyses the resulting traffic.

The cumulative distribution of waiting times produced by the analyzer part of the simulation equipment has been used as a basis for the comparison of the performance of the various system configurations under varying conditions of use.

4. Basic characteristics for a dispatch system in the USSR

In the USSR, a portion of the spectrum in the band 300-340 MHz is reserved for shared channel dispatch systems using automatic channel selection. The basic technical parameters of the system equipment are established by the State standards in force in the USSR but the network configuration and other parameters of such systems are not standardized.

The following describes a dispatch system in the mobile radio service with automatic channel selection as developed in the USSR. The system is designed for the organization of dispatch radiotelephone communication with stationary and mobile stations. A distinctive feature of the system is the fact that a number of users of the same type combine to form groups which are controlled by a single control post. Groups may vary considerably in size, and the number of mobile stations in each group is not fixed. Moreover, the system makes provision for priority users who can call individually. In practice, the number of subscribers to the system is determined by the number of control posts (user groups) and the number of priority mobile stations. In order to maintain the required service quality, limitations are imposed on the period of radio-channel occupancy, i.e. the system is protected against the possibility of excessively long service delays. In the interests of efficient frequency planning in an area, the system uses two-frequency simplex operation.

Technical characteristics

4.1 The equipment operates with eight fixed frequencies (four for reception and four for transmission) forming a group (RF trunk) of four two-frequency channels.

4.2 25 kHz frequency spacing then provides ten groups in a 1 MHz bandwidth.

The minimum frequency separation between the adjacent channels of one trunk is 25 kHz and the maximum separation between the end channels of a trunk is 175 kHz. Frequency spacing within a trunk is not uniform and excludes third-order intermodulation products. This frequency grid structure reduces intra-system interference to a minimum.

4.3 Within a single trunk, three channels are equally accessible and a fourth is used for privileged access by a limited number of mobile stations (priority mobile stations).

4.4 Seizure of the three equally accessible channels of a trunk is performed by automatic call with free channel finding. For automatic free channel finding, a marker tone is transmitted on all the channels that are free at any given time. A mobile station requiring access is thus automatically connected to any one of the three channels which happens to be free.

- 4.5 Priority mobile stations and control posts have the option of access to the fourth, priority, channel if the other three are occupied.
- 4.6 Up to four connections between mobile stations can be set up simultaneously within one trunk, and one mobile station can be put on stand-by.
- 4.7 The system contains a device which clears the radio channels after 20-30 s in the absence of any speech signal.
- 4.8 The channel equipment of the base station may be up to 5 km from the central dispatch point and is connected to it by cable.
- 4.9 The channel equipment of the base station are designed for 24-hour service in reception and transmission. When the traffic drops in the evening, one transceiver is left in operation. If this radio channel is occupied, other transceivers are switched on automatically.
- 4.10 The mobile station automatically switches over to watch-keeping operation if the subscriber fails to lift his handset within 10-20 s of receiving the ringing tone.

5. A trunked mobile radio dispatch system installed in Japan

5.1 Composition of the system

The system consists of a base station including functions of a switching centre, control posts and mobile stations. A typical user has one control post and a number of mobile stations. A control post is connected to a base station by a radio channel, that means a control post works as a mobile station in a fixed position. A base station employs one channel as a signalling channel and 15 channels for conducting voice communication. The maximum time for voice communication is restricted to 60, 120 and 180 s, according to the occupancy of the system. A queuing strategy is adopted for the call reservation of up to 30 calls. The system provides operational functions such as individual calls, group calls or party calls and it also permits facsimile and data transmission. The standardized specification allows, if necessary, any mobile station to receive the same trunked radio service throughout the country (multi-zone service).

5.2 Operational status

This system has been installed nation-wide, since the first one was installed in October 1982 in Tokyo, and the number of systems in Japan are 59 with approximately 9,100 control posts and approximately 128,000 land mobile stations as of the end of October 1987. These 59 systems are operated in the major cities of Japan by eight different operating entities.

According to the operational data of the system in Tokyo as of October 1987, the total number of calls per system is between 21,000 and 24,000 per day. This means that the average number of calls per day for each station is less than eight. The busiest period is between 1300 and 1800 h and for each station, the mean holding time in this period is approximately 48 s.

6. Common-relay system used in France — RADIOCOM 200

France has brought into service a system with the following characteristics.

6.1 General characteristics

Dispatch-type operation, in which operators' positions are connected by a radio channel at the time of the call to a "base station" which retransmits the call to the mobile stations. Calls can be placed in the control post-to-mobile station or the mobile-station-to-control post direction.

The services provided are:

- calling and routing calls between two mobile stations or between one mobile and a group of mobile stations. From the technical point of view, the control post is a mobile station in a fixed position;
- calling from a mobile station to the switched telephone (PSTN) network and *vice versa*.

6.2 *Characteristics of the mobile station* [CNET, 1984]

The synthesized mobile station has access to 8 channels chosen from the 100 or so available.

The system may include portable stations with a power of 2 W.

If a switched-on mobile station does not reply, the number of the caller is stored in the memory of the called mobile station (dispatch application).

6.3 *Characteristics of base stations*

Each base station is designed to cover an area corresponding to the potential propagation. The mobile station is served in this area only. One base station controls a group of 8 channels. The transmitters are coupled with low losses to a single antenna.

6.4 *Signalling*

The code in Appendix 39 to the Radio Regulations is used for signalling. Signals are sent on a signalling channel transmitting a modulated carrier. In the event of saturation, the common channel may be used as a traffic channel and the first channel that becomes available will then be used as a signalling channel.

The numbering capacity is 1000 codes.

6.5 *Frequency efficiency*

It is estimated that this type of 8-channel network will be able to deal with up to 500 mobile stations, about 100 of them having access to the PSTN.

6.6 *Charging*

The fixed station switching centre contains a device for magnetic recording of charges, which are then processed for itemized billing.

6.7 *Number of subscribers, commissioning*

Since the system was first brought into operation (December, 1982) the number of subscribers in Paris has reached about 3000 (May, 1985).

The system capacity in Paris is approximately 6000.

Systems have also been set up in Marseilles (November, 1984) and Lyon (March, 1985).

6.8 *Special features of the service*

Up to 9 sub-fleets can be established in a group or fleet of mobile stations, and one mobile station can belong to several sub-fleets.

Connection to the network is on an individual basis and can be effected by using a subscriber line to a private automatic switchboard or to the PSTN.

Certain mobile stations in a fleet may have access to the network while other mobile stations in the same group may not have such access.

7. **Trunked mobile dispatch systems in Australia**

In Australia, trunked mobile radio systems have been in operation since 1983 in the 800 MHz band, and since 1987 in the 400 MHz band. Plans are being developed to support trunked systems in both the 80 and 160 MHz bands using 12.5 kHz channelling.

Systems using a single base site are initially licensed to use five channels, with additional channels being assigned only if 350 mobiles are supported, i.e. 70 mobiles per channel. Wide area systems (both simulcast and non-simulcast) may use up to four base sites in any 100 km x 100 km area with an initial 20 radio channels. Expansion beyond 20 channels is permitted only if 2,000 mobiles are supported, i.e. 100 mobiles per channel.

Trunked operators may use any signalling format for system control; for example some users employ a dedicated signalling channel whilst others utilize channels used for voice communication.

Large scale systems to provide national coverage are under consideration.

Interconnection to the PSTN is permitted.

TABLE I

General system specifications

System Specifications	United States	Sweden 3)	Netherlands & United Kingdom	USSR	Japan	France	Australia
Class of emission	F3E, F2D	F3E, F2D	F3E	F3E	F3E, F2D	G3E	F3E, F2D
Transmit frequency band (MHz)				300-340			
- base station	a) 935-940 b) 851-866	76-77.5	201-207 ¹⁾		850-860	200.5-201.3	a) 865-870 b) 415.55-418.05
- control post	896-901 806-821	-	193-199 ¹⁾		905-915	-	820-825 406.10-408.60
- mobile station	896-901 806-821	81-82.5	193-199 ¹⁾		905-915	192.5-193.3	820-825 406.10-408.60
Communication mode	Semi-duplex and full duplex	Semi-duplex	Semi-duplex ²⁾	Semi-duplex	Semi-duplex	Duplex	Two frequency simplex or full duplex
Channel spacing (kHz)	a) 12.5 b) 25	25	12.5	25	12.5	12.5	a) 25 b) 12.5
Total number of channels available for assignment	a) 400 b) 600	-	408 ¹⁾	-	799	72	200
Transmitted power e.r.p. (W)							
- channel equipment	1,000 maximum	25 typical	25 or less	-	40 typical	18 $\pm$ 1 dB	50
- control post	25 typical	-	5 or less	-	10 or less	-	1
- mobile station	25 typical	15 typical	25 or less	-	30 or less	10 $\pm$ 1 dB	25
Typical service area (km-rad)	50	40	15-30	5-15	20-30	15	Single base site = 30 Wide area system = 100
Voice signals							
- type of modulation	FM	FM	FM	FM	FM	FM	FM
- peak deviation (kHz)	a) $\pm$ 2.5 b) $\pm$ 5	$\pm$ 5	$\pm$ 2.5		$\pm$ 2.5	$\pm$ 2.5	a) $\pm$ 5.0 b) $\pm$ 2.5
Control signals							
- type of modulation	FSK	MSK-FM	FFSK	-	MSK-FM	Multi-tone	a) FSK b) Multi-tone sequential
- peak deviation (kHz)	a) $\pm$ 1.5 b) $\pm$ 3	$\pm$ 3.5	$\pm$ 2.5	-	$\pm$ 2.5	-	$\pm$ 2.5 $\pm$ 2.0
- code form	Various	NRZ code	NRZ code	-	NRZ code	-	Various -
- transmission speed (bit/s)	Various	1,200	1,200	-	1,200	-	Up to 9,600 -
Error correction code	Various	63,48	CRC	-	Hagelbarger	-	Various -

Notes:

- 1) Applicable to the United Kingdom only. In Germany (Federal Republic of) and Netherlands, base stations operate in the frequency band 420 - 430 MHz and control posts and mobile stations in the band 410 - 420 MHz. In Finland the bands are 446.5 to 449 MHz and 440 to 442.5 MHz with up to 200 channels.
- 2) The duplex mode is available in Finland.
- 3) An example of MOBITEK implementation in Sweden.

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REPORT 901-2

**FREQUENCY ASSIGNMENT METHODS FOR TRUNKED
MOBILE RADIO SYSTEMS**

(Question 37/8)

(1982-1986-1990)

1. Introduction

This Report deals with proposals from several administrations on the assignment of channels to trunked systems so as, among other reasons, to reduce intermodulation product interference between systems and within the trunked system itself.

2. Channel groupings with fixed channel separations

In this method, used in the 400 MHz band in Canada and in the 800 MHz band in Canada and the United States of America, channels are grouped into blocks and each channel is separated from the others in the block by a fixed number of channels of nominal channel spacing (usually 25 kHz). Consequently each successive channel block is offset by one channel. Sub-blocks can also be identified to provide flexibility in the number of channels assignable to any individual system. Minimum separations between assigned channels within a trunked system are 250 kHz and 100 kHz in the 800 MHz and 400 MHz bands respectively.

Table I gives an example of a channelling plan, used in the United States of America in the 800 MHz band, using a 200-channel allocation. This total allocation of 200 channels could be divided into 10 blocks of 20 channels with 10-channel spacing between frequencies within the block. This arrangement also sub-divides each 20-channel block into 5-channel groups with 40-channel spaces between successive frequencies in each 5-channel group. In addition, these 5-channel groups are arranged into 4 blocks with two 20-channel spacings and one 10-channel spacing between 5-channel groups. This illustration also offsets successive 20-channel blocks by 1 channel to form ten 20-channel blocks.

TABLE I — *Frequency assignment plan for trunked systems in the 800 MHz band in the United States of America*

Block	Channels	Block	Channels
1	1-41-81-121-161 21-61-101-141-181 11-51-91-131-171 31-71-111-151-191	6	6-46-86-126-166 26-66-106-146-186 16-56-96-136-176 36-76-116-156-196
2	2-42-82-122-162 22-62-102-142-182 12-52-92-132-172 32-72-112-152-192	7	7-47-87-127-167 27-67-107-147-187 17-57-97-137-177 37-77-117-157-197
3	3-43-83-123-163 23-63-103-143-183 13-53-93-133-173 33-73-113-153-193	8	8-48-88-128-168 28-68-108-148-188 18-58-98-138-178 38-78-118-158-198
4	4-44-84-124-164 24-64-104-144-184 14-54-94-134-174 34-74-114-154-194	9	9-49-89-129-169 29-69-109-149-189 19-59-99-139-179 39-79-119-159-199
5	5-45-85-125-165 25-65-105-145-185 15-55-95-135-175 35-75-115-155-195	10	10-50-90-130-170 30-70-110-150-190 20-60-100-140-180 40-80-120-160-200

Tables II and III give channelling plans used in Canada for the 800 MHz and 400 MHz bands respectively. Expansion beyond 50 channels would utilize an approach ensuring progressive deployment of trunked systems, while at the same time providing the flexibility of reserving part of the spectrum for future expansion of a given trunked system to multiple blocks of five channels each or the introduction of new technologies that might require different channel bandwidths and channelling plan.

With the methods outlined above, intra-system interference is minimized, in particular since intermodulation products coincide with other frequencies in the same channel block and do not fall in between the channels. The signal from a multi-channel base station on a particular frequency will at any point in the coverage area exceed intermodulation products falling in the same channel, since they result from transmissions from the same base station (but see also Report 739).

TABLE II — *Frequency assignment plan for trunked systems in the 800 MHz band in Canada*

Block	System	Channels
1	1	1 - 11 - 21 - 31 - 41
	2	2 - 12 - 22 - 32 - 42
	3	3 - 13 - 23 - 33 - 43
	4	4 - 14 - 24 - 34 - 44
	5	5 - 15 - 25 - 35 - 45
	6	6 - 16 - 26 - 36 - 46
	7	7 - 17 - 27 - 37 - 47
	8	8 - 18 - 28 - 38 - 48
	9	9 - 19 - 29 - 39 - 49
	10	10 - 20 - 30 - 40 - 50

TABLE III — *Frequency assignment plan for trunked systems in the 400 MHz band in Canada*

System	Channels
1	1 - 5 - 9 - 13 - 17
2	2 - 6 - 10 - 14 - 18
3	3 - 7 - 11 - 15 - 19
4	4 - 8 - 12 - 16 - 20

REPORT 1153

FUTURE PUBLIC LAND MOBILE TELECOMMUNICATION SYSTEMS

(Question 39/8)

(1990)

1. Introduction and summary

Interim Working Party 8/13 was set up by Decision 69 in 1985 with the task of investigating the form of Future Public Land Mobile Telecommunication Systems (FPLMTS) with particular regard to the overall objectives, suitable frequency bands and the degrees of compatibility or commonality and which has resulted in the preparation of a new Recommendation 687 on FPLMTS.

This Report summarizes the result of the study.

From the beginning of the study it was apparent that a light-weight personal pocket radio would be a dominant feature of FPLMTS. Most administrations advocated the desirability of such a personal station (PS) being usable anywhere world-wide, with the inherent implication of a requirement for a single radio interface standard. However, considerations of the varieties of radio coverage scenarios for personal communications (from very short range ones for use within buildings or urban areas through medium range suburban and longer range rural coverage, together with the desire for links to aircraft or satellites) indicated that a number of RF transmission requirements and frequency bands are likely to be required. Moreover, a tandem link could well be involved from the PS via another mobile connection. There is a possibility that the final link to the PS could be the same. However, the situation of the vehicle mounted mobile accessing terrestrial links is different, and some administrations emphasized the much reduced requirements for inter-regional roaming (although international roaming within a region will be a requirement). Hence, it was agreed that greater flexibility for regional standards for roaming is needed. Nevertheless, it was also recognized that a common world-wide radio interface and frequency band may increase the possibility of reduced costs of network and terminals and availability of equipment.

The frequency allocations made by WARC-MOB-87 for the land mobile satellite service also influenced the study on FPLMTS, with the realization of the opportunities that satellite links could bring to such systems.

A number of objectives for FPLMTS has been agreed. It was found desirable to partition them into 19 primary ones and 10 secondary. The secondary status allows more freedom of choice for implementation.

The FPLMTS could be considered as either an adjunct to or an integral part of the fixed network (PSTN/ISDN). The particular characteristics of the radio channel will, however, impose some constraints on the services offered as well as providing opportunities for others such as dispatch, group call, etc. Service should be provided to a variety of mobile terminals ranging from the PS to vehicle mounted mobile stations (MS). Additionally FPLMTS should be usable as a temporary or permanent substitute for the fixed network. The standards of services provided should be comparable with those of the fixed network, and particular attention will have to be paid to the need for enhanced privacy/security on the radio channel (not only for speech, but also for billing, etc.).

Two complementary scenarios are postulated. The first is concerned with land-based systems. This involves the PS operating in five modes, i.e. accessing a private "domestic" base, accessing a private office base, accessing a public personal base, accessing a public mobile base directly or communicating via an MS. The MS is linked to a base station (BS) for its mobile service and operates either in its own right or as a relay for a PS. The second scenario involves satellite links to mobile earth stations (MES) either in their own right or for relay to a PS. Also envisaged are satellite and terrestrial paging systems.

An architecture is defined that shows interfaces within the FPLMTS and with the fixed network. The functionality of the mobile equipment is also indicated.

The factors which affect the choice of desired frequency bands of operation are discussed. Starting with services to be offered and traffic models, estimates are made of the requirements of such factors as modulation, coding, re-use, etc., in order to assess the bandwidth requirement. This Report also contains comments on the choice of frequency bands including any biological factors.

The desire for compatibility within and between FPLMTS is commented on in several sections of the present Report as are the benefits of commonality. Considerations of compatibility with the ISDN give rise to the concept of mapping its B (bearer) and D (signalling) channels onto I (information) and C (control) channels for FPLMTS.

2. Objectives for future public land mobile telecommunication systems

FPLMTS should aim to achieve the following objectives which are classed as primary and secondary. Secondary objectives are those which some administrations or regions may not wish to include. Within each class the numbers are for reference purposes only and do not convey an order of priority.

PRIMARY:

- P1. To make available voice and non voice telecommunication services to users who are on the move or whose location may change (mobile users).
- P2. To provide these services over a wide range of user densities and geographic coverage areas.

- P3. To make efficient and economical use of the radio spectrum consistent with providing service at an acceptable cost.
- P4. To provide, as far as practical, a service of high quality and integrity, comparable to the fixed network.
- P5. To accommodate a variety of mobile terminals ranging from those which are small enough to be easily carried on the person (the personal pocket radio), to those which are mounted in a vehicle.
- P6. To provide a framework for continuing extension of mobile network services, and access to services and facilities of the fixed network (PSTN/ISDN) subject to the constraints of radio transmission, spectrum matters and system economics.
- P7. To admit the connection of mobile users to other mobile users or fixed users, using the fixed network (PSTN/ISDN) or other telecommunication networks as appropriate.
- P8. To permit the use of the FPLMTS for the purpose of providing its services to fixed users, under conditions approved by the appropriate national or regional authority, either permanently or temporarily, either in rural or urban areas.
- P9. To admit the provision of service by more than one network in any area of coverage.
- P10. To allow mobile and fixed network users to use the services irrespective of location (i.e. national and international roaming).
- P11. To provide for the required user authentication and billing functions.
- P12. To provide for unique user identification and PSTN/ISDN numbers in accordance with appropriate CCITT Recommendations.
- P13. To support integrated communication and signalling.
- P14. To establish signalling interface standards in terms of the Open System Interconnection (OSI) model.
- P15. To provide an open architecture which will permit the easy introduction of technology advancements, as well as different applications.
- P16. To enable each mobile user to request particular services, and initiate and receive calls, as desired.
- P17. To allow the co-existence with, and interconnection with, mobile systems which use direct satellite links taking into consideration CCITT Recommendation E.171.

- P18. To provide for a unique equipment identification scheme.
- P19. To provide a modular structure which will allow the system to start from as small and simple configuration as possible and grow as needed, both in size and complexity within practical limits.

SECONDARY:

- S1. To provide for additional levels of security (for voice and data services) compared to that contained in P4. In addition, to allow for the provision of end-to-end encryption for voice and data services.
- S2. To provide service flexibility which permits the optional integration of services such as mobile telephone, dispatch, paging and data communication, or any combination thereof.
- S3. To provide an indication to the paying party of added call charges, e.g. due to roaming.
- S4. To support terminal interfaces which allow the alternative use of terminal equipment in the fixed ISDN network.
- S5. To support equipment and component design that can withstand typical rural conditions (rough roads, dusty environment, extreme temperatures and humidity, etc.).
- S6. To allow the system to be configured for special conditions where mobility between cells, or even within a cell, is not required; ~~or where a high traffic~~ per user is required.
- S7. To take account of the communications requirements for road traffic management and control systems.
- S8. To accommodate the use of repeaters for covering long distances between terminals and base stations, providing this does not constrain the specification of the radio interfaces.
- S9. To allow the connection of PABX's, or small rural exchanges to mobile stations.
- S10. To allow for an extension of the cell size in rural or remote areas, providing this does not constrain the specification of the radio interfaces.

3. Services

3.1 *Introduction*

The concept of a service conferring mobility is one which is desired by users whether they are served by the mobile network or the fixed one. Thus it is part of future (fixed) telecommunications systems and FPLMTS. The degree of mobility is different between the networks:

Mobile Network Mobility

Mobility involves the ability of the user to be in continuous motion (including the stationary condition) whilst accessing and using telecommunication services. This requires the telecommunication services to be available throughout a spatial volume and ideally at all times.

Fixed Network Mobility

Mobility is conferred by flexibility of access to telecommunication service provision, which is available at discrete locations, in such a way that the user identifies with, and may configure, any one of these fixed terminals to meet his requirements. These requirements may then be relocated from terminal to terminal without restriction or extent.

FPLMTS will give continuous mobility within a coverage area whilst the fixed network will give discrete mobility.

FPLMTS can also benefit from this flexibility of access concept. These "higher layers" of service are offered by FPLMTS (mobility, flexibility of access) in addition to the conventional telecommunication services.

Services which previously could be considered as specific to separate mobile (radio) systems, such as dispatch services, could now be included in the supplementary telecommunication services of FPLMTS.

Services will be described under seven main headings: General Requirements, Mobility, Service Access, Telecommunication Services, Basic Speech Quality, Security, Quality of Service.

Note: The CCIR considers that the differences from a user perspective between the services provided by the mobile and fixed networks will become less

into the future. It will become increasingly difficult for users to differentiate between these two networks.

3.2 *General requirements*

FPLMTS may be either an adjunct to or an integral part of the PSTN/ISDN. The services offered in the PSTN/ISDN and other public networks should as far as possible be available in FPLMTS bearing in mind the differences in the characteristics of the fixed network and mobile radio environment. FPLMTS may also offer additional services, taking into account the special nature of mobile communications.

FPLMTS should be designed to allow international operation and automatic roaming of mobile subscribers and stations. The degree to which the roaming facility between networks (or countries) shall be automatic is not yet decided. There will also be a need for FPLMTS to have the capability to restrict roaming and access to and from unauthorized mobile subscribers or stations.

FPLMTS should be designed in such a way that the location of a roaming user should not be disclosed to a calling user without the agreement of the called party. Conversely, the location of a calling user, when roaming, should not be disclosed to the called party without the calling party's agreement.

The services and facilities offered in FPLMTS may not always be the same over all the system.

The FPLMTS should be designed so that under favorable circumstances services requiring high information rates can be provided. For example if the mobile terminal is stationary, or moving slowly, and is close to a base station which is not heavily loaded otherwise, it may be possible to provide services up to the primary ISDN rate of 1536 or 1920 kbit/s. Under less favourable circumstances (e.g. mobile terminals at greater distances from the base station or when the base station is carrying a heavy load of traffic) the services may be restricted to those using lower rates.

It should be possible for terminal stations in FPLMTS to be used in the maritime and aeronautical environment, to the extent permitted by national or international regulatory authorities.

FPLMTS should be capable of providing service to a variety of mobile terminals ranging from those which are small enough to be easily carried on the person to those which are based in a vehicle [De Brito, 1981].

It should be possible for FPLMTS to be used as a temporary or permanent substitute to fixed networks where fixed network facilities are limited or not available, or more generally where reasons of convenience or economics make this desirable. The system should then be capable of adaptation to these conditions (i.e. higher traffic per user, no movement between cells or even within a cell).

3.3 *Continuous mobility*

3.3.1 *Definition*

Continuous mobility is the ability to access and use telecommunication services whilst in continuous motion (including whilst stationary), by the provision of telecommunication facilities which are available everywhere within a spatial volume and ideally at all times.

3.3.2 *Constraints.*

Within this definition the degree of continuous mobility may be constrained. In general the constraints will depend on the type of terminal used to access the service. The following identifies the constraints on the mobility of the terminals in order to have the FPLMTS services available.

- (i) The personal station PS (e.g. pocket-sized portable) can be used in various ways with identifiable constraints. Within buildings the rate of motion will be limited to walking speed. A similar speed constraint applies when the PS is used in a moving vehicle. In this case, the vehicle can provide a tandem link to the network via an MS or MES located in the vehicle. When the PS is used directly outdoors, consideration needs to be given to its use for rates of motion up to that of slow-moving vehicles in urban and suburban areas.
- (ii) The mobile station MS (e.g. vehicle mounted): within defined coverage areas (e.g. urban, suburban, rural) for rates of motion of say high speed trains.
- (iii) The mobile (satellite) earth station MES: within defined coverage areas (e.g. rural and remote areas, with restricted availability in urban and suburban areas) for rates of motion up to say supersonic aircraft speeds.

- Notes:
- (1) Some administrations may wish to implement networks which allow the PS to have the mobility of the MS. The mobility constraints will also vary depending on the actual implementation technologies employed.
 - (2) It will be necessary to consider how to provide telecommunication services with aircraft over land which does not use an MES (see Report 1051).

3.4 *Service access*

3.4.1 *Introduction*

The FPLMTS can provide an additional degree of mobility and flexibility to the user when the Personal Telecommunication Number (PTN), as defined below, is used.

In most of the existing networks, telecommunication services are provided to the users using the identity of the equipment to be used. Often several users access the same equipment on a shared basis.

The implementation of the concepts presented in this section makes this sharing more formal: the various users that may access a particular equipment at a particular moment will not always "roam" together, making it necessary therefore to have a differentiated management for each one. As a result, precise definitions of all the concepts handled in this section have to be given.

3.4.2 *Definitions and comments*

All the following definitions need to be considered together; a number of them have been based on or derived from CCITT definitions.

The order in which these definitions are presented does not depend on the importance of them. The order was chosen to provide a progressive introduction to the various concepts being presented. Some definitions have been reproduced here for completeness and/or to avoid any possible misunderstanding of the new concepts.

Some of these concepts are illustrated in Fig. 1.

3.4.2.1 *User*

a) *Definition*

The user is a person (or a person's logical function) or device/machine designated by the subscriber, individually or by class, as having access to the service and having such authorization, individually or by class, as may be required by the network operator or an authorized agent concerned. (see CCITT Recommendation I.112)

b) *Comments and examples*

As an implication of the concepts presented in this section, it has to be made clear that the term "user" may cover only a particular aspect of those using the network. To illustrate the consequences of this, the following examples of "users" are given:

- a person
- a particular aspect of the activity of a person (the differentiation could be made for billing purposes or for call forwarding purposes), corresponding to e.g. his/her private life, business activity (e.g. driver of a truck)
- a vehicle as such (e.g. for the purpose of managing a fleet of vehicles)
- the load of a vehicle (e.g. for the purpose of managing a fleet of containers).

In many cases the different users presented in the above example could share, for some period of time, the same mobile equipment and then roam in independent ways requiring therefore specific means of identification (see PTN here below).

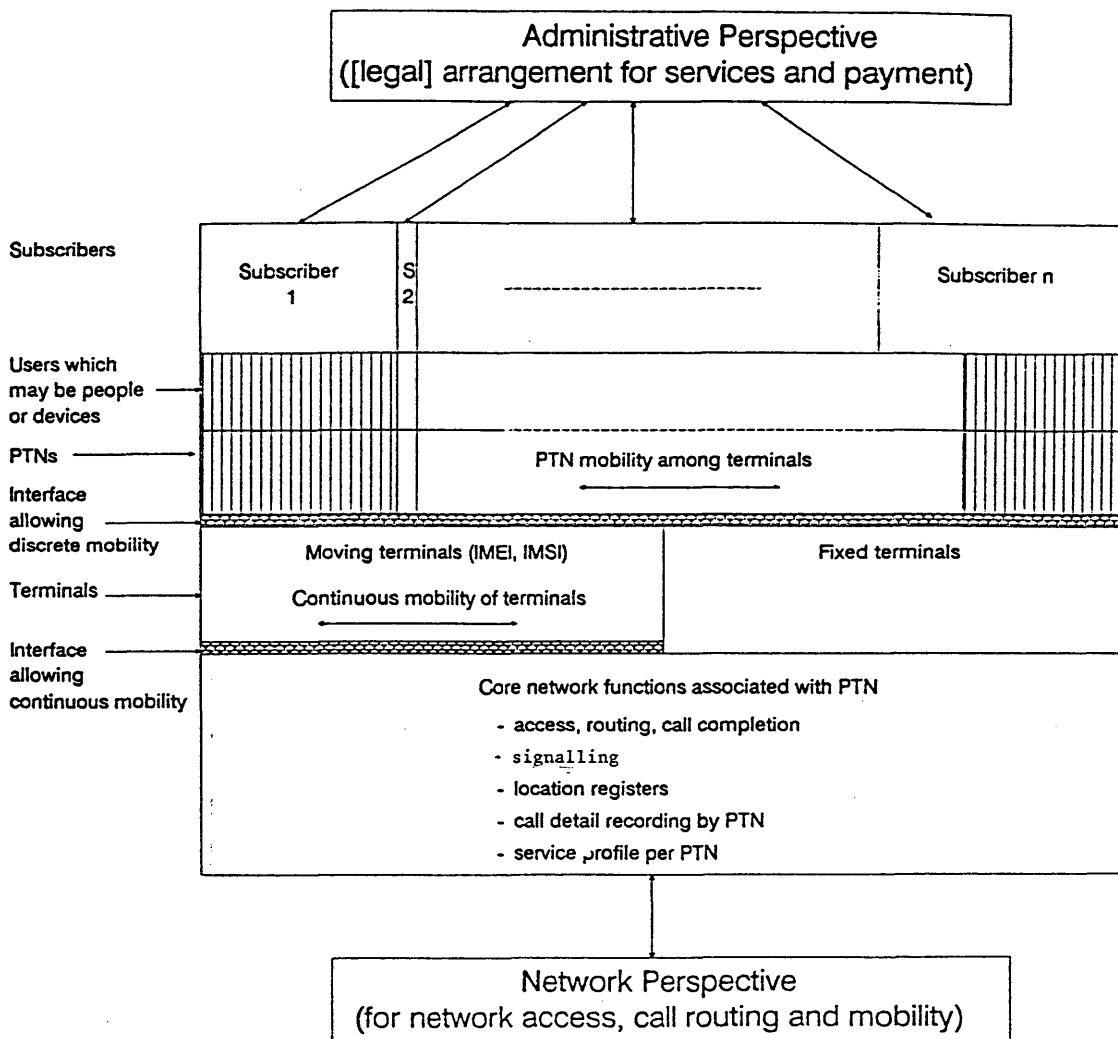


Fig. 1 The PTN Concept

3.4.2.2 *International mobile station equipment identity (IMEI)*

a) Definition

The IMEI uniquely identifies the mobile station as a piece or assembly of equipment.

b) Comments and/or examples

The IMEI could be a serial number of the mobile equipment, attached permanently to it.

3.4.2.3 *International mobile subscriber identity (IMSI)*

a) Definition

The IMSI uniquely identifies the mobile station as authorized by a network operator. It can be mapped directly to the existing directory number. (See CCITT Recommendation E.212)

b) Comments

In many cases, there is a one to one correspondence, at a particular moment in time, between IMEI and IMSI.

3.4.2.4 *PTN (Personal Telecommunication Number) and PTN service*

a) Definition

The PTN Service is a service supported by the FPLMTS that provides to the network the capability of identifying a particular user to be served, independently of the identity of the physical equipment (both IMEI and IMSI) serving that user.

The identifier to be used to access a particular user is referred to as the PTN of that user.

b) Comments and examples

There is a one to one correspondence between users (see definition 3.4.2.1) and PTNs.

Several PTNs may be accessed through the same terminal unit.

One PTN may be served by different terminal units for different services.

One subscriber may have several PTNs, each authorized to separate users.

By using the PTN service, users will be able to benefit from telecommunication services as independently as possible from both the physical equipment to be used and the network transporting the information. In particular, there should be no need for a calling party to know with which network the mobile is able to communicate at that particular moment.

Using the PTN, it is expected that the user can, in principle, obtain telecommunication services via telecommunication facilities belonging to any public telecommunication network (FPLMTS, PLMN, ISDN, PSTN, etc.) or associated private networks (i.e. private networks supporting the FPLMTS and PTN services).

This requires that the desired services are supported by the particular combination of network and terminal.

3.4.2.5 *Log-in*

a) Definition

In this Section 3 log-in is the action by which the user indicates to the network that all or part of the calls addressed to him (to his PTN) are to be sent to a particular telecommunication terminal equipment.

b) Comments and/or examples

In order to get service from a network with a particular physical equipment, a user can log-in using his PTN through that terminal (see 3.4.2.6).

This log-in can be done automatically when an intelligent device is connected to the terminal equipment, or manually by the user (by keying in the PTN plus some password or by the application of an intelligent/memory device).

In some cases a subscriber or his representative may also perform the log-in for a user.

The same telecommunications equipment may be shared by several users (corresponding to several PTNs). Therefore several users may simultaneously be logged in through it.

3.4.2.6 *PTN service profile*

a) Definition

The PTN service profile is the information relative to one particular PTN, that is needed in order for the network to be able to complete the call processes involved with that PTN. PTN service profiles complement any other network profiles or discriminations.

b) Comments and/or examples

There is a one to one relation between PTNs and PTN service profiles.

The services to be provided to a particular user depend upon the PTN service profile associated with his PTN.

Part of the PTN service profile will depend upon information supplied through the subscription, while another part may be updated dynamically by the user.

By using or modifying his PTN service profile, a user can also request to a network that some services addressed to his PTN be served by specified terminals (e.g. calls to be forwarded to another terminal).

In some cases a subscriber or his representative may modify the PTN service profile for a user.

The PTN service profile can also be used for selectively screening incoming calls.

3.4.2.7 *Personal Identification Number (PIN)*

a) Definition

Confidential information which may be used for the authentication of the user.

3.4.3 *Features of the PTN service*

A user can be accessed through the usage of his PTN.

The assigned PTN is independent of:

- (i) physical network, including access means (e.g. radio, wire) and terminals

- (ii) services (e.g. voice, data) and
- (iii) geographic location.

The PTN service will provide the following:

- (i) service access
- (ii) location registration
- (iii) pursuit routing, alerting and call delivery
- (iv) authentication and
- (v) charging.

The PTN is essential for automatic inter-network roaming and inter-terminal roaming.

3.4.4 *Format of the PTN*

The PTN format is subject to CCITT studies. One example of PTN format can be as follows :

CC + SI + OI + PN

CC: Country code

SI: Service Indicator (flag "PTN" or "not PTN")

OI: Operator Identity (Administration or RPOA)

PN: Personal number

3.5 *Telecommunication services*

3.5.1 *Introduction*

The telecommunication services to be supported by the FPLMTS should reflect the needs of users to communicate in a mobile environment.

In this section the services are firstly classified from the user perspective in terms of the categories and classes of services and types of information to be communicated.

The combinations of these factors lead to specific network requirements in the form of offered teleservices and bearer capacity. The network services are subsequently defined in terms of specific attributes.

Some typical services so defined are summarized in Table I.

3.5.2 Service categories and classes

Three main service categories have been identified: mobility services, interactive services and distribution services as shown in Fig. 2. As indicated in this figure, these three service categories are further subdivided into nine service classes.

The mobility services are those which specifically result from the user need for mobility. The interactive and distribution services are closely aligned to those defined by CCITT for the fixed network in Recommendation I.121. The addressed one-way services have been added to the distribution service category.

Table I
FPLMT Services

USER PERSPECTIVE				FPLMTS NETWORK REQUIREMENTS	
SERVICE CATEGORIES AND CLASSES	TYPE OF INFORMATION	SERVICE DESCRIPTION	FPLMTS APPLICATION EXAMPLE	TELESERVICE	BEARER CAPABILITY
1. Mobility Services 1.1. Locating	Voice	Voice announcement indicating location of user	- *		
	Audio	*	-		
	Text	Text information indicating location of user	- location information to dispatcher	- text	≤ 8 kbit/s
	Image	Image data indicating location of user	- Mobile navigation to vehicles or dispatcher	- data	
	Video	*	-		
	Signalling	Signalling information based on user location	- presentation of location specifics to enable user to reconfigure equipment or service profile	- data	≤ 8 kbit/s
1.2 PTN	Voice	*			
	Audio	*			
	Text	*			
	Image	*			
	Video	*			
	Signalling	Sending of users PTN or called PTN for call routing	- dialling and personal identification	- data	≤ 8 kbit/s
1.3 Dynamic Reassignment	Voice	*			
	Audio	*			
	Text	*			
	Image	*			
	Video	*			
	Signalling	Allocation of communication facility to user based on PTN profile	- facility availability signalling (available, busy, or unavailable)	- data	≤ 8 kbit/s

* For further study to determine potential new FPLMTS user services.

Table I (cont.)
FPLMT Services

USER PERSPECTIVE				FPLMTS NETWORK REQUIREMENTS	
SERVICE CATEGORIES AND CLASSES	TYPE OF INFORMATION	SERVICE DESCRIPTION	FPLMTS APPLICATION EXAMPLE	TELESERVICE	BEARER CAPABILITY
2. Interactive Services 2.1 Conversational (real-time 2-way)	Voice	End-to-end 2-way voice connection	- 2-person telephone call - conference telephone call	- speech	8 - 64 kbit/s
	Audio	End-to-end 2-way audio connection	- audio conference - interactive data using modems or DTMF tones	- programme sound - data	64 - 384 kbit/s 8 - 64 kbit/s
			- Control and monitoring or medical data instruments with A/D and D/A conversion	- data (telemetry and control)	8 - 64 kbit/s
	Text	End-to-end data connection for 2-way presentation of text	- 2-person data call for screen sharing	- text	8 - 64 kbit/s
			- data conference call	- text	8 - 64 kbit/s
	Image	End-to-end 2-way image connection	- Connectionless short message conversation	- data	8 - 64 kbit/s
			- 2-way fax	- telefax	8 - 64 kbit/s
	Video	End-to-end 2-way video connection	- 2-way compressed video	- video phone	64 - 1920 kbit/s
	Signalling	End-to-end 2-way signalling connection	- remote control and status acquisition	- signalling	connectionless packet
Note: - There may be various combinations of services which are not symmetric in both directions but which together constitute a conversational service. - There may also be combinations of classes of services in a given use of telecommunication services.					
2.2 Messaging (store and forward)	Voice	Store and forward voice	- voice mail box	- speech	8 - 64 kbit/s
	Audio	Store and forward audio	- *		
	Text	Store and forward data/text	- e-mail - text paging	- text - text	100 bits/s - 64 kbit/s low bit rate
	Image	Store and forward images	- fax mailbox	- telefax	8 - 64 kbit/s
	Video	Store and forward video	- video mail	- video mail	64 - 1920 kbit/s
	Signalling	Store and forward signalling	- call alerting - calling number ID	- data - data	100 bits/s - 64 kbit/s 8 - 64 kbit/s
* For further study to determine potential new FPLMTS user services.					

**Table I (cont.)
FPLMT Services**

USER PERSPECTIVE				FPLMTS NETWORK REQUIREMENTS	
SERVICE CATEGORIES AND CLASSES	TYPE OF INFORMATION	SERVICE DESCRIPTION	FPLMTS APPLICATION EXAMPLE	TELESERVICE	BEARER CAPABILITY
2.3 Retrieval (access to stored information)	Voice	Access to stored voice messages	- telephone access to a voice menu - voice based stock quotation	- speech retrieval - speech retrieval	8 - 64 kbit/s 8 - 64 kbit/s
	Audio	Access to stored audio messages	- programme sound	- audiotex	64 - 384 kbit/s
	Text	Access to stored text messages	- electronic bulletin board	- teletex	8 - 64 kbit/s
	Image	Access to stored images	- real estate home listings with pictures	- telefax/image retrieval	8 - 64 kbit/s
	Video	Access to stored moving pictures	- teleshopping	- video	64 - 1920 kbit/s
	Signalling	Access to stored signalling	- network feature activation (call forwarding, etc.)	- data	8 kbit/s
			- polling of telemetering station	- data	8 - 64 kbit/s
3. Distribution Services					
3.1 Distribution (real-time broadcasting or narrowcasting)	Voice	Voice message broadcasting or narrowcasting	- public service announcements - taxi dispatching	- speech - speech	8 - 64 kbits 8 - 64 kbit/s
	Audio	Audio broadcasting	- *		
	Text	Text broadcasting or narrowcasting	- *		
	Image	Image broadcasting	- *		
	Video	Video broadcasting	- *		
	Signalling	Signalling information broadcasting	- *		
3.2 Distribution service with user presentation control	Voice	*			
	Audio	*			
	Text	Selection of one text message out of several being broadcast simultaneously on a channel	- city information service	- text	8 - 64 kbit/s
	Image	Selection of one image (etc.)	- teleshopping	- telefax/image	8 - 64 kbit/s
	Video	Selection of one moving picture (etc.)	- teleshopping	- compressed video	64 - 1920 kbit/s
	Signalling	*			

* For further study to determine potential new FPLMTS user services.

Table I (cont.)
FPLMT Services

USER PERSPECTIVE				FPLMTS NETWORK REQUIREMENTS	
SERVICE CATEGORIES AND CLASSES	TYPE OF INFORMATION	SERVICE DESCRIPTION	FPLMTS APPLICATION EXAMPLE	TELESERVICE	BEARER CAPABILITY
3.3 Addressed one-way (real time)	Voice	Voice message	- voice paging, individual or group call	- speech	8 - 64 kbit/s
	Audio	*			
	Text	Text message	- radio paging with text display, individual or group call	- data	8 - 64 kbit/s
	Image	Addressed image	- telefax, point-to-point or point-to-multipoint	- telefax	8 - 64 kbit/s
	Video	*			
	Signalling	Signalling message	- radio paging, alerting only	- data	≤ 8 kbit/s
* For further study to determine potential new FPLMTS user services.					

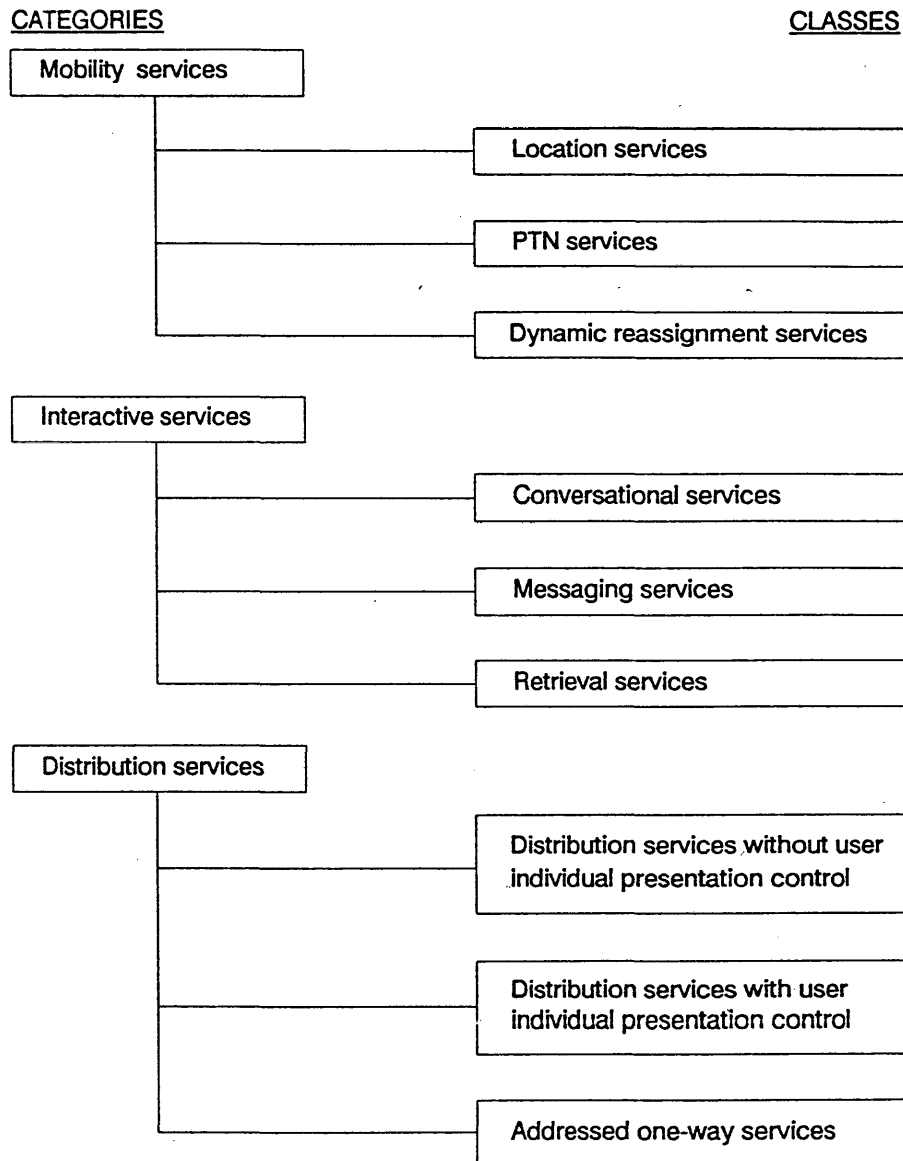


Fig. 2
Classification of mobile services

3.5.3 *Definition of the various service classes*

The definitions for the mobility services are based on the concepts of mobility introduced in 3.1 through 3.4. The definitions for the interactive distribution categories of services which follow are extracted from CCITT Recommendation I.121 and are presented here for completeness. The definition of the addressed one-way services are based on actual network services, such as radio paging with real time voice.

3.5.3.1 *Location services*

For successful operation, FPLMTS must know the location of the mobile unit to some level of accuracy. This capability is the cornerstone for world-wide personal telecommunications. Location information may be provided to authorized users by FPLMTS, e.g. to relevant authorities in cases of emergency calls or for vehicular traffic management. In order to protect the privacy of the user, the access to location information must be restricted to specific applications authorized by the customer and the administration concerned.

3.5.3.2 *PTN services*

The PTN services are defined in 3.4. They provide flexible access to either the fixed network or FPLMTS for users who require mobility as part of their communication needs.

3.5.3.3 *Dynamic reassignment services*

One of the essential features which mobility offers a user is the ability to receive and/or originate a call wherever he may be located. This concept implies the need for dynamic call reassignment services to track the whereabouts of the user in both space and time.

3.5.3.4 *Conversational services*

Conversational services in general provide the means for bidirectional dialogue communication with real-time (no store-and-forward) end-to-end information transfer from user to user or between user and host (e.g. for data processing). The flow of the user information may be bidirectional symmetric, bidirectional asymmetric and in some specific cases (e.g. such as video surveillance), the flow of information may be unidirectional combined with a signalling/control link in the reverse direction.

Examples of mobile conversational services are telephony, videotelephony, teleconference and data transmission, and short connectionless text messages.

3.5.3.5 *Messaging services*

Messaging services offer user-to-user communication between individual users via storage units with store-and-forward, mailbox and/or message handling (e.g. information editing, processing and conversion) functions.

Examples of mobile messaging services are message handling services and mail services for images, text and audio information.

3.5.3.6 *Retrieval services*

The user retrieval services can retrieve information stored in information centres and in general provided for public use. This information will be sent to the user on his demand only. The information can be retrieved on an individual basis. Moreover, the time at which an information sequence is to start is under the control of the user.

Examples are mobile retrieval services for image, audio information, and archive information.

3.5.3.7 *Distribution services without user individual presentation control*

These services include broadcast services. They provide a continuous flow of information which is distributed from a central source to an unlimited number of authorized receivers connected to the network. The user can access this flow of information without the ability to determine at which instant the distribution of a string of information will be started. The user cannot control the start and order of the presentation of the broadcast information. Depending on the point of time of the user's access, the information may or may not be presented from its beginning.

Examples are mobile broadcast services for image, text and audio programs.

3.5.3.8 *Distribution services with user individual presentation control*

Services of this class also distribute information from a central source to a large number of users. However, the information is provided as a sequence of information entities (e.g. frames) with cyclical repetition. So, the user has the ability of individual access to the cyclical distributed information and can control the start and order of presentation. Due to the cyclical repetition, the information entities selected by the user will always be presented from its beginning.

One example of such a service is teletext.

3.5.3.9 *Addressed one-way services*

Services of this class distribute information from a control source to a large number of users. However each message is addressed to one or more users in the group.

Examples of such services are one-way voice, text, audio and video, and radio paging without receipt.

3.5.4 *Types of information*

The following types of information should be capable of being carried by the FPLMTS:

Voice	(A or μ -law)
Audio	(3.1/7/15 kHz, including voice band modems, DTMF, etc.)
Text	(ASCII, etc.)
Image	(Variable resolution up to 1000 dpi, color)
Video	(Limited motion color)
Signalling	(Variable data rate)
Data	(Asynchronous and synchronous at various bit rates)

It should be possible to carry combinations of these information types as appropriate.

3.5.5 *Description of network services*

Telecommunication services can be divided into two broad categories:

- basic services and
- supplementary services

Basic services are modified or supplemented by supplementary telecommunication services. Consequently, supplementary services cannot be offered to a customer as stand-alone services. They must be offered together with or in association with a basic telecommunication service. The same supplementary service may be common to a number of telecommunication services.

These basic and supplementary services are further subdivided into:

- bearer services
- teleservices

Bearer services provide the capability for the transmission of signals between user-network interfaces. Teleservices provide the complete capability including terminal equipment functions, for communication between users according to protocols established by agreement between administration and service providers. The teleservices utilize the capabilities of one or more bearer services.

For each type of service, bearer, teleservice and supplementary service there will be a set of attributes which specify the characteristics of the service.

The services which should be offered by the FPLMTS are described in the sections which follow. Many of them are based on CCITT definitions contained in the I.200 series of recommendations. The text identifies aspects of the services which may require modification to enable support within the FPLMTS. FPLMTS, in their ability to support subscriber mobility, will support additional services that do not exist within the fixed network.

As new telecommunication services are defined in the ISDN, it should be verified whether these can be made available to mobile users by means of new FPLMTS services or via service/network interworking with existing FPLMTS services.

Due to spectrum limitations, some of the services available in the ISDN may have to be offered in a slightly different manner in the FPLMTS (e.g. 64 kbit/s ISDN speech to be transmitted at bit rates lower than 16 kbit/s.)

3.5.5.1 *Bearer services*

The constraints imposed by radio transmission, spectrum availability and economics make it difficult to provide full ISDN B channels and D channels with an acceptable grade of service. Lower bit rate channels (or sub-rate channels) should be defined for the FPLMTS equivalent to the B (bearer) channel and the D (signalling) channel. In the FPLMTS, the I (information) and the C (control) channels have been identified to refer to the lower rate channels offering a functionally equivalent service to the B and D channels respectively.

Provision of new bearer services is possible when the protocol is based on the D channel protocol.

3.5.5.2 *Circuit - mode bearer services*

Bearer services can be defined over a range of bit rates.

Circuit-mode N kbit/s unrestricted,
8 kHz structured bearer service category,
where for example N = 8, 16, 32, 64, (2x64), 384, 1536 and 1920.

Bearer services with data rates < 64 kbit/s (sub-rate channels) can be used. Bearer services at data rates < 8kbit/s require further study but may be important in satellite applications.

ISDN bearer services described in CCITT Recommendations I.231.1, I.231.5, I.231.6, I.231.7 and I.231.8 correspond to data rates of 64, (2x64), 384, 1536 and 1920 kbit/s respectively.

The data rates for circuit-mode bearer services depend on the mix of services, spectrum availability and market demand and requires further study. It is likely that in initial implementations data rates greater than 64 kbit/s will be restricted to indoor environments. As technology improves higher bit rates may be possible and coverage may be increased.

Circuit-mode N kbit/s, $N = 8, 16, 32$ and 64
8 kHz structured bearer service category,
usable for speech information transfer ISDN service.

CCITT Recommendation I.231.2 corresponds to $N = 64$ kbit/s.

This bearer service is similar to the unrestricted service discussed above except that FPLMTS and the fixed network may use processing techniques appropriate for speech. Hence, bit integrity is not assured.

Circuit-mode 64, 32, 16 kbit/s and lower rates,
8 kHz structured bearer service category,
usable for 3.1 kHz audio information transfer

CCITT Recommendation I.232.3 defines a 64kbit/s service. Performance at 32 kbit/s, 16 kbit/s and lower rates requires further study.

This bearer service is suitable for modems and telefax applications as well as speech.

3.5.5.3 *Packet-mode bearer services*

Virtual call and permanent virtual circuit bearer service category-B channel (CCITT Recommendation I.232.1) or I channel

These services offer access to packet-switched data services.

The user information access channel within the FPLMTS may be a sub-rate channel. Maximum packet size and quality of service may be restricted. Further study is required to determine how the benefits of packet service can be given to FPLMTS data users.

Connectionless packet bearer service category on a D channel (CCITT Recommendation I.232.2) or C channel

The access channel in the FPLMTS may be a sub-rate D channel.

User signalling bearer services category (CCITT Recommendation I.232.3).

3.5.5.4 *Teleservices*

FPLMTS should support the teleservices listed below. These include teleservices derived from Figure C-2/I.210 of CCITT Recommendation I.210 and a variety of additional services.

Many of the services listed are based on definitions from CCITT Recommendations with appropriate modifications to suit the FPLMTS.

a) **Telephony**

The telephony service provides users of the FPLMTS with the ability for real-time two-way speech conversation. The service will be based upon CCITT Recommendation I.241.1 appropriately modified for application to the FPLMTS. Application to FPLMTS needs further study.

b) **Programme sound**

The programme sound service provide users of the FPLMTS with the ability to deliver information with sound programme quality.

c) **Text**

Text teleservices are outlined in the CCITT X.400 series of Recommendations Application to FPLMTS needs further study.

d) **Teletex**

The teletex service provides users of the FPLMTS with the ability to exchange office correspondence in the form of documents containing teletex coded information on an automatic memory-to-memory basis. (CCITT Recommendation I.241.2)

e) **Paging**

FPLMTS should be able to provide paging services integrated with telephony and data services, to the extent permitted by individual administrations. Mobile satellites may offer wide area paging beyond the range of terrestrial systems.

FPLMTS should be able to provide paging services in several different modes:

- "open loop transmission" (i.e. page sent to a device that does not transmit any acknowledgment upon reception of the message)

- "closed loop transmission" or "with network acknowledgment" (i.e. page sent to a device that will acknowledge reception)
- "with user acknowledgment" (i.e. the acknowledgment will be sent when the user indicates to the device that he has received the message)

The sender of the page should have the possibility of choosing the preferred mode (whether the sender will receive an acknowledge or not, will depend upon the hardware of the pager and of the propagation conditions).

f) Videotex

The videotex service will provide a retrieval service for text and image information. The service should be based on CCITT I.241.5 Recommendations suitably modified for application to the FPLMTS.

g) Telefax

The telefax service will enable users to exchange correspondence in the form of documents containing telefax coded information on an automatic memory-to-memory basis via the FPLMTS. Standards for this service should be based on CCITT Recommendation I.241.2 suitably modified for application to the FPLMTS.

h) Programme video

The programme video service will enable users to deliver video via the network. It is likely that all programme video carried via the FPLMTS will be compressed.

i) Video surveillance

The video surveillance service will provide users of the FPLMTS with the ability to deliver video for remote surveillance purposes.

j) Video telephony

The video telephony service will provide users of the FPLMTS with the ability for real-time two-way combined speech and video conversation via the network. It is likely that all video telephony carried via the FPLMTS will involve compressed video.

k) Data

Several synchronous and asynchronous data services are standardized on the PSTN including 300, 1200, 2400, 4800 and

9600 bit/s. Issues needing resolution include interfaces according to U, I, and X Recommendations, interworking with fixed networks and error control on the radio link.

The FPLMTS should be designed so that under favourable circumstances services requiring high bit rates can be provided. For instance if the mobile terminal is stationary, or moving slowly, and is close to a base station which is not heavily loaded otherwise, it may be possible to provide services up to the primary ISDN rate of 1536 or 1920 kbit/s. Under less favourable circumstances, e.g. mobile terminals at greater distances from the base station or when the base station is carrying a heavy load of traffic, then the services may be restricted to those using lower rates.

l) Teleconference

The teleconference service will provide users of the FPLMTS with the ability for real-time two-way speech, data and/or video between two or more locations simultaneously. It is likely that all video information carried via the FPLMTS will be compressed.

m) Short messages

These connectionless services allow the exchange of messages of limited length (say one or several 32 byte blocks) between a storage system connected to the fixed network and a mobile station, or between mobile stations, in real-time. It can be a point-to-point or point-to-multipoint service.

The teleservices defined above are divided into two categories: primary and secondary as listed below:

Primary:

- Telephony
- Text
- Teletex
- Paging
- Videotex
- Telefax
- Data
- Short Message

Secondary:

- Programme Sound
- Teleconference
- Programme video
- Video surveillance
- Video telephony

Any implementation of an FPLMTS should support as minimum, the primary teleservices. The degree to which the secondary teleservices are also supported by the FPLMTS may vary between administrations and should be subject to further study.

3.5.5.5 *Supplementary services*

Supplementary services for mobile systems are under discussion in CCITT. It is expected that CCITT Recommendations Q.1011 through Q.1019 describing some supplementary services will be issued during the 1988-92 study period. Examples of supplementary services are listed in Table 1/I.250 in CCITT Recommendation I.250.

The following supplementary services support mobility:

a) Point-to-multipoint

FPLMTS should provide this service for dispatch, group calling, closed user groups and other applications, to the extent permitted by individual administrations.

b) Separation of answering from alerting

In current public telecommunication systems, the alerting function always resides in the same terminal used to answer the call. However, in the FPLMTS, it is envisaged that the device on which the alert is received, e.g. pager, personal station (PS), etc., is not necessarily the one used to answer the call. The called party will be able to use any terminal of his choice (e.g. telephone or mobile station (MS /PS) to answer the incoming call. This implies that the delivery of a signal to an alerting device is not a completed activity, rather, just a part of the total activity associated with establishing a call.

This service concept could have implications on the sequence of call-establishment signals as well as call completion time delays. The details of this service as well as user acceptability need further study by a wide range of experts.

c) Charging service

FPLMTS should be able to decide the distribution of the charges between the parties involved with a particular call based on the personal telecommunication service profiles associated with each PTN used by these parties respectively.

The paying party (or parties) should be able to receive the respective charge information before, during, or after each call when it is feasible.

This charging service will provide all users the option to obtain charge information of a particular call without compromising the other users' privacy.

d) End-to-end encryption

The FPLMTS should support the use of end-to-end encryption associated with any teleservice which uses an unrestricted circuit-mode or packet-mode bearer service. As a supplementary service, FPLMTS should also support end-to-end encryption of speech by the use of an unrestricted circuit-mode bearer via the ISDN. This service may have an impact on the ISDN and requires further study.

e) Mobile-to-base station encryption

The FPLMTS should also support mobile-to-base station encryption, which in some cases will offer sufficient protection whilst being easier to implement than end-to-end encryption.

f) Location

The provision of information to the calling or called party as to the location of the corresponding party. (See also 3.5.3 and 3.5.5c.)

g) PTN

The provision of a PTN service as defined in 3.4 to a user including the feature offered by the PTN service profile.

h) Automatic message box status indication

Automatic message box status indication (e.g. new message, urgent message, empty) reduces loss of communication whilst a user is moving.

3.5.5.6 *Service attributes*

Telecommunication services are classified using their static characteristics described by attributes. Service attributes for ISDN services are described in CCITT Recommendation I.210. The service attributes for FPLMTS can be grouped into 4 categories:

- Information transfer attributes
- Access attributes
- General attributes
- Mobility attributes

Table II indicates some possible values for the FPLMTS service attributes.

Table II Possible FPLMTS service attributes

Attributes (Note 1)		Possible values of attributes											
A. Information transfer attributes		Circuit									Packet		
A1. Information transfer mode													
A2. Information transfer rate		Bit rate kbit/s									Throughput		
(*: specific to FPLMTS)		8	16	32	64	2x64	384	1536	1920	Other values for further study		Options for further study	
A3. Information transfer capability		Unrestricted digital information				Speech		3.1 kHz audio	7 kHz audio	15 kHz audio	Video	Others for further study	
A4. Structure		8 kHz integrity				Service data unit integrity				Unstructured		TSSI (2)	RDTD (2)
A5. Establishment of communication		Demand				Reserved				Permanent			
A6. Symmetry		Unidirectional				Bidirectional symmetric				Bidirectional asymmetric			
A7. Communication configuration		Point-to-point				Multipoint				Broadcast			
B. Access attributes		I*, C*, D(16), D(64), B, H0, H11, H12 (*: specific to FPLMTS)									Others for further study		
B1. Access channel and rate													
B2. Signaling access protocol (all layers)		For further study (specific to FPLMTS)											
B3. Information access protocol attributes		For further study (specific to FPLMTS)											
B4. Higher layer protocols		For further study (specific to FPLMTS)											
C. General attributes		- supplementary services provided - quality of service - interworking - OA + M				For further study (specific to FPLMTS)							
D. Mobility attributes		- location - handovers - user profile (PTN) - alerting - call completion				For further study (specific to FPLMTS)							

Notes:

- Attributes are intended to be independent of each other.
- TSSI: Time Slot Sequence Integrity
RDTD: Restricted Differential Time Delay
- This table follows Figure 8-2/1.210 (CCITT) except where noted.

3.6 *Basic speech quality*

The quality of speech services in FPLMTS should be comparable to that of the PSTN/ISDN as far as practicable.

The effective utilization of the frequency spectrum is an important factor for FPLMTS and, accordingly, it is necessary to develop speech coding methods that provide high quality speech at low bit rates. The practical design of the speech coder will be strongly influenced by the requirement to achieve reasonable overall transmission delay. This overall delay should also take into account delay introduced by the RF channel and signal processing.

Speech services that are sensitive to delay may need to be redefined.

The quality of speech supported within the FPLMTS should be nominally that recommended in the CCITT P series Recommendations. However it is acknowledged that in some applications where low bit rate coding is employed some minimal degradation of speech quality may occur.

3.7 *Security*

FPLMTS should provide a level of security comparable to that of the PSTN/ISDN and public data networks. Cryptographic techniques can provide both privacy and authentication. Due to the basic nature of the radio transmission, added protection against unauthorized access to the system (authentication) must be provided to protect customers and the integrity of the system. An authentication method is outlined in CCITT Recommendation X.509. The use of the PTN service provides another means for authentication. Application to FPLMTS requires further study.

FPLMTS may be capable of offering additional encryption of user information but any such facility should not increase the costs incurred by mobile subscribers who do not require such facility.

It should be possible for the users themselves to provide end-to-end encryption, subject to the technical limitations of the network.

3.8 *Quality of service*

The quality of service offered by the FPLMTS should be comparable to that of the PSTN/ISDN and public data networks as far as practicable.

Consideration will need to be given to maintaining quality of service during handover between cells and any implications from this on cell size, frequency requirements, protocols, etc.

Mobile satellite systems have a number of characteristics which are different from land mobile systems such as wide area coverage and propagation delays on the order of 0.3 s or more.

To maintain the quality of services for Group 3 and Group 4 telefax machines, additional error control facility for the radio communication path may be required.

4. System architectures and scenarios

4.1 *System architecture*

The basic structure for the FPLMTS can be provided by a system architecture which evolves from present-day fixed and mobile networks toward the goal of world-wide communications mobility for people wherever they may be, at home or away, in transit (aircraft, trains, cars, ships), etc., with services interworkable with those offered by the fixed networks such as PSTN/ISDN. FPLMTS may comprise one integrated system or several separable systems, as further study will indicate.

The support of service interoperability will require the provision of network functions and the definition of signalling protocols for use both over the radio system signalling channels and within the fixed network. CCITT has developed a three-stage method for the development of protocols (see CCITT Recommendation I.130, Blue Book), used effectively in the development of the ISDN, which may be useful for the design of FPLMTS.

4.1.1 *Service access*

The PTN service provides a flexible means of accessing FPLMTS services as described in 3.4.

The FPLMTS should provide universal personal telecommunication services through a universal service access mechanism such as PTN, by which new technology and various regional constraints may be accommodated [De Brito, 1981]. The FPLMTS should perform service origination and termination regardless of user location within the FPLMTS or the PSTN/ISDN.

Given a PTN, any user terminal can be used as a means for access; consequently, the terminal may vary as access is provided through either mobile or fixed means.

A user-network interface standard needs to be defined so that user terminals will be transparent to transmission technology.

4.1.2 *Description of interfaces*

The major functional components of the FPLMTS are shown in Fig. 3 (This is not intended to represent a specific implementation scenario).

These functional components in turn define a group of interfaces (1-10) which may be physical or conceptual. Furthermore, these interfaces should be based on appropriate CCITT Recommendations after unique requirements of FPLMTS are incorporated. The generic radio link interface 1 will be discussed in more detail in 4.3.

A possible partition of basic MS/PS signalling functions is shown in Fig. 4, which consists of:

- Call Control;
- FPLMTS Call Control Adaptation;
- Mobility Management; and
- Channel and RF Transmission Management.

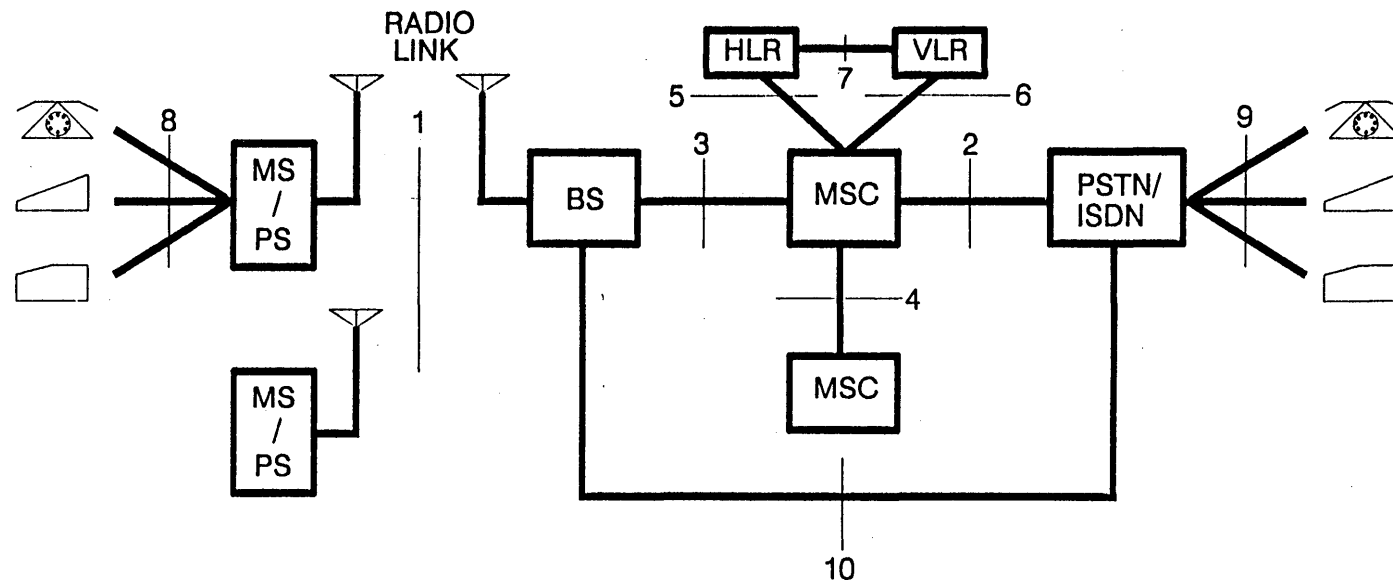
The signalling reference points between these functions are designated as S/R, Xm, Sm and Um, and are shown in Fig. 4. These correspond to the same designations as in CCITT Recommendation Q.1061.

Call Control functions are those standard call controlling functions available in any CCITT Standard Terminals (e.g. X.25 Data Terminals, ISDN User Terminals, PABX etc.) or any future FPLMTS terminals. FPLMTS Call Control Adaptation functions are those functions necessary to convert standard call control signalling into call control signalling appropriate in the FPLMTS environment. Mobility Management functions are those functions necessary to support the mobility of user and terminals. Channel and RF Transmission Management includes those functions associated with providing a wireless physical connection between the user terminal and a base station, such as selecting the channel coding, controlling transmitter power, seizing a channel, changing a channel, etc.

4.1.3 *Integration with mobile satellite systems*

FPLMTS users roaming over a wide area may access (directly or indirectly) either terrestrial systems or mobile satellite systems (See Figs. 5 and 6, also see Report 1177.) An architectural feature could be automatic diversion to mobile satellites for FPLMTS service when the user is beyond the range of terrestrial system coverage. Mobile satellites may also support:

- A satellite paging function which could reduce the amount of location registration data to be transferred.
- Paging terrestrial mobile users beyond the range of the terrestrial system. The equipment required for this one-way function is simpler than that for a two-way mobile satellite function.
- Linking remote base stations or where temporary (e.g. emergency) system extension is needed.



- MS — Mobile Station
- PS — Personal Station
- BS — Base Station
- MSC — Mobile Services Switching Centre
- HLR — Home Location Register
- VLR — Visitor Location Register

Fig. 3 Major Functional Components and Interfaces of the FPLMTS

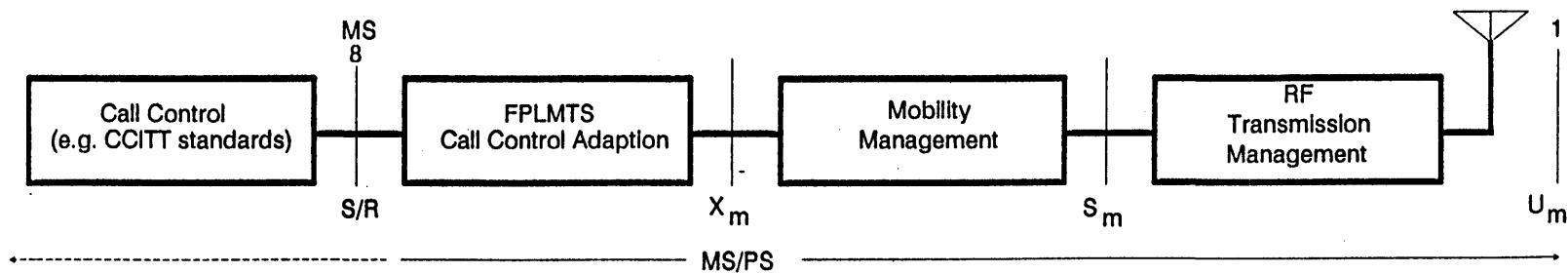
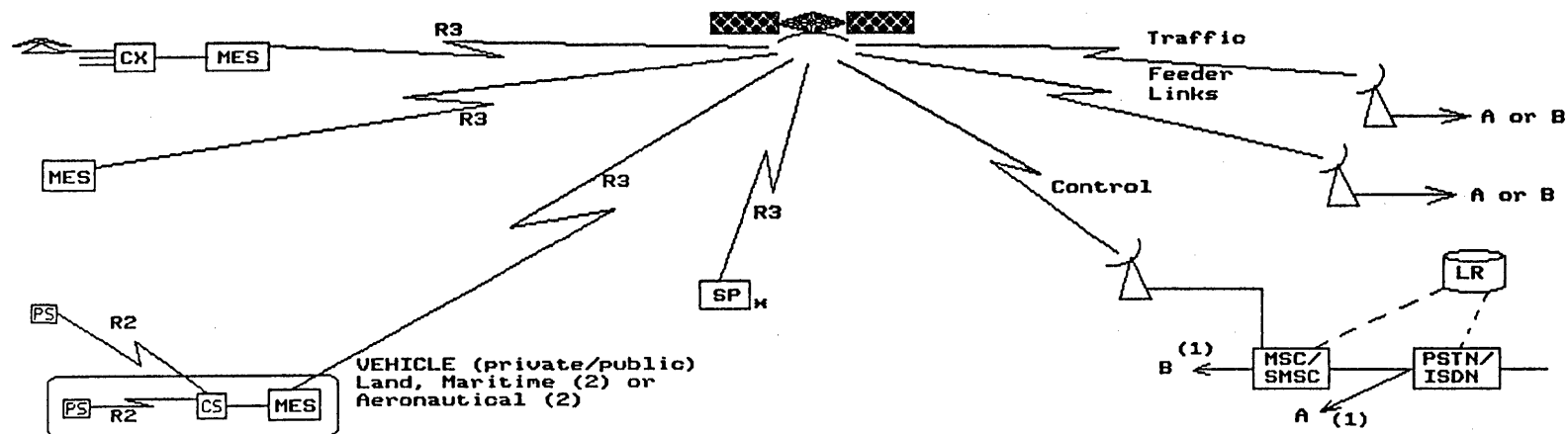


Fig. 4 MS/PS Signalling Reference Points



R1-R3: Radio Interfaces
 PS: Personal Station (R2)
 CS: Personal Base Station (Cell Site for PSs)
 MES: Mobile Earth Station
 SP: Satellite Pager (R3)
 MSC: Mobile services Switching Centre
 SMSC: Satellite Mobile services Switching Centre
 LR: Location Register
 CX: Small rural exchange, etc.
 : Telephone

Note (1): Access to and the scope of the location registration functions will vary with system evolution and network operator requirements. This is reflected in network interfaces A and B.

Note (2): Terrestrial based systems have different radio interface requirements from R3 which need to be considered.

X: Can be co-located/integrated with the PS.

Fig. 6 Scenario for personal communications with FPLMTS
(Satellite component)

4.2 *Scenario for personal communications within FPLMTS*

A scenario for the evolution and implementation of personal communications is illustrated (Figs. 5 and 6) and tabulated (Table III). These figures and the table provide a quick survey of the arrangement of facilities and typical dimensions for such a scenario.

The scenario discussed identifies a broad approach to evolving personal communications. It presents a preliminary framework for understanding the implications of other areas of the Report and a basis for identifying areas of commonality.

Some of the distinguishing characteristic assumptions of high density PS communications are small coverage area per personal base station (CS) and less dependence on a single base station to minimize shadowing.

In Table III, the dimensioning of cell areas, traffic, antenna height and other parameters are typical values averaged or estimated by various administrations. The dimensions are not maximum or minimum values which might be required in some circumstances.

As cell areas are reduced, the number of handover increases. The subjective user tolerance is dependent on the frequency and duration of handover breaks. This may limit the ultimate minimum cell area.

Station output powers are shown as an average power value which is a fair measure of range requirements. Peak and average values are normally the same for systems with FDMA, but the peak value could be higher for Time Division Duplex (TDD) and TDMA "duty cycle" systems.

Table III
Example of the characteristics for personal communications within FPLMTS
(High density area)

CHARACTERISTIC	TYPE OF STATION		
	Mobile	Personal Station (PS) ⁽¹⁾	
Cell Plan	MS Outdoor	PS Outdoor	PS Indoor
Cell Area:	(min) 0.5 - 1 km ²	(typ) 16000 m ²	(typ) 600 m ²
Base Station Antenna Height	50 m	<10 m	<3 m ⁽²⁾
Service Area Reliability	90%	>90%	99%
Base Station Installed Indoor/outdoor	No/Yes	Yes/Yes ⁽³⁾	Yes/Yes ⁽³⁾
Voice traffic per station ⁽⁶⁾	0.10 E	0.04 E	0.20 E
Non-voice traffic per station	0.05 E	0.004E	0.11 E
Voice traffic per km ² ⁽⁶⁾	500 E	1500 E	20000 E ⁽⁴⁾
Non-voice traffic per km ² ⁽⁶⁾	82 E	150 E	5000 E ⁽⁴⁾
Blocking	2%	1%	<0.5%
Station: ⁽⁵⁾			
Volume	Vehicle mounted	<~200 cm ³	<~200 cm ³
Weight	or portable	<~200 g	<~200 g
Highest Power	5W	50 mW	10 mW

- (1) The same PS provides access to cell plans PS outdoor and PS indoor, and also to cell plan MS if MS and PS have the same radio interface (i.e. R1 = R2).
- (2) Or leaky feeder
- (3) Usual case
- (4) Per floor
- (5) A range of terminal types will be available to suit operational and user requirements.
- (6) See Section 5.



4.3 *Radio interface considerations*

4.3.1 *Interface types*

From the scenario of Fig. 5 and Table III, the radio interface considerations are:

- The personal station (PS) interface R2 is used in two separate cell plans, one for mainly outdoor and one for mainly indoor coverage. The two cell plans may use different sets of frequencies within the same band.
- A separate interface R1 and separate frequencies may be envisaged for the mobile station (MS) as distinct from the personal station. However, a common radio interface ($R1 = R2$) may be possible, as well as a common frequency band for the different cell plans.
- The radio interfaces should be designed to allow different applications to use the same interface where this can be shown to be technically and economically feasible.
- If the same radio interface cannot be used for all applications then the individual interfaces should have maximum commonality to allow interworking with the minimum extra complexity.
- With the low transmit powers considered for the personal station, coexistence with other services (e.g. private wireless PABX) is a possible consideration.
- It is generally assumed that the location of the personal station changes slowly with respect to the personal base station.
- The wide area paging interface R4 is an additional signalling interface used for alerting (e.g. paging) a station in the case of a mobile terminated call.
- When using R4, the location area will conform to the paging area of the paging base station.

The scenario of Fig. 6 introduces interface R3, which is integrated in the signalling functions of FPLMTS.

- Interface R3 is a satellite communications interface used for both signalling and traffic.
- When using R3 for alerting (e.g. paging), the location areas will conform to the coverage areas of the satellite station.

Real time interaction with CCITT Common Channel Signalling No. 7 (CCS7) requires special consideration, especially when alerting and answering are considered to be separate functions joined by CCS7 (see 3.5.5.5).

As shown in Figs. 5 and 6, a mobile station (MS) or a mobile earth station (MES) can be envisaged to provide a tandem link to personal stations (PS) inside a vehicle.

Other interfaces may be required for terrestrial aeronautical and maritime mobile systems.

A common world-wide radio interface may be a desirable but not essential goal to achieve mobility, although it may be desirable from the point of view of cost reduction and wider availability of equipment.

4.3.2 *Signalling protocols*

The signalling protocols should support ISDN compatible services and services specific to the FPLMTS, and should facilitate interworking between PSTN/ISDN and FPLMTS. Studies and refinements of telecommunications services definitions should be undertaken for the development of signalling protocols (e.g. requirement to support packet data on control channels).

Reference configurations and signalling protocol models should be aligned as closely as possible with OSI and ISDN reference models. In addition, account should be taken of signalling protocols developed for current and planned mobile systems including cellular systems, cordless telephones, mobile satellite systems and terrestrial air-to-ground systems.

Signalling protocols must be consistent with the propagation and spectrum constraints of the radio environment. However, the functionality of ISDN signalling should be maintained to the maximum extent practicable.

The signalling functions over the radio interface is envisioned to consist of three layers.

Signalling Layer 1 should provide a reliable RF connection between a base station and a mobile/personal station. Examples of functions are RF channel structure, RF modulation and demodulation, framing/timing, power adjustment, etc.

Signalling Layer 2 should provide reliable signalling data link between a base station and a mobile/personal station.

Signalling Layer 3 should provide a reliable application message communication between a base station and a mobile/personal station. It can be subdivided into three functional entities: Call Control (CC), Mobility Management (MM) and RF Transmission Management (RT). The CC entity would contain the functionalities for support of basic and supplementary services

(Call Setup, Call Release, etc.). The MM entity would contain functions like Location Updating, PTN Attach/Detach, Authentication, etc. The RT entity would contain functions like Channel Assignment, Handover, Channel Release, Measurement Reporting, etc.

The details of the signalling protocols will be jointly studied and defined by both CCIR SG 8 and CCITT SGXI.

4.4 *System capabilities for a PTN service*

A personal telecommunication number (PTN) service which enables call connections, billings etc., on a personal basis is defined in 3.4.2.4.

In order to offer a PTN service, the following system capabilities are necessary:

- To provide a PTN number for each user ,
- To admit registrations and dismissals of users to and from the system by the PTN number,
- To enable routing to the registered location by the PTN number,
- To charge on the basis of the PTN number,
- To authenticate an user by the PTN number.

In order to provide the PTN service, the following major topics should be considered:

- Traffic for PTN registration and dismissal of a person to the network and related aspects,
- Man-machine interface, including authentication,
- System operational aspect,
- Roaming service ability.

The CCITT Study Groups are investigating a broader application of the PTN service which covers both fixed networks and radio networks.

5. **Frequency considerations**

5.1 *General*

A fundamental goal in the design of a FPLMTS is the provision of adequate capacity to serve its intended customers. It is also important to ensure that the most efficient use of radio spectrum (maximum Erlangs of information transfer per MHz per km²) is made.

To that effect, considerations on spectrum should take into account the traffic, the available and foreseeable techniques, the propagation characteristics and the timescale for meeting the users' needs to the greatest possible extent. All the techniques considered need to be balanced with particular attention to the cost, size, weight and power consumption of the mobile and personal stations.

5.2 *Users' needs*

The need for mobile communications available in any place and time is increasing rapidly.

An expansion rate as high as 100% a year has been experienced for current systems based on analogue technology, in a number of countries. It is expected that the number of subscribers to public mobile systems will exceed 10 million in 1990. With the introduction of digital technology it is foreseen that the penetration of mobile units in Europe and North America in the late 1990's will be around 5% of the population.

In addition demands for personal communications using lightweight portable equipment for both indoor and outdoor communications will increase considerably the penetration of public mobile communications. It is expected that this category of users will generate a need for worldwide roaming. The number and diversity of services is also expanding. Systems under consideration will include voice and non-voice services with similar levels of importance.

For all categories of customers, particularly those using personal stations, compatibility of the services provided by different networks is of primary importance. Commonality, particularly in frequency matters, is a very important factor for FPLMTS. Cost and technological factors are also of importance to make available cost effective and, when necessary, low weight, low power consumption equipment. Within a service area, a reliability of at least 90% is expected and may reach 99% for home and business applications.

5.3 *Traffic considerations*

5.3.1 *General*

The maximum demand for "personal" telecommunication services is in large cities where different categories of traffic can be found, i.e. that generated by mobile stations (MS), vehicle-mounted or portable and personal stations (PS), outdoor and indoor. The number of customers in these different cases may be estimated, with some variations, according to the nature of the cities.

The traffic generated by the paging service in FPLMTS has not yet been estimated but is considered proportionally insignificant.

The traffic of the Land Mobile Satellite Service has not been considered. It is relevant to Decision 81.

5.3.2 *Voice traffic*

It is estimated that in a traffic jam the number of vehicles per km of street length is around 600 if they are stationary or 350 if they are moving slowly. Assuming a mean value of 400 vehicles per km of street length, 50% of the vehicles equipped with a mobile station, each generating 0.1 E, the traffic density will be 20 E/km of street length or 300 E/km² for vehicle-mounted mobile stations on the street. Adding about the same amount of traffic for portable mobile stations carried by pedestrians, the combined traffic, for Mobile Stations (MS), would be around 500 E/km² in the most dense city areas.

The peak traffic density for personal stations outdoor is estimated at 1500 E/km², assuming 3000 pedestrians per km of street length, 80% penetration of the personal station and 0.04E/station.

For personal stations indoors the traffic may increase by a factor of 10 or more in a multi-storey office building. An estimation is 1 station per 10 m² active floor area with a traffic of 0.2 E per station. This corresponds to 20,000 E/km²/floor.

It is estimated that the peak to mean ratio of traffic for pedestrians in busy streets of large cities has a value around 3.

5.3.3 *Non-voice traffic*

In the future, non-voice services will constitute an increasing proportion of total traffic. Some data services need more transmission capacity than a duplex voice channel, increasing the necessary spectrum. On the other hand, some non-voice services can be handled by using queuing procedures, which improve the spectrum utilization.

5.3.3.1 *Mobile stations*

- Circuit-switched services are considered as being relevant mainly for vehicle-mounted MS. Telefax is an example of such a service. Assuming 3000 terminals/km², of which 15% are equipped with a telefax unit, a call holding time of 6 minutes per hour, per terminal; the estimated traffic amounts to 45 E/km².

Interactive data services are likely to employ packet transmission. The assumption is a cumulative channel occupancy of 15 s/hour for a hand portable (10 pages per hour with 8 kbit per page, at a transmission rate of 4.8 kbit/s) and 30 s/hour occupancy for a vehicular mobile (4.5 and 9 mE respectively). Assuming 5000 terminals per km² (3000 vehicular and 2000 portables), the estimated traffic amounts to 37 E/km² [Turner, 1986; Goodman et al., 1988].

5.3.3.2 *Personal stations outdoor*

Traffic generated by circuit-switched data services, eg. telefax, has been considered insignificant. Therefore, only short-message interactive data communications are considered in the present case.

The assumption is a cumulative channel occupancy of 5 s/hour (10 pages per hour with 8 kbit per page at a transmission rate of 16 kbit/s) corresponding to 1.4 mE/station.

Assuming 2400 stations/km of street length (37,500 stations/km²), as in the case of voice, the amount of traffic would be 50E/km². To account for other data services, this estimate is increased by a factor of 3 to 150E/km². The non-voice traffic is then 10% of the voice traffic.

5.3.3.3 *Personal stations indoor*

For telefax application: assuming 25% of stations having a telefax and 6 minutes call holding time per hour, per telefax terminal; the estimated traffic is 25 mE per station i.e. one-eighth of the voice traffic, or 2500 E/km².

For interactive application: considering all stations are using this application and assuming 20 interactive sessions per hour, with a cumulative channel occupancy of approximately 2 s per session, we arrive at 0.01 E/station, equivalent to 1000 E/km². In order to take account of contention due to packet transmission, a factor of 2 is assumed, leading to 2000 E/km².

In total, taking into account batch data application, data base retrievals (10% overhead) a total amount of 5000 E/km² is assumed for non-voice services indoor.

5.4 *Spectrum Requirements*

5.4.1 *Bandwidth estimation method*

A general equation can be derived for circuit-switched services giving the bandwidth 'BW' (MHz), assuming a relatively uniform traffic load over a reuse cluster of cells.

$$'BW' = 10^{-3} \cdot F(a \cdot E, P_b) \cdot b \cdot n$$

where 'F' is the appropriate traffic formula (e.g. Erlang B or C for voice). 'E' is the traffic load in Erlang/km², 'a' is the cell area in km², 'n' is the number of cells (channel sets) per reuse cluster, 'P_b' is the blocking probability in percent, and 'b' is the equivalent bandwidth per duplex channel in kHz, including signalling and other overheads.

5.4.2 *Technological and other factors influencing overall spectrum requirements*

There are many factors that affect the variables in the bandwidth calculation equation above and hence the overall spectrum requirements.

5.4.2.1 *Bandwidth of a duplex channel*

The bandwidth of a duplex channel 'b' can be reduced significantly by using low bit rate speech codecs. In addition, the use of packet transmission, digital speech interpolation (DSI) and adaptive system parameters offer the possibility of further reductions [Turner, 1986; Goodman et al., 1988]. In contrast, future demands for high quality speech or the requirement to support ISDN services with, for example, more stringent BER thresholds or limits of transmission delay would increase channel bandwidths, as would additional overhead e.g. for dynamic power control.

5.4.2.2 *Number of cells/cluster*

A decrease in the number of cells (channel sets) per reuse cluster 'n' could be achieved by a combination of micro diversity (space/frequency hopping) and macro-diversity, coupled with both inter-cell and intra-cell rapid handover procedures. The implementation of adaptive system techniques such as electronically steerable antennas, dynamic power control and "leaky feeders" (mainly for indoor cells) could also reduce the number of cells/cluster. However, non-ideal transmitter locations and imperfect cell geometry, combined with a need to carry less interference tolerant signals e.g. high-quality speech, would increase the number of cells/cluster. An increase in service area reliability and/or adverse propagation conditions could also increase the effective cluster size.

5.4.2.3 *Cell area*

Rapid handover procedures are essential if cell areas 'a' are to be reduced and the consequent improvements in spectrum efficiency realized. However, non-ideal cell geometry would reduce the overall improvement.

5.4.2.4 *Blocking probability.*

The choice of blocking probability 'P_b' for any particular cell type or environment is a quality-of-service target. However, a change in 'P_b' will only marginally affect the overall spectrum requirements.

5.4.2.5 *Trunking efficiency*

Trunking efficiency is influenced by traffic patterns, services and system design. By using techniques such as dynamic channel allocation, trunking efficiency and hence spectrum utilization can be improved. However, other factors such as multiple operators splitting the spectrum allocation into different blocks, would reduce trunking efficiency and increase spectrum demands.

5.4.3 *State of techniques*

5.4.3.1 *General*

The design of systems for carrying traffic to mobile users is directly related to spectrum efficiency and so the system parameters should be optimized to that effect.

Any improvement in spectrum efficiency should be balanced against system design constraints e.g. power consumption and cost effectiveness. A number of system parameters are considered below.

5.4.3.2 *Speech transmission*

A transmission bit rate of 16 kbit/s or below is desirable to improve spectrum utilization. Good quality codecs with robustness and low delay are required. Linear predictive coding (LPC) techniques combined with adaptive quantization and adaptive prediction offer, at this point in time, the most attractive possibilities for FPLMTS in terms of speech quality at low bit rates. For fast fading channels, the most efficient and economical system would have matched speech and channel codings.

The power consumption and size of codecs is particularly critical for personal stations.

5.4.3.3 *Data transmission*

Practical and frequency economy considerations would probably limit the allowable data rate to below 1 Mbit/s for mobile stations. However, megabit rates may be useful and practicable for short range systems within buildings.

For relatively narrowband transmission (e.g. less, or of the order of the coherence bandwidth) and for fast-fading, forward error correction (FEC) is effective to improve spectrum efficiency and to reduce the RF output power for high quality transmission (average BER is less than 10^{-3} to 10^{-4}). For relatively low quality transmission (average BER is 10^{-2} or more), FEC gives no improvement and can reduce spectrum efficiency. For the slowly fading channel, error detection and automatic repeat request (ARQ) may be more effective than FEC.

5.4.3.4 *Modulation and equalization*

Digital modulation schemes should be used which combine low bandwidth requirements with good C/I and C/N tolerances. Modulation techniques having transmission efficiencies in excess of 1 bit/s/Hz are desirable candidates for FPLMTS. Signal processing techniques such as adaptive equalizers might be necessary for wide band transmission.

5.4.3.5 *Channel coding*

Channel coding combined with interleaving is a spectrally efficient technique to provide adequate transmission quality, particularly for mobile applications. Channel coding techniques can be divided into two families: block codes and convolutional codes. Block codes, such as Reed-Solomon, are used for both error detection and correction, while convolutional codes are primarily used for error correction. Fade duration, inter-fade interval and random error rate can influence the choice of channel coder.

5.4.3.6 *Multiple access*

Multiple access techniques, such as FDMA, TDMA and CDMA, should be considered for FPLMTS. FDMA/TDMA schemes for second generation cellular systems have been adopted in both Europe, North America and Japan. (See Report 1156).

FPLMTS could provide a mixture of narrowband and broadband services. A flexible multiple access technique and an efficient management system for the traffic associated with the various different services are both desirable to maximize the overall spectrum efficiency of FPLMTS. In all cases, demand assignment will improve spectrum efficiency.

5.4.3.7 *Adaptive techniques*

In order to improve performance, systems may adapt such parameters as channel bit rate, bandwidth and frequency/time/coding arrangements, and use techniques such as diversity, channel assignment and multipath equalization to adapt to actual propagation, interference and traffic conditions. Such adaptive techniques also have the potential to minimize radio network planning requirements, especially as cells become smaller. [Acampora and Winters, 1987; Dornstetter and Verhulst, 1987; Maseng, 1986; Stjernvall and Uddenfeldt, 1987].

5.4.4 *Estimate of required spectrum*

5.4.4.1 *Parameter assumptions*

In this section we will introduce some values for parameters relevant for spectrum estimation (cell area, cluster size and duplex channel bandwidth).

It should be noted that, although the calculations assume that mobile stations and personal stations are accessing different cells of the network; it may be envisaged that they can access the same cells, for example if interface $R_1 = R_2$, or by use of dual-mode mobile/personal stations.

- a) Voice services
 - i) Mobile stations (R_1 interface)

Two future possible solutions are considered in order to give a range of estimates depending on the availability of certain

techniques. In both cases, the distance between base station sites is taken as 1.75 km.

A conservative estimation can be based on the example of a system with a speech codec of 7 kbit/s with a duplex bandwidth of 25 kHz per voice channel associated with a cluster size of 9 (a cellular arrangement of 3 sites with $3 \times 120^\circ$ -sectors per cluster). This cellular arrangement, combined with a site separation of 1.75 km, leads to a cell area of 0.94 km^2 .

A more optimistic estimation can be based on the example of a system with a 4 kbit/s speech codec and a duplex bandwidth of 10 kHz per voice channel; associated with a cluster size of 24 (a cellular arrangement of 4 sites with $6 \times 60^\circ$ -sectors per cluster). This cellular arrangement, again combined with a site separation of 1.75 km, leads to a cell area of 0.47 km^2 . This implies the availability of speech codecs of approximately 4 kbit/s with the same resistance to errors as present ones and of equivalent quality.

ii) Personal stations outdoor (R2 interface)

The personal stations outdoor are assumed to be served by microcells at street level with one base station per intersection spaced at 125 m. Each cell covers 250 m of street length. This corresponds to a cell area of 0.016 km^2 .

Due to the severe limitations on cost and power consumption for personal stations, a higher bandwidth of 50 kHz per duplex voice channel is assumed.

The cluster size of $n = 16$ for personal stations outdoor assumes a structure of 4×4 cells.

iii) Indoor personal stations (R2 interface)

The same bandwidth per voice channel of 50 kHz as for other personal stations is assumed. The cluster size of $n = 21$ is used, corresponding to a cluster size of 7 per floor repeated every 3rd floor [Porter, 1985].

The indoor personal stations are assumed to be served by indoor base stations each covering adjacent rooms along a corridor. An indoor cell size of $40 \times 15 \text{ m}$ equal to 600 m^2 is assumed.

b) Non-voice services

This section will discuss the required duplex bandwidth for data channels and derive, from the figures of 5.3.3 estimations of traffic needs in terms of equivalent 25 kHz duplex channels.

i) Duplex bandwidth considerations

- fast-fading channels (macro cells)

In this case, a powerful forward error correction (FEC) technique is needed to achieve high quality. It was assumed, taking into account future systems, that a 50 kHz duplex channel could support a 2.4 kbit/s user data with a BER of 10^{-6} and 9.6 kbit/s user data with a BER of 10^{-3} .

- slow-fading channel (micro cells)

In this case it is assumed that a 50 kHz duplex channel can convey 16 kbit/s user data with high quality (BER of 10^{-6}). However, error correction coding is needed to achieve this level of quality.

ii) Mobile stations

The telefax traffic is assumed to be carried at 4.8 kbit/s and therefore requires 100 kHz duplex channels. (4 equivalent 25 kHz duplex channels). Therefore the traffic (see 5.3.3) accounts for 180 equivalent 25 kHz channels/km².

The interactive data traffic is assumed to be carried at 4.8 kbit/s rate (2 equivalent 25 kHz channels) with an ARQ protocol. The total traffic (see 5.3.3) for this application is therefore equal to 74 equivalent 25 kHz channels/km².

The total amount (254 equivalent 25 kHz channels/km²) for non-voice services is therefore estimated as 50% of the number of 25 kHz channels (or 100% of the number of the 10 kHz channels) needed for voice.

iii) Personal stations

The transmission in these cases is carried on 50 kHz bandwidth channels (same assumption as for voice). The traffic in 5.3.3 can be directly compared to voice traffic.

5.4.4.2 *Presentation of results*

Using the traffic and services assumptions of 5.3.2, 5.3.3, and 5.4.4 the estimate of required spectrum bandwidth is presented in Table IV for voice services, and Table V for non-voice services.

For voice services, the total required bandwidth ranges from approximately 160 MHz (scenario 1) down to approximately 110 MHz (scenario 2) according to future possible technical solutions presented in 5.4.4.

For non-voice services the total required bandwidth amounts to 65 MHz. In this case the two scenarios lead to the same result, since the duplex channel bandwidth for non-voice services is the same in both scenarios and the increased cluster size in scenario 2 is assumed to be compensated by higher C/I and resultant higher data transmission efficiency. The total bandwidth therefore ranges from approximately 230 MHz (conservative projection of future technology), down to approximately 180 MHz (optimistic projection of future technology).

In a number of countries an allocation of 50 MHz of spectrum below 1 GHz is used for existing cellular mobile systems, which are presently less spectrally efficient than FPLMTS scenario 2. If these existing systems remain in service, the traffic they carry should be subtracted from the FPLMTS traffic assumptions. The additional required bandwidth can then be estimated. The need for new spectrum (likely to be above 1 GHz) ranges therefore from approximately 180 MHz (scenario 1) down to approximately 130 MHz (scenario 2).

More detailed calculations must take into account specific technologies employed by existing systems. For example, if a technology comparable to scenario 1 was used by an existing system below 1 GHz and the technology of scenario 2 was used by FPLMTS, the overall additional needed bandwidth would be approximately 150 MHz.

It must also be noted that the estimation has been conducted taking into account dense metropolitan areas. Outside those areas the spectrum need is less and frequency sharing with fixed services should be considered.

The spectrum requirement for the paging service in FPLMTS has not been estimated since it is considered insignificant, compared with the spectrum requirements of other services in FPLMTS.

The spectrum requirements for the Land Mobile Satellite Service have not been included. This is considered relative to Decision 81.

Table IV
Spectrum estimation for voice services
(example of scenario as described in 5.3.2 and 5.4.4.1a)

	MS (interface R1)		PS Outdoor (Interface R2)	PS Indoor (Interface R2)
	scenario 1	scenario 2		
Radio Coverage (%)	90	90	>90	99
E (E/km ²)	500	500	1500	20000 ⁽¹⁾
a (km ²)	0.94	0.47	0.016	0.0006
P _b (%)	2	2	1	0.5
n	9	24	16	21
b (kHz)	25	10	50	50
Traffic per cell (E)	470	235	24	12
Number of channels per cell	493	249	34	23
BW (MHz)	111	60	27	24

E: Traffic density
a: Cell area
P_b: Blocking probability
n: Cluster size
b: Duplex bandwidth per channel
BW: Bandwidth
(1) Note: per floor

scenario 1: Conservative projection of future technology
scenario 2: Optimistic projection of future technology

Table V
Spectrum estimation for non-voice services
(example of scenario as described in 5.3.2 and 5.4.4.1b)

	MS (interface R1)		PS outdoor (interface R2)		PS indoor (interface R2)	
	Circuit switched	Packet switched	Circuit switched	Packet switched	Circuit switched	Packet switched
Traffic density (E/km ²)	45	37	insignificant	150	2500 (1)	2500 (1)
Duplex bandwidth per channel (kHz)	100	50	50	50	50	50
Bandwidth (BW) (MHz)	56		3		6	

(1) Note: per floor

5.5 *Factors for choice of frequency band(s)*

5.5.1 *Propagation characteristics*

Various factors affect the propagation characteristics. They can be summarized in Table VI and as follows:

5.5.1.1 *Outdoor propagation*

For outdoor propagation the law for decrease of the power as a function of propagation is around r^{-3} to r^{-5} for large cells and r^{-2} to r^{-6} for small cells. As frequency increases, fading rates and Doppler spread of the signal increase. There are limitations to the bandwidth of the transmissions resulting from these effects. The multipath propagation characteristics introduce a delay spread in the transmission which limits the achievable transmission rate.

The above effects also limit the coverage area. As the operating frequency is increased, received power is decreased due to increased path attenuation and reduced antenna aperture. Some experimental data with 20 m base station antenna heights indicate that this decrease is about 4 dB between 100 MHz and 10 GHz. Other data indicate a decrease of 6 dB between 900 MHz and 1.5 GHz, and 5 dB between 1.5 GHz and 2.2 GHz for antenna heights of 30 m in reasonable agreement with Report 567-3.

5.5.1.2 *Indoor propagation*

Propagation measurements at 900 and 1700 MHz show that frequencies in this range are suitable for low-power indoor communications. An increase in frequency from 900 to 1700 MHz results in 5 dB increased loss due to reduction in antenna aperture. Penetration loss through walls and floors is greater at the higher frequency by about 3 to 6 dB depending upon the building's construction, which may affect favourably the frequency reuse in a multi-storey building. When the effects of floor and substantial walls have been removed, signal strength decreases at the same rate at both 900 MHz and 1700 MHz. For a mean effective radiated power (e.r.p.) of 5 to 10 mW the typical range, at 1700 MHz, for satisfactory speech communication will vary from 10m through two floors to 100 m on the same floor. The exact communication range depends on the nature of the building and the precise location of the transmitting and receiving antennas.

Overall propagation factors indicate that the use of a band below 3 GHz would be suitable.

Table VI
Propagation characteristics around 1 GHz

APPROXIMATE VALUES
 (Available data as of 1988)

	Outdoors		Indoors
	100m Antenna height (Large cell)	10m Antenna height (Small cell)	
DELAY			
SPREAD			
-Median	3 microsec	0.15 microsec	0.15 microsec
-Typical max.	16 microsec*	0.5 microsec	0.3 microsec
PROPAGATION			
PATH LOSS			
Small Scale			
- Statistics	Rayleigh	Rician/Rayleigh*	Rician/Rayleigh*
- Coherence distance	wavelength/4	wavelength/4	wavelength/4
- Cross-polarization			
*Coupling	information	0-6 dB	0-6 dB
*Correlation	not available	0	0
Large Scale			
- Statistics	LOG - NORMAL	LOG - NORMAL	LOG - NORMAL
- Power law	r^3 to r^5	r^2 to r^6	r^2 to r^6
- Std. Dev	6-8 dB	10 dB	10 dB
- Coherence distance	30 m*	10 m*	3 m*

* DEPENDING ON SPECIFIC ENVIRONMENT

5.5.2 *RF technology*

The cost and power efficiency of the RF technology available to implement FPLMTS hardware are both adversely affected as the frequency band of operation is increased. The resulting consequences on battery operating durations for portable and personal stations and on cell structure must be considered when choosing frequency bands. It is not currently clear what frequency limit there will be for integrated circuit technology, in order to yield adequate performance in high volumes and low cost, in the late 1990s.

While it is desirable to have spectrum to meet demand cost-effectively, a fundamental technical constraint to its usability may be frequency agility of equipment. It is therefore advantageous to minimize the required frequency agility; moreover, it is generally accepted that the greater it is, the more expensive the equipment.

5.5.3 *Technical and functional compatibility/commonality*

5.5.3.1 *General*

Frequency commonality should allow for the desired level of operational compatibility on the system(s). In principle, a complete commonality of one frequency band on a worldwide basis is desirable, but reasonable commonality can also be obtained through a common signalling band and sufficient overlap of the traffic bands to ensure compatibility.

5.5.3.2 *Mobile station (Interface R1)*

Compatibility of an MS with different national or regional FPLMTS networks implies that the frequency bands of the MS should be at least partially overlapping with the frequency bands in which the FPLMTS networks are operating. Internationally agreed frequency bands also facilitate the planning of national networks and reduce the risk of harmful interference with other radio services. To determine suitable frequency bands for inter-regional mobility it is necessary to consider whether the allocation in a given ITU Region is primary or secondary and to consider geographic sharing.

5.5.3.3 *Personal station (Interface R2)*

Frequency bands separate from the band used by mobile stations in FPLMTS may be envisaged, although sharing with other radio services might be possible. One continuous common band on a worldwide basis would be desirable, but reasonable commonality could be obtained through a common signalling band and sufficient overlap of the traffic bands to ensure compatibility of low-power microcell PS with different national or regional FPLMTS networks. This would minimize the required frequency agility and could reduce RF equipment cost, which is an important factor for the realization of an affordable universal personal station.

To accommodate different cell sizes for indoor and outdoor applications the frequency bands allocated to FPLMTS could be divided into a number of adjacent sub-bands. Indoor and outdoor cells in the same area could use adjacent sub-bands in order to deal with the case in which an outdoor cell blocks channels in many adjacent indoor cells.

5.5.3.4 *Mobile earth station (Interface R3)*

Mobile users operating over wide areas would benefit by having direct access to both mobile satellite and terrestrial systems. This could be most readily achieved by the use of adjacent satellite and terrestrial mobile bands.

Given the prospect of more than one mobile satellite system, it is desirable that technical and/or functional compatibility of mobile earth stations be a goal of the FPLMTS, so that mobile users are able to roam regionally and/or worldwide with the same piece of equipment.

It is noted that WARC-MOB-87 Resolution No. 44 urges administrations to encourage the development and manufacture of compatible user equipment in the mobile-satellite service.

5.6 *Radio frequency radiation hazard considerations*

Research to date has suggested, in general, that vehicular mobile radios do not cause radio frequency exposures in excess of currently accepted safety guidelines when factors such as intermittency of transmission and antenna location with shielding afforded by the metal body of a vehicle are taken into account. However, frequency-related power absorption studies in test animals and models of man with a constant power density of 1 mW/cm^2 indicate that, in general, average specific absorbed power increases by a factor of approximately 100 for every increase in frequency by a factor of 10 in the range 1 MHz to 400 MHz.

These results suggest that there may be a range of combinations of frequency bands used and power density required which places limits on the safe operation of FPLMTS equipment. As frequency is increased, adverse propagation effects would require a compensating increase in FPLMTS radio power levels so that equivalent cell coverage can be maintained.

Question 52/1 of Study Group 1 requests the study of methods of predicting field strengths in the presence of transmitters for the purpose of allowing the assessment of radiation hazards by competent authorities (see Report 671). It is noted the IEC technical committee 12 has activities in this area [WHO, 1981; ANSI, 1982; IRPA, 1984; NCRP, 1984].

5.7 *Conclusions of frequency considerations*

5.7.1 *Evolution of traffic characteristics*

The current studies have identified the potential needs of the users for mobile telecommunications, in particular public mobile telecommunications.

These potential needs point to new categories of service and additional traffic, with voice and data being regarded as of similar importance in the future. A new category of service is appearing which requires low power microcells and generates estimated amounts of traffic in the order of thousands of Erlangs per km². However, this category of traffic is not the most spectrum demanding due to the intensive frequency reuse. The consequence is that there is likely to be an increased demand for spectrum usage in the field of public mobile communications.

5.7.2 *Consideration of spectrum needs*

Consideration of spectrum needs in the mobile field shows that the designation of frequency spectrum should take into account different factors.

Use of a frequency band between 1 GHz and 3 GHz is consistent with propagation and RF technology factors.

The concept of regional and/or worldwide roaming is an integral part of FPLMTS. Therefore the inclusion, at least partly, of a worldwide common frequency band would appear to be the first choice to enable universal access, particularly for personal stations.

Regional and national differences can be recognized in order to meet the varying magnitude of the demand for FPLMTS in different geographical areas. It will not be necessary to use the entire available spectrum in all locations.

It should also be noted that the equipment which would benefit from worldwide spectrum allocation, such as personal hand-held equipment, would likely have a limited range.

5.7.3 *Sharing possibilities*

The latter observations offer more possibilities for sharing with other services which are presently using the spectrum between 1 and 3 GHz. It should be noted that FPLMTS is to be an advanced system where control of the equipment will allow for adapting to various situations. In particular dynamic power control and frequency agility would aid conformance with planning criteria and should be further investigated. Frequency sharing or coexistence with other radio services may be studied in more detail by developing suitable algorithms and models.

5.7.4 *Timescales*

It is expected that existing mobile regional systems will likely be saturated around the end of the century.

The availability of systems with worldwide universal access is desirable. To this end it will likely be necessary that spectrum be designated for FPLMTS

on a worldwide basis. This could be addressed at the next competent World Administrative Radio Conference.

Studies should be undertaken so that spectrum could become available and sharing criteria determined with a view to satisfying the potential evolving needs of FPLMTS around 1998 and beyond.

6. Compatibility and commonality

6.1 *Compatibility*

6.1.1 *General*

Compatibility enables a service to be provided between equipment on both sides of an interface. These generic interfaces are described in 4.1.2 and shown in Fig. 3. See, in particular, interfaces 1, 3 and 8.

6.1.2 *PSTN/ISDN - FPLMTS network and service compatibility*

Network compatibility refers to all interfaces except 8 and 9 of Fig. 3.

Tables VII and VIII detail some points of service compatibility between FPLMTS and ISDN. They introduce the concept of mapping ISDN B and D channels onto FPLMTS I and C channels.

6.1.3 *MS/PS, terminal and equipment compatibility*

This refers to interfaces 1, 8 and 9 of Fig. 3 and is mentioned in section 4.1.1.

6.1.3.1 *FPLMTS - ISDN terminal compatibility*

Interface 8 in Fig. 3 recognizes that, in addition to terminals specifically designed for FPLMTS, ISDN user terminals or other CCITT standard terminals should be usable in FPLMTS (via appropriate adapters).

6.1.3.2 *Radio interface compatibility*

Interface 1 of Fig. 3 refers to the generic radio link. Frequency allocation considerations, as they affect the generic radio link, are discussed in 5.5.3. Specific radio interface considerations are discussed in 4.3.

6.1.4 *Billing and charging*

Similarly, a consistent structure for billing/charging is desirable for compatibility, but this may limit the flexibility of the system operator for innovative marketing.

6.1.5 *Further study*

In general, the detailed form of compatibility should be the subject of further study following a more precise definition of the services of FPLMTS.

The OSI layer model should be applied here in a form suitable for mobile communications.

6.2 *Commonality*

Commonality of technical elements of FPLMTS, for example, speech codecs, RF components, etc., enables cost benefits to be passed on to network operators and users. The specifications of FPLMTS have not as yet been defined in enough detail for meaningful comment on particular areas of desired commonality, but the attention of system designers is drawn to the benefits of this subject.

Table VII
ISDN - FPLMTS Compatibility (General)

ISDN	FPLMTS (General)
Digital	Digital at interface between FPLMTS and ISDN network (as a minimum) (interface 2 of Section 4.1.2, Fig. 1).
Basic Access (integrated voice and data) made up of 2B + D where B = 64 kbit/s (Bearer channel) and D = 16 kbit/s (Signalling channel)	B Channel Functionality (flexible bit rate). For voice (at 2B + D baseband) 32 kbit/s, more likely 16 kbit/s or lower; based on spectrum and coding efficiency trade-offs.
Consider B and D channels as information content building blocks.	Compatible with ISDN D channel standards. D channel functionality may be spread over a number of radio channels depending on the multiple access techniques used. For FPLMTS, I (Information) and C (Control) will be used in place of B (Bearer) and D (Signalling) respectively, to signify the above differences within otherwise similar structures.
Error performance (64 Kbit/s) is stated in CCITT Recommendation G.821. This will be acceptable for data transmission (therefore very acceptable for speech.)	Voice quality will be a function of the coding and channel bit error ratio. Source and channel coding may be optimized for speech. Acceptable error performance for data traffic on mobile radio channels is a subject for further study.
Bearer Services at 64 kbit/s can be used to support voice and data services.	Extension of this capability to the mobile station would require provision of up to 64 kbit/s through the FPLMTS. Information on true user data rates to and from ISDN would make the provision of compatibility between FPLMTS and ISDN more efficient in spectrum usage.

Table VIII
ISDN - FPLMTS Compatibility (Specific)

ISDN	FPLMTS (Specific)	
	Mobile Station/ Personal Station	Mobile PABX/ Mobile Concentrator
Basic Access (2B + D) 144 kbit/s	2I + C, I + C, C to 2B + D (Note 1)	
Primary Access 1536 kbit/s (23B + D) or 1920 kbit/s (30B + D)		mI + nC (Note 2)

Note 1: Full extent of ISDN Basic Access at 144 kbit/s (2B + D) could be provided by FPLMTS at the expense of reduced FPLMTS service availability due to the increased RF resource required to support such mobile use.

Note 2: The exact values for m and n are subject for further study

7. Further work

The form of Future Public Land Mobile Telecommunication Systems (FPLMTS) in accordance with Decision 69 has been investigated.

The present Report and Recommendation 687 should be regarded as the first step in a long process of refinement. Many topics require further investigation; those which have been specifically identified are as follows:

7.1 Services

- Service Access (Section 3.4)
- Mobility Services (Section 3.3, 3.5)
- Network services (Section 3.5.5)
 - * Bearer service including data service (Section 3.5.5; 1 - 3)
 - * Teleservices (Section 3.5.5.4)
 - * Supplementary services (Section 3.5.5.5)
- Speech quality (Section 3.6)
- Security (Section 3.7)
- Quality of service (Section 3.8)

7.2 System architecture

- Degree of FPLMTS integration and the level of integration with PSTN/ISDN (Section 4.1)
- A user-network interface standard (Section 4.1.1)
- The evolution and implementation of personal communications within FPLMTS (Section 4.2)
- Radio Interface requirements (Section 4.3)
- Signalling protocol (Section 4.3.2)
 - * Refinement of Telecommunications Service Definitions

- * Identification of all necessary functionalities needed for the radio interface.
- * Identification of information and message flows between the above mentioned functional entities and their peer entities.

- Compatibility or commonality between PTN service in FPLMTS and PSTN/ISDN (Section 3.4, 4.4)

7.3 *Frequency considerations*

- Traffic considerations (Section 5.3)
- Radio Propagation characteristics (Section 5.5.1)
- Sharing possibility (Section 5.7.3)
- Spectrum availability (Section 5.7.2)

7.4 *Compatibility and Commonality*

- Detailed form of compatibility (Section 6.1.5)
- Structure of Information (I) and Control (C) channels (Section 6.1.5)
- Specification of FPLMTS and commonality (Section 6.2)

7.5 *Others*

- Operations, administration and maintenance

In addition to these areas of further work there are the matters of liaison and close cooperation with CCITT Study Groups, IEC, etc. To this end, it is anticipated that a number of CCIR Study Group 8/CCITT joint expert working meetings will be necessary.

A separate Report 1155 has been prepared highlighting the needs and interests of developing countries.

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REPORT 1157

INTEGRATION OF PUBLIC MOBILE RADIOCOMMUNICATION SYSTEMS

(Question 52/8)

(1990)

1. Introduction

This Report describes the general principles for the integration of public mobile radiocommunication (terrestrial) systems. Recommendation 622, Report 742 and Report 1051 are considered helpful in designing integrated public mobile radiocommunication systems. The integration between terrestrial and satellite systems has been discussed in Report 1177.

2. Concept of system integration2.1 Model

Among public mobile communication systems, land, maritime, aeronautical and satellite mobile systems are candidates for integration.

A model of a public mobile radiocommunication system is shown in Figure 1. This model is composed of a mobile/portable station, a base station, a mobile switching service centre (MSC) and an operational and maintenance (O & M) facility. Control functions such as signalling, location registration, hand over, numbering and charging are also important for system operation. These components or functions can be combined in several ways and their careful integration can contribute to system economy and operational simplicity.

2.2 Integration level

Integration levels among terrestrial systems are classified as follows:

i) Total compatibility

When all the equipment and the O & M facilities are fully compatible with respect to every service, there is total compatibility. This compatibility, in which all services are provided by an integrated system, is the highest level.

ii) Partial compatibility

When diverse systems do not share all components or control procedures, there is partial compatibility. Within the concept of the partial compatibility, various integration levels are possible.

A practicable integration level should be determined by taking into account regulatory, operational, and technical characteristics. Report AZ/8 gives more detail on these and other issues.

3. Considerations for system integration

3.1 Operational characteristics

i) Different frequency bands are assigned to each service according to the present Radio Regulations. However, most suitable frequency bands should be discussed from a technical point of view.

ii) Service quality, such as blocking probability, speech quality and delay differs between mobile service access systems. Where the quality of service differs significantly from the PSTN standards, care must be taken that customers do not blame the PSTN operator for poor quality.

iii) Parameters related to the call processing, service areas and so on differ according to the requirements of each system.

iv) The charging rate depends on system cost and generally varies from service to service.

v) A common numbering plan should be used for all services.

vi) O & M facility compatibility contributes to the economic system production and operation.

3.2 Technical characteristics

i) The technical characteristics of transceivers depend on the assigned frequency. The transmitting power and the receiver sensitivity are determined according to the required zone radius and the required speech quality.

ii) The environmental specifications of the mobile stations are determined according to the circumstances of each service.

iii) The cell structure and channel re-use are designed according to the traffic densities, required frequency efficiencies, propagation characteristics and quality requirements.

iv) The location of a base station is determined according to the required service area, the geographical features and propagation characteristics.

v) The control channel configuration is designed according to traffic density, signalling reliabilities and frequency efficiency.

4. Practicable integration level

The practicable integration level should be determined considering the following items:

- equipment;
- control procedures;
- operational features.

Based on the Japanese experience, an example of an integrated system of land, maritime and aeronautical services is shown in Annex I of this report.

5. Conclusion

This report illustrates the general principles for the integration of public mobile radiocommunication systems.

Practicable integration levels, frequency bands for integrated systems, operational considerations and relationships with satellite systems are matters for further study.

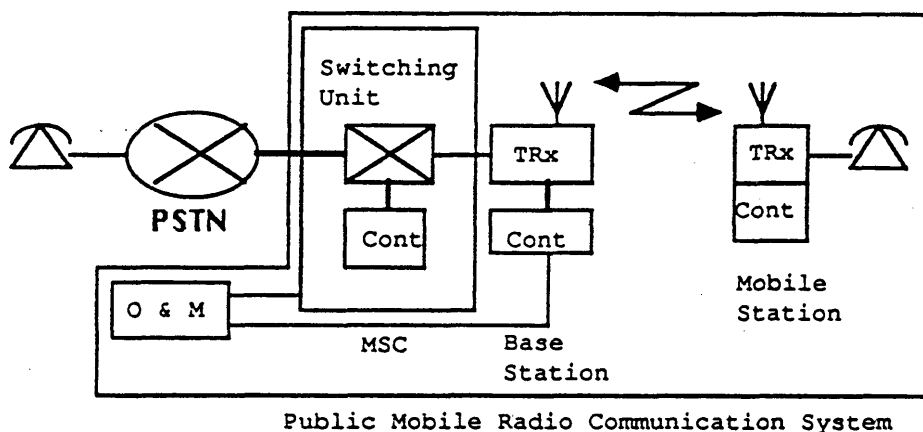


FIGURE 1

System configuration (terrestrial systems)

ANNEX I

INTEGRATION OF PUBLIC MOBILE RADIOCOMMUNICATION SYSTEMS IN JAPAN

The aeronautical (Report 1051) and the maritime mobile systems [Kuramoto and Ogawa, 1987] were integrated into the public land mobile telephone system (Report 742) in May 1986 and November 1988, respectively. In this integrated system, control procedures, O & M facilities and related equipment are shared in order to improve system economy and to simplify system operation.

1. System configuration

The system configuration is shown in Figure 2.

The system is composed of the following equipment:

- mobile stations/portable stations dedicated to each service;
- base stations dedicated for each service;
- mobile switching service centre comprising a mobile control unit and a switching unit;
- operation and maintenance (O & M) facilities.

2. General characteristics

The operational and technical characteristics of the integrated system are based on Recommendation 622.

The major characteristics of this integrated system are shown in Table I.

3. Integration level

Most items, except for the radio frequencies, have been integrated. The detailed integration levels are as follows.

3.1 Equipment

The following equipment is compatible:

- switching unit and mobile control unit;
- telephone part of the mobile station;
- O & M equipment.

The following equipment remains dedicated because of differences in assigned frequencies, locations of base station and coverage areas:

- transceivers in mobile/portable stations or base stations.

3.2 Control procedure

The control procedure for the radio-path is compatible except for the control channel scheme and the service quality parameters. In the maritime and aeronautical systems, the paging channel and access channel are combined into one radio control channel because of low traffic.

The control procedure between base stations and switching unit is compatible.

3.3 Operational features

The following elements are shared:

- charging principle, except for the charging rate;
- numbering plan (commonly used in the newly introduced public land mobile telephone system, as well as in the above-mentioned system);
- supervision and control of all equipment, radio paths and wire lines;
- O & M.

REFERENCE

KURAMOTO, M. and OGAWA, K. [October, 1987] - Integration of mobile communication systems in NTT. Fifth World Telecommunication Forum, Vol. III, pages 51-59.

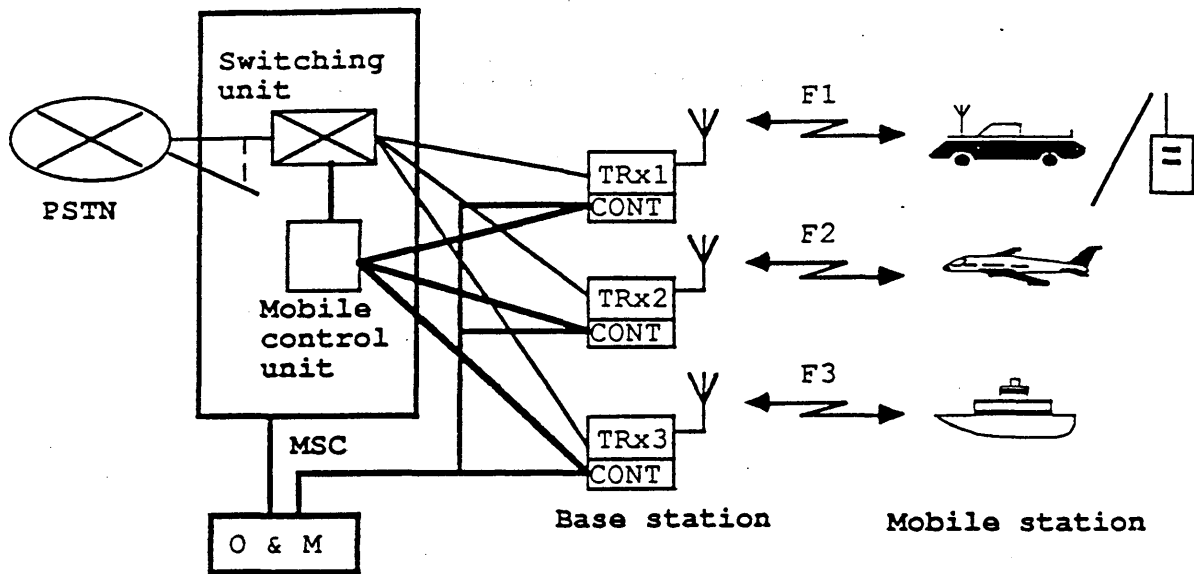


FIGURE 2 - Configuration of the integrated system

TABLE I - Technical characteristics of the integrated system

ITEM	LAND	MARITIME	AERONAUTICAL
Frequency (MHz)	800-900	250	800-900
Channel spacing (kHz)	25	12.5 interleave	25
Maximum base station e.r.p. (W)	50	40	130
Nominal mobile station transmitting power (W)	5	5	10
Zone radius (km)	3 -10	50 - 100	400
Numbering plan	Common		

REPORT 742-3

PUBLIC LAND MOBILE TELEPHONE SYSTEMS

(Questions 37/8 and 39/8)

(1978-1982-1986-1990)

1. Introduction

Public land mobile telephone systems are defined as land mobile systems for public correspondence via radio stations connected to the public switched telephone network (PSTN).

Part A of this Report deals with the general principles of public land mobile telephone systems (both conventional and cellular) and, in particular, with system parameters and technical characteristics which are important for international operation. The basic concept of the cellular techniques is outlined in Report 740.

Part B of this Report considers the implementation of international systems.

The major characteristics of some existing or imminent public land mobile telephone systems are given in Part C of this Report, together with brief descriptive material on the status of these systems and other unique aspects of their design.

PART A**GENERAL PRINCIPLES OF PUBLIC LAND MOBILE TELEPHONE SYSTEMS****1. Operational aspects**

The following general operational aspects should apply:

- automatic setting up and charging of calls to and from the mobile station;
- for international systems, the ability to set up calls between the mobile stations and any fixed telephone subscriber or any other mobile telephone subscriber within the system;
- costs should be charged in a manner consistent with charging principles in the public switched telephone network;
- the introduction of the system should not necessitate any significant changes in the fixed telephone networks.
- blocking probability should be kept within limits similar to that of the PSTN services at all stages of development;
- continuous control of call quality should be maintained, with automatic hand-over between base stations if needed.

Any specific system must have at least two well-defined interfaces:

- the radio interface between the land-based system and the mobiles it serves; and
- the wire-line interface to the public switched telephone network.

Other internal interfaces may also be defined such as:

- the billing system;
- the interface between controllers for the handling of roamers;
- the methodology for communicating between base sites and controllers; and
- the man-machine interface for operational purposes including maintenance.

2. Interworking with the PSTN

CCITT Recommendation Q.70 specifies the necessary interworking requirements between the PSTN and the mobile network.

* The Director of the CCIR is requested to bring this Report to the attention of the CCITT.

3. Numbering plan and routing

A difficult problem in a fully automatic mobile telephone service is to select a numbering plan for the mobile subscribers which is compatible with the established fixed network. A numbering plan based on the conventions built up in the land network may impose severe limitations on mobile routing especially since the subscriber number in a mobile system no longer is referred to the location of the subscriber.

One solution is to allocate a special access number to the mobile service, in which case all dialling to the mobile network is handled free from the routing routines of the telephone service.

Since national numbering plans vary in their capability to manage long numbers and to allocate codes to special services, an international numbering plan needs to be established. CCITT Recommendations E.212 and E.213 deal with this subject.

4. Roaming

To be able to automatically set up calls between the mobile stations and any fixed telephone subscriber regardless of country, a fully automatic roaming facility is necessary. The mobile system has to be informed of the location of the mobile subscriber as he moves from one location to another.

Further studies are required before adequate technical and operational roaming procedures can be defined. For this reason Recommendation 624 has been written.

Roaming also requires compatibility in frequency bands, channel spacing, and operational signalling protocols and codes.

5. Charging

The principle for charging in mobile networks varies in different national networks.

The question of charging therefore needs to be studied by the CCITT to achieve international agreement.

6. Radio-frequency considerations

Current systems use the 450 MHz band, the 800-900 MHz band, and other bands according to existing possibilities.

Conventional trunked systems can use only a small number of channels, but cellular systems are only efficient when a larger allocation (e.g. 300 channels) is available since the allocation must be divided into sub-sets in order to implement the cellular plan. Further studies may be necessary before the choice of a radio-frequency band for an international public mobile telephone service can be made.

7. Signalling

The system functions which are required in an automatic public mobile telephone system require a fairly complicated signalling system. The mobility of the subscribers creates a need for information transfer of a kind that has no equivalence in the fixed telephone network. This applies both to the updating procedure, performed when the mobile station enters a new location area, and to the hand-over procedure when a call in progress is switched from one base station to another.

Signalling errors in the radio channel may result in loss of control and the inability to establish a traffic channel between mobile and base stations. The reliability with which a traffic channel may be obtained should be adequately high, since:

- radio channels correspond to lines in the public-switched telephone network, and the signalling and supervision reliability should be as good as in that network;
- to fail to establish and maintain a traffic channel means that the base station loses control of the mobile station. The probability of failure therefore should be made as small as possible;
- billing accuracy is of high importance.

The reliability of control channels can be improved by (for example) diversity, error-control coding, message exchange protocols and simulcast (see Reports 903 and 1022 and Question 67/8).

Attempts to seize a channel may fail due to simultaneous requests by two or more mobile stations. The problem can be reduced by a polling technique, but in a high-capacity system it is more efficient to mark the busy/idle status of the channel [Fluhr and Porter, 1979; Okasaka, 1978].

8. Reduction of ineffective air-time

In all public radiotelephone systems, including cellular, a significant delay resulting from dialling, switching, and ringing time occurs between the initiation of a call and the start of the conversation. Some cellular systems store the dialled digits in the mobile unit prior to transmission (called "pre-origination dialling") which significantly reduces the ineffective air-time.

The traffic capacity can be increased by assigning traffic radio channels only when both parties are ready to speak [Tridgell, 1977], or by queueing calls on a first-come-first-served basis until they materialize and a free traffic channel becomes available.

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PART B

CONSIDERATIONS FOR THE DEVELOPMENT OF INTERNATIONAL SYSTEMS

1. General

This part identifies considerations that are appropriate for the development of international public land mobile telephone systems and services that may be adopted by many countries around the world. Systems that provide common services amongst a few nearby countries have been implemented.

2. Implementation of international systems

To allow for the implementation of international systems, it would be ideal to select a common band of frequencies with common world-wide service and technical standards. As the development of these agreements could take a long time, it may be necessary to choose methods involving interim stages which immediately take advantage of some of the benefits of international compatibility. Such implementations may be possible by the addition of agreed narrow-band or wide-band systems [Murtonen, 1985].

Narrow-band systems may be added with channels directly adjacent to those of existing systems, or in the case of some existing systems, by interleaving the channels of both systems. In both cases there would be systematic expansion of new channels into existing system channel assignments as over time older systems are retired.

Similarly, wide-band systems (spread spectrum, TDMA, etc.) may be added using new or cleared spectrum adjacent to the channels of existing systems. Expansion of new systems is possible when the older systems are retired.

Additionally, it may be possible to overlay some spread-spectrum systems onto existing assignments and to simultaneously operate both the new and the existing systems; however the overlay of spread-spectrum systems requires further study, especially in the case of a heavy concentration of users of these systems.

3. Factors needing international agreement

- channel allocations and spacing;
- class of emission;
- modulation characteristics;
- transmitter and receiver standards;
- signal processing techniques, such as companding;
- use of diversity reception;
- signalling and supervision methods and protocols.

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PART C

SYSTEMS BEING INSTALLED OR PLANNED IN THE NEAR FUTURE

1. High-capacity cellular public mobile telephone services

1.1 Introduction

High-capacity cellular systems have been developed in the United States of America, in Canada, in the United Kingdom, in the Nordic countries, in Japan, in the Federal Republic of Germany and in Italy to provide a high-quality, nationwide-compatible radiotelephone service.

Systems are now in operation or being installed in the United States of America and Canada, where 666 channels split into two sub-bands and allocated to two systems per city, have been set aside. These consist of a varying number of cells, depending on the area to be served. The system capacity can be expanded to meet the anticipated subscriber demands which are expected to be more than 100 000 in larger cities. Annex I provides more details on the general characteristics of these systems.

Two nation-wide systems (TACS), similar to those in North America but based on channels spaced at 25 rather than 30 kHz have been developed in the United Kingdom, and were brought into use in January, 1985. Each system will offer coverage to at least 90% of United Kingdom inhabitants by 1990. In November, 1985, the total number of subscribers on the two systems exceeded 35 000. Annex V provides more details on the general characteristics of the system.

The Japanese system MCS-L1 now provides nation-wide service to more than about 70% of all Japanese cities, using 9 mobile telephone switching centres connected to the common channel inter-office signalling network. _____

The numbers of subscribers in the Tokyo metropolitan and Osaka areas are expected to reach the MCS-L1's maximum capacity in the near future. To meet demands, such as handling capacity increase, lower rates and up-graded services, the new 900 MHz band high capacity land mobile communications system (MCS-L2) has been introduced. The system handling capacity is expanded to about 100.000 subscribers in the large cities, by improving spectral efficiency to about three times the present system. Annex II provides more details on the general characteristics of the systems.

The Nordic mobile telephone system (NMT-450) was put into operation in the participating countries, Denmark, Finland, Norway and Sweden during the end of 1981 and the beginning of 1982 with country-wide continuous coverage and full roaming capability. During the last year, cell splitting has been introduced in the big cities in order to cover the traffic demand. Cells as small as 1.0 km cell radius are used. In October, 1985, the number of subscribers exceeded 200 000 corresponding to 0.9% of the population. During 1986, this system will be expanded to Iceland with full roaming capability between all the countries. The system will be expanded into the 900 MHz band in late 1986 and is designed for 2000 channels including the utilization of interleaving. At start-up, 799 channels were used (NMT-900). Annex III provides more details on the general characteristics of the system.

In Belgium, the Netherlands and Luxembourg, a cellular system providing international roaming is in operation. This system is an NMT-450 system with modifications such as 20 kHz channel spacing and the use of a syllabic compandor.

The C450 network in the Federal Republic of Germany was brought into service in 1985. It provides a complete coverage of the Federal Republic of Germany. The ultimate subscriber capacity is estimated to be at least 200 000. Specific features are: off-air call set-up and queueing operation. Annex IV provides more details on the general characteristics of the system.

The Italian second-generation mobile telephone system operating in the 450 MHz frequency band was brought into service in the Rome and Milan areas beginning in September, 1985. It is expected to provide a complete coverage of the Italian territory by 1988. Annex VI provides more details on the general characteristics of the system.

1.2 System features

The following system features are common to all described systems unless stated:

- cellular structure in both base-to-mobile and mobile-to-base directions with frequency re-use and inter-cell hand-over (switching call-in-progress);
- automatic, two-way, direct-dial operation;
- full duplex communications;

- voice channel supervision via continuous tone (North American and United Kingdom systems, NMT system, Italian system), or digital control channel (C450, MCS-L2 system);
- automatic roaming;
- dedicated control channels used for call set-up (North American and United Kingdom system, Japanese system, C450 system, Italian system);
- pre-seizure dialling;
- cell splitting during system growth.

1.3 *Conclusions*

Cellular radiotelephone systems using the parameters and features described above have been installed or are planned for the near future. There is a large demand for a land-mobile radiotelephone service in many parts of the world, and it is considered that systems with the above characteristics are capable of meeting that demand.

2. **Hybrid systems combining public mobile telephone system and dispatching type networks**

2.1 In 1985 France opened a system with the main characteristics described below.

2.1.1 *General characteristics*

- Dispatch type operation for half-duplex private network application offering the following links:
 - base-to-mobile or fleet (or sub-fleet);
 - mobile-to-base;
 - mobile-to-mobile or fleet (or sub-fleet);
 - access for certain mobiles to the PSTN;
- duplex public radiotelephone operation; a unique number is used to call the mobile wherever it is located;
- cellular structure covering a wide geographical area with automatic roaming using medium-sized cells (15 to 20 km radius); cell splitting possible during system growth;
- digital data transmission is under study;
- the system provides for priority subscribers;
- hand-over is under study.

Annex VII provides more details on the general characteristics of the system.

TABLE 1 - General system specifications

Feature	North American system	Japanese system MCS-L1	Japanese system MCS-L2	NMT-450 system	NMT-900 system	C450 system	United Kingdom system - TACS	Italian 450 MHz system	French system RADICOM 2000
Class of emission	Voice 40K0G3E Control 40K0G1D	Voice 16K0G3E Control 16K0F1D	Voice 8K50G3E Control 8K50F1D	Voice 16K0G3E Signalling 16K0G2D	Voice 16K0G3E Signalling 16K0G2D	Voice 14K0G3E Control 14K0F1D	Voice 32K0G3E Control 32K0F1D	16K0G3E	Voice 11K0G3E
Transmit frequency bands (MHz)									
- base stations	870-890	870-885	870 - 885	463-467.5	935-960	461.3-465.74	935-950	460-465	202.7-207.5 207.5-215.5 424.8-427.9 194.7-199.5 215.5-223.5 414.8-417.9
- mobile stations	825-845	925-940	925 - 940	453-457.5	890-915	451.3-455.74	890-905	450-455	
Duplex separation (MHz)	45	55	55	10	45	10	45	10	8 10
Channel spacing (kHz)	30	25	12.5	25	12.5	20 (10)	25	25	12.5 12.5
Total number of duplex channels	666 (333 in each of two sub-bands, including 21 signalling channels)	600	1 200	180	1999	222	600 (300 in each of two sub-bands, including 21 signalling channels)	196	1024 256
Maximum base station e.r.p. (W)	100 (1)	50	20	50	100	100 (adaptive control)	100	25/2.5	25 to 70
Nominal mobile station transmitter power (W)	3	5	1	15	Mobile 6 Hand-held 1 (with autonomous power control)	15 (+3 dB antenna gain adaptive control)	7 - class 1 mobile	10/1	11
Typical cell radius (km)	2-20	3 (urban area) (1) 10 (suburban area) (1)	3 (urban area) (2) 10 (suburban area) (2)	1-40	0.5-20	2-30	2-20	5-20	20
Voice signals									
- type of modulation	PM	PM	PM	PM	PM	PM	PM	PM (FM, if the band inverter is used)	PM
- peak deviation (kHz)	± 12	± 5	± 2.5	± 5	± 5 (including supervisory signal)	± 4	± 9.5	± 5	± 2.5
- processing	2:1 syllabic compandor	2:1 syllabic compandor	2:1 syllabic compandor	-	2:1 syllabic compandor (CCITT Rec. G.162)	2:1 syllabic compandor	2:1 syllabic compandor	2:1 syllabic compandor (CCITT Rec. G.162)	Syllabic compandor base to mobile

TABLE I (continued)

Feature	North American system	Japanese system MCS-L1	Japanese system MCS-L2	NMT-450 system	NMT-900 system	C450 system	United Kingdom system - TACS	Italian 450 MHz system	French system
Control signal: - type of modulation - peak deviation (kHz) - code form - transmission bit rate (kbit/s) - effective information transmission rate (kbit/s), (depends on message type)	FSK ± 8 Manchester 10	FSK ± 4.5 Manchester 0.3	FSK ± 2.0 Manchester 2.4 1.3 - 1.64	FFSK ± 3.5 NRZ 1.2	FFSK ± 3.5 NRZ 1.2	FSK ± 2.5 NRZ 5.28	FSK ± 6.4 Manchester 8	FM ± 4 Multifrequency (two of seven) ⁽¹⁾ About 0.1	FFSK ± 1.7 NRZ 1.2
Control signal for in-service: - type of modulation - peak deviation (kHz) - code form - transmission bit rate (kbit/s) - effective information transmission rate (kbit/s), (depends on message type)	0.27-1.2	0.12-0.18	FSK ± 0.6 Manchester 0.1 0.04	About 0.46	About 0.46	1.82	0.22-0.96	About 0.1	0.46
Error protection coding Control signal: - base-to-mobile - mobile-to-base Control signal for in-service: - base-to-mobile - mobile-to-base	Shortened (63:51) BCH repeated ⁽²⁾ (40:28) BCH (48:36) BCH	Shortened (63:51) BCH; shortened (23:12) BCH (43:31) BCH (43:31) BCH-access; (23:12) BCH-paging	Shortened (63:51) BCH; (40:28) BCH (40:28) BCH (23:12) BCH; (23:12) BCH	Type B1 burst error-correcting convolutional code (Hagelbarger)	Type B1 burst error-correcting convolutional code (Hagelbarger)	(15:7) BCH	Shortened (63:51) BCH repeated ⁽²⁾ (40:28) BCH (48:36) BCH		Hagelbarger code (6:19)
Error detection	Min. 11 per Max. 89 200 bits	Min. 3	Min. 3			Min. 40 per 150 bits	Min. 11 per Max. 89 200 bits		
Error correction	Min. 5 per Max. 83 200 bits	1 error	1 error for 2.4 kb/s 2 error for 0.1 kb/s	Min. 6 with 19 bit guard space	Min. 6 with 19 bit guard space	Min. 20 per 150 bits	Min. 5 per 200 bits		Min. 6
Message protection	Recycled control signal transmissions; repeated control signal transmissions with bit-by-bit majority voting	Recycled control signal transmissions; simultaneous transmissions from the base stations in a control zone	Recycled control signal transmission; simultaneous transmissions from the base stations in a control zone	Frame acceptance procedure depending on message category	Frame acceptance procedure depending on message category	Adaptive message repetition in case of error	Recycled control signal transmissions; repeated control signal transmissions with bit-by-bit majority voting	Autocorrelation control of the encoded message with repetition, depending on message type, in case of error	Frame repetition

⁽¹⁾ Exceptions may be allowed, depending on circumstances.⁽²⁾ A control zone covers about 10 cells.⁽³⁾ The modulation rate is 50 ms/character.⁽⁴⁾ Repeated 5 to 11 times, depending on message type, with bit-by-bit majority voting. In addition, to achieve decorrelation, two message streams are interleaved on the common paging channel.

ANNEX I
NORTH AMERICAN SYSTEM

PART I
GENERAL DESCRIPTION

1. Operational and system characteristics

1.1 Purpose

The land mobile telephone system is designed to permit an automatic exchange of traffic with the public switched telephone network (PSTN) and offers a service, from the user's point-of-view, similar to that of land-line calls: high voice quality, high reliability, low blocking, and relatively low cost.

1.2 Cellular properties

- Growth from large cells at start-up to small cells at maturity, with a mix of several sizes, is permitted.
- Hand-off is allowed, up to at least one per call per minute.
- For spectrum efficiency (channel re-use), small values for the number of channel sets are used: appropriate to the multipath (Rayleigh) fading typical of urban mobile channels, the terrain variability, the antenna patterns selected and the RF quality desired. A local median C/I target of 17 dB at the 90th percentile is suggested. (If the standard deviation is 8 dB, this is equivalent to a median C/I of 27 dB, at the nominal boundary of the cell.) Other quality objectives are of course permissible.

1.3 Signal processing

- The widest deviation and receiver pre-detection bandwidth appropriate to the cellular concept (which permits separation of the spectrum into channels by both frequency division and spatial separation) are used.
- Voice processing using a 2:1 compressor/expander is employed to enhance the perceived quality of the received signal.
- The coding of digital signalling uses burst-error detection and correction.
- Design for a peak-to-r.m.s. ratio of between 6 dB and 20 dB for the demodulated voice signal at the receiver is possible; that is, a range of quality comparable to land-line service should be accommodated.

1.4 Service protection

- A transmitted serial number prohibits the use of stolen units or use by unauthorized callers.
- The introduction of means to ensure privacy of communication is not precluded by the design.
- Means for preventing intelligible crosstalk is included in the system design.
- The successful completion of the entire call takes place on at least 99% of the calls on which an initial two-way connection was established between calling and called parties.

1.5 Services offered

- Voice message telephone service with automatic calling to and from the PSTN.
- Data services.
- Vehicular-mounted equipment.
- Hand-held equipment.
- Other enhancements consistent with mobile services: call forwarding, repertory dialling, message waiting, etc.

2. Signalling formats and codes

See Part 2.

3. RF equipment

- 3.1 The RF equipment allows full duplex operation.
- 3.2 Frequency tolerance: base station – 1 part in 10^6 ; mobile – 2.5 parts in 10^6 .
- 3.3 RF power limit: 100 W e.r.p.; more when adjacent systems are not expected.

- 3.4 Power tolerance: ± 2 dB.
- 3.5 Mobiles are tunable to each of the allocated channels.
- 3.6 Receiver sensitivity: -116 dBm from a 50Ω source applied to the antenna terminals should produce a 12 dB SINAD (C-message weighting).
- 3.7 Diversity: optional at mobile; also optional, but highly recommended, at base station.
- 3.8 Antenna height: 30 to 45 m, higher when adjacent systems are not expected to interfere. With careful design, according to local practice, greater heights may be used.

4. Land-based control equipment

- 4.1 Connection to the PSTN is accommodated on a fully automatic basis.
- 4.2 Mobile-to-land, land-to-mobile, and mobile-to-mobile calls are permitted.
- 4.3 Hand-off of mobiles is a feature of the system plan (see § 1.2).
- 4.4 Supervision of the radio channel is sufficiently robust so that accurate billing timing can be expected (accurate to ± 5 s); billing to the proper number takes place with more than 0.9999 probability.
- 4.5 The system can provide for automatic roaming within an administration's territory.
- 4.6 The system is designed for blocking objectives similar to the PSTN.

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PART 2

COMPATIBILITY SPECIFICATION FOR CELLULAR CODES, FORMATS, AND PROTOCOLS*

1. Modulation interface

1.1 Voice

- 2 : 1 syllabic compandor (3 ms attack time, 13.5 ms recovery time).
- Pre-emphasis: 6 dB per octave, from 300 to 3000 Hz.
- Deviation limiter: ± 12 kHz.
- r.m.s. deviation: ± 2.9 kHz.

1.2 Control

- Manchester bit encoding.
- Deviation: ± 8 kHz for digital signalling; ± 2.0 kHz for continuous out-of-band supervision.

2. Baseband interface

2.1 Unique mobile identification

- Mobile unit identification number: 10 decimal digits coded into 34 bits, via specified algorithm.
- Serial number (not changeable): 32 bits, intended as an anti-fraud, anti-theft means.

* These standards are specified in precise detail in the United States Federal Communications Commission Document OST-53 and in the Canadian Department of Communications Document RSS-118, Annex A, dated October 1983, respectively; these are both compatible with EIA Interim Standard CIS-3A, available from the Electronic Industries Association, Engineering Dept., 2001 I (Eye) Street, Washington DC 20006. This specification describes a minimum capability; manufacturers and system providers have also written enhanced versions of this specification to describe capabilities above and beyond this minimum capability. This document comprises about 60 pages; in lieu of publishing that specification in its entirety, this Part 2 summarizes its major points, as they relate to codes, formats, and protocols.

2.2 *System-specific memory*

- Station class.
- Access method.
- First paging channel to use.
- Home system identification: 15 bits.
- Preferred system. The overall plan allows for one or two separate systems (A, B) in any city, and the unit must remember its assignment.

2.3 *Supervision*

Voice channels use a continuous supervisory audio tone (SAT); control channels and digital signals on the voice channels use a digital equivalent called "colour code".

- SAT frequencies: 5970, 6000, 6030 Hz.
- Digital "colour code": 2 bits coded into 7 bits, using a (7 : 4) Hamming code; used to identify the base station to which the message is intended.

In addition to these supervisory signals, a 10 kHz tone (deviation: ± 8 kHz) is used to indicate "on-hook".

2.4 *Malfunction*

Each mobile has a timer running independent of other logic functions, which will shut down the mobile transmitter within 60 s if the main logic ceases to function.

2.5 *Call processing*

Several primary tasks are specified; most important of these are (in order):

- d.c. power start-up and initialization;
- idle;
- activation:
 - origination;
 - paging response;
 - order response;
 - autonomous registration;
- conversation;
- release.

Other tasks, called secondary, are performed concurrent with, and controlled by, these primary tasks; these include:

- control data transmission and reception;
- control channel scanning;
- RF power control;
- response to system control messages;
- user interface management.

The primary tasks can be described briefly:

Initialization: when the unit is first turned on, the logic is made to enter the proper sequence; the memory and other mobile functional units can optionally be audited, and then the potential control channels are scanned, using a logarithmic measurement of signal strength, to find out which one will be able to tell the unit about its current proximity. Depending on whether the unit is "home" or "roaming", it may register its existence with the land system. It should also determine a control channel over which it can be paged if it receives a call.

Idle: without radiating, the unit receives and decodes the continuous stream of system control ("over-head") and paging messages, using the error-detecting and error-correcting algorithms available. During this time, it also monitors signal strength and reinitializes itself if the signal becomes unusable.

Activation: if the unit receives a call or gets a command from the user interface to place a call, the control channels should be rescanned, the idle state of the selected control channel determined, and a seizure of the system attempted. The unit then identifies itself to the system and a voice channel assignment will be given to it by the land system.

Conversation: if the mobile is being called, a ringing sequence is initiated; otherwise, the conversation state begins immediately. During the call, "blank-and-burst" messages (comprising a brief blanking of the voice signal and a burst of control data) may be received, instructing the mobile to change power and/or channel. Supervision via the out-of-band and SAT continues.

Release: when the call is ended normally, the transmitter is turned off and the unit is made to revert to the idle task again. If channel conditions of trouble or low-signal are encountered, release should also take place, so that positive system integrity is maintained.

2.6 Signalling

A variety of messages are required to administer the system; these are:

TABLE II

Channel	Types of messages
Base transmit: Control channels Voice channels	Paging messages Channel assignments Overhead information Filler text Hand-offs Orders
Mobile transmit: Control channels Voice channels	Page responses mobile address serial number Origination mobile address serial number called number Orders Order confirmation Order confirmation Called numbers

The fundamental signalling format used is a shortened (63 : 51) BCH word in which the first 28 or 36 bits are the message and the final 12 bits are parity check bits calculated from the message bits. Of the message bits, the first four serve a control or "book-keeping" function and the rest form the specific messages. 28 information bits are used in the base-transmit direction and 36 in the mobile-transmit direction, so that the resulting codes are (40 : 28) BCH and (48 : 36) BCH, respectively.

A second level of redundancy — message repetition when transmitting and bit-by-bit majority voting when detecting — is layered on top of the basic block coding. Interleaving of two message streams adds further de-correlation of bit error probability, where this is possible. Messages are typically repeated 5 times, except on the base-to-mobile voice channel when hand-off messages are sent; in this case, the message is repeated 11 times.

Bit synchronization is augmented by providing a long 101010... sequence before each message. Word synchronization employs an 11-bit Barker sequence (11100010010), which possesses unique minimum distance properties.

2.7 Seizure

The effects of collisions during control channel contention are mitigated by two devices:

- between each 10 bits on the base-to-mobile control channel, an eleventh bit is inserted; its state informs the mobiles of the busy/idle status of the control channel, and its change relative to the timing of a seizure attempt tells the mobile logic whether or not its seizure was clean;
- a precursor of 48 bits is added to the mobile's asynchronous seizure attempt on the mobile-to-base control channel to
 - provide bit and word synchronization and
 - signify the base station to which the attempt is directed.

This latter scheme lowers false seizures caused by co-channel interference.

3. Other

- Power classes:
 - Class I: +6 dBW max.
 - Class II: +2 dBW.
 - Class III: -2 dBW.
- Power control: 7 steps, 4 dB per step.
- Other specifications: typically state of the art.

ANNEX II

GENERAL DESCRIPTION OF THE JAPANESE SYSTEM

PART I - MCS-L1

1. Radio link design objectives

1.1 Service quality

- Blocking probability: Radio channels are allotted to make the radio path blocking probability less than 3%.
- Speech quality: The design objective of the speech quality is to realize more than 80% sound articulation score, which can produce 100% sentence intelligibility, in the service area.

1.2 Radio zone configuration

Coverage reliability of 80% sound articulation score in more than 90% of the service area. To get this reliability:

- Maximum cell size is determined so that 90% of the sites at the zone boundary provide reliable reception. Median $C/N = 17$ dB.
- Channel re-use is determined so that 90% of the sites at the zone boundary provide a reliable wanted/unwanted signal ratio. Median wanted/unwanted (W/U) = 15 dB.

2. Control channel configuration

2.1 Control channel allotment

The control channels are dedicated and divided into sub-sets of the following two channels:

- paging channel: mobile stations always seize this channel and receive the paging signal, location data and access channel data;
- access channel: a mobile station randomly originating calls is controlled with this channel. To prevent double seizures, the base stations detect the access signal from mobile stations and broadcast the busy status on this channel and then prevent the signals from other mobile stations (ISMA: idle signal multiple access).

2.2 Multi-cell control technique

A control channel is assigned to cover a group of adjacent traffic cells, and simultaneously transmitted from each base station in the control area.

2.3 *Seizure*

The busy/idle status in the control channel is used to mitigate the effects of collisions during control channel contention.

2.4 *Reduction of co-channel interference*

In order to prevent co-channel interference, control zones and radio zones using the same frequencies are discriminated by J-code and K-code, respectively.

3. *Control sequence*

3.1 *Mobile origination*

The features required in a mobile origination call are:

- numbering from the mobile station that uses the same format as the PSTN;
- locations of mobile stations are detected at base stations by the field strength of access signals received from mobile stations;
- speech channels are assigned by the access channel and a loop check signal is exchanged on the assigned speech channel between mobile station and base station;
- dialling signals are sent through the speech channel after the speech channel is set up.

3.2 *Mobile termination*

- Numbering for the mobile station is as follows:
0 + A0 + (7 digits subscriber number), where A is charging class code and 3 or 4 is selected for it.
- A paging signal which is checked with the subscriber data at the home memory office, and is then simulcast from each base station in the control zone. If there is an answer from the mobile station, the next process is similar to the mobile origination. When there is no answer, i.e. if the mobile station is off-power or out of the service area, the originating subscriber will get a service announcement of the situation from the exchange (MSC).

3.3 *Hand-over*

A new speech channel is reassigned to mobile stations crossing a zone boundary:

- the base station detects *S/N* deterioration from the speech channel and asks the control station for hand-over;
- the control station orders an *S/N* check of the mobile to the original and neighbouring base stations;
- the control station selects a new zone and a new speech channel, after comparing the signals from the ordered base stations, and then assigns a new channel to the mobile station. In order to shorten disconnection time during this procedure (less than 500 ms), the land-line is reserved. If the mobile station cannot change to a new speech channel successfully, it returns to the previous speech channel.

4. *Signalling code*

Signalling code includes the following characteristics:

- telephone number: 7 decimal digits coded into 24 bits binary;
- serial number (not changeable);
- station class;
- first paging channel to use;
- home area identification: 2 decimal digits.

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PART II - MCS-L2

1. Introduction

The second generation mobile telephone system (MCS-L2) that operates in the same 900 MHz band as the MCS-L1 has been developed in Japan. It is intensively aimed to expand the system handling capacity, to reduce the system cost and to diversify the various services.

The handling capacity is expanded to about 100,000 subscribers in the large cities by techniques of 12.5 kHz channeling analogue FM transmission, diversity reception, intra/inter-zone channel reassignment, adaptive transmitting power control and semi-off-air call set-up.

Full detailed specifications for the system and their components are given in the reference list. The following are the additional features and specifications, compared with the MCS-L1.

2. Radio link design objectives

See PART I, § 2.

3. Radio-frequency interface

3.1 Traffic channel

Maximum frequency deviation is limited to 2.5 kHz for 12.5 kHz channel spacing.

3.2 Radio channel control

Radio channel control utilizes both 2 400 bit/s signalling and 100 bit/s channel-associated control signalling. The 100 bit/s data channel that transmits various signals between a base station (BS) and a mobile station (MS), such as C/I and C/N information from MS, hand-over information from the BS, etc., is transmitted at sub-audio frequency, maintaining overall radio spectrum compactness for 12.5 kHz channel spacing.

4. Control channel configuration

4.1 Control channel allotment

The control channels are dedicated and divided into sub-sets of the two channels, paging channel and access channel.

— **Paging Channel:** A new transmission method, Multi-transmitter Simul/Sequential-Casting (MSSC), is used. Signals common to all radio zones in a control zone, such as mobile terminated call information, are simultaneously broadcast from multiple base stations. Signals peculiar to each radio zone, such as each radio zone's access channel number, are broadcast sequentially.

— **Access Channel:** The Idle signal-Casting Multiple Access with Data-slot Reservation (ICMA-DR) scheme is used in multiple-access radio channels. A data-slot reservation packet is sent first from a terminal through the upward radio channel. Then, the MS sends the information packet after it has received an information request signal from the BS. This method shortens the period of wasted transmission caused by packet collision, and can improve random access channel efficiency.

MSSC transmission secures advantages of wide area paging by simulcasting, more accurate location by sequential broadcasting to allocate exclusively access channels to individual radio zones.

4.3 Control zone configuration

The basic unit is the radio zone. This corresponds an access zone. Several such radio zones together form a paging zone which corresponds to a registration zone. Several paging areas are combined in one switching zone.

5. Switching and Control System

5.1 Signalling

Common channel signalling system No 7 is adopted between the mobile control centres (MCCs) and between the MCC and the PSTN. A dedicated common channel signalling system has been used between the MCC and the BS.

5.2 Roaming

As in the MCS-L1, the MCS-L2 system maintains connectability from/to the PSTN wherever the subscribers go. The MCS-L2 system can also provide the roaming capability to the MCS-L1 system.

5.3 Dialling method

See PART I, § 3.2.

5.4 Speech channel supervision

The quality of a on-going call is continuously supervised by the BS and MS using measurements of RF signal level, BER of 100 bit/s in-service control data signal and desired and undesired signal level ratio.

5.5 Hand-Over procedure

A new speech channel is reassigned quickly and noiselessly to mobile stations in the following three conditions.

- the BS or MS detects C/I deterioration and the BS asks to the control stations for intra-zone hand-over;
- the base station detects C/N deterioration or excess C/N and the BS asks to the control stations for inter-zone hand-over;

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ANNEX III

GENERAL DESCRIPTION OF THE NORDIC MOBILE TELEPHONE SYSTEM (NMT)

1. Introduction

The development and specifications of the international Nordic mobile telephone system (NMT) has been carried out by cooperation between Denmark, Finland, Norway and Sweden. The international aspects of the system were taken into account from the very beginning, making the system a true international public land mobile network from the start, e.g. roaming between countries and full access to the international public switched telephone network and its services.

The rapid growth of the number of subscribers in the NMT-450 system in the Nordic countries has led to the need for expanding the system into the 900 MHz band (NMT-900). Therefore, a new specification has been brought forward, taking the experiences from the NMT-450 into account.

Full detailed specifications for the systems and their components are given in the reference list.

2. Technical characteristics**2.1 RF equipment requirements**

In accordance with Recommendation 478, the adjacent-channel selectivity requirement and adjacent-channel power requirement are met at 25 kHz channel spacing.

2.2 Cell structure and channel re-use

It is possible to use large cells (20-40 km radius) in rural areas as well as small cells (down to 0.5 km radius) in urban areas. The base station receiver sensitivity can be adjusted for symmetry in coverage range for the, two propagation directions taking base station (BS) and mobile station (MS) e.r.p.'s and (MS) receiver sensitivity into account.

In peak traffic density areas (e.g. city centres) it is possible to use a "sector" cell structure with directional antennas and a channel concentration at the centre of the traffic peak area.

2.3 Control signals

The use of 1200 bit/s FFSK signalling enables generation and detection of the FFSK signal at the mobile service switching centre (MSC) and the use of ordinary telephone network circuits to convey the FFSK baseband signal. At the base stations, the FFSK baseband signal is processed by the ordinary audio signal circuits (including pre- and de-emphasis). The same signalling is used to control BS channel equipment.

The capacity of the control signal is large enough to handle the traffic of at least 25 000 mobile stations per location area (LA) with a traffic load of 0.015 erlang per mobile station.

3. Operational characteristics**3.1 Channel search procedure**

All channels may be used as calling channels (CC) or traffic channels (TC). The CC of a base station is used mainly to page the mobile stations on calls from the PSTN while the TCs are used to handle the calls and for the call set-up procedure for calls from a mobile station.

The channel search procedure of a mobile station is carried out in order to find either a new CC (current CC may be changed either because current base station has chosen a new CC or when entering a new cell) or to find an idle TC on call set-up.

The channel locking criteria is set in accordance with the frequency planning-criteria. The scan of the band is performed at three receiver sensitivity levels.

3.2 Location registration (roaming)

The mobile station initiates the location registration procedure when necessary. Full automatic roaming is implemented between mobile service switching centres (MSC) and between countries in accordance with Recommendation 624.

3.3 *Call set-up procedure to MS*

The MS is locked to a CC of a base station in the LA where it was last updated in the system. A paging call is transmitted on the CC of all base stations in the area location. On reception of the call, the MS transmits an acknowledgement on the return frequency of the CC of the current base station. On reception of this signal, the MSC sends a channel switching order to an idle TC on that base station and the call is established on that channel.

3.4 *Call set-up procedure from MS*

MS searches for a free-marked TC (NMT-450 and NMT-900) or an access channel (NMT-900). The channel is seized by transmitting a call set-up request on that channel.

3.5 *Speech channel supervision*

During a call, one out of four supervisory tones with a frequency of about 4000 Hz and a deviation of ± 300 Hz is inserted by the BS together with the voice signal. This tone is looped by the MS and the quality of the tone after looping is evaluated by the BS.

3.6 *Hand-over procedure*

The quality of an on-going call is continuously supervised by the BS using measurements of RF signal level (NMT-900) and supervisory tone quality (NMT-450 and NMT-900). Should the quality fall below a pre-set level, the mobile service switching centre (MSC) is informed. The MSC will order neighbouring base stations (up to 16) to measure the signal quality on the used TC of the old BS, using a special measuring receiver, tunable to all channels. The measuring results from all neighbouring base stations is evaluated by the MSC. If one of the neighbouring base stations provides better reception than the current one, the MSC will allocate a free TC on the new BS for the call, send a channel switch order to the MS on the TC in use for the call on the old BS and re-route the call to the allocated TC on the new BS. Hand-over can also be carried out between base stations in adjacent location areas.

3.7 *Call release procedure*

Calls are released immediately upon reception of clearing signal from the mobile station. If the PSTN party hangs up but not the mobile station, the normal time-out function of the PSTN clears the call. Forced call clearing is carried out if the signal quality at the base station falls below a pre-set level for more than 20 s and hand-over is impossible. In the mobile station, there is an autonomous time-out, switching the transmitter off when the RF signal level is below a certain level for more than 30 s.

3.8 *Numbering plan*

In accordance with CCITT Recommendation E.213, the mobile station is identified in the public land mobile network (PLMN) by a unique seven-digit mobile subscriber number. In NMT-900 the mobile subscriber number is extended with a secret three-digit password.

3.9 *Signalling between MSCs and PSTN interface*

CCITT Signalling System No. 7, permitting hand-over between MSCs by use of mobile use part (MUP), or CCITT Signalling System R2.

The PSTN interface is in accordance with CCITT Recommendation Q.70. The differences between various national PSTNs are covered by the MSCs. Connection to the PSTN is made on trunk exchange level. MSC and trunk exchange can be integrated.

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 4. Technical specification for the base station equipment, NMT Document 3.
- NMT-900: 1. System description, NMT Document 900-1;
 2. Technical specification for the mobile telephone exchange, NMT Document 900-2;
 3. Technical specification for Signalling System No. 7 mobile user part, NMT Document 900-2, Annex 3;
 4. Technical specification for the mobile station, NMT Document 900-3;
 5. Technical specification for the base station equipment, NMT Document 900-4;
 6. Technical specification for the system simulator, NMT Document 900-5.

Available at any of the Telecommunications Administrations of Denmark, Finland, Norway and Sweden.

ANNEX IV

GENERAL DESCRIPTION OF THE C450 SYSTEM

1. Summary

The C450 system design combines optimum transmission quality and high capacity with very efficient utilization of the frequency spectrum. This is achieved by adaptive power control, distance measurements for cell assignment and hand-over and special switching capabilities like off-air call set-up and queueing operation. All relevant CCIR and CCITT Recommendations are met.

2. Radio-frequency interface

The radio-frequency interface is consistent with Recommendation 478. The system is designed for 20 kHz channel spacing and adjacent channels can be operated at one base station. Channel addresses for 25 kHz spacing are also provided in the equipment, as well as for 10 kHz and 12.5 kHz interleaved channels.

3. Permanent control of connection

Traffic channels have a permanent digital sub-channel control for continuous identification, power and signal quality control, etc. The power of base and mobile transmitters is adaptively controlled with 8 steps over a range of 35 dB.

Field strength and signal-to-noise ratio (phase jitter) are monitored for transmission quality assessment, both from the mobile and base station. As soon as the quality tolerances are violated, an intra-cell hand-over to another channel will be initiated.

During connection, the distance of a mobile station is continuously determined from its base station, and this information is transmitted on the digital sub-channel. By means of a special scanning receiver, each base station observes field strength and distance information of mobile stations in neighbouring base station areas as these approach the boundary of its own cell. In this way, the hand-over is prepared, and will be effected when the boundary is passed. The use of a channel is thus restricted to the defined base station area, which improves channel re-use and spectrum utilization.

A permanent exchange of identifications between mobile and base station prohibits another mobile station from accidentally entering into the connection.

The digital sub-channel is provided by time compression of speech signals, leaving 1.14 ms for the data burst within a period of 12.5 ms. Time compression has no perceptible influence on speech quality.

4. TDMA common control channel

All base stations of the C450 system operate on a common time-shared control channel, so that the mobile station can compare signals of the surrounding cells. The right base station is selected, either by comparison of relative distance (delay) — which is the preferred mode of selection — or by field-strength measurement. Each base station uses 1 time slot out of 32. In areas with high traffic, additional frequency channels can be added for common control (roaming), up to a total number of 8. The switch-over from the first common control channel, which is used in the whole service area, to another control channel is initiated by command from the relevant base station.

The TDMA frame with 2.4 s duration consists of 32 time slots, each containing 396 bits.

If a base station is in the status "blocked waiting queue", the mobile stations will check surrounding cells to determine whether support is possible.

5. Identification of mobile and base station*Mobile station:*

— nationality	3 bits
— home mobile switching centre	5 bits
— remaining digits of subscriber number	16 bits

Base station:

— nationality	3 bits
— number of the mobile switching centre	5 bits
— remaining digits	16 bits

Up to 16 decimal digits are transmitted in one message.

6. Location registers

Location registration is consistent with Recommendation 624. There are three types of location registers.

The home mobile switching centre (MSC) has a file of all subscribers assigned to its area, with the following features: class of mobile station, priority status, roaming location, etc.

The visited mobile switching centre keeps a file of the active mobile stations including those originating from other MSCs.

The base station contains a register of active mobile stations within its cell boundaries. On average once in every 4 min, the mobile stations are polled and requested to answer to the base station. This updating procedure discovers the switched-off mobile stations and avoids "dead" entries in the file.

7. Security

Personal subscriber identification cards are used to allow operation of different mobile stations by the same subscriber. The call charge is assigned to the holder of the card. A special security code prohibits the use of lost cards by unauthorized users.

Privacy is provided by means of inversion of the audio frequency band. For data transmission, this band inversion can be switched off at the mobile station.

The mobile station has a built-in monitoring function which will shut down the transmitter when a severe malfunction occurs.

8. Switching capabilities

CCITT Signalling System No. 7 is used between base station and mobile switching centre. The numbering plan is in accordance with CCITT Recommendation E.213. The PSTN interface complies with CCITT Recommendation Q.70.

The time slot capacity of the common control channel is designed for a minimum percentage of access collisions. All other message dialogues are performed under control of the base station.

Optimum utilization of the speech channels is obtained by queueing and off-air call set-up during traffic peaks.

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ANNEX V

GENERAL DESCRIPTION OF THE TOTAL ACCESS COMMUNICATION SYSTEM (TACS)

1. Operational and system characteristics

1.1 Purpose

The public land mobile network (PLMN) is designed to permit an automatic exchange of traffic with the public switched telephone network (PSTN) and offers service, from the user's point-of-view, similar to that of land-line calls: high voice quality, high reliability, low blocking, relatively low cost.

1.2 Cellular properties

- Growth from large cells at start-up to small cells at maturity, with a mix of several sizes, is permitted.
- Hand-over is allowed, up to at least one per call per minute.
- For spectrum efficiency (channel re-use), small values for the number of channel sets are used, appropriate to the multipath (Rayleigh) fading typical of mobile channels, the terrain variability, the antenna patterns selected, and the RF quality desired. A local median C/I target of 17 dB at the 90th percentile is suggested. (If the standard deviation is 8 dB, this is equivalent to a median C/I of 27 dB, at the nominal boundary of the cell.) Other quality objectives are of course permissible.
- Enhanced mobile station registration techniques are employed to allow automatic roaming both nationally and internationally.

1.3 *Signal processing*

- The widest deviation and receiver pre-detection bandwidth appropriate to the cellular concept (which permits separation of the spectrum into channels by both frequency division and spatial separation) are used.
- Voice processing using a 2:1 compressor/expander is employed to enhance the perceived quality of the received signal.
- The coding of digital signalling uses burst-error detection and correction.
- Design for a peak-to-r.m.s. ratio of between 6 dB and 20 dB for the demodulated voice signal at the receiver is possible; that is, a range of quality comparable to land-line service should be accommodated.

1.4 *Service protection*

- A transmitted serial number prohibits the use of stolen units or use by unauthorized callers.
- The introduction of means to ensure privacy of communication is not precluded by the design.
- Means for preventing intelligible cross-talk is included in the system design.
- The successful completion of the entire call takes place on at least 99% of the calls on which an initial two-way connection was established between calling and called parties.

1.5 *Services offered*

- Voice message telephone service with automatic calling to and from the PSTN and other mobile users.
- Data services.
- Vehicular-mounted equipment.
- Transportable equipment.
- Hand-held equipment.
- Other enhancements consistent with mobile service: call forwarding, repertory dialling, message waiting, etc.
- Charging information transmitted during a call to allow direct display of call charges: coin-box mobiles, taxi phones etc. are thus facilitated.

2. **RF parameters**

The major RF parameters of the system are detailed in Table I of Part C of this Report.

Other specific RF parameters are as follows:

2.1 *Modulation characteristics*

The peak FM deviation is around twice that normally associated with 25 kHz channel spacing. This has the major advantage of an improved resistance to co-channel interference, the minimum usable mean carrier-to-interference (C/N) ratio being around 10 dB rather than 15 dB. This means that for the same voice quality, a shorter frequency re-use distance, and hence a smaller frequency re-use pattern, can be employed. There is the effect, however, that adjacent channels cannot be used in the same cell, but this is not a problem for cellular radio systems because only a small proportion of the available channels (typically 1/7th) is used in any one cell.

2.2 *Total number of channels — up to 1000*

2.3 *Control channels*

In order to allow for more than one system operator in a given location, two sets of dedicated control channels are identified. There are 21 control channels in each set and each block of control channels is contiguous.

The dedicated control channels for system A are channels 23 to 43.

The dedicated control channels for system B are channels 323 to 343.



2.4 Mobile station power

There are 4 classes of mobile stations corresponding to the following powers:

TABLE III

Class of station	Type of mobile station	Nominal e.r.p. (W)
1	Very high power mobile	10
2	High power mobile	4
3	Mid-range power hand portable	1.6
4	Low power hand portable	0.6

Adaptive power control is employed with 7 steps of 4/8 dB.

2.5 Supervisory tones

Supervisory audio tones (SAT) are sent by the system over an assigned voice channel and transponded by the mobile station on the duplex voice channel forming a closed identification loop.

SAT frequencies — 5970 Hz; 6000 Hz; 6030 Hz.

Signalling tone (ST) is an 8 kHz tone transmitted on the assigned voice channel from mobile to base station. It is sent to indicate the status of the mobile handset:

- handset on hook — tone sent;
- handset off hook — tone not sent.

3. Signalling code and format

Signalling occurs in both directions on control channels and voice channels to ensure that the mobile station is always under system control. All data is generated at an 8 kbit/s rate.

Each of the four signalling paths carry different types of information and their operational conditions differ, this would require many compromises if a common signalling format were adopted. Each path is therefore treated independently.

Synchronization (sync.) sequences are chosen to minimize the risk of a random appearance within the data, and to provide sufficient time for the mobile/base station equipment to achieve sync. With the exception of the forward control channel (FOCC), used by the base station, transmission occurs as a burst of data when required.

All channels are subject to fading and interference. To give adequate error protection, each data word is transmitted a number of times and incorporates forward error correction bits to ensure integrity. At the receiving end, the repeats are stored sequentially and a majority decision made bit-by-bit on five stored words (the eleven repeats sent on the forward voice channel ensure that at least five will be received by the mobile station).

A BCH code generator is used to produce 12 bits of parity which are added to the end of the data to produce the overall data word. The forward channels (FOCC; FVC) use 28 information bits and 12 parity bits per word, the reverse channels (RECC; RVC) have 36 information bits and 12 parity bits per word. This code structure is capable of correcting 1 bit errors and detecting 4 bit errors.

Transmission of control data over the voice channels (FVC; RVC) is accomplished by muting the audio paths while transmitting the data burst. This is a very short duration activity not noticeable to the user.

Before FSK transmission, the data is Manchester encoded. This ensures that sufficient data transitions occur to permit accurate bit synchronization within the signal. This is particularly necessary for synchronizing sequences with large strings of zeros and ones.

The data words are complex packets of information sub-divided into groups of bits or single bits each defining a system parameter: serial number, dialled digit, etc. The exact format within a word is dependent on the type of channel in use and the type of message.

4. International mobile station identity

A 34 bit binary mobile identification number is derived from the 10 digit international mobile station identity which comprises a 3-digit mobile counting code, a 1-digit mobile network code and a 6-digit mobile station identification number (see CCITT Recommendation E.212).

A unique 32 bit binary serial number is assigned to each mobile station. The number is non-changeable and is intended as an anti-fraud, anti-theft means.

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ANNEX VI

GENERAL DESCRIPTION OF THE ITALIAN 450 MHz SYSTEM

1. System structure

The Italian territory is sub-divided into 10 *calling areas*. A calling area is characterized by a single calling channel; calls to mobile stations present in that area are transmitted in a simulcast mode by all the base stations included in that area. Adjacent calling areas use different calling channels. The same calling channel is re-used in distant calling areas.

Calls directed to mobile stations are routed through the main mobile control centre (MMCC) of the calling area where the mobile station is present.

When a mobile station crosses the border between two calling areas, it automatically informs the MMCC of the new calling area of its presence. This MMCC will update the location registration of all the other MMCCs, so that each MMCC knows the calling area in which each subscriber is present.

A calling area may either be coincident with a *conversation area* or sub-divided in some conversation areas. A conversation area is characterized by the fact that mobile station originated calls enter the telephone network directly in the satellite mobile control centre (SMCC) of the conversation area, apart from the conversation area where the MMCC is located. The hand-over procedure is only possible within a conversation area. Each conversation area will include several base stations in order to cover the most populated areas and the main roads.

The Italian territory will be sub-divided into 16 conversation areas; the average size of a conversation area will be about 20 000 km².

The MMCCs and the SMCCs will be connected to the tertiary centres of the telephone network.

2. Radio-frequency characteristics

2.1 Frequency band

The system is planned to operate in the 450 MHz band. There are 200 duplex channels, the highest four channels are utilized as unidirectional calling channels.

2.2 Base stations and channel grouping

Each base station will be equipped with a redundant calling transmitter, two monitor receivers utilized for the hand-over procedure and a number of traffic transceivers dependent on the traffic expected in the coverage area of the base station.

The maximum number of duplex channels for each station will be 64; several groups of 64 channels can be used at the same site. Traffic channels utilizing the same antenna system are usually grouped in a self-masking intermodulation arrangement. Normally eight channels with equal spacing of 600 kHz are grouped together; such a spacing allows the utilization of small transmit branching filters.

2.3 *Transceiver characteristics*

Most transceiver performances are in line with those of equipment utilized in other advanced public mobile systems. However, it seems suitable, from the system point of view, to mention the following characteristics:

- the nominal output power of the base station transmitters is 25 W and that of the mobile transmitter 10 W. A 10 dB reduction of the output power is possible. The power reduction of the mobile station is automatic;
- a syllabic compandor is utilized in both directions of transmission in order to improve speech quality;
- band inverters are provided which can be inserted and deactivated by the mobile subscriber during the conversation.

3. **Numbering and call procedures**

3.1 *Numbering*

The national telephone number of a mobile station consists of a service access code "0333" followed by a six digit subscriber number. The first of the six digits is the same for all the subscribers, and is utilized for control purposes, the second and third digits are related to the MMCC to which the subscriber belongs.

Usually the telephone number is the same as the mobile station identification number; however, there is the possibility for about 10% of subscribers to have an identification number different from the mobile station telephone number; this possibility allows a subscriber whose mobile equipment has been stolen, to keep the same telephone number.

3.2 *Call to a mobile subscriber*

A call to a mobile subscriber is made by dialling his telephone number, including the service code. The service code connects the calling party to the nearest MMCC. This MMCC analyzes the mobile station number with regard to validity, subscriber category and location registration. If the mobile station is in another calling area the call is routed to the corresponding MMCC.

The MMCC of the calling area where the mobile station is present, pages it through all the calling transmitters included in that area.

When the mobile station receives its identification, it starts to search for an idle traffic channel with the same procedure as for mobile station originated calls (see § 3.3). When a free traffic channel is found, the mobile station identification and the acknowledgement signal are transmitted. If the MMCC does not receive the acknowledgement within 10 s, it sends a second paging call.

3.3 *Mobile station originated calls*

The pre-seizure dialling, or off-air call set-up, is employed in order to reduce the channel occupancy due to dialling information. The mobile station searches for a free traffic channel with a double threshold search; in the first attempt a search is only made for a good quality free traffic channel, in the second attempt a search is made for a channel with only acceptable quality. The double threshold search improves the system quality and reduces the number of hand-over events.

Once a traffic channel is found, the mobile station identification and the dialled number are transmitted. The SMCC or the MMCC analyzes the subscriber validity and category and then the call is set up.

4. **Hand-over**

If during a conversation the quality falls below a given value, the involved base station informs the corresponding SMCC or MMCC which will initiate the hand-over procedure by ordering a field-strength measurement on the relevant traffic channel to the monitoring receivers of adjacent base stations. If one of the base stations offers better transmission quality and at least one traffic channel is free in that station, a command is sent to the mobile station via the previous traffic channel to switch to the new channel.

A confirmation from the mobile station is sent on the new channel and then the switch-over is completed by the control centre.

5. **Signalling on the radio path**

The main signalling utilized in the radio path is a two-of-seven in band multi-frequency code suitable for operation in a mobile service environment. It has been derived from the code already used in the first generation Italian system for which a long field experience has been gained.

A low frequency pilot with amplitude modulation is used for information that is transmitted during conversation, e.g. band inverter insertion and deactivation command, and call charge pulses to the mobile station.

Up to three different audio frequency pilots can be utilized to identify the same (frequency) channel employed in the three different adjacent clusters.

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ANNEX VII

GENERAL DESCRIPTION OF THE FRENCH NATIONAL SYSTEM

1. Frequency band and class of emission

The system uses blocks of channels (up to 256 per block) with a channel spacing of 12.5 kHz, class of emission 11K0G3E. The frequency bands used in France are the 200 MHz (duplex spacing 8 MHz) and 400 MHz (duplex spacing 10 MHz) bands.

2. Signalling

FFSK modulation at 1200 bit/s; Hagelbarger coding.

3. Characteristics of the mobile stations

The station may be multi-service or only offer some of the services described in § 2 of Part C of this Report.

The mobile station has access to 256 channels.

4. Characteristics of the base stations

The base station comprises a switching unit connected to the PSTN as a time-division PABX with automatic direct routing capabilities ("direct dialling-in").

The base station may have up to 24 transceivers cavity coupled to two or three antennas. The radio channels are managed by queueing. Off-air call set-up is used.

5. Architecture of the system

The PSTN switching exchanges are connected by PCM links to the relays which are comprised of the relay management unit, the switching and radio units. Remote radio units may be connected to the relays, for instance to cover low density areas.

Information exchange and re-routing are carried out using the PSTN.

6. Capacity of the system

The numbering plan provides for 500 000 subscribers.

7. Frequency efficiency

A block of 256 channels in a cell may serve 7000 public telephone type subscribers, 20 000 dispatch type subscribers or any combination of these two types.

8. Charging

Charging is carried out by the relays, and then retransmitted by 1200 bit/s links to an operating centre.

9. Commissioning

Operation started at the end of 1985, and the total subscribers were 100 000 at the beginning of 1989 and 150 000 on 1 October 1989.

It is foreseen that 85% of the surface area of metropolitan France will be covered, and 98% of the French population at the end of 1990.

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REPORT 902-1

LEAKY-FEEDER SYSTEMS IN THE LAND MOBILE SERVICE

(Question 36/8)

(1982-1990)

1. Introduction

Leaky feeders are also referred to sometimes as 'radiating cables' but this term is inaccurate and should be avoided. The alternative expression 'leaky cable' is acceptable in specific regard to the feeders themselves, rather than the system, in the vast majority of cases where the feeder does in fact take the form of a cable rather than an open-wire transmission line.

A leaky feeder is a form of transmission line that enables radio communication to take place with or between mobile sets in its vicinity through its leakage fields, while substantially providing the linear range of the system through its internal propagation properties (Delogne, 1982; Martin, 1982).

Such systems are used in confined spaces, such as tunnels, mines or large buildings, where natural propagation is inadequate. They may be completely self-contained or extensions of conventional surface systems otherwise using free radiation: in the latter case the feeds for the underground coverage may either be taken 'off air' or share a base station with the surface antenna directly. Conversely, a primarily underground system may interface with a surface antenna to provide local cover, for example, of the precincts of a mine.

Leaky feeders may also be used on the surface, to confine radio coverage and thus improve spectrum efficiency.

The term 'leaky feeder' can also be taken to include a non-leaky cable with periodic discrete radiators or mode converters (Delogne and Liegeois, 1971).

2. Classification of leaky feeders

The following types of leaky feeder are commonly used:

- (a) bifilar lines;
- (b) continuously leaky coaxial cables;
- (c) coaxial cables with periodic apertures;
- (d) cables with mode converters.

Types (a) and (b) are intrinsically non-radiating in the sense that a cable of infinite length extending in free space can only carry waves guided by the structure. However, any discontinuity along the cable causes mode conversion and radiation.

In type (c), the periodic apertures are radiating discontinuities and act like elements of an antenna array. Maximum radiation is obtained in oblique directions determined basically by the ratio of the spatial interval to the wavelength.

In type (d) the mode converters or radiating elements are separate discontinuities acting in isolation.

2.1 Basic performance parameters

The performance of a leaky feeder system may be characterized by two parameters:

- longitudinal attenuation,
- coupling loss.

The longitudinal attenuation is governed primarily by the factors which apply to normal transmission lines, such as construction, conductor size and dielectric. Additionally there is a small loss component attributable to the leakage (or mode converters).

The coupling loss is, in general terms, the power loss between the feeder and a mobile antenna in its vicinity. For the commonly used coaxial types of leaky cable it is dependent on the degree of shielding in the feeder construction, the configuration of the shield or conductors and the permittivity of the dielectric. For a given cable construction it should also be noted that the coupling loss is also dependent upon:

- the environment in which the cable is mounted;
- the cable mounting position;
- the characteristics, position and orientation of the mobile antenna;
- the operating frequency.

The longitudinal attenuation of cables designed for a low coupling loss can increase substantially with close mounting of the cable to a wall, structure or other cables or with surface contamination, except for cables specifically designed to minimize this effect such as the triaxial (tri-coaxial) cables and inherently non-leaky cables using discrete mode converters. In the same process as the increase in attenuation the coupling loss is usually decreased (i.e. the received signal is increased locally).

For a given coupling loss, various types of cable are not necessarily subject to the same increase in attenuation by these effects of proximity and contamination. Bifilar lines are by far the most sensitive, followed by longitudinally slotted cables and lastly by the various types of cable with numerous small holes. Some braided types of cable are so insensitive to proximity effects that they may be run in normal hangers along with power cables, telephone lines and other such conductors without undue degradation of their performance in the VHF range.

2.2 Bifilar lines

Bifilar lines (nominally balanced) in general have a low coupling loss and a lower longitudinal attenuation for a given conductor size in comparison with coaxial cables. For these reasons they have generally the lowest cost of all leaky cables. However, they can be sensitive to mounting position and surface contamination at VHF and more especially at UHF.

If a bifilar cable is given a tight twist (e.g. several turns per wavelength) the balance is normally improved against proximity effects, thereby improving the longitudinal propagation, but in the same process the coupling loss may be increased by up to 15 dB (Martin, 1975).

2.3 Coaxial cables

2.3.1 Continuously leaky coaxial cables

This type (with either two or three coaxial conductors) includes the loosely braided cables, cables with continuous slots and cables with discrete apertures or slots separated by distances much smaller than the wavelength.

Since these coaxial cables have an imperfect outer conductor, part of the transmission-line energy travels outside the cable as a leakage field (Delogne and Safak, 1975; Fernandes, 1979).

Basic cable properties that (together with environmental factors) determine the coupling loss are (a) the internal propagation velocity and (b) the surface transfer impedance of the outer conductor. It is preferable to characterize cables by quoting these quantities, which can be measured in the laboratory, rather than an arbitrarily and artificially defined 'coupling loss'. Propagation velocity is determined by the internal dielectric constant, while surface transfer impedance is determined by the structure of the outer conductor and is increased, for example, by increasing the size or number of apertures.

Increase of surface transfer impedance results in a proportional increase in the coupled signal, and is normally accompanied also by an increase in longitudinal attenuation depending on the precise cable structure. The sensitivity of the longitudinal attenuation to mounting position also becomes more marked, but there is evidence that this is more a function of 'hole size' than of surface transfer impedance in itself, and so a large number of smaller holes is better than fewer large holes in this respect.

Increase of internal cable velocity (e.g. through the internal dielectric constant) increases the total energy in the leakage field, though this relationship is not linear and involves other parameters. The field close to the cable is actually reduced, but the spread of the field increases. The congestion radius, within which most of the power of the coaxial-mode leakage field is contained, is of the order of 60 cm at 100 MHz and varies in inverse ratio to the dielectric constant of the internal cable insulation. It is also directly proportional to wavelength.

The mechanism by which a mobile antenna outside the congestion radius is coupled to the coaxial mode is the diffraction of the coaxial-mode leakage fields by the inhomogeneities within the congestion radius; these are the inhomogeneities and irregularities in the environment, cable construction and suspension brackets and obstacles of all kinds. This diffraction is a random process: furthermore, in tunnels the resulting coupling loss does not significantly depend on the distance of the antenna from the cable; this is because the diffracted fields are carried by numerous waveguide modes.

2.3.2 Coaxial cables with periodic apertures

Coaxial cables having discrete apertures with dimensions and/or periodicity comparable with the wavelength are a source of radiation.

A coaxial cable with zig-zag slot arrays can be treated as an antenna array (Nakahara and Kurouchi, 1968) and the transmitted energy in the cable is little affected by the radiation from the slots. The radiating field propagates as a cylindrical wave.

Properly located zig-zag slots in coaxial cables improve radiation in the transverse direction and it is possible to manufacture cables with different coupling losses suitable for grading (§3.2.1).

2.4 Cables with mode converters, radiating devices or antennas

These are ordinary bifilar or coaxial lines fitted with devices to convert part of the energy carried by the line into guided modes propagating outside the line or into radiated spherical waves; the devices are installed with either regular or irregular spacing at points determined by the propagation conditions prevailing in the surrounding environment (Deryck, 1972; Delogne, 1973 and 1976).

A radiating device for coaxial cables can take the form of a slot cut around the circumference, forming a complete interruption of the outer conductor to which circuit elements are added so that only part of the power transported by the cable is radiated (Delogne and Liegeois, 1971).

The spacing of the mode converters or radiators may vary from 100 m to 1 km depending on the system input power, the position of the cable in the tunnel and the single-wire mode or tunnel-mode attenuation (whichever is predominant in the conditions). The conversion rate may be adjusted by a suitable choice of circuit components. A common rate is approximately 10%, which gives rise to an insertion loss of about 0.5 dB per converter (Delogne, 1982).

Mode converter systems have been constructed using a short section of continuously leaky cable inserted in the conventional 'non-leaky' cable. Such a section acts as a continuous network of circumferential slots.

3. Systems aspects

3.1 Basic system

A basic leaky feeder system for two-way communication comprises a base-station transmitter and receiver connected to a leaky cable which provides communication to conventional mobile stations, as shown in Fig. 1. By placing the base station at the centre of the cable the longitudinal range can be effectively doubled relative to an end-fed system.

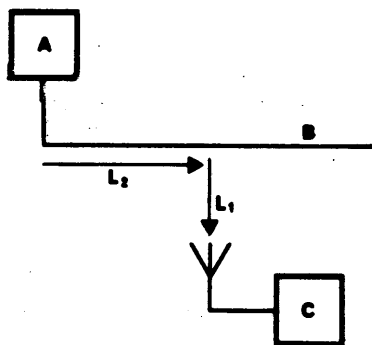


FIGURE 1 - Basic system

A: base station	L_1 : coupling loss
B: leaky cable	L_2 : cable insertion loss
C: mobile	

3.1.1 Communication range

The basic equation relating the parameters (in dB) of a leaky-feeder system is:

$$\text{SYSTEM LOSS} = \text{LINE LOSS} + \text{COUPLING LOSS}$$

where

SYSTEM LOSS is the ratio of the transmitter output power to the power at the receiver input terminal,

LINE LOSS is the longitudinal attenuation of the line between the base station and the point on the line nearest the mobile set;

COUPLING LOSS is the ratio of the power in the line at the point nearest the mobile set to the power at the antenna terminal of the mobile set.

In the case of centre-fed systems, the splitting loss of 3 dB has to be first subtracted from the system loss.

This relationship enables the approximate range of a simple leaky-feeder system to be predicted provided the basic parameters of the cable, radio sets and the particular environment are known. Typically, ranges of between 1 and 2 km are achievable.

Coupling losses as low as 30 dB have been recorded in favourable conditions (e.g. with bifilar lines in a poor state of balance), but more typical figures range from 60 to 90 dB. Short-distance variations of 20 dB, to a Rayleigh-type distribution, are common in tunnels and mines and result from multipath effects. To take these into account it is normally wise to work to a coupling loss figure of at least 90 dB for systems using a cable that is reasonably insensitive to mounting arrangements.

In the case of widely spaced discrete radiators or antennas rather than a continuously leaky cable the natural propagation of the tunnel has to be considered between radiators. Below cut-off frequency of the tunnel a different mechanism applies and the single-wire-mode propagation of the feeder has to be considered instead.

3.2 More complex systems

3.2.1 Graded cable

The basic range of a single base station may be extended by suitably grading the cable along its length, so that coupling loss reduces (at the expense of longitudinal attenuation or proximity effects) with distance from the base station, this technique has been used extensively in Japan. (Okada *et al.*, 1975).

3.2.2 Multiple base stations

The range of the basic system may be extended as required by deploying further base stations, which may be separately controlled or under common control with the first. Lack of synchronism between successive base transmitters gives rise to objectionable 'overlap' effects, but these may be minimized by leaving a gap between feeder extremities in the regions affected. The transmitters may instead be synchronized by passing a common crystal-frequency signal, already modulated, between them over the feeder itself.

If the base stations are to carry separate information, leaving a suitable gap between their feeder extremities will help the normal capture effect of the receivers to minimize the interference between the sections to a few tens of metres.

3.2.3 Use of repeaters

Line amplifiers or repeaters can be inserted at frequent intervals to compensate for line losses in the leaky feeder as shown in Fig. 2. Some system problems and complication in the design of the repeaters for two-way communication can be avoided by separating the base transmitter and receiver as shown in Fig. 3. Using these techniques the power radiated by the mobile or fed into the feeder can usually be less than 100 mW per channel (Martin and Haining, 1979).

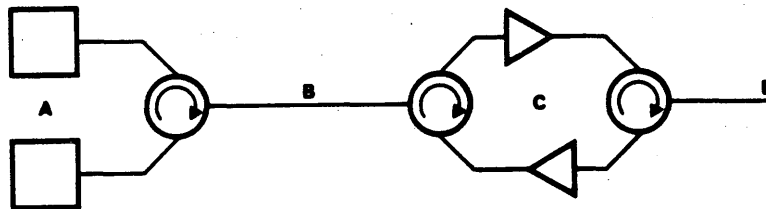


FIGURE 2 - *Extended system using two-way repeaters*

- A: base station transmitter and receiver
- B: leaky cable
- C: two-way repeaters

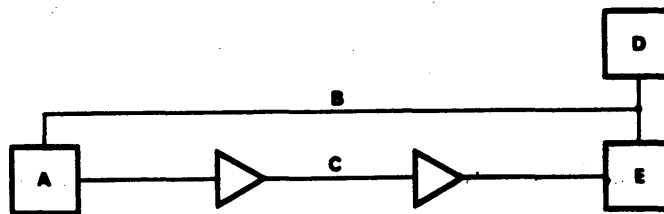


FIGURE 3 - *Daisy-chain system with one-way repeaters*

- A: base station transmitter
- B: audio line connection
- C: leaky cable with one-way repeaters
- D: control point
- E: base station receiver

The simple daisy-chain arrangement of Fig. 3 using one-way repeaters shares a disadvantage of the simple multiple-base-station systems in requiring a separate linking telephone line; it is also inconvenient to apply in a heavily branched form. Both these objections may be overcome by incorporating small line-powered frequency converters at the extremities of the system, in conjunction with a lower-frequency link signal carried over the same cable, allowing the base station itself to be re-united in one place (Martin, 1982).

There are many possible variations in the use of repeaters (Martin, 1982). The systems in general are characterized by a remarkable consistency of signal level and quality through the system. In the largest systems, noise generated by the repeaters may have to be taken into account in considering the transmission from mobile to base.

Repeatered systems are particularly economic in multi-channel operation, since the same repeaters can accommodate all channels providing suitable precautions are taken against intermodulation effects. When the number of channels exceeds about six it usually becomes worthwhile to duplicate the feeder and employ separate chains of repeaters for the two directions of communication. Such an arrangement can provide a useful measure of redundancy, for in the event of failure of one repeater chain the system can revert to simple daisy-chain operation with fewer channels or restricted operation.

3.3 Applications in mines and tunnels

3.3.1 Single-wire conductor link

An extremely rudimentary type of leaky feeder is a single-wire conductor insulated or spaced off from the wall of the tunnel. Its existence in a tunnel substantially modifies propagation because it gives rise to a quasi-TEM mode. This has no cut-off frequency and may therefore be used at low frequencies.

The attenuation of this single-wire mode depends in a complex way on the form and dimensions of the tunnel, the electrical parameters of the wall, and especially on the distance of the wire from the wall and the frequency. Acceptable attenuations are obtained below about 30 MHz. Such systems are confined to wholly underground use (Delogne, 1982).

3.3.2 Bifilar line

This type of leaky feeder has been in use for a long time in underground systems but has to be used with extreme care, especially with regard to mounting position and the possibility of surface contamination by grime or moisture. Generally, its characteristics in regard to longitudinal attenuation and coupling loss are most unstable. Such feeders, however, can be particularly useful in temporary or emergency applications.

3.3.3 Coaxial cables

All types of continuously leaky cables have been successfully used underground, but where severe moisture or difficult mounting conditions are encountered loosely braided cables have given the best results.

Coaxial cables with periodic apertures have been successfully used in railways and motorway tunnels (Suzuki *et al.*, 1980).

3.3.4 Cables with mode converters or radiators

Many installations have been made underground. No special care is needed in cable laying except for a few metres on either side of the radiating devices or mode converters. For the rest, any convenient way of running the cable may be used.

3.3.5 Transverse variation of the field in a tunnel

Since several modes are present in the tunnel when leaky cables are used above tunnel cut-off frequencies, a multipath situation arises. The median value of the field is approximately constant across the tunnel and the statistical variation of the field strength follows a Rayleigh law. With the twisted bifilar line or mode-converter system this variation of the field strength is reduced.

3.4 Applications in railways and motorways

Train radio systems are in use at 400 MHz using coaxial cables with periodic apertures installed alongside the track (Okada *et al.*, 1975).

Lateral measurements of the radiated field strength in an open area along a railway using this type of feeder has shown that coupling loss increases by approximately 3 dB per distance doubling. For other types of leaky cable in similar situations the coupling loss has been measured to increase by about 6 dB per distance doubling (Cree and Giles, 1975).

3.5 Applications in buildings

Leaky cables have been used to provide radio propagation within buildings. In some cases the base station working into the leaky feeder has also been connected to a normal radiating antenna to provide external coverage also. In some other cases the local base station itself has been replaced by an elevated antenna that interfaces the leaky-feeder system to a remote base station by taking off-air signals.

The principles described for tunnels are applicable to any confined spaces such as corridors, stairwells, lift-shafts and car-parking areas. But the large variety of shapes and dimensions of these spaces calls for some caution in system design and the need for empirical solutions is more common than in the case of simple tunnels.

4. Preferred frequency bands

Leaky-feeder systems have been used successfully over a wide range of frequencies, at least down to 3 MHz (Delogne, 1982) and as high as 800 MHz (Suzuki *et al.*, 1980).

Generally, economic factors favour the lower part of the VHF range, but other factors that may influence the choice are :

- the availability of channels for a particular scheme (especially if there is some surface cover involved);
- the size and configuration of a tunnel or building;
- if the system is intended as an extension of a conventional mobile radio system;
- the required linear (end-to-end) range;
- any required lateral range from the feeder down a side tunnel not itself equipped with feeder;
- the availability of suitable radio sets;
- the size of antenna on the personal sets.

Leaky cables tend to be more sensitive to mounting position and to surface contamination at the higher frequencies. Bifilar cables are affected the most in this way.

A factor that favours the higher frequencies is the possibility of employing tunnel propagation in suitable places to augment or extend the cover by feeder. Such extended range, however, will usually be dependent on the tunnel remaining unobstructed by vehicles.

In some situations the use of leaky feeders may be completely avoided by relying on such natural propagation at frequencies above the tunnel cut-off frequency (Emslie *et al*, 1975; Delogne 1982). Again, however, blockage of the tunnel by vehicles between the base-station and mobile antennas can interfere with communication, to an extent depending on the tunnel and vehicle sizes.

5. Sharing with other services

Leaky-feeder systems used completely underground may operate in frequency bands allocated to other services without mutual interference, because the environment provides a high degree of shielding.

For surface application, systems may be designed to have a reduced overall path loss, such as with closely-spaced in-line repeaters (Martin and Haining, 1979). This enables the transmitter powers to be minimized and receivers to be desensitized, thus reducing the co-channel interference potential and susceptibility and facilitating sharing with other services.

In assessing an interference risk involving leaky feeders it should be noted that the interference sources (and susceptibilities) tend to be concentrated at the base station and repeaters, and at discontinuities such as the system extremities and sharp changes of direction.

In using a two-frequency channel within possible interference range of a conventional system it may be beneficial to transpose the frequencies, that is to use the base transmission frequency for mobile transmission and vice versa. This is because in a leaky-feeder system it is generally the mobile that has the greater interference potential and susceptibility rather than the base station as in conventional systems.

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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].

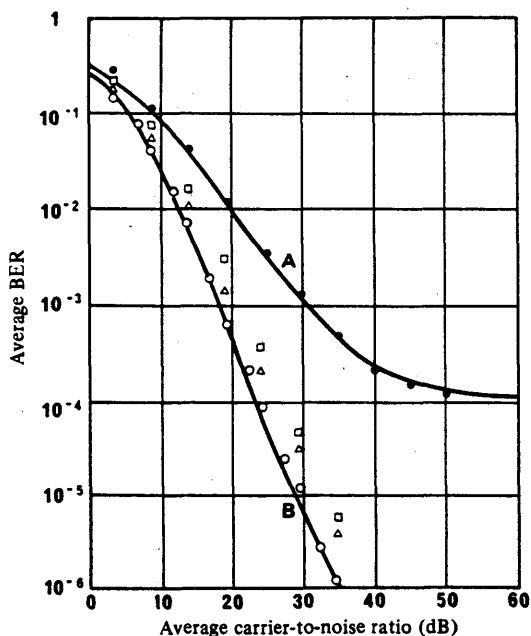


FIGURE 1 – Thermal noise performance of GMSK transmission with correlated two-branch selection diversity

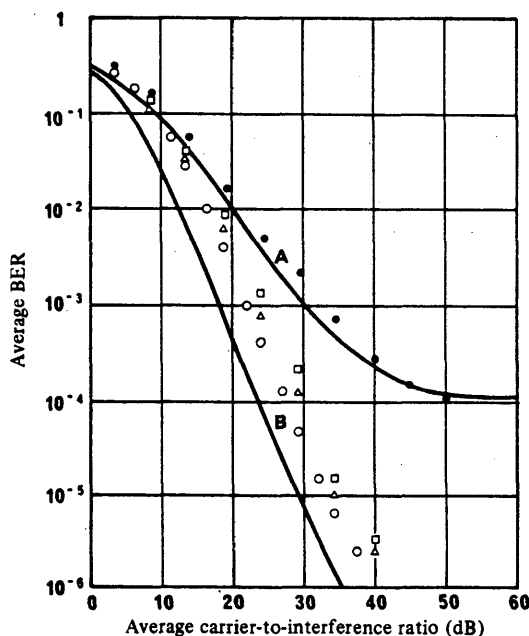


FIGURE 2 – Co-channel interference performance of GMSK transmission with correlated two-branch selection diversity

Curves A: no-diversity

B: 2-branch selection diversity

○: $\rho = 0$

△: $\rho = 0.5$

□: $\rho = 0.8$

16 kbit/s – GMSK ($B_b T = 0.25$)

$\cos 2T$ 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 μ V)) 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. $S \geq 20$ dB) errors are caused solely by co-channel interference and the error probability [French, 1981] is as follows:

$$\begin{aligned} P_e &= 0.5 && \text{when } S < I \\ P_e &= 0 && \text{when } S > I \text{ or} \\ \therefore P_e &= 0.5 && \text{Prob } (S < I) \end{aligned}$$

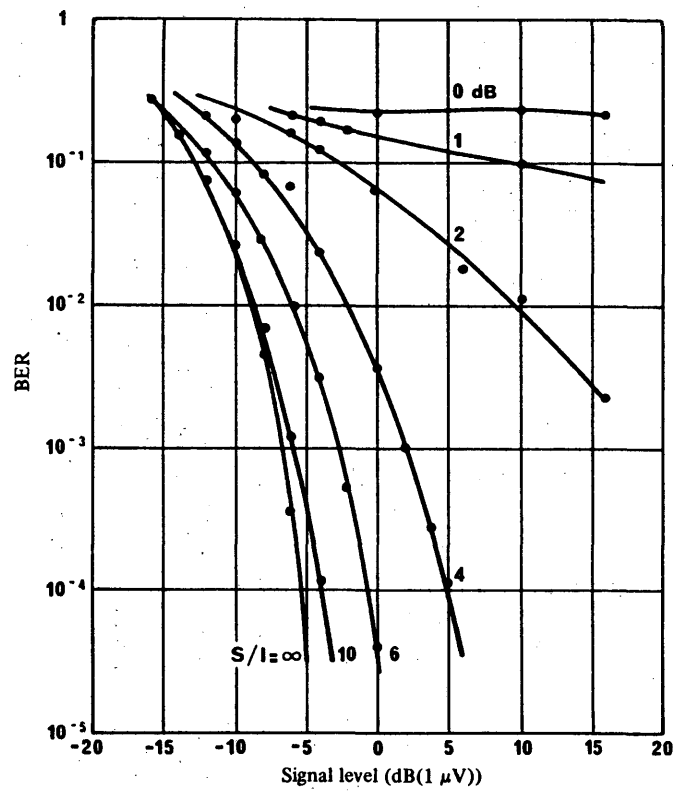


FIGURE 3 - BER due to co-channel interference, stationary, FM 1200 bit/s FFSK

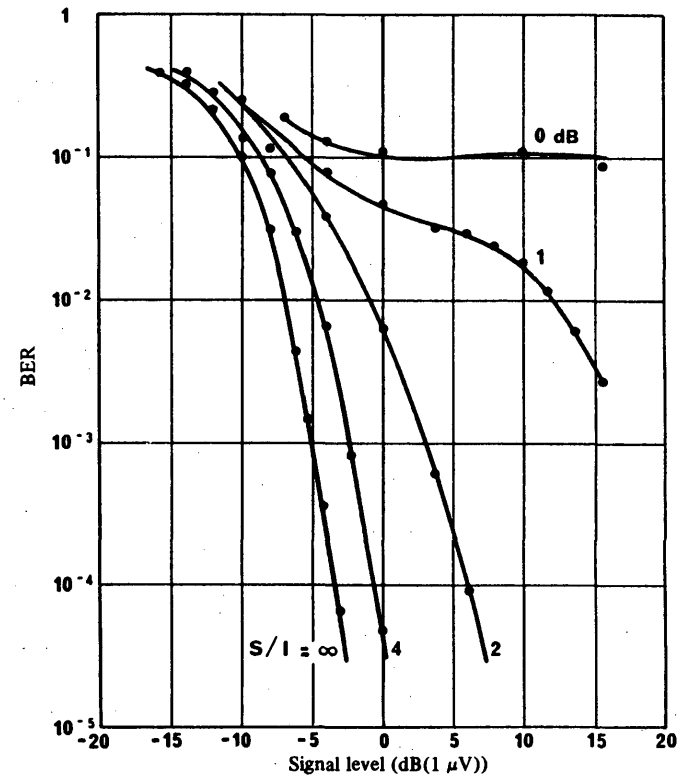


FIGURE 4 - BER due to co-channel interference, stationary, AM 1200 bit/s FFSK

[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.

* Also known as Gaussian filtered fast frequency shift keying.

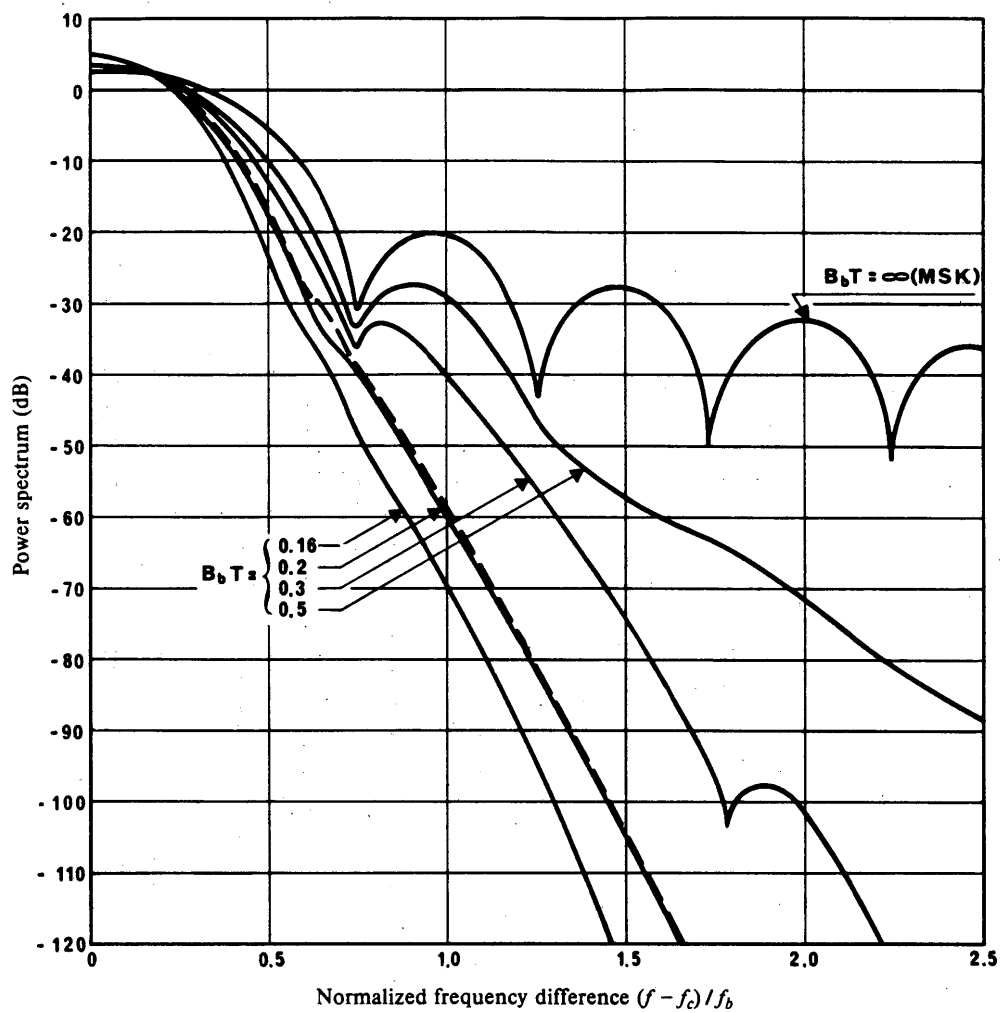


FIGURE 5 - Power spectra of TFM and GMSK

-----: 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)

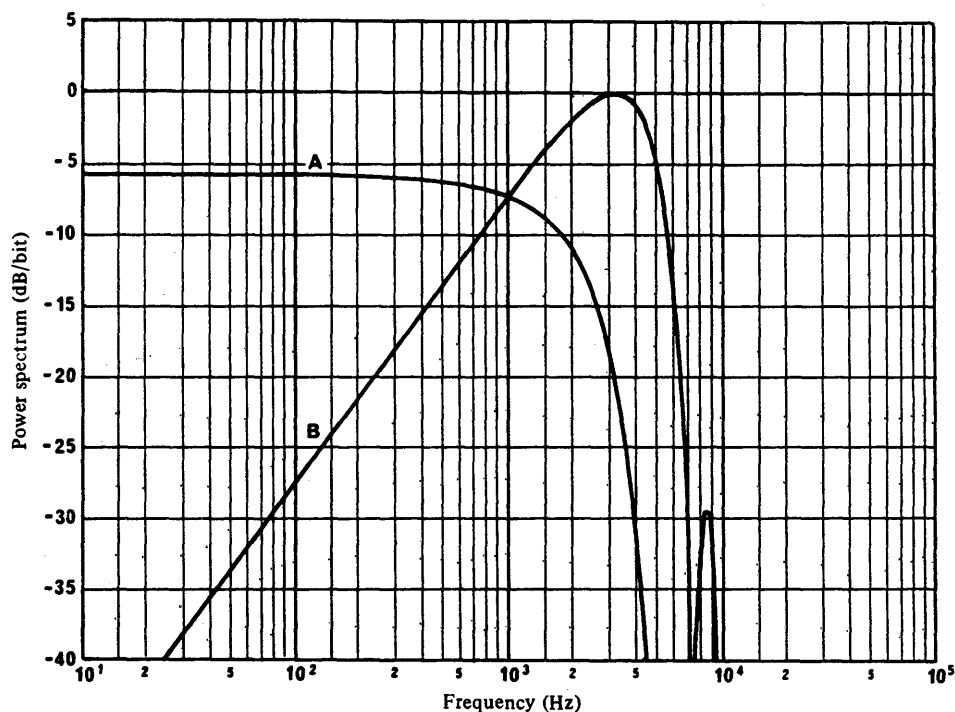


FIGURE 6 – Power spectra of filtered baseband

Curves A: baseband spectrum

B: filtered FFSK (GMSK)

Data and GMSK

Data rate: 4800 bit/s

GMSK: 2400 Hz/4800 Hz

$B_b T$: normalized bandwidth of pre-modulation filter

$B_b T = 1$

Baseband $B_b T = 1/2$

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 orthogonal 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].

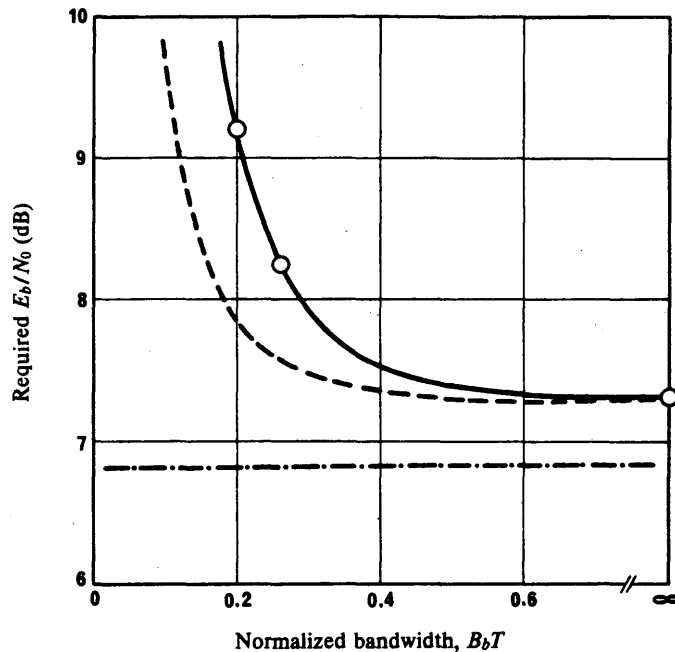


FIGURE 7 - Required E_b/N_0 versus B_bT

- : orthogonal coherent detection with differential decoding
- : maximum-likelihood detection with differential decoding
- · - · - : ideal orthogonal coherent detection
- : 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.

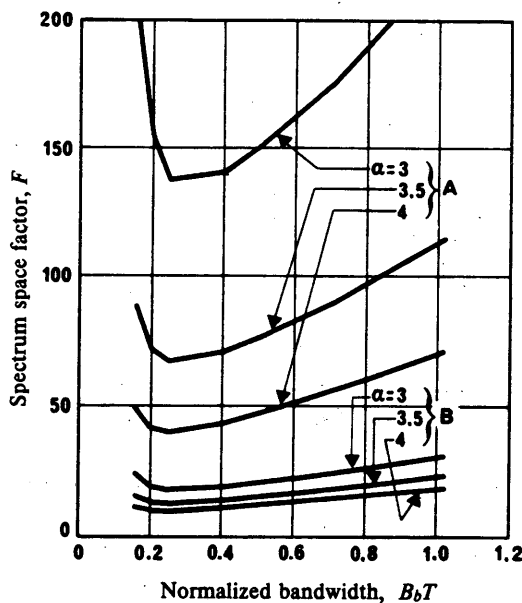


FIGURE 8 - Spectrum space factor of GMSK

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.

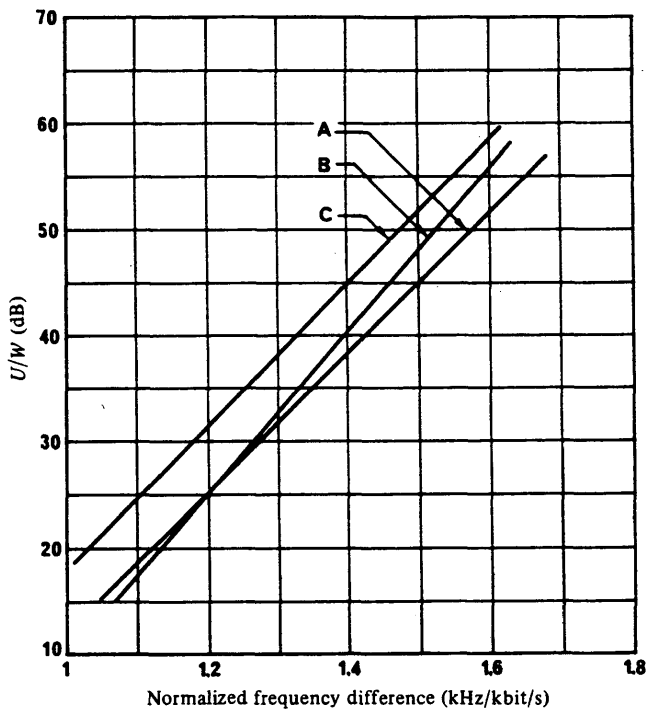


FIGURE9 - Adjacent-channel interference performance

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

Unwanted signal: U-level corresponds to $BER = 1 \times 10^{-2}$ 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.

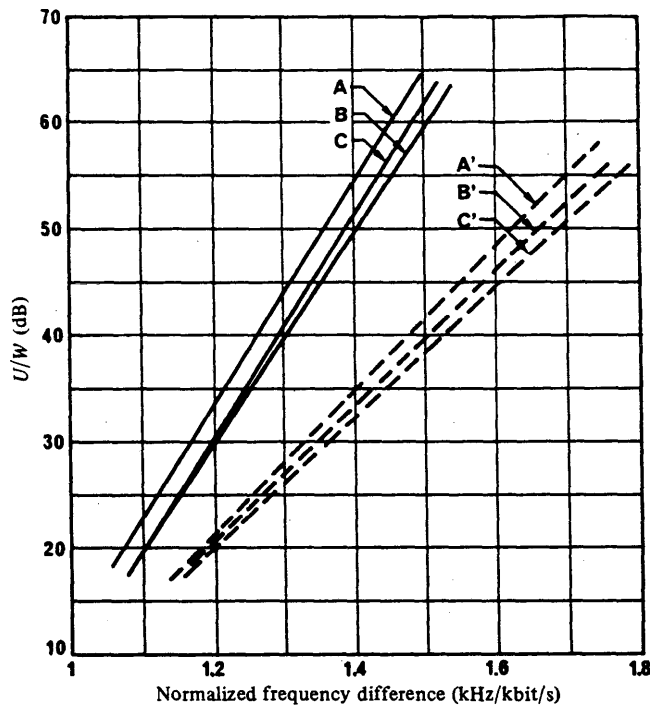


FIGURE 10—Adjacent-channel interference performance

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

Unwanted signal: U-level corresponds to $\text{BER} = 1 \times 10^{-2}$ 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 in 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.

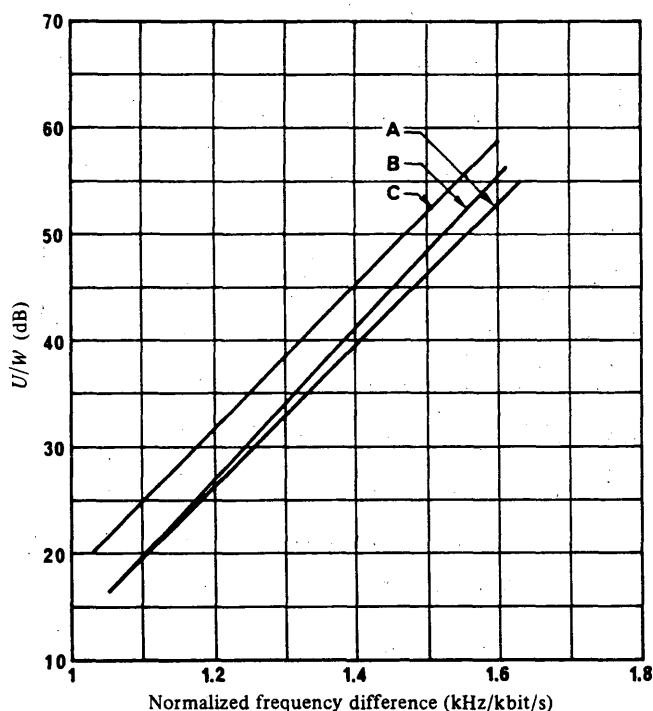


FIGURE 11 - Adjacent-channel interference performance

- 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.

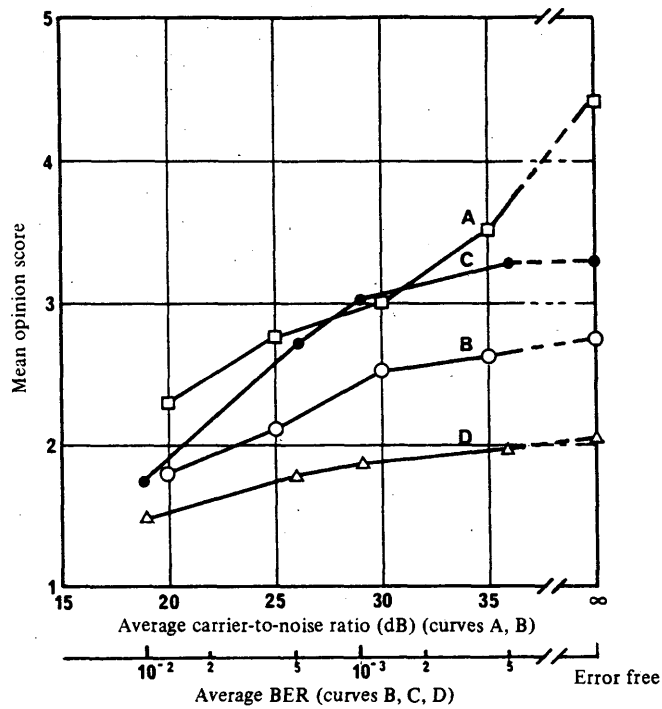


FIGURE 12—Comparison of digital and analogue voice transmission; result for thermal noise

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

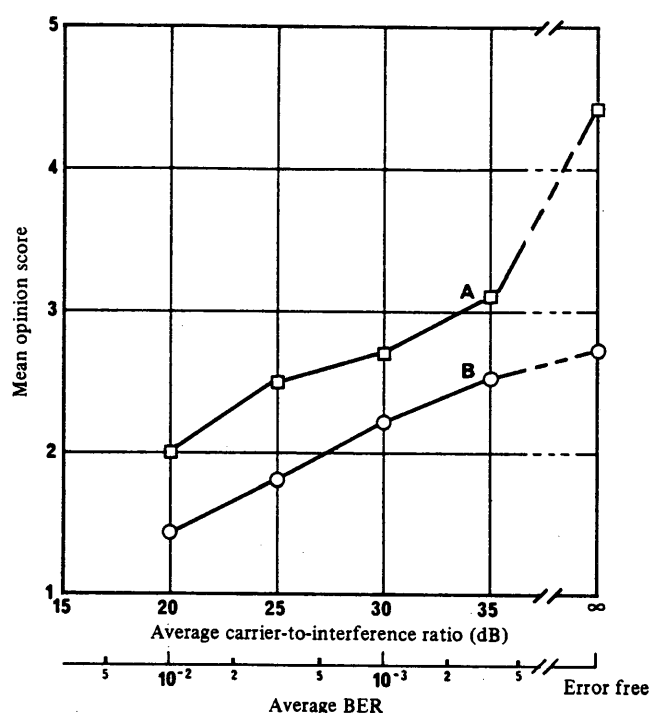


FIGURE 13— Comparison of digital and analogue voice transmission; result for co-channel interference

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 digital data messages are encoded and repeated several times at the source. The coding used on all radio channels is a shortened (63 : 51) Bose-Chaudhuri-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 ≥ 0.9 for input levels ≥ 0 dB(μ V) e.m.f. without fading and ≥ 10 dB(μ V) 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

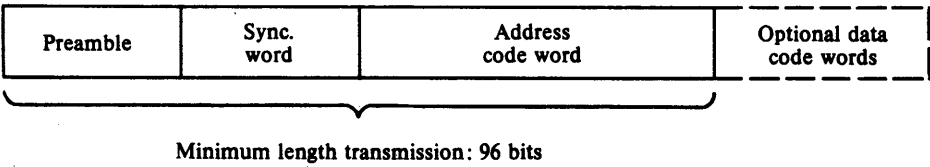


FIGURE 14 - The format

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)

21-32: addressor’s identity (i.e. from)

33-48: data

49-64: check bits (see § 7.2.1.7)

} optional

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.

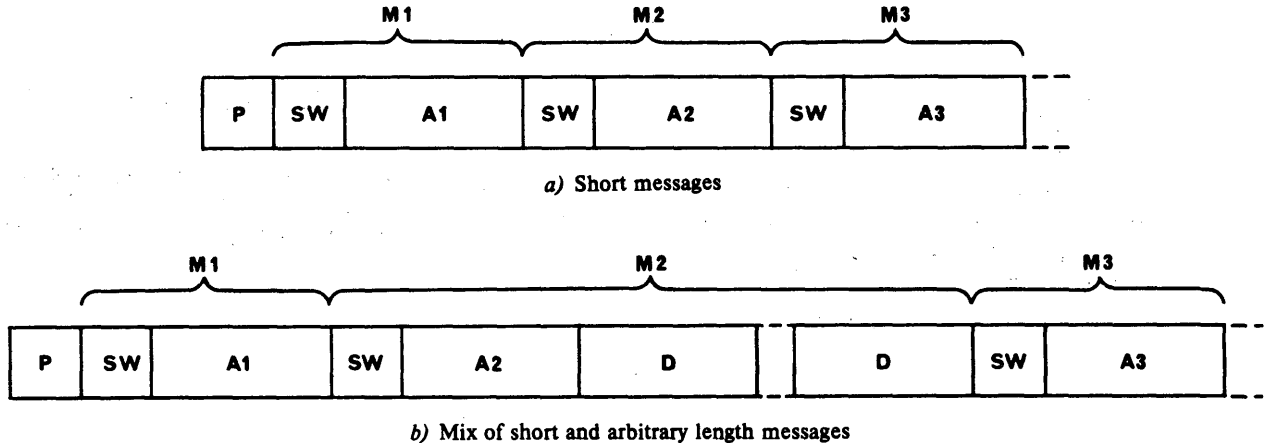


FIGURE 18 - Concatenated messages

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.

$$P_f \approx P(0, s) \cdot P(\geq d, n) \cdot 2^{-r} \quad (1)$$

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

$P(\geq d, n) \cdot 2^{-r}$ 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 ($r = 15$). $P(\geq d, n)$ is the probability that an n bit word contains d errors or more.

For independent errors

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

Because the code guarantees detection of all odd numbers of errors, $P(\geq d, n)$ 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 ($\sigma = 4$ dB) with a vehicle speed of about 50 km/h, without ignition noise present, are given in Fig.20.

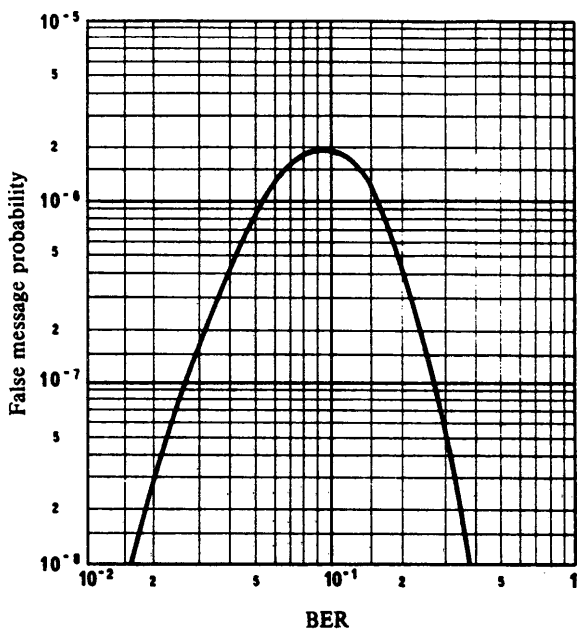


FIGURE 19- False message probability with a steady signal

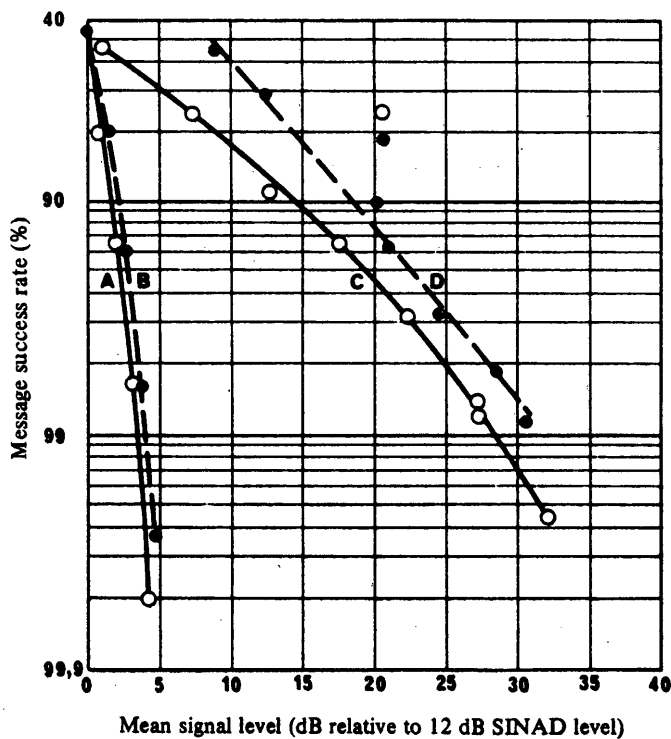


FIGURE 20 - Field measured performance

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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REPORT 1021

**EQUIPMENT CHARACTERISTICS FOR DIGITAL TRANSMISSION
IN THE LAND MOBILE SERVICES**

(Study Programme 7A/8)

(1986)

1. Introduction

This Report describes major characteristics of land mobile digital modulation equipment: BER performance, adjacent signal selectivity, and acceptable bandwidth.

2. BER performance (sensitivity)

The measured BER performance of a bit rate of 8 kbit/s under no-fading conditions using different modulation methods is shown in Fig. 1. The reference sensitivity is defined as E_b/N_0 (signal energy per bit/noise power density) corresponding to BER of 1×10^{-2} . The reference sensitivities were less than 12 dB for these modulation methods. Similar results were obtained for other transmission bit rates, such as 2.4 kbit/s, 4.8 kbit/s and 16 kbit/s.

The $E_b/N_0 = 12$ dB corresponds to the receiver's input level of $(\sqrt{R}/2) \mu\text{V}$ (R : bit rate in kbit/s) when the receiver noise figure is equal to 13 dB.

3. Adjacent signal selectivity

Typical adjacent signal interference performance of bit rates of 8 kbit/s is shown in Fig. 2. The measurements were conducted by setting the level 3 dB above the sensitivity level given in § 2, and adjusting the unwanted signal level until the bit error ratio degraded to 1×10^{-2} . Similar results were obtained for transmission bit rates of 2.4-16 kbit/s.

At the normalized frequency difference (the ratio of frequency difference to transmission bit rate) of 1.5 kHz/kbit/s, the ratio of unwanted to wanted signal level (U/W) becomes greater than 45 dB.

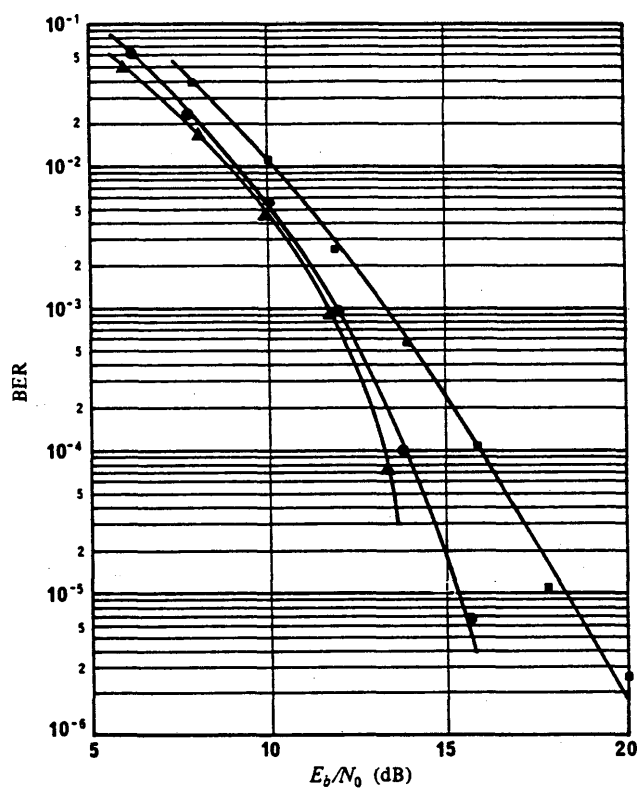


FIGURE 1 – BER performance under no-fading condition (measured)

Modulation:

- : GMSK with coherent detection
- : 4-level FM with discriminator detection
- ▲ : PLL-4-PSK with discriminator detection

Transmission bit rate: 8 kbit/s

 E_b/N_0 : signal energy per bit/noise power density

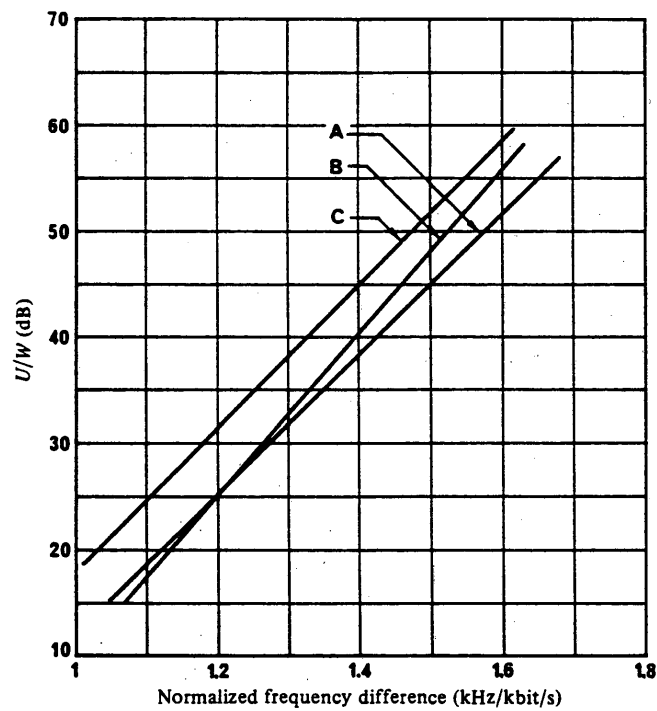


FIGURE 2 – Adjacent signal interference performance (measured)

U/W : ratio of unwanted to wanted signal level
Wanted signal: W -level corresponds to $BER = 1 \times 10^{-2}$
Unwanted signal: U -level corresponds to $BER = 1 \times 10^{-2}$ when wanted signal level is 3 dB in excess of W -level
Modulation: wanted and unwanted signals are modulated by
A: GMSK;
B: 4-level FM;
C: PLL-4-PSK.

4. Acceptable bandwidth

The acceptable bandwidth of a receiver can be defined as the frequency bandwidth within which a bit error ratio of less than 1×10^{-2} is obtained when the signal level is set at the level 6 dB above the sensitivity level given in § 2.

Typical measured values of the acceptable bandwidth for 2.4-16 kbit/s are shown in Fig. 3. All measured values were more than 0.4 kHz/kbit/s. Taking account of performance fluctuations due to production, the acceptable bandwidth should be specified as wider than $0.3R$ kHz (R : bit rate in kbit/s).

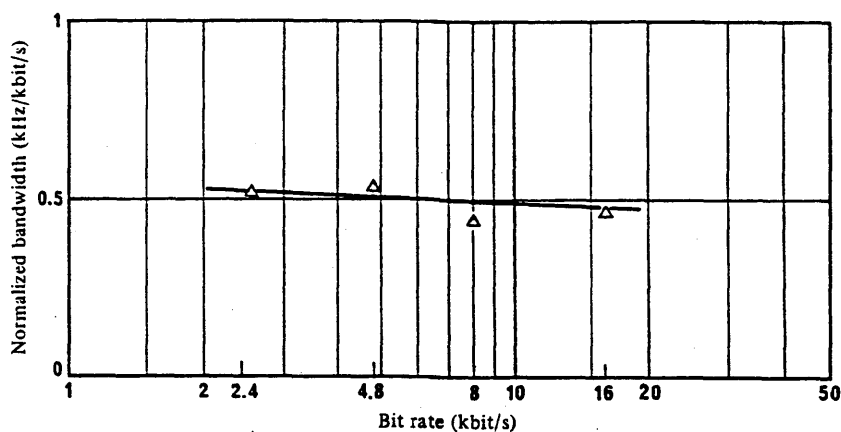


FIGURE 3 – Acceptable bandwidth (typical measured value)

4-level FM with discriminator detection

$BT = 1.0$

(B : bandwidth; $T = 2R$; R : bit rate in kbit/s)

5. Conclusion

As mentioned above, the following specification is desirable:

- the sensitivity should be such that for a 1×10^{-2} bit error ratio, the input signal level should not be greater than $(\sqrt{R}/2) \mu V$ (R : bit rate in kbit/s);
- the adjacent signal selectivity should be more than 45 dB at the normalized frequency difference of 1.5 kHz/kbit/s;
- the acceptable bandwidth should be wider than $0.3R$ kHz (R : bit rate in kbit/s).

In addition to these specifications, the following technical characteristics require study in accordance with Study Programme 7A/8:

- frequency tolerance;
- occupied bandwidth;
- adjacent-channel power;
- spurious response;
- spurious emissions;
- radio-frequency intermodulation;
- low-pass filtering modulation signal.

REPORT 904-2

**AUTOMATIC DETERMINATION OF LOCATION AND GUIDANCE
IN THE LAND MOBILE SERVICE**

(Question 51/8)

(1982-1986)-1990)

1. Summary

In an automatic vehicle location (AVL) system the location of a mobile in a fleet is determined automatically as it moves around, within a given geographic area.

An AVL system comprises the location subsystem, data transmission subsystem and control and data manipulation subsystem. In dispatch operations including police, fire, public transportation, taxi, etc., a large percentage of the voice communications transmitted on land mobile radio channels is composed of routine information, the greater part of which is related to vehicle location status.

AVL techniques that can meet operational requirements of land mobile users can be classified into five major categories: proximity, dead reckoning, hyperbolic, satellite and combinations of two or more of these techniques, each of which has its advantages and disadvantages. The selection of the most effective technique is dependent on the type of user and operations involved [Hansen, 1977].

In the past few years, dedicated land mobile AVL systems have been introduced in many countries. These systems together with the radiodetermination satellites which are due to be fully deployed in 1990's and the already established area navigation systems would certainly provide a wide range of choices to the users. An AVL and position reporting system has recently become operational in the U.K. and test results from this system are reported in Annex I of this Report.

Operational AVL systems in Japan and Australia are described in Annex II and Annex III respectively.

Along with the development of AVL systems, numerous research and development projects have also been carried out in the field of vehicle navigation and guidance. The operation of these systems is based on the same techniques as used in AVL systems, but the significant difference is that in navigation and guidance systems, users are given advice on how to proceed to the chosen destination. Vehicle navigation and guidance systems have received attention from many countries. European collaborative projects such as Drive and Prometheus have been initiated. A description of these systems is given in Annex IV.

2. Introduction

Increasingly over the past few years, police and public transportation operators around the world have realized the potential benefits associated with the knowledge of vehicle locations. Some operators in Europe, Japan and North America have already installed AVL systems.

With the continuing increase in the costs of land mobile dispatch services resulting from an overall increase in operating and maintenance costs and the rising demand by the public for improved services, it is becoming more important to find a means of reducing costs.

One of the most critical elements in dispatch services is the knowledge of current precise location of every vehicle in the operational fleet.

An AVL system with its data reduction facility can provide the information necessary to control and adjust the operation of a dispatch fleet on a real time basis, thus resulting in a more efficient and productive utilization of personnel and equipment.

3. Operational considerations

There is no authoritative body which can define the operational requirement for the many different kinds of AVL use. The following broad requirements have been stated by potential users.

3.1 Positional accuracy

From 100 to 200 metres for many services. Some require accuracies of 10 m; others (e.g. large area trucking dispatch systems) would be satisfied by accuracies of about 1 km.

3.2 Frequency of up-dating

Ideally once per minute or so for vehicles which are required to be quickly deployed in limited areas (e.g. police, fire tenders, ambulances) but less frequently for dispatch operations over larger areas or defined routes.

In practice, attainable updating intervals may be governed by the rate at which choices of route are presented to each vehicle. These rates are proportional to the occurrence of road junctions in the area of travel and to vehicle speeds.

3.3 Coverage area

For many systems (e.g. police, fire, ambulance, passenger omnibus, taxi services) operational areas up to 100 km × 100 km are common. Some operations are confined to much smaller areas up to 10 km². Others may require continental coverage.

4. Cost/benefit considerations

Potential benefits [Wilson, 1977] that can be achieved for the two different types of operation namely: fixed route, as in public transportation, and random route, such as in police and taxi operations, are summarized in the following sections.

4.1 Potential benefits in fixed route operations

- Reduction in checking and control personnel.
- More even distribution of passengers between vehicles.
- Reduction in (lay-over) time resulting in reduction in number of buses and personnel.
- Increased on-time service.
- Improved efficiency of response during emergencies and for dispatching a replacement vehicle.
- Increased passenger loads due to more convenient and up-to-date location information available to the public.

The AVL requirements of many fixed route systems can be satisfied by non-radio means.

4.2 Potential benefits in random route operations

- Reduction in response time to emergency and service calls.
- Reduction in the number of vehicles while maintaining the same coverage area.
- Reduction of unnecessary travel.

4.3 AVL benefits-to-cost ratio

Results based on a computer model for AVL benefit-to-cost [Symes, 1979] have indicated benefit-to-cost ratios of up to 7 to 1 are possible for the public transportation operation and as much as 13 to 1 are possible for police operations.

Although all the benefits stated above can potentially reduce the cost of running dispatch services, to date AVL systems that have been installed in Europe and North America have the objective of improving service reliability in general and reducing response time to emergency and service calls in police services in particular.

In some public transportation systems in Europe as a result of the installation of AVL systems, a reduction in the required number of buses was experienced; however these were retained to accommodate service expansion [Herrman and Zimmerman, 1974].

A benefit-to-cost analysis was conducted in Canada, which shows that there is a definite benefit ranging up to a 2 to 1 ratio [Fujaros, 1976].

5. Spectrum efficiency considerations

With the installation of an AVL system, location information, which constitutes a large percentage of the voice communications [Fujaros, 1976], is automatically transmitted from vehicles to control centres in data form. This on-board capability facilitates transmission of other routine status messages as well, resulting potentially in an increase in spectrum utilization efficiency in the land mobile services. It has been estimated [Cortland, 1986] that 20 to 25% reduction in voice communications between vehicles and control posts can be achieved.

Some systems do not generate demand for any additional spectrum allocation. These systems use the existing transmissions of radionavigation aids, which are designed for ships and aircraft, to determine their positions. Examples include Loran-C and Decca systems. However some of these systems only provide limited coverage particularly on land.

Satellite systems like GPS, described below, together with the two systems mentioned above are basically broadcast systems which can accommodate an unlimited number of users.

AVL systems can be applied to improve the spectrum efficiency of other mobile services as in the following examples.

5.1 Application to cellular systems

It has been shown that AVL can be a very useful measuring tool [Meek, 1988] which can provide accurate coverage data for optimising cellular telephone networks. Furthermore, accurate mobile station location information could provide enhanced system control in terms of originating calls on the appropriate base station and effecting handover control. This may need to be considered against increase in system complexity.

5.2 Microcellular system control

Many narrowband and wideband techniques are being developed to improve spectrum utilisation. One approach is to use a microcellular system. The area covered by a microcell is small and frequent handovers may therefore be required. With the implementation of an AVL system, the channel carrier frequency can be switched as a mobile unit moves across a known microcell boundary [Towaij, 1983]. In order to operate the microcellular system,

the location of every mobile unit must be known to the network controller. A proximity technique is used to determine the location of every mobile as it passes the base stations. Each base station is designated with a unique ID code. Short low power transmissions are emitted from each base station once every two seconds. Each burst contains a flag, base ID and control signals.

6. AVL techniques

AVL techniques that might be capable of meeting operational requirements of land mobile radio users can be classified into five categories: proximity, dead reckoning, hyperbolic, satellite and combinations of two or more of these techniques.

6.1 Proximity techniques

The location of the vehicle is determined as it passes roadside signposts, sometimes referred to as roadside beacons whose locations are precisely known. A wide range of techniques has been employed for the communication of information between roadside signposts and passing vehicles including inductive loops, UHF radio transmission, microwave transmission and infra-red transmission. There are two different types of signposts depending on the direction of information flow:

- active signpost: location signals are transmitted from signpost to vehicles;
- passive signpost: signals are transmitted from vehicle unit to signpost.

There are two methods of assessing proximity characterised by one of the following procedures:

- direct proximity: the active signpost transmits its location to the vehicle, which then transmits its data by radio link to the central computer;
- inverse proximity: the vehicle transmits its data to the passive signpost which then relays the data to the central computer by wire or radio link.

The precision of proximity techniques is directly related to the spacing between signposts.

6.2 *Dead reckoning techniques*

These techniques employ heading and distance travelled (e.g. odometers) sensors for calculating the location of vehicles relative to fixed known location references. The computation may be made on board the vehicle or in the centre computer. Location determination accuracies depend on the sensing devices, frequency of reference updates and the severity of external factors such as magnetic field variations, wheel slippage and road camber, etc.

6.3 Hyperbolic techniques

The locations of vehicles are determined from distance differences of vehicles from three or more fixed sites. These differences can either be expressed by phase differences between received signals (phase multilateration) or differences in the time of arrival of leading edges of synchronized pulse signals (pulse multilateration) producing hyperbolic lines of constant phase or time differences. The location of vehicles can be determined from the intersection of these lines. The positioning techniques can employ dedicated systems for land mobile application or utilize existing navigation systems such as Loran-C or Decca.

6.4 Techniques using satellites

The position of a mobile is established by the precise measurement of the distances between the receiver and the satellites at an instant in time. Four satellites are required to establish a three dimensional fix. Their positions must be known precisely and be available at the instant of measurement if overall system accuracy is to be maintained. It may be noted that the positions of satellites may not be as predicted by recent orbital information due to drag by abnormal solar winds at times of solar eruptions. Corrections may not be applied for some hours after such deviations.

In general, there are two types of systems with different characteristics of mobile user equipment:

- active equipment: two-way communication between the mobile and its ground control station is provided by satellites. The position calculations are performed in the ground station and the resulting location data is transmitted back to the mobile.

- passive equipment: navigation signals are broadcast to the mobile. Vehicle location is determined in the mobile equipment.

6.5 Combination of techniques

Two or more techniques may be combined to improve upon the performance available using an individual technique and to form an integrated system.

7. Applicable AVL systems

7.1 LF hyperbolic system

Within the coverage of existing transmitters, position data can be provided for many mobile users. The test results in Annex I were obtained using this system in London.

7.2 UHF hyperbolic system using spread-spectrum

A terrestrial-based AVL system utilizing paged spread-spectrum beacons on the vehicles to be located has been developed and installed in Australia. Specifically developed for urban environments, the system has demonstrated positional accuracies of the order of 30m on moving or stationary vehicles with 1 second measurement time. A unique frequency division multiplexing scheme endows the system with a high location throughput. (See Annex III).

7.3 Signpost/Dead reckoning systems

A pilot system which is installed in London consists of about 700 UHF roadside signposts. The on-board vehicle equipment is initialised when it first comes to the coverage area of a signpost. Between signposts, the vehicle location is established by dead reckoning based on the last initialisation. Whenever the vehicle approaches another signpost, the on-board equipment is reinitialised by the reference signals continuously transmitted from the signpost so that the accumulated errors are removed. The vehicle location is automatically stored in the on-board vehicle memory even when the engine is switched off. This system can provide useful location information within the network. The provision of the considerable number of signposts required for high locational accuracy can be costly.

7.4 Omega

The international VLF global radionavigation aid OMEGA is available without charge and continuation is assured into the next century.

An extensive programme of field trials on OMEGA, operating differentially, has demonstrated that over a wide range of environment, including mountains and city centre, the practical accuracy is 300m 2σrms (95-98% probability), or better [Stratton 1987].

The accuracy of differential OMEGA can decrease as the distance between the differential monitor and mobile increases beyond 100km; errors can also increase substantially at night. The cause has been identified as diurnal variations in the local velocity and direction of propagation from the standard values used in converting the differential phase measurements into position. By linking data from three to four monitoring stations the variations in azimuth and velocity can be continuously measured and correction applied [Stratton 1988].

An AVL system using differential OMEGA is operational in the Middle East.

7.5 Radio-determination satellite systems

From 1990 onwards, several radio-determination satellite systems will be available, a description of these systems follows.

TRANSIT, [Blanchard, 1983] a currently operational satellite radionavigation system, is of very limited use to land vehicles because of its long measuring time. By 1996, TRANSIT will be replaced by GPS.

The GPS system which is expected to be fully deployed after 1992, will provide three-dimensional position and velocity information. The technical characteristics of the NAVSTAR GPS are described in ANNEX I of Report 766.

GLONASS [IMO, 1989] is a satellite system which will provide position and velocity information to civilian users when fully deployed between 1991 and 1995. The design of this system is very similar to that of the GPS.

A proposed radiodetermination satellite system being developed for operation in the United States of America starting in 1991-1993 is expected to be capable of providing two functions : localisation and communications. Three-dimensional accuracies at the order of 30-40 m can be achieved, with a measuring time of less than 1 s. The system, which will utilize geostationary satellites in the radiodetermination satellite service, is described in Report 1050. A similar radiodetermination satellite system which has been proposed to provide coverage in Region 1 is now under development [Hernandez, 1987].

8. Guidance systems

Guidance systems employ the same basic techniques for location as used by AVL systems with additional information being provided to the moving vehicle (see Annex IV).

9. Conclusions

AVL should be viewed as a technology that could have significant influence on the effectiveness and productivity of dispatch operations as well as on spectrum utilization efficiency.

Satellite systems can provide global coverage which will benefit all the land, maritime and aeronautical users and hence may have significant implications on spectrum saving. However, in urban areas, the performance can be degraded due to signal blocking. To overcome this problem the mobile equipment may have to include a dead reckoning system so that continuous tracking can be maintained. In the near future, several autonomous satellite systems will become available; the potential for mutual interference between these systems and for interference with other services needs further study.

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ANNEX I

TRIALS CONDUCTED WITH DATATRAK AVL SYSTEMS IN THE UNITED KINGDOM

1. Datatrak, a low frequency (LF) hyperbolic system, has been in operation since April 1988. The present system provides coverage of the south east of England and the Midlands, and coverage of the whole United Kingdom is expected to be available in early 1991.

2. The network consists of a large number of UHF base stations (for mobiles reporting to base stations), a vehicle control centre and 16-20 time-multiplexed LF transmitters which are approximately 160 km apart. Each LF transmitter comprises a 50 metre monopole aerial with radial earth mat, the radiated power is approximately 50-100 watts e.r.p.

2.1 The transmitters all operate at the same pair of frequencies and time sharing is achieved with Time Division Multiplexing technique. Navigation frequencies are 135kHz (f1) and 145kHz (f2) both of which are class of emission 1k00 P0N. The navigation timing diagram is illustrated in Fig. 1. The pattern of the time slots 1-8 is repeated every 1.68s, whereas time slots 9-24 are effectively at half rate and are repeated every 3.36s. The faster time slots are assigned for urban areas while the slower time slots are for rural areas.

3. The vehicle equipment comprises a locator unit, a status keypad and an antenna.

3.1 The navigation processor is the central processing unit of the vehicle locator. After the lines of position are established from the phase data derived from the navigation transmissions, the position of a vehicle is converted into Ordnance Survey National Grid coordinates and the status data are collected via the bidirectional data bus connected to the keypad. The position and status data are formatted by another processor before being transmitted over the UHF channel (460MHz) to the base stations at a data rate of 3600 bit/s.

3.2 Each locator unit is assigned one or more 30ms time slots within a 30 minute timing cycle and there are approximately 54000 slots available. The locator units and base stations are synchronised to the same timing cycle so individual vehicles can be identified at the base stations by its slot position. The 30ms vehicle slot data format as shown in Fig. 2 begins with a 1ms guard band. Synchronisation is achieved by the following 10 bits preamble and the 16 bits Gold code. The status data and the vehicle's latest position are also transmitted within this timing cycle.

3.3 The locator unit has a power consumption of 14W and requires 12V or 24V d.c. supply. The unit can be powered by the vehicle battery via the ignition switch, so it is switched on automatically when the engine starts. The locator unit has a built in time delay device, so the vehicle's position continues to be monitored for 20 minutes after the engine is switched off.

3.4 The status unit indicates the coordinates of the current vehicle position. Communication from vehicle to the central control is established by the status keypad which activates the locator unit to transmit up to 250 user defined status codes. It is not possible to send data in the reverse direction.

3.5 The vehicle antenna for LF reception is an active monopole. With a diplexing facility, reception of the LF navigation signal and data transmission at UHF is accomplished with a single antenna.

3.6 With this vehicle location technique the vehicle does not need to transmit in order to obtain its position. This implies that an unlimited number of vehicles may be used to obtain their positions using the system. However, for fleet management and control purposes data transmission to the control centre is needed as described in Sections 4 and 5 below and this imposes an upper limit to the numbers using a given system. A block diagram of the system is shown in Fig. 3.

4. Equipment at the vehicle control centre comprises a computer, VDU and keyboard and a microprocessor controlled modem to couple the computer to the data gathering and distribution centre.

4.1 At the radio base stations, vehicle data are re-formatted and tagged with the slot identity at the base station. Then these data are transmitted onto the control centre via the data gathering and distribution centre. Transmission between the base station and the control centre is conducted by the means of leased telephone lines.

4.2 The data transmissions over the UHF channel are received by more than one base station, this redundancy technique is employed to ensure that vehicles' data will still be received even if one station fails to receive the signal correctly. At the control centre, data information is sorted and logged by the data centre computer and any redundant data is removed.

4.3 Vehicle positions, identities and directions of travel are overlaid on digitized map stored within the high-resolution display. Areas of special interest can be displayed in greater detail with Zoom facilities.

4.4 In addition to the graphic display, each vehicle's data information are displayed in tabular form on a VDU. These data include vehicle identity, current coordinates, direction of travel, speed and its last reporting time.

4.5 Unallocated alarm slots are reserved for emergency signalling. In this event, the vehicle's identification together with its status and current position are transmitted. The alarm facility can be initiated either automatically (e.g. in the case of a vehicle collision) or by the action of the driver. The alarm slot data format is illustrated in Fig. 4.

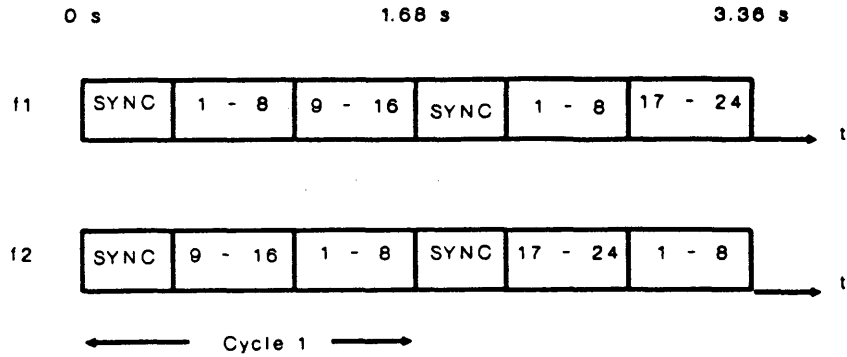
5. Tests

5.1 Both dynamic and static tests have been conducted. The dynamic testing was conducted by driving the vehicle in a normal manner along public roads and observing its progress against a digitized map background at the control centre. For the static tests the vehicle was parked at a known location and its position data were continuously being monitored over a period of time.

5.2 A specimen computer printout of recorded data from the dynamic tests is shown in Fig. 5.

5.3 Static tests were made by recording data from a stationary vehicle. The "observed position" was estimated using Ordnance Survey maps. The "computed position" was derived from the recorded hyperbolic coordinates converted to Ordnance Survey National Grid reference. The "position error" is the distance between "observed position" and "computed position" in Eastings and Northings.

5.4 Static tests have been conducted at Richmond, a residential area between Central London and Heathrow Airport. The test location was approximately 70 km away from the nearest LF transmitter. The results of the tests showed that a typical standard deviation for a 24 hour period was 9 metres. So the 95% circle of position was of radius 18 metres.



1. Numbers shown are the slot numbers of transmitters
2. Slot numbers 1-8 repeat every 1.68s on each frequency
3. Slot numbers 9-24 repeat every 3.36s on each frequency

Fig. 1 Timing diagram - navigation chain

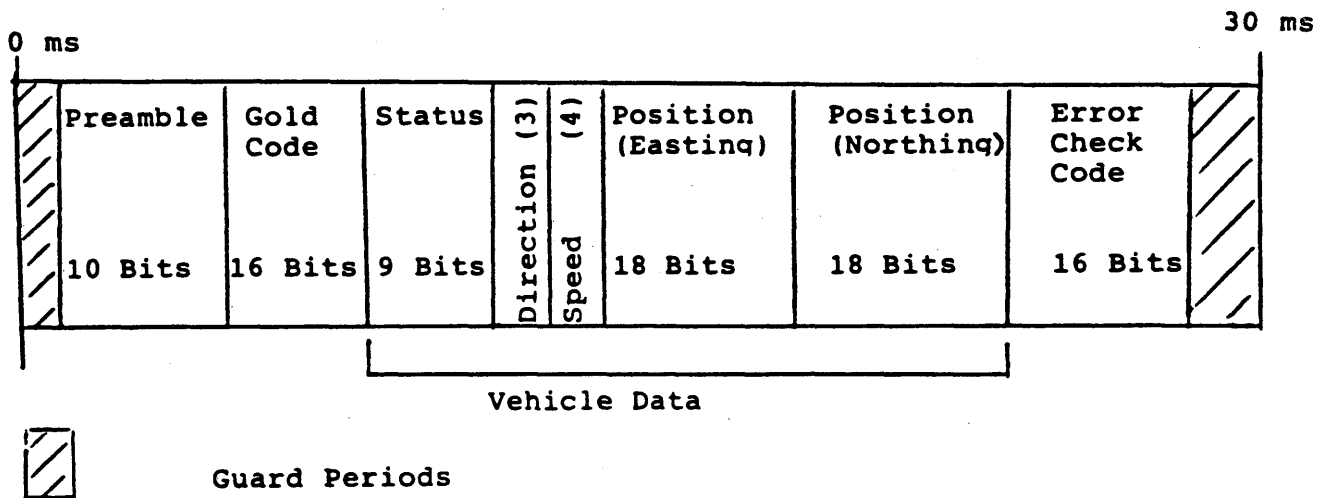


Fig. 2 Vehicle Slot Data Format

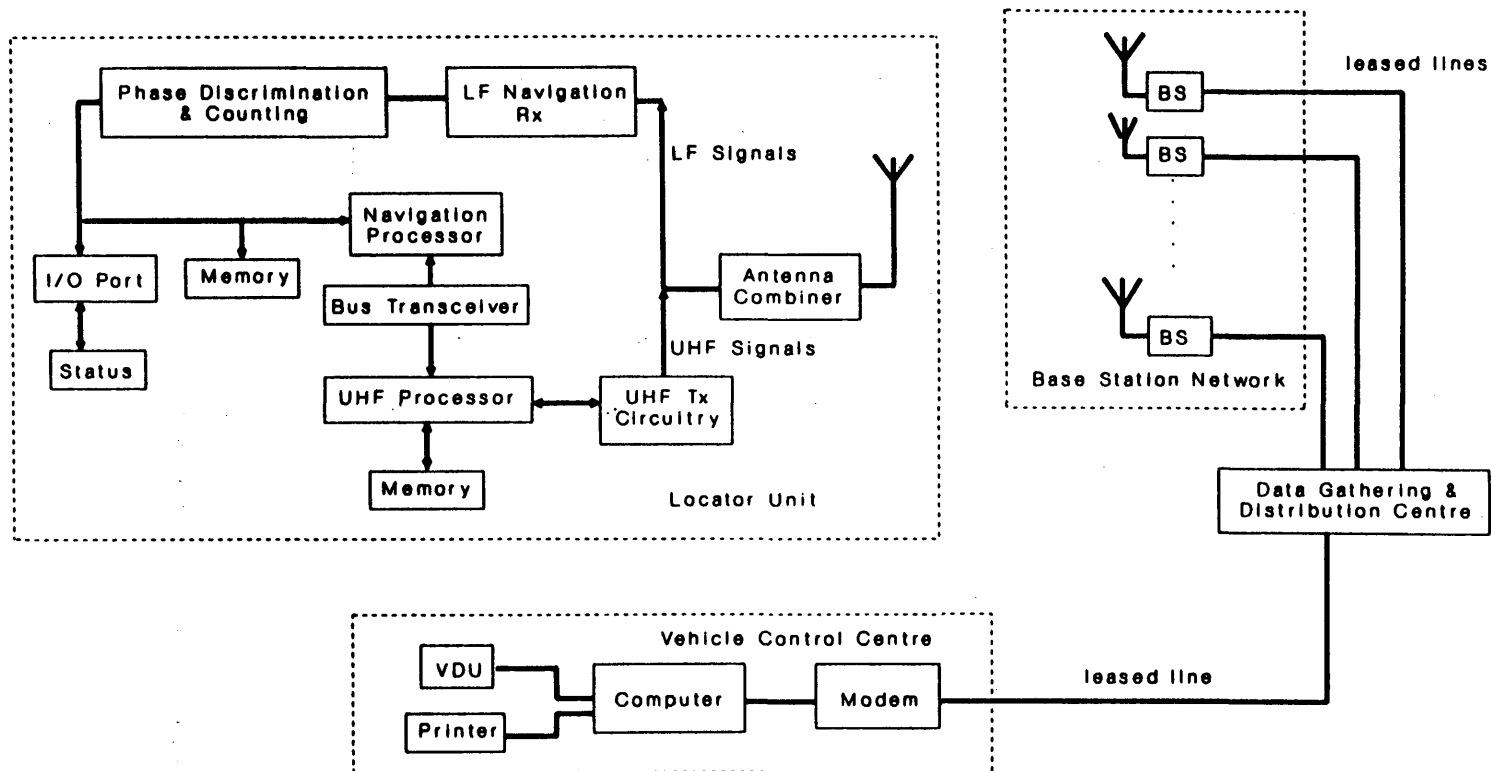


Fig. 3 Vehicle location system

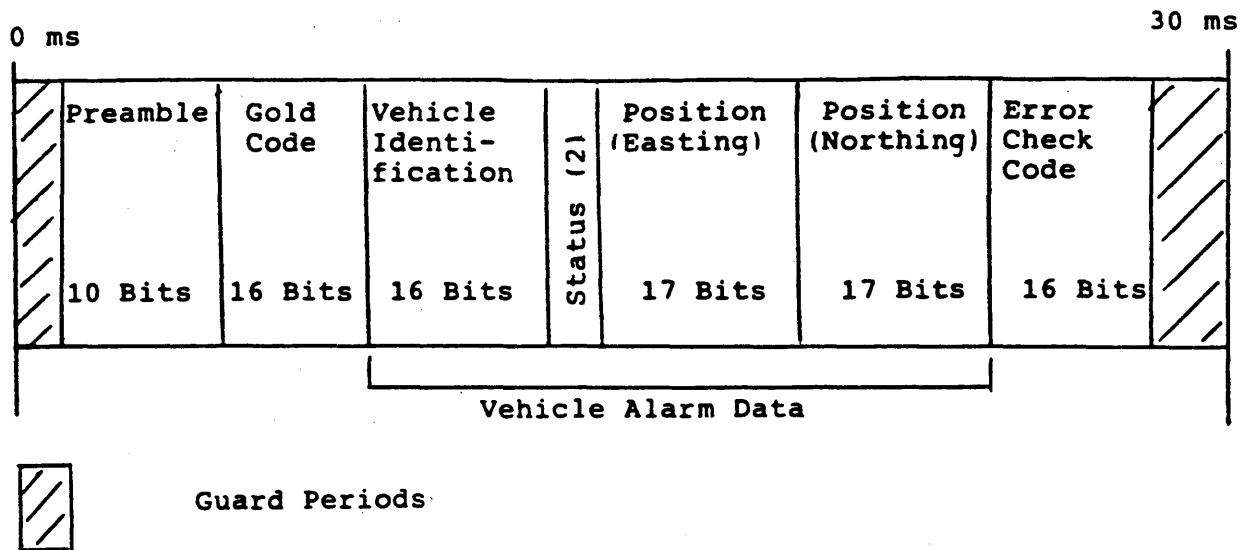


Fig. 4 Alarm Slot Data Format

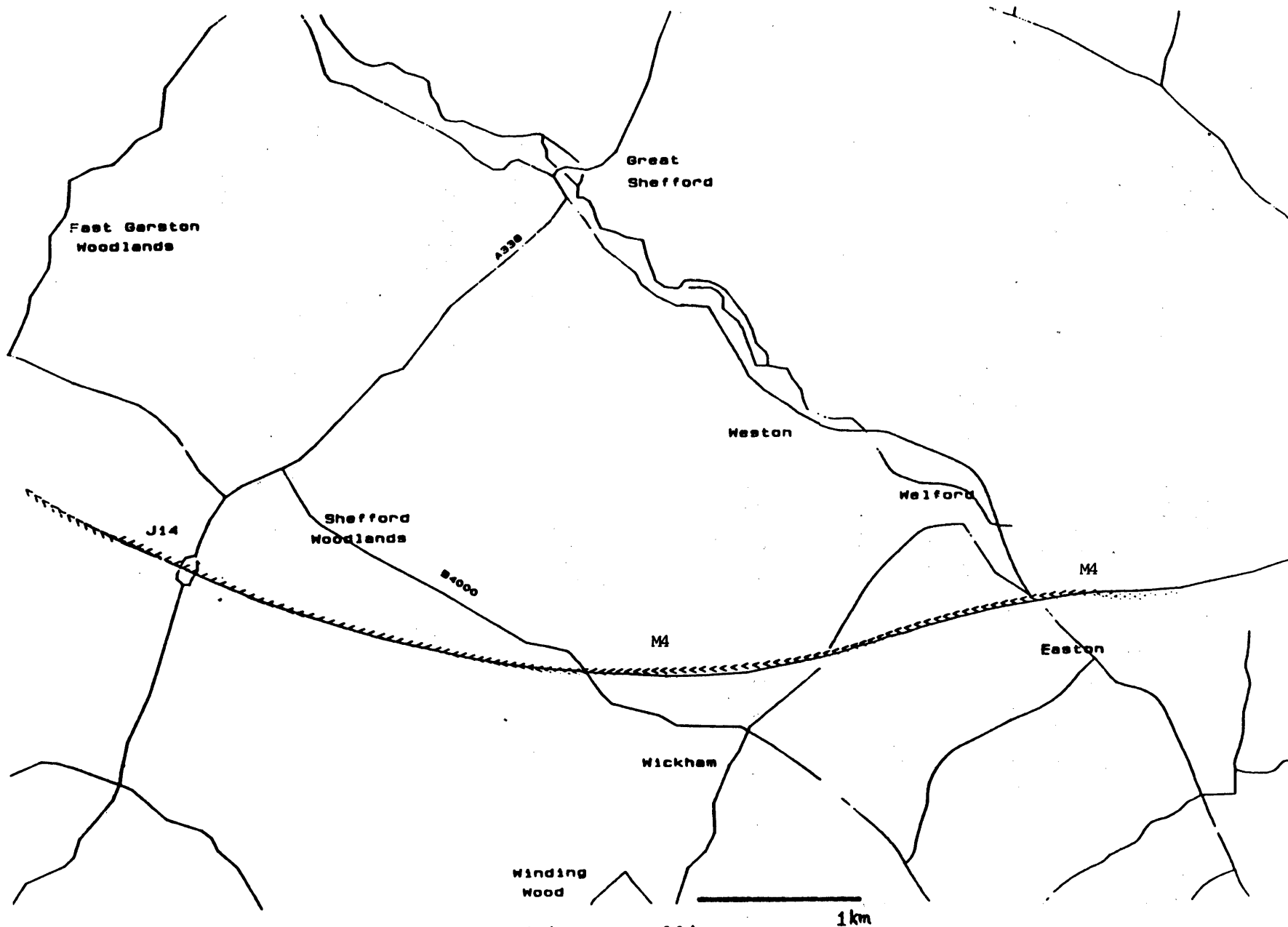


Fig. 5 Results of a vehicle test drive travelling West along motorway M4 in West Berkshire.

ANNEX II

AVL SYSTEMS IN JAPAN

Since 1980 Japan has introduced direct proximity AVL systems in 7 cities and inverse proximity systems in 16 cities.

In the Tokyo metropolitan area, a direct proximity AVL system with 29 signposts is used by about 5400 taxis (October, 1983). For direct proximity systems, the country is divided into 50 km squares, each with an area code and a reference point in the North-West corner. Each signpost is assigned an area code (2 digits) and an x and y position code (2 times 3 digits) relative to the reference point.

The signposts transmit on 426 MHz with a power of 0.01 to 1 W using sub-carrier FM modulation (1500 Hz, 1900 Hz FSK at 200 bit/s). A 15 bit frame synchronization signal is followed by the area and position code in which each digit is BCD coded with added parity bit.

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ANNEX III

SPREAD-SPECTRUM AVL SYSTEM DEVELOPED IN AUSTRALIA

1. In the Sydney metropolitan area of Australia, an automatic vehicle location system called QUIKTRAK has been operational since late 1987 [Hurst, G., 1989]. The system has been designed to provide coverage over an area of some 2000 square kilometres with an accuracy of about 30 metres. The system has the capability to support many independent users and many thousands of vehicles with a tracking capacity of 30 000 locations per hour. Spread-spectrum, phase-multilateration techniques are used to provide accurate locations on moving vehicles in the severe multipath environment encountered in heavily built-up areas.

2. The QUIKTRAK system consists of:

- . Vehicle-mounted transponders,
- . Multi-channel, spread-spectrum receivers at remote sites,
- . Transponder control transmitters co-located with some of the receiving sites,
- . A system timing-reference transmitter,
- . A central processing site,
- . User terminals.

2.1 Transponders. The transponder mounted in a vehicle to be located consists of a low-power, spread-spectrum transmitter and a low-current-drain receiver to pick up system commands and control the spread-spectrum transmission. The control receiver receives and decodes standard paging data to activate the transmitter and to select the spread-spectrum channel. The transmitter produces a direct-sequence spread-spectrum signal with a main-lobe bandwidth (null-to-null) of 2 MHz and a power output of 25mW. Within the QUIKTRAK system, spread-spectrum channels are defined and selected by their carrier frequencies which are offset from each other by small amounts relative to the signal bandwidth. All channels are accommodated within essentially the same 2 MHz system bandwidth located in the UHF Radiolocation band.

2.2 Receivers. Remote receiving sites are located throughout the system coverage area at approximately 15 km spacing. Each site contains a multi-channel spread-spectrum receiver and a remote site computer. The receivers acquire and track the spread-spectrum signals and measure the phase of the pseudo-random spreading code with respect to a system-wide reference signal. In addition to controlling site operation, the remote-site computer relays the signal time-of-arrival information (code phase) from all channels to the central computer via telephone line. Some remote sites also include a transponder control-signalling transmitter.

2.3 Transponder Control Transmitters. Transponder control is accomplished using conventional wide-area paging techniques. Consequently the transponder control transmitters consist of simulcasting 25W paging transmitters. QUIKTRAK transponder control signalling is located in the UHF land-mobile band.

2.4 Timing Reference Transmitter. System synchronisation is achieved by transmitting a separate spread-spectrum signal, from a centrally located site, which all remote sites receive and track. The signal is similar in structure and power output to that radiated by a transponder but is located in a separate 2 MHz band. By locking to this system-wide, low-power signal it is possible for all remote-site receivers to be synchronised to nanosecond precision and thus highly precise comparisons may be made on time-of-arrival of transponder signals at the various sites.

2.5 Central Processing Site. All sites send transponder time-of-arrival information to the central processing site by telephone line. The central computer processes all available information to calculate a best estimate of transponder location. In addition to calculating positions, the central computer schedules transponder activation in response to user requests and passes transponder locations to the relevant user terminals via telephone lines.

2.6 User Terminals. The user terminals are located on the users' premises and are connected to the central computer via telephone lines. The terminals usually take the form of a Track Display Station (TDS) which consists of QUIKTRAK software running on a personal computer. The TDS accepts operator inputs for vehicle location requests and displays vehicle locations on colour video maps stored within the TDS. A complete suite of maps for the system coverage area is stored within each TDS.

REFERENCE

Hurst, G. [September 1989] Quiktrak: a new AVL system developed in Australia. IREECON INTERNATIONAL MELBOURNE 1989 Convention Digest.

ANNEX IV

GUIDANCE SYSTEMS

1. Introduction

Increasingly the potential benefits associated with the use of guidance systems are being recognised. Many such systems have been developed in Europe and North America.

European collaboration has been agreed for the development of electronic systems for road traffic. The European Commission has funded a planning exercise called DRIVE (Dedicated Road Safety Systems & Intelligent Vehicles in Europe). This programme [Commission of the European Communities, 1988] is aimed to assess if information and telecommunication technology could make a major contribution to the improvement of road safety. The main emphasis is being placed on making the system available as widely as possible. Two of the major studies are of vehicle guidance and communication.

Also, PROMETHEUS (PRO-gramme for European Traffic with Highest Efficiency and Unprecedented Safety), is a programme initiated by the European car industry. One of the subprogrammes, Pro Road, suggests the creation of roadside communication and information equipment to provide data to a computer on a vehicle for use in higher level management systems. One important task is the standardization of a European digital road map.

2. Benefits

Research by the British traffic researchers [Belcher, 1989] has shown that saving of about 10% of journey costs could be achieved with a route guidance system. Average journey times could also be reduced by about 10%.

3. Types of guidance systems

Guidance systems can in general be classed as Autonomous systems and Infrastructure-supported systems.

3.1 Autonomous systems

The major advantage of autonomous systems is that they do not require any external navigation information and so they are self-contained. There are two types of autonomous system - simple directional aids and map display systems.

Simple directional aids are essentially dead reckoning systems. The on-board processor computes the vector connecting the initial location and the destination. Drivers are only provided with heading and distance information but the route of travel is decided by the driver.

Map display systems show the driver his position on a map display. A data base containing digitised cartographic data is required. In some map display systems, an automatic route search facility is available in an on-board computer which presents to the driver the optimum travel route. However these driving recommendations may in fact be sub-optimum because changes in the road network or actual traffic conditions have not been taken into account for route calculation. This problem might be overcome by receiving information from an external broadcast traffic channel such as RDS (Radio Data System) [European Broadcasting Union, 1984].

3.2 Infrastructure supported system

In an infrastructure supported system, a large number of roadside signposts are installed at strategic locations. Guidance information between signposts is provided by dead reckoning techniques. The vehicle on-board computer measures journey time between signposts and this information is transmitted to a central control room via two-way communication signposts which have the combined nature of the active and passive signposts. This information together with the traffic conditions such as the effects of incidents and time-of-day variations in traffic patterns are taken into account by a central computer which then determines the best route to the destination for each individual user. This recommended route data is transmitted to drivers via the signpost network.

Guidance and up-to-the minute traffic information are available in this system, but the information is only valid within the coverage area of the infrastructure network.

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REPORT 1155

**ADAPTATION OF MOBILE RADIOCOMMUNICATION TECHNOLOGY
TO THE NEEDS OF DEVELOPING COUNTRIES**

(Question 77/8)

(1990)

1. INTRODUCTION AND SUMMARY

Interim Working Party 8/13 was set up in 1985 by Decision 69 to study Future Public Land Mobile Telecommunication Systems (FPLMTS). It considered interalia that advances and cost reduction in radio and VLSI technologies could make the radio approach a competitive alternative for voice and non-voice telecommunication services, with special emphasis on personal communications, a concept of great interest to developed countries.

Recognizing the disparity that exists in telecommunication infrastructures in the world, this Report points out the potential of cellular technology (and its evolution into FPLMTS technologies) to help developing countries bridge the gap.

Also in 1985, the desire of developing countries in the Americas Region, through the plan-committee for Latin America, to take advantage of cellular technology to satisfy their own needs, including its use in fixed applications, led to Questions 77/8 and 38/9.

Report 1153 and Recommendation .687 include some of the aspects raised by these Questions with the idea that future wireless systems - FPLMTS - can serve the needs of developing as well as developed countries.

The objective of this Report is to emphasize the needs and interests of developing countries by promoting the application of FPLMTS for fixed services. It should furthermore be stressed that the use of FPLMTS for such applications is also attractive to developed countries.

Key objectives for FPLMTS are identified that could benefit developing countries: fixed service; standardization as a possible means of reducing costs; flexibility to start from a small, simple configuration and grow as needed as well as to accommodate special needs; commonality with remote areas of developed countries; large cells and repeaters.

2. THE TELECOMMUNICATIONS GAP

The role of telecommunications in the development process as a means to increase productivity and efficiency as a substitute for or complement to transportation and to save energy, etc. [ITU, a, b, c, 1988] is today more important than ever, but the gap between developed and developing countries is wider than ever. This has been clearly indicated by the Independent Commission for World Wide Telecommunications Development set up by the ITU in 1983, in its report [ITU, d, 1984]:

- 75% of the world's telephones are concentrated in 9 countries.
- 75% of the world's population live in countries with less than 10 telephones for every hundred persons.
- 50% of the world's population live in countries with less than one telephone for every hundred persons.

Faced with this reality one can only agree with the Commission's conclusion: "Neither in the name of common humanity nor on grounds of common interest is such a disparity acceptable."

The main reason for this situation is the limited financial resources available to a developing country to satisfy all its needs and the resulting allotment of resources to telecommunications which is not sufficient to close the gap as much — and as fast — as is required.

3. THE POTENTIAL OF FPLMTS TECHNOLOGIES

Present cellular technology and its evolution into the FPLMTS technologies by the end of the century, offer great potential to help developing countries bridge the gap in a more effective way [India, 1987]. Some of the aspects that make FPLMTS an attractive alternative to traditional wireline systems for telecommunications services are:

- Capability for rapid provision of service in new areas.
- Growth capacity and flexibility.
- Cost reductions resulting from technology improvements, modular design and mass usage of FPLMTS.
- Capability for covering wide geographical areas.

A number of developing countries are currently planning to use cellular technology to solve some of their problems such as lack of service in rural and remote areas or lack of capacity to satisfy demand promptly in urban areas.

4. OBJECTIVES AND ISSUES

Objectives have been set in Report 1153 for FPLMTS related to quality, mobility, flexibility, efficiency, connectivity, adaptability, security, identification, suggestions for charging (billing), etc.

The relative importance that each objective has to a developing country depends, on the one hand, on its stage of development and national goals, and on the other, whether effective cost savings will result from not implementing a specific feature in a particular situation.

Nevertheless, it is important to point out the features and characteristics that stand out as very important to developing countries, and deserve special notice: fixed service requirements, standardization and flexibility.

4.1 FIXED SERVICE

A very important objective defined for FPLMTS, from the point of view of developing countries, is that they can be used for the provision of service to fixed users either in rural or urban areas.

Two main factors that cause a radio solution to be considered are:

High installation and maintenance costs associated with rural wireline networks due to long distances, difficult terrain and climatic conditions.

The high and often difficult-to-foresee growth that limits the ability to plan the outside plant network properly in urban areas.

It is therefore of great importance for situations where wireline facilities are not available, that an FPLMTS can be used as a temporary or permanent substitute for the wireline network.

This objective requires FPLMTS to meet certain conditions and characteristics:

It should, to the maximum extent possible, support equipment and component design that can withstand rural conditions typically found in developing countries, such as rough roads, dusty environment, extreme temperatures or high humidity, while at the same time providing long equipment lifetime with minimum maintenance. It is, however, not intended that the basic FPLMTS equipment satisfies all the needs of the fixed service, while at the same time catering to the FPLMTS. Rather, the basic FPLMTS structure and technology should permit adaptation to fixed service requirements.

It should take into account the effect on reliability and availability of special propagation conditions such as, in mountainous terrains, areas of high rain precipitation or dense forests.

It should allow for large cell sizes.

Since the service to be provided is an extension of the national and international network, it should be possible to integrate it into the network without any, or with only very minor limitations. The objective that FPLMTS should provide a service with quality, integrity and security comparable to that provided in the fixed network is therefore very important to developing countries.

In some developing countries, the service provided by an FPLMTS may well be the only telecommunications pathway, and is likely to act as a lifeline, particularly for the provision of storm and flood warnings or similar public service activities. Service can also be provided to tourist areas, sports events and for other special needs.

The possibility of accessing both terrestrial and satellite systems, through the use of the same band or adjacent bands, could be important to countries with large, thinly populated rural areas. One possible application would be to link rural and remote base stations.

The use of an open architecture is of considerable importance for FPLMTS usage for fixed services as this would permit the administration in developing countries to choose the equipment based on performance needs.

CCIR Reports 380 and 1057 describe methods of providing fixed rural telecommunications and may be referred to for further information.

4.2

STANDARDIZATION

Report 1153 includes a discussion of a common radio interface standard from the point of view of personal communications. The advantage of such a standard in providing for regional and/or world wide roaming of mobile and personal stations as well as the coordination of spectrum allocations and planning are recognized. However, the limitations of a common standard in providing sufficient flexibility to meet diverse national or regional objectives, including the delay inherent in the standards process, are also noted.

From the point of view of some developing countries, standardization of the many interfaces associated with FPLMTS, including the radio interface, may offer important advantages:

- a) Reduction of costs of networks and terminals: Economies of scale resulting from the mass production of equipment generally lead to lower costs. Significant cost differentials, however, may not exist between FPLMTS equipment manufactured to regional standards as opposed to world-wide standards due to the large regional markets that exist.

In any case, lower cost equipment could lead to its use on a wider basis in developing countries.

- b) Increased equipment availability may facilitate the procurement process which could stimulate the introduction of FPLMTS in developing countries. Some developing countries may have an interest in the local manufacture of certain products unique to their markets for FPLMTS and may benefit from the flexibility provided by minimal standards. The use of standard elements within the system, such as speech codecs, RF components, etc., could contribute to the reduction of overall system costs.

4.3

FLEXIBILITY

Great importance has been given to an open, flexible architecture able to match network investment to revenue growth, adapt readily to environmental factors, different applications and new developments.

Of special importance to developing countries is to define the simplest possible system (e.g. speech alone), both from the point of view of hardware (terminals, base stations) and software. This may reduce cost and simplify maintenance. Of further special interest is the capability of the system to start from a small (e.g. stand-alone base station) and simple "stripped-down" (e.g. no roaming between cells or within a cell) configuration and grow as needed, both in size and complexity; thus enabling very low subscriber densities and low traffic rates to be catered for. This option, could allow a subset of the system's full service capability to be provided with minimum initial investment.

The use of a modular structure to permit simple configurations and future growth is of particular importance in developing countries.

Furthermore, there are special needs for fixed services such as: repeaters for covering long distances between terminals and base stations; PABX's, concentrators or small rural exchanges with wireless trunks; and different types of terminal equipment (2 or 4 wire terminals, coin boxes, etc.). The interface No. 8, defined in Figure 1 of Recommendation 687 and Report 1153 may be suitable to link MS to a PABX. This means, among other things, that the system should be able to be configured for situations of high traffic per terminal.

A graphical representation of what an FPLMTS is expected to achieve, accommodating both portable and fixed needs, is included in Report 1153, Figures 3 and 4.

5.**SERVICES**

The services that are proposed in Report 1153 for FPLMTS surpass the immediate needs of a number of developing countries. Furthermore, there are great differences in the stage of development and needs between developing countries.

The following services have been identified as more important to the developing countries:

- Voice
- Point to multipoint
- Short messages
- Paging
- Facsimile
- Text
- Data

The most immediate and largest requirement will be centered around voice communications.

Also, the FPLMTS should have sufficient charging (billing) flexibility to adapt to different charging schemes and be capable of being configured for special conditions where mobility between cells, or even within a cell, is not required.

6.**COMMON CHARACTERISTICS WITH RURAL AND REMOTE AREAS OF DEVELOPED COUNTRIES.**

The possibility of using mobile technologies for fixed applications is not only attractive and important to developing countries but also to some developed countries, especially those with extensive territories, difficult terrain and thinly populated areas.

Developed countries that must provide service to areas with such conditions face some of the same difficulties as developing countries.

7. LARGE CELLS AND THE NEED FOR REPEATERS

7.1 Large cells: area of radio coverage

Three aspects in particular distinguish the fixed service from FPLMTS:

- the cells tend to be large; the distance to be covered is often greater than is possible in one cell, while the number of subscribers is very low in all cells,
- the radio paths are fixed and well-defined, and
- the average traffic intensity per subscriber is typically 3 to 4 times higher than in the mobile service.

7.2 The need for repeaters

In a cellular mobile system, the cell centres are normally connected to a switching system by point-to-point microwave, cable, or fibre. In the fixed service, the total number of subscribers generally makes these means of connecting cells expensive. A means of employing a repeater to provide service to groups of subscribers well beyond line-of-sight has proven to be cost effective in today's point-to-multipoint communications systems. Without adversely affecting the mobile application the possibility of a repeater should be included in the basic structure of the FPLMTS.

Repeaters carrying multiple voice channels are inherently easier to design and build in a TDMA system than in an FDMA or CDMA system. The complexity and usefulness of the repeater depends on how many circuits the TDMA system employs on a radio carrier. The lower the number of circuits per RF carrier, the more difficult it is to design and build a repeater that can repeat all the circuits in a cell since more than one RF carrier may be needed to provide the desired number of circuits.

Another factor when implementing a repeater is the allowable radio path delay that arises from the longer distances and the processing delay in the repeaters themselves. It should be noted that a number of repeaters in series may be used to cover large distances or cope with mountainous terrain. The FPLMTS design should allow for such delays, perhaps on an optional basis.

8. ANTENNAS AND ANTENNA POLARIZATION

8.1 Antennas

Antennas used for fixed services, being stationary, can be directional and so have high gain.

Fixed service subscriber stations would use a single directional antenna to optimize the link to the base station which is normally either an omni or sectoral type. Mounting arrangements are kept as simple as possible (low height; existing structures) at the subscriber end whilst the base station should use an antenna mounted as high as possible to provide wide coverage. An additional difference for the fixed service is that connections between the transceiver and the antenna can sometimes be about 30 m in length, and is accomplished by coaxial cable feeders.

8.2 Polarization

Generally mobile services have used vertical polarization. Fixed service uses also permit horizontal polarization to be used where it is advantageous.

At a repeater location the discrimination obtainable between horizontal and vertical polarization can be very effectively used.

9. NEED FOR FURTHER STUDIES

The adaptation of mobile radiocommunications technology to the needs of the developing countries and the fixed service use of that technology requires ongoing analysis and inputs from all parties involved: the developing countries themselves, the developed countries with special interest in this issue, and the producers of technology and systems.

The areas which have been identified for further investigation are indicated in Report 1153.

Developing countries are particularly interested in:

- Propagation aspects
- Software and equipment simplification
- Rugged, simple to maintain terminal units for a wide range of environmental conditions
- Use of different types of terminals
- Use of concentrators, PABXs and Rural exchanges
- Services
- Flexibility and modularity
- Interference questions
- Evolution scenarios for FPLMTS
- Use of repeaters

Large cells
Reliability and availability
Performance objectives

Annex 1 to this report provides some preliminary considerations for performance objectives for circuit quality in the fixed service whilst Annex 2 provides some specific information regarding possible fixed service applications in India.

10. CONCLUSIONS

The ITU's Independent Commission for World-wide Telecommunications Development pointed out that industrialized and developing countries alike should join their efforts in improving and expanding telecommunications networks world-wide, fully exploiting the benefits of new technologies.

FPLMTS could be an effective means to help developing countries bridge the telecommunications gap by accelerating the expansion process of their networks and integrating their rural areas in to the national social and economic development process.

The objectives and architecture of FPLMTS noted in Report 1153 accommodate this need and requirements for the fixed service are summarized in Table I.

This report is intended to focus greater attention on this matter and encourage more analysis towards the unified Recommendation for FPLMTS.

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ITU b: ITU Publication, "Benefits of telecom to the transportation sector of developing countries", March 1988

ITU c: ITU Publication, "Telecom and the National Economy", May 1988

ITU d: "The Missing Link", Report of Independent Commission for World- Wide Telecommunications Development, ITU, 1984

TABLE 1 - REQUIREMENTS FOR THE FIXED SERVICE

ITEM	REQUIREMENT FOR FIXED SERVICE
1. VOICE ENCODING - BIT RATE - CIRCUIT NOISE - VOICE QUALITY	- FPLMTS SHOULD OFFER PERFORMANCE COMPARABLE TO THAT ACHIEVED IN THE FIXED NETWORK
2. RADIO COVERAGE - REPEATERS	- FPLMTS ARCHITECTURE SHOULD PERMIT THE POSSIBILITY OF REPEATERS. IF TDMA IS USED, REPEATER DESIGN IS SIMPLIFIED.
3. DESIGN LIFE 4. RELIABILITY 5. ENVIRONMENT 6. POWER CONSUMPTION 7. ANTENNAS	15 to 20 Years required for permanent fixed service applications MTBF for subscriber stations and base stations should be very high to achieve an acceptable maintenance cost. Some equipment may be exposed to the outdoor environment. It will have to withstand rain, snow, dust, sand, corrosion, insects, and a wide temperature and humidity range. As low as possible for solar and other sources. Directional at subscriber stations (in some cases base stations) for optimized radio path design using both vertical and horizontal polarization.

NOTE: It is important that items 1 and 2 above be taken into account in the basic design of the FPLMTS. Some of the areas where the fixed service application requires special design are listed in items 3 through 7.

Annex 1

Preliminary performance objectives for circuit quality for the fixed service

The following preliminary considerations on circuit quality were put forward by one administration.

1. VOICE ENCODING SCHEME

The voice encoding scheme determines the ultimate quality of voice communications. Lower bit rate encoding, while conserving spectrum, can introduce performance degradations and limitations when compared with 64 kbit/s PCM. It is reasonable to expect that substantial improvements in low bit rate encoding will be made over the next 10 years.

The driving force for low bit rate voice encoding in the FPLMTS is the need for a way to use the spectrum more efficiently in heavily congested urban areas, whereas when used in the fixed service, especially in remote areas, such congestion is not likely and use of higher encoding bit rates for such applications may be a desirable approach.

2. VOICE QUALITY

Voice quality when the FPLMTS is applied to the fixed service should be as close as possible to present day voice quality in fixed networks. The FPLMTS coding scheme should not significantly limit overall voice quality and intelligibility, reduce the possible number of tandem serial encodings in the network, nor cause overall network delay requirements to exceed established limits. As it will be an integral part of the fixed telecommunication network, the fixed service application of FPLMTS should not compromise the network's overall performance.

3. CIRCUIT NOISE

Levels of circuit noise that permit full integration with the national and international network without degradation beyond limits established for such connections by CCITT recommendations are necessary.

Idle noise levels — the noise not exceeded for about 99% of the time — of about 100 pWp are easily achieved with radio systems using 64 kbit/s PCM employing low cost, mass produced, encoding devices. A performance requirement of a similar order is desirable when FPLMTS are applied to the fixed service.

Annex 2

Some specific information relating to possible FPLMTS applications in the fixed service in India

1 TRAFFIC ESTIMATES FOR FIXED SERVICE APPLICATION IN INDIA

1.1 Introduction

Provision of telephone facilities in rural areas has gained momentum in recent years, and a number of low cost solutions are evolving for rural communications. Presently, the traffic generated by the Village Telephones (VTs) in India is quite low because the number of villages having a telephone facility is also very low (6.5%). The economic condition of the villagers and lack of tariff subsidies are other main reasons for the low traffic generated by the VTs. As the number of villages provided with a telephone facility increases, and also if tariff subsidies are provided, the traffic will grow substantially.

1.2 Available Data

No detailed studies have been done on the traffic growth patterns so far, mainly because the number of villages having telephones is very low. The only study that has been carried out was by NCAER (National Council of Applied Economic Research) on Long Distance Public Telephones (LDPTs) in 20 selected Telecom districts. Further details of the traffic information and the VT program are given in the publication, "A Phone in Every Village" by B.S. Murty, Dept. of Telecommunications Government of India.

2. SOME SPECIFIC POTENTIAL REQUIREMENTS FOR THE FPLMTS IN THE FIXED SERVICE IN INDIA

2.1 Incoming Calls to VTs

There are a few problems in handling incoming calls to the VTs. If the call is long distance, it is not possible to hold the calling party until the called subscriber arrives at the VT location to answer. In such a case, where the VT is primarily a community phone, the particular person (PP) has to be sent a message that he is required to answer the telephone. It may take a few minutes for a messenger to locate the PP. The calling party can then make a repeat call, say after 15 minutes, when the PP has reached the VT. Alternatively, he can initiate a call if the calling party's identity is known and if the PP is willing to pay for the call. In most cases the PP may only wish to receive the call but be unwilling to call back and pay.

To facilitate such "re-calling", it is suggested that the software be flexible enough to permit delaying the commencement of billing to the calling

party for the first 15 seconds so that this period can be used to leave a message regarding the identity of the PP. It is expected that the administration will consider such changes bringing relevant regulatory measures so that the volume of rural communications increases. There can be some scope for misuse of this facility for the passing of quick, short messages without paying, and this needs to be observed and remedies worked out.

It is expected that such implementation will be supported by PTN (Personal Telecommunication Number) service defined in FPLMTS. Also, low-cost hard-wired voice mail systems can be introduced to provide additional service to the villagers.

2.2 Quasi-stationary applications

There are a few cases which would provide fixed applications on a "quasi-stationary" basis. The concept is that any vehicle or person moving on some assigned tasks can carry a phone to the village or villages and allow the public to use the phone. Even though the vehicle is itself mobile, the phone is used and communications are carried out, only when the phone is stationary. The constraints of limited antenna gain and limited height are overcome in this mode since the antenna can be directed. In this case it is a general mobile used as a special mobile which puts the phone to use only when the vehicle is not moving. It is also not necessary to permanently fix the phone in the vehicle.

At present, only 6.5% villages have public phones while 93.5% of villages remain to be provided with telephones. To provide such a large number of villages with telephones will take some time. Therefore the quasi-stationary mode of operation described above will immediately improve the situation because one phone can be used in four to five villages at different times of the day.

3. LARGE CELL CONCEPTS FOR FIXED APPLICATIONS IN INDIA

3.1 General

To provide telephone facilities under the "a phone in every village" scheme, a number of low cost solutions are being developed and deployed in the villages. Presently, a very small percentage of villages are provided with telephones and even in these, the traffic generated is very low. Increasing the number of villages having telephones will increase the traffic; but at the same time, because of the economic conditions of the villagers, tariff subsidies need to be provided for rural communication. In addition a number of anomalies exist in the tariffs being charged for rural telephone subscribers as compared to urban telephone subscribers. Such anomalies need to be removed not only for promoting rural communication but in the interests of attaining the objectives of the

ITU Independent Commission for World-wide Telecommunications Development.

In an urban area, a telephone subscriber has access to 0.5 million to 1.0 million subscribers spread over a large geographical area, typically 40 km radius. For example in Bombay, nearly 0.7 million subscribers can be accessed over a 40-45 km belt at the cost of a local call. Also, there is no limit on the local call duration.

By contrast, a village subscriber, for all practical purposes, can be considered to not use a local call facility at all, because within a village, the number of telephones is very low and local switching activity is insignificant. So, each outgoing call from a village telephone will be a long-distance call which will also be metered.

3.2

FPLMTS coverage and its influence on the Unit Fee Zone (UFZ)

For fixed applications of FPLMTS, the coverage areas could be divided into three categories:

Category A consists of urban areas in which the mean traffic will be up to 25 calls/day/telephone and perhaps as high as 25 to 50 calls/day in special cases.

Category B consists of suburban areas in which the traffic will be 5 to 25 calls/day/telephone.

Category C consists of rural areas where the traffic will be 1 to 5 calls/day/telephone.

With the introduction of FPLMTS in a fixed mode and their coverage of urban and rural areas, it is expected that some of the anomalies mentioned above, will be overcome. The concept of UFZ needs to be redefined to fix the tariff so that the UFZ covers a large enough area with a group of villages and nearby towns. This region forms a "logical" area and all calls within the UFZ can be considered as local calls.

The FPLMTS can influence a new definition of UFZ as covered by the multiple cells connected to the main exchange (Mobile Switching Centre, MSC). Such a new definition can simplify the billing task. In addition, it will extend the "logical" area benefit to the village phones. A call originating from one UFZ and terminating in another UFZ is considered as a long-distance call. A time limit can also be put on the call duration for local calls.

This new definition of the UFZ concept will remove the anomalies between urban and rural subscribers because a village subscriber can then have ready access to the nearby villages and the nearest town.

A preliminary examination of one area in India indicates that cell radii of from 15 to 25 km will be typical.

REPORT 778-2

WIRELESS COMMUNICATION SYSTEMS FOR PERSONS WITH IMPAIRED HEARING

(Question 49/8)

(1978-1982-1990)

1. Introduction

Many forms of hearing impairment cannot be satisfactorily improved by the provision of amplification only. Difficulties such as distortion in the residual hearing, loss of binaural directivity, emphasis of environmental noise and room reverberation lead to the use of systems incorporating the placement of the microphone near to the speaker rather than to the listener.

2. Transmission systems

A number of means have been used to transfer the speech signals from the microphone to the listener's hearing device. The means include infra-red radiation, the magnetic induction field internal to current loops, VHF radio and the external induction field of a radiating antenna.

The use of infra-red radiation is of particular interest as it does not occupy allocated radio spectrum.

The radio induction field system is of particular interest as its realization leads to the following advantages:

- efficient use of the radio spectrum;
- ease of incorporation, with an acoustic hearing aid, into a single device;
- simulation of normal conditions of hearing;
- satisfactory use within the school, home, industrial or external environment.

On the other hand, VHF systems are employed to take advantage of the following:

- large coverage areas;
- relative immunity to natural and man-made noise.

3. System concepts

3.1 Radio induction field system

The mobile-to-mobile induction-field hearing assistance system exploits the FM capture effect to permit co-channel operation with selection by proximity. This pattern of selection closely parallels that used in ordinary conversation.

When an induction-field wireless hearing aid receiver is operated in the vicinity of two co-channel transmitters using a medium deviation FM transmission, the rapid change in field strength together with the FM capture effect ensures that there is a rapid changeover in reception from the more distant transmitter to the nearer transmitter with little subsequent breakthrough of consequence. For example, for a frequency deviation of 12 kHz and 75 μ s receiver de-emphasis, as reported for the Australian system in Annex I, it can be shown that, at a field strength ratio of 8 : 1, the maximum breakthrough from the more distant transmitter is 34 dB (unweighted). Within the region of inverse cubic decay of the induction field, the unwanted transmitter need only be at twice the distance of the wanted one to achieve this result. The field decay rate is illustrated in Fig. 1.

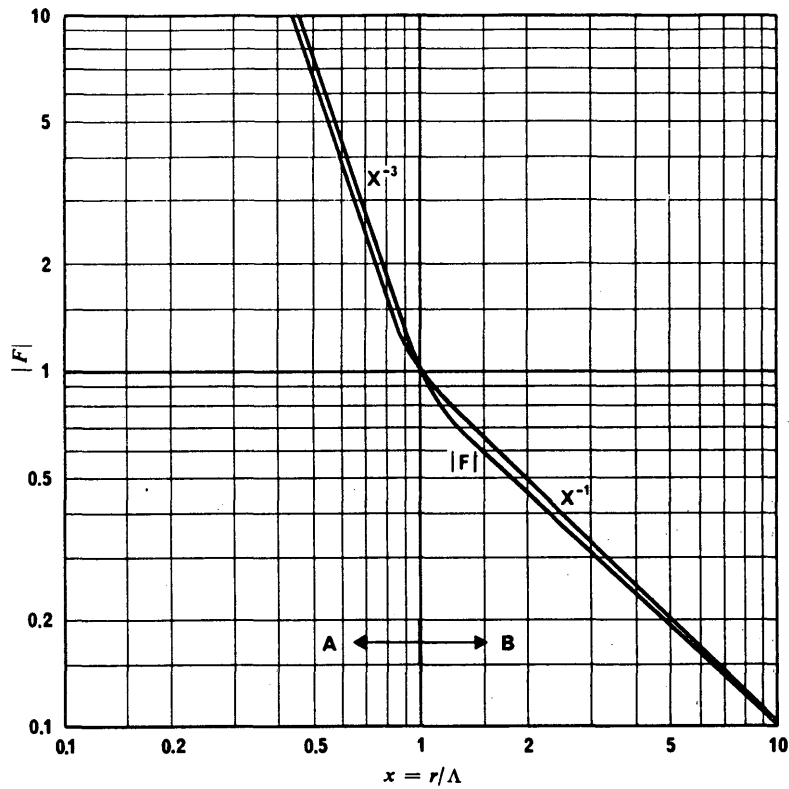


FIGURE 1 — *The field in free-space near a small dipole*

The field intensity $|F|$ in the equatorial plane is proportional to

$$|1/r^3 + j/\Lambda r^2 - 1/\Lambda r^3|.$$

Λ is the radian wavelength $= \lambda/2\pi \approx 48$ metres divided by frequency in MHz

A: Induction
B: Radiation

A magnetic induction field is preferred as it is less perturbed by conducting objects such as the human body, and is compatible with the use of compact ferrite rod antennas. The measured decay of a magnetic induction field is shown in Fig. 2.

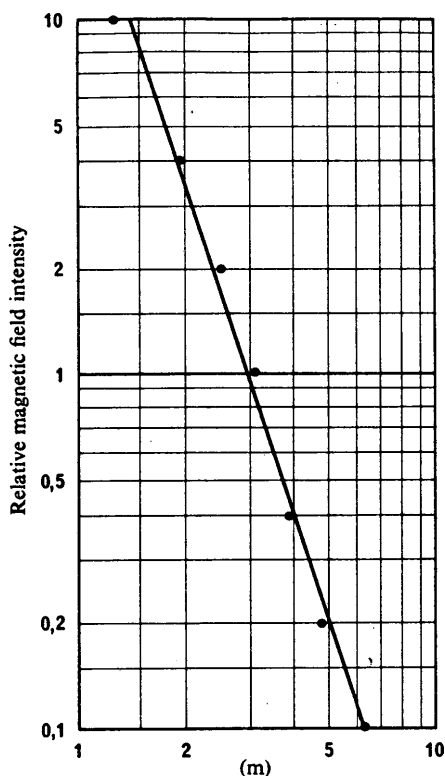


FIGURE 2 — *The measured decay of a magnetic induction field*

The points represent the measured values of the field; the straight line is an exact inverse cubic decay. The measurements were made in the laboratory in proximity to large metal objects. A frequency of 3.6 MHz was used.

The design of the induction-field wireless hearing aid proceeds from the following four principles:

- The upper limit for the carrier frequency is about 4 MHz; at higher frequencies the extent of the rapidly decaying induction field is less than 12 m, which is insufficient.
- The lower limit to the maximum frequency deviation is taken as 12 kHz as with lesser deviations, excessive breakthrough from nearby co-channel transmission occurs.
- The lower limit of the carrier frequency is taken as 3 MHz. The quality factor (Q) of tuned windings on ferrite rod antennas is of the order of 200. At lower carrier frequencies the bandwidth of the tuned antenna circuits cannot accommodate the required frequency deviation.
- The mean carrier frequency of all transmitters should be stabilized to within 20 Hz of their nominal channel frequency to avoid the production of sustained audible beat notes in receivers operated near more than one co-channel transmitter. Since the carrier frequency has been set below 4 MHz, the required degree of stabilization can be obtained by reference against quartz crystal oscillators operating at ambient temperature.

3.2 VHF system

Systems employing VHF radio transmission are capable of providing communication over distances greater than those using the radio-induction field system, as they employ transmission via a radiation field which decays less rapidly with distance than does an induction field. As a consequence, VHF radio transmission systems require that each transmission in any locale, such as a school and its environs, be assigned a separate frequency channel. This requirement is met with available frequency assignment methods, and is not a significant factor in the operation of the system.

VHF reception is generally less susceptible to interference from natural and man-made noise than is reception at lower frequencies, and systems employing VHF radio transmission may be useful in certain circumstances to avoid local problems of interference which may affect the operation of the radio-induction field system.

Radiocommunication systems intended only for short-range communication are capable of producing high field strengths at their required working distances, without radiating significant levels of power. Exploitation of the resulting possibilities of shared spectrum usage results in improved spectrum utilization, and may allow large numbers of channels to be made available to satisfy the requirements of large schools for children with impaired hearing.

The requirements of VHF auditory training systems used in the United States of America [CCIR, 1978-82] are also shown in Annex I.

ANNEX I

SYSTEM CHARACTERISTICS

1. Radio induction field system used in Australia

Using the system design principles described in the body of this Report, an operational induction-field wireless hearing aid system has been developed and is now in widespread use in Australia.

Evaluation of the system has indicated it provides substantial benefits including:

- greatly improved speech discrimination in noisy environments;
- virtual elimination of the problems of co-channel interference from adjacent systems as a result of the FM capture effect;
- greater flexibility for educational use. For example, with careful placement of pupils, a single frequency can be used in open plan classrooms with more than one teacher;
- the number of channels required in locations where there are many groups of users is reduced to four. Four channel transmitters and receivers have been developed for this purpose which also:
 - simplify frequency changes;
 - enable children to use the devices in different classrooms by selecting the appropriate frequency;
 - overcome difficulties associated with mixing groups of children with devices operating on different frequencies.

Some interference to reception has been experienced on the 3 175 kHz frequency (the frequency of the single channel devices) from high powered (10 kW) transmitters operating on 3 184 kHz at distances up to 30 km. This has been resolved by using four channel devices on other frequencies and in one case by changing the frequency of the high powered transmitter.

There has also been interference to the reception of signals on 3 175 kHz from the seventh harmonic of intermediate frequency ($7 \times 455 \text{ kHz} = 3 185 \text{ kHz}$). This spurious signal is generated within the receiver and degrades the quality of the received signal to noise ratio by adding to the receiver noise level. The problem has been overcome by altering the circuit board layout to minimize the interaction between the radio frequency and audio frequency signals.

The parameters are as follows:

Transmission medium:	Magnetic dipole induction field
Modulation:	FM
Frequency deviation:	± 12.5 kHz
Carrier frequencies:	3 175, 3 225, 3 275, 3 325 kHz
Frequency tolerance:	± 20 Hz
Audio frequency range:	100 Hz - 5 kHz
Audio pre-emphasis:	6 dB/octave
Transmitting antenna:	Ferrite rod, 127 mm x 10 mm, disposed vertically
Transmitter final stage power:	60 mW
Field strength produced at 3 metres:	11 mV/m (measured at a frequency of 3 175 kHz)
Transmitter radiated power:	38 nW (calculated from above)
Transmitter spurious emission:	Undetectable, but calculated as 0.1 pW
Transmitter dimensions:	145 mm x 53 mm x 18.5 mm
Receiving antenna:	Ferrite rod, 57 mm x 10 mm, disposed vertically
Receiver type:	Single conversion superheterodyne
Receiver dimensions:	80 mm x 53 mm x 18.5 mm (four channel device) 70 mm x 53 mm x 18.5 mm (single channel device)
Intermediate frequency:	455 kHz
System range:	12 m (subject to environment)

The low carrier frequency, which is specified to ensure that transmission takes place via an induction field, confers other benefits. It assists in keeping the receivers' battery consumption low and allows good image rejection to be obtained without recourse to double conversion superheterodyne techniques.

The use of a self-contained ferrite rod antenna is particularly convenient in a transmitter designed to be handed informally to another person.

2. VHF radio systems used in the United States of America

Systems have successfully shared the 72-76 MHz and the 88-108 MHz frequency bands for many years, with the type of radio services to which these frequency bands are allocated by the Radio Regulations.

2.1 72 to 76 MHz

Channel bandwidth:	50 kHz for a narrowband device 200 kHz for a wideband device
Frequency tolerance:	$\pm 0.005\%$ (Transmitter)
Frequency stability:	$\pm 0.005\%$ (Receiver)
Field strength produced at 30 m:	Not to exceed 8000 $\mu\text{V/m}$
Transmitter radiated power:	1170 μW (calculated from above)
Modulation requirements for FM:	± 20 kHz maximum (narrowband) ± 75 kHz maximum (wideband)
Out of band emissions:	25 kHz or more from carrier, no more than 150 $\mu\text{V/m}$ at 30 m for narrowband. 150 kHz or more from carrier, no more than 150 $\mu\text{V/m}$ at 30 m for wideband.
Receiver selectivity:	40 dB minimum, adjacent channel
Receiver image rejection:	40 dB minimum.

2.2 88 to 108 MHz

Channel bandwidth:	200 kHz
Field strength produced at 15 m:	Not to exceed 50 $\mu\text{V/m}$
Transmitter radiated power:	0.011 μW (calculated from above)
Out of band emissions:	100 kHz or more from the carrier, no more than 40 $\mu\text{V/m}$ at 3 m
Receiver standards:	Comply with normal receiver standards for this band.

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REPORT 1022 -1

**MULTI-TRANSMITTER RADIO SYSTEMS USING QUASI-SYNCHRONOUS (SIMULCAST)
TRANSMISSION IN THE LAND MOBILE SERVICE**

(Question 67/8)

(1986-1990)

This report consists of two parts: A and B. Part A deals with the technical consideration of quasi-synchronous (simulcast) transmission and Part B deals with an application of these techniques for digital transmission in the land mobile service.

PART A**TECHNICAL CONSIDERATIONS OF QUASI-SYNCHRONOUS
(SIMULCAST) TRANSMISSION****1. Introduction**

Special care must be taken with multi-transmitter systems in which the same message is transmitted at the same time in the same radio channel by two or more transmitters (such systems are known as quasi-synchronous or simulcast). They are used where:

- a large service area must be covered by transmitters of moderate power, in which case there will be a small overlap in the coverage of the transmitters;
- intensive coverage is needed, e.g. in some paging systems. In this case there will be a substantial overlap in the coverage of the transmitters which are used to provide diversity against shadowing (slow fading).

Note. - No diversity advantage is obtained against multipath fading.

Multi-transmitter simulcast transmission can be classified into three categories as follows:

- carrier frequency offset method [Hattori and Hirade, 1978];
- waveform offset method [Hattori and Ogose, 1980; Hattori *et al.*, 1982]; and
- modulation index offset method [Adachi, 1979].

It is necessary to study these methods with respect to transmission performance under fading conditions and their applicability to mobile radio transmission.

The following parameters are important:

- difference between the carrier frequencies of the transmitters;
- relative timing of the modulation at the transmitters caused by different delays in telephone lines and modulation circuits;
- differences in modulation depth and frequency response.

2. Error performance of digital transmission to moving receivers

With a small carrier frequency difference (< 10 Hz) and accurate modulation timing (< 0.2 bit period) the BER at a moving receiver, in Rayleigh fading conditions, in a multi-transmitter system is approximately the same as with a single transmitter, for a given received signal level, as shown in Fig. 1 (for the case of direct frequency modulation of the RF carrier). Typical error distributions are shown in Fig. 2 [French, 1980].

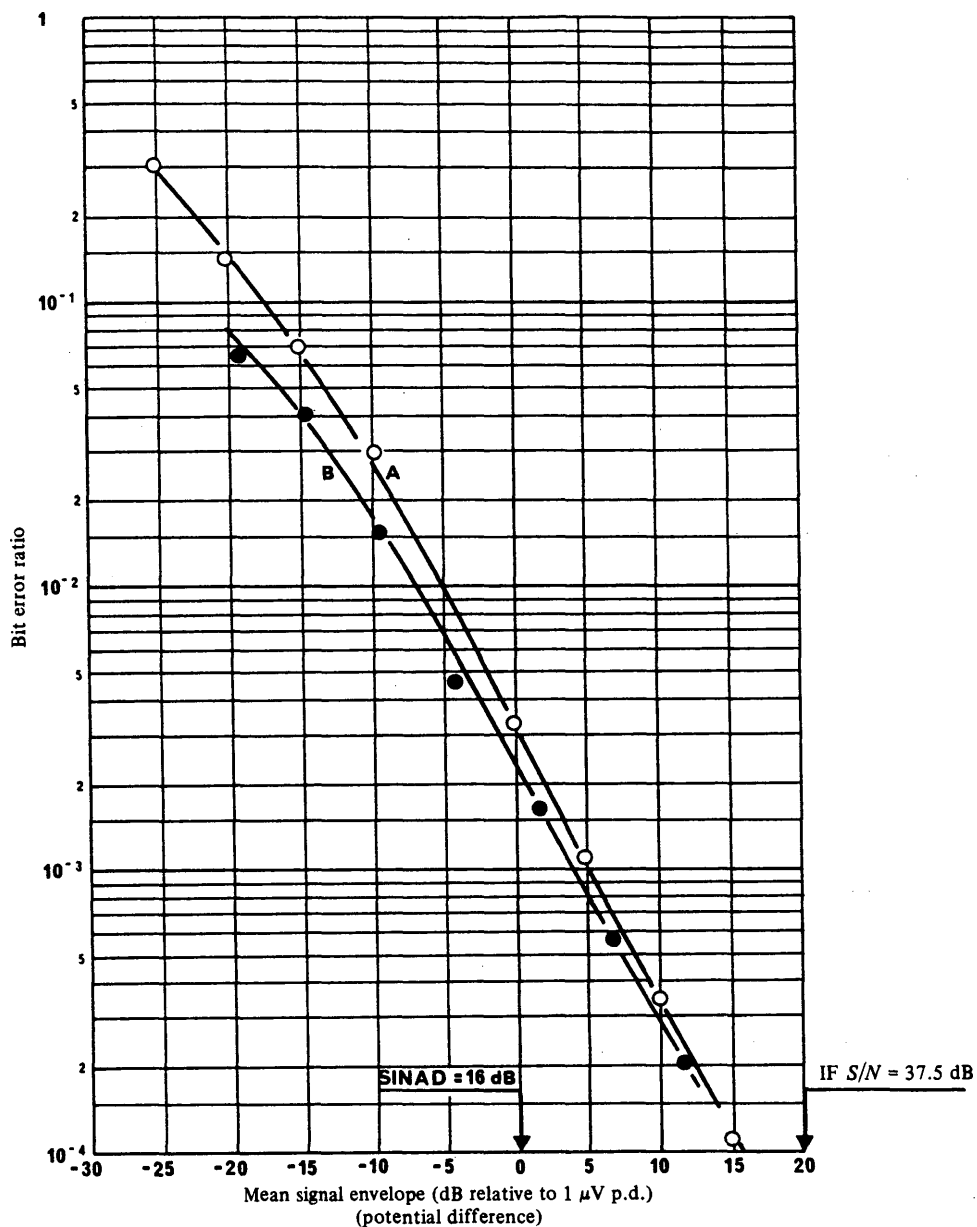


FIGURE 1 – Error performance, mobile, 1200 bit/s, direct FM

SINAD: 12 dB at $0.5 \mu\text{V pd}$

Curves A: two transmitters

B: single transmitter

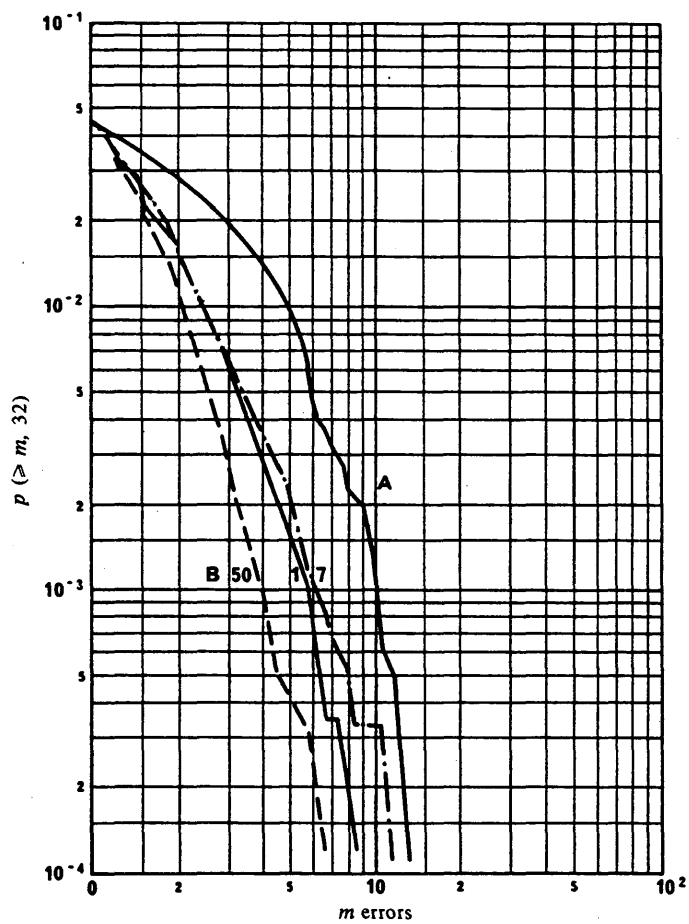


FIGURE 2 - Error distribution, mobile, 1200 bit/s, direct FM

$$\text{BER} \approx 3 \times 10^{-3}$$

Curves A: one transmitter

B: carrier frequency difference Δf_c

To avoid an excessive BER due to difference in modulation timing, the timing difference should be less than 0.3 bit period for direct frequency modulation of the carrier, as indicated in Fig. 3. With sub-carrier data modulation the timing difference must be small compared to the period of the sub-carrier.

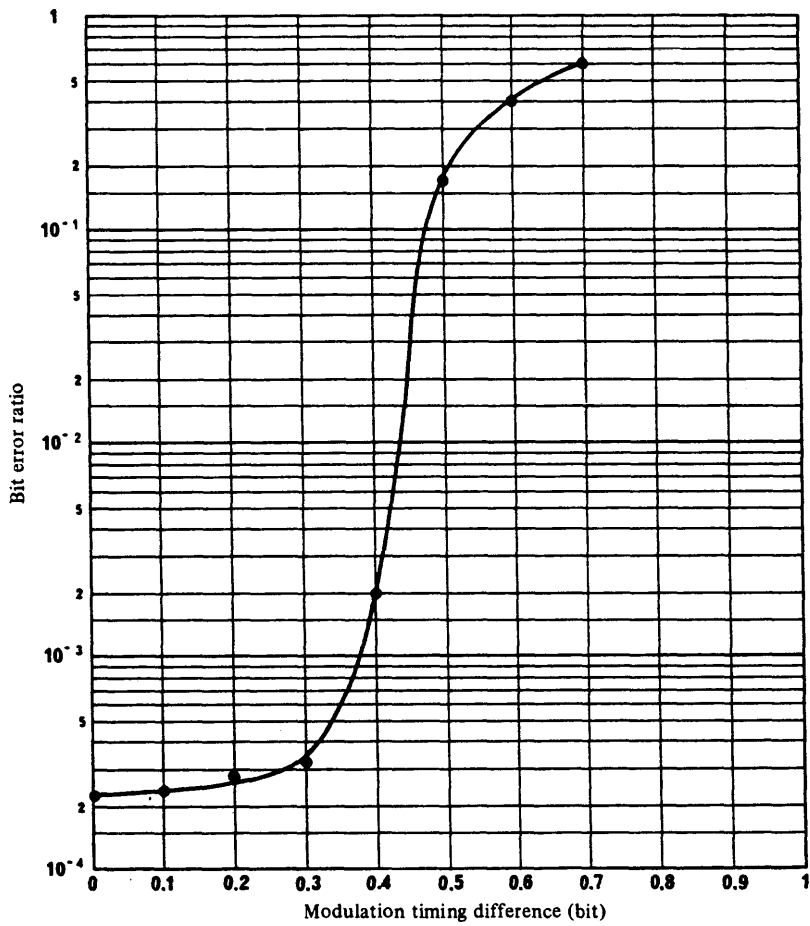


FIGURE 3 – Bit error ratio modulation timing, mobile, 1200 bit/s, direct FM
 $\Delta f_c = 7 \text{ Hz}$

Increasing the carrier frequency difference to approximately the bit rate, reduces the BER by two orders of magnitude, as in Fig. 4 [French, 1980]. At higher bit rates this difference in carrier frequency reduces spectrum efficiency.

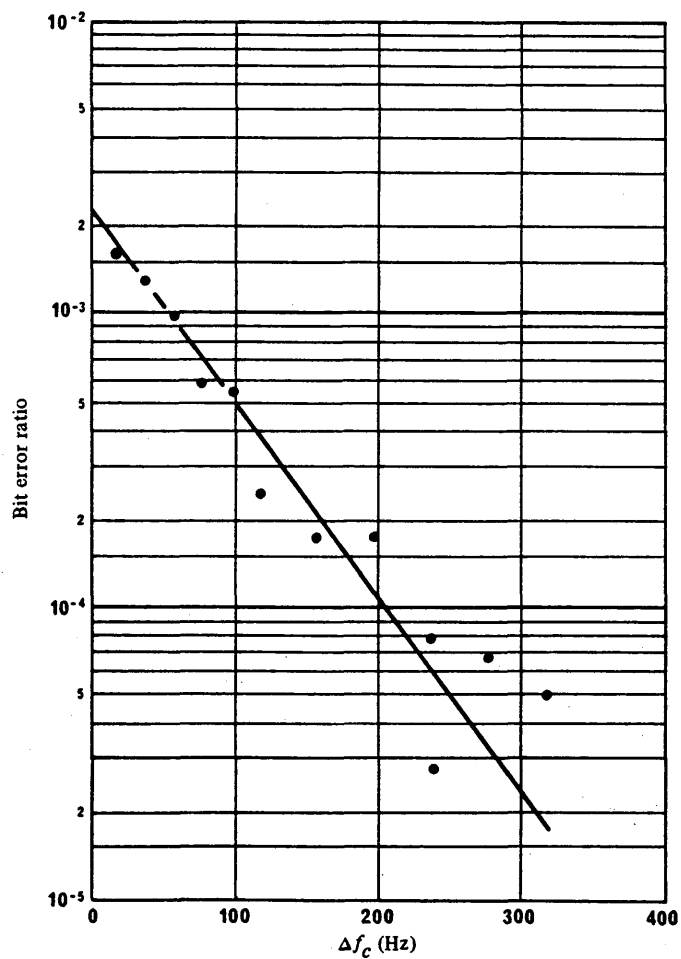


FIGURE 4 – Bit error ratio against carrier frequency difference, 300 bit/s, direct FM

Laboratory test results concerning offset methods of carrier frequency, waveform and modulation index are shown in Figs. 5 to 9 inclusive. As shown in Fig. 5, each method results in performance improvement (diversity effect of more than 10 dB) when the average equivalent levels of two received signals are combined.

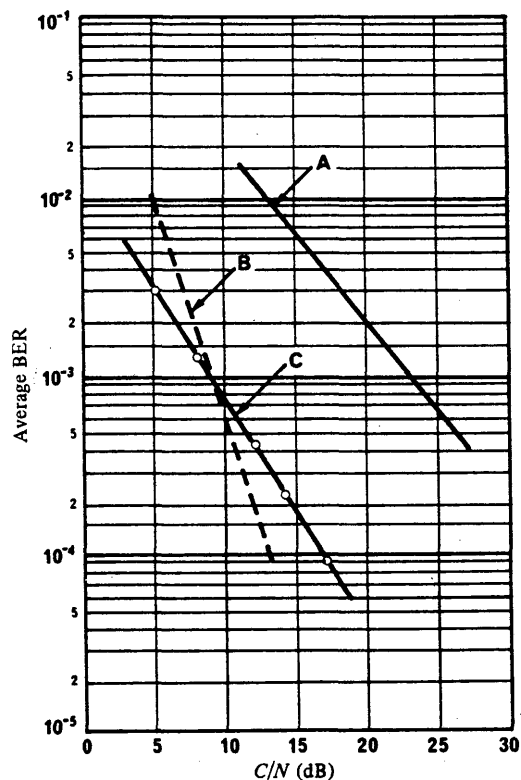


FIGURE 5a – BER performance versus C/N of carrier frequency offset method and waveform offset method for 300 bit/s Manchester coded signal with direct FM

Curves A: single transmitter ($\Delta f_D = \pm 3$ kHz)

B: two transmitters with carrier frequency offset ($\Delta f_C = 1$ kHz, $\Delta f_D = \pm 3$ kHz)

C: two transmitters with waveform offset ($\Delta f_d = \pm 3.5$ kHz, $\Delta f_{dh} = \pm 1$ kHz)

Maximum Doppler frequency: 40 Hz

Cross-correlation coefficient: 0

Δf_C : carrier frequency difference

Δf_D : maximum frequency deviation

Δf_d : average frequency deviation of offset waveform method

Δf_{dh} : frequency deviation corresponding to the peak amplitude of the sinusoidal wave ($= \Delta f_D - \Delta f_d$)

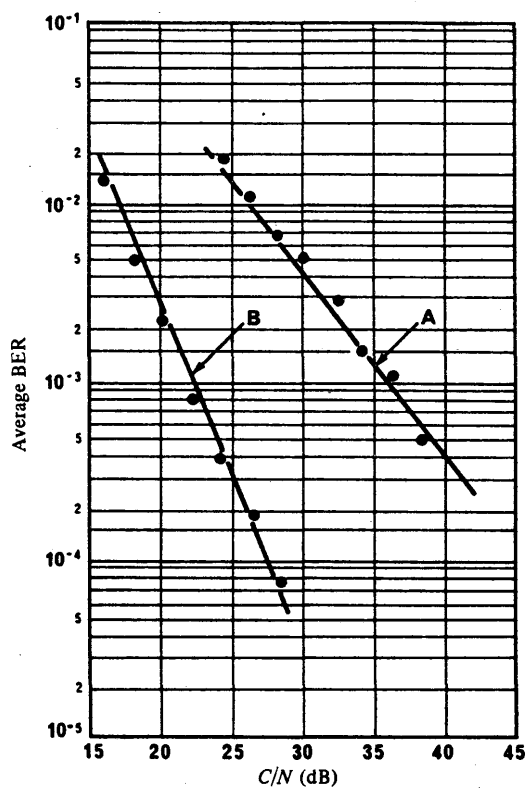


FIGURE 5b – BER performance versus C/N
of modulation index offset method
for 600 bit/s differentially encoded
Manchester coded signal with direct FM

Curves A: single transmitter (modulation index $\beta = 9.7$)

B: two transmitters with modulation index offset
(modulation index $\beta_1 = 11.7, \beta_2 = 9.7$)

Maximum Doppler frequency: 40 Hz

Cross-correlation coefficient: 0

•: experimental results

If the carrier frequency fluctuates, performance improvement using the frequency offset method changes greatly in the range from 0 to 2 kHz, as shown in Fig. 6 by the broken line. The waveform offset method results in more stable and better performance, as shown in Fig. 6 by the solid line.

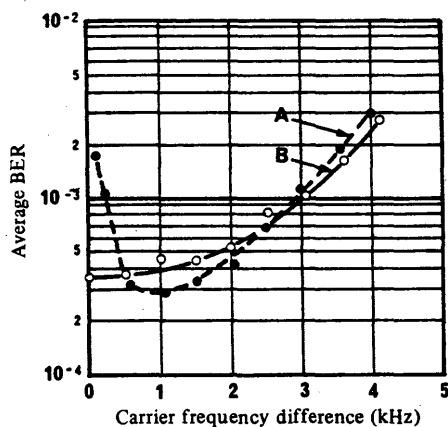


FIGURE 6 – BER performance versus carrier frequency difference

Curves A: carrier frequency offset method ($\Delta f_D = \pm 3.5$ kHz)

B: waveform offset method ($\Delta f_d = \pm 3.5$ kHz, $\Delta f_{dh} = \pm 1$ kHz)

Maximum Doppler frequency: 40 Hz

Transmission bit rate: $f_b = 300$ bit/s

The diversity effect decreases as the modulation signal phase difference increases. The improvement characteristics of the phase difference for the frequency offset and the waveform offset methods are similar. In addition, the allowable value for each method is approximately 60° (0.16 bit duration), as shown in Fig. 7.

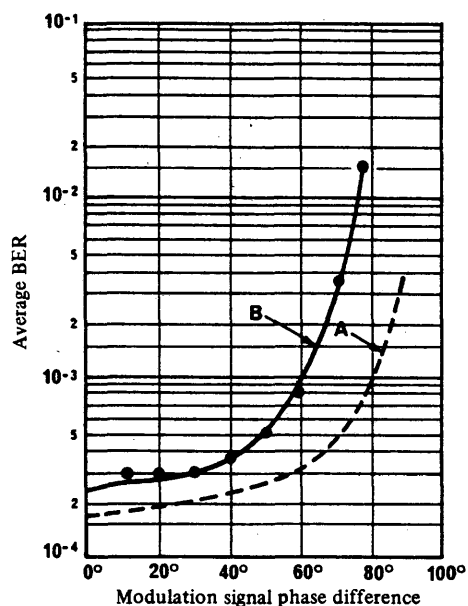


FIGURE 7 – Effect of modulation signal phase difference on BER performance

Curves A: carrier frequency offset method
 $(\Delta f_C = 1 \text{ kHz}, \Delta f_D = \pm 3 \text{ kHz})$

B: waveform offset method
 $(\Delta f_d = \pm 3.5 \text{ kHz}, \Delta f_{dh} = \pm 1 \text{ kHz})$

Transmission bit rate: $f_b = 300 \text{ bit/s}$

Maximum Doppler frequency: 40 Hz

Cross-correlation coefficient: 0

In the modulation index offset method, BER is reduced by an order of magnitude and the optimum index value difference is 2, as shown in Fig. 8.

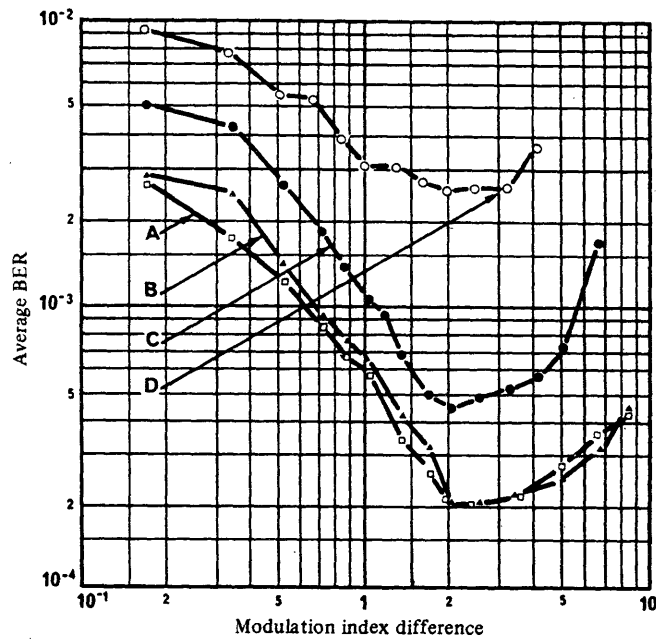


FIGURE 8 – BER performance in the modulation index offset method for 600 bit/s differentially encoded Manchester coded signal with direct FM

Modulation index of one transmitter is set as:

Curves A: $\beta_1 = 13.3$

B: $\beta_1 = 11.7$

C: $\beta_1 = 8.3$

D: $\beta_1 = 5$

$C/N = 26$ dB

Modulation index difference: $\Delta\beta = \beta_1 - \beta_2$ ($\beta_1 > \beta_2$)

Cross-correlation coefficient: 0

The waveform offset method has an advantage in that it maintains good diversity effect uniformly in a wide range of phase difference between superimposed sinusoidal signals, as shown in Fig. 9. This results in uniform and good performance in all the overlapping surrounding cell areas when a phase difference of 60° among base stations is chosen. The carrier frequency offset and waveform offset methods are successfully used in the Japanese land mobile telephone systems.

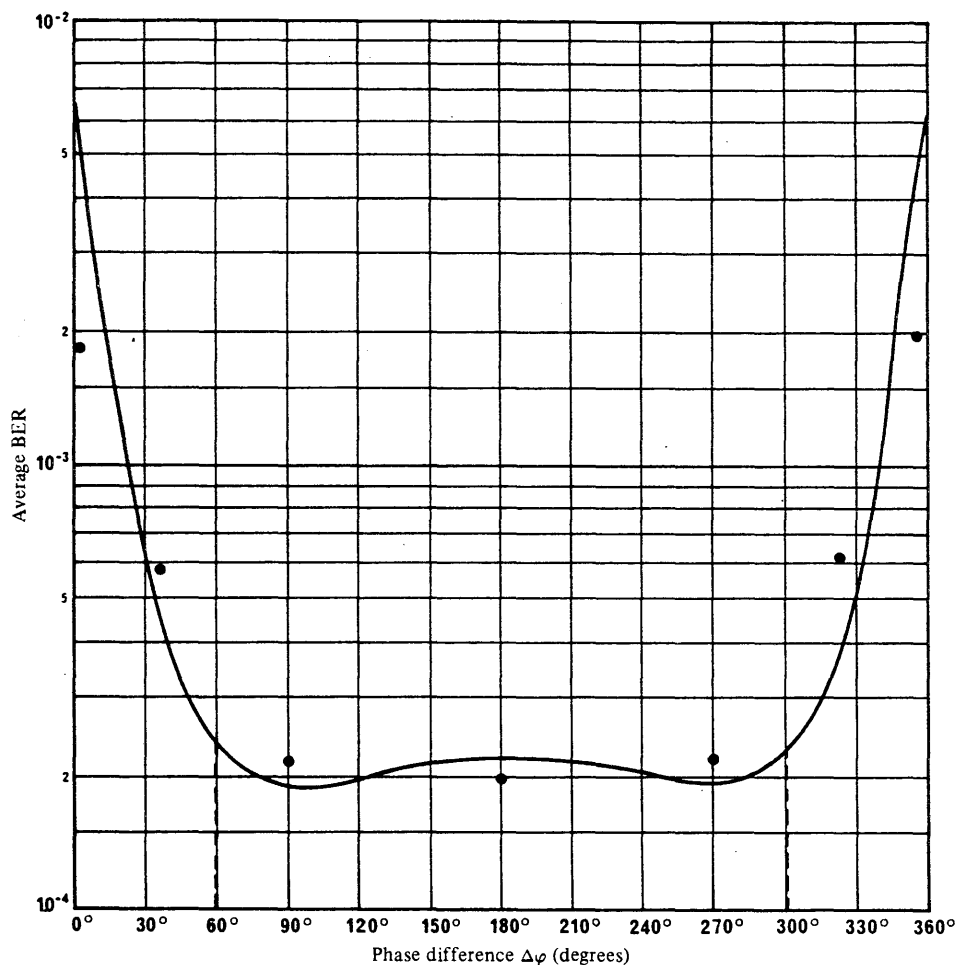


FIGURE 9 – Effect of phase difference between two superimposed sinusoidal waves

Waveform offset method ($\Delta f_d = \pm 3.5$ kHz, $\Delta f_{dh} = \pm 1$ kHz)

Transmission bit rate: $f_b = 300$ bit/s

Maximum Doppler frequency: 40 Hz

Cross-correlation coefficient: 0

3. Error performance of digital transmission to stationary receivers

At a stationary receiver the steady signals received from say two transmitters will suffer carrier interference, leading to a periodic variation of the amplitude of the resultant signal at the carrier difference frequency. The measured BER as a function of the absolute and relative levels of the two received signals, compared with theory, is shown for amplitude modulation (sub-carrier 1200 bit/s FFSK) in Fig. 10. Similar results are shown for frequency modulation (sub-carrier 1200 bit/s FFSK) in Fig. 11.

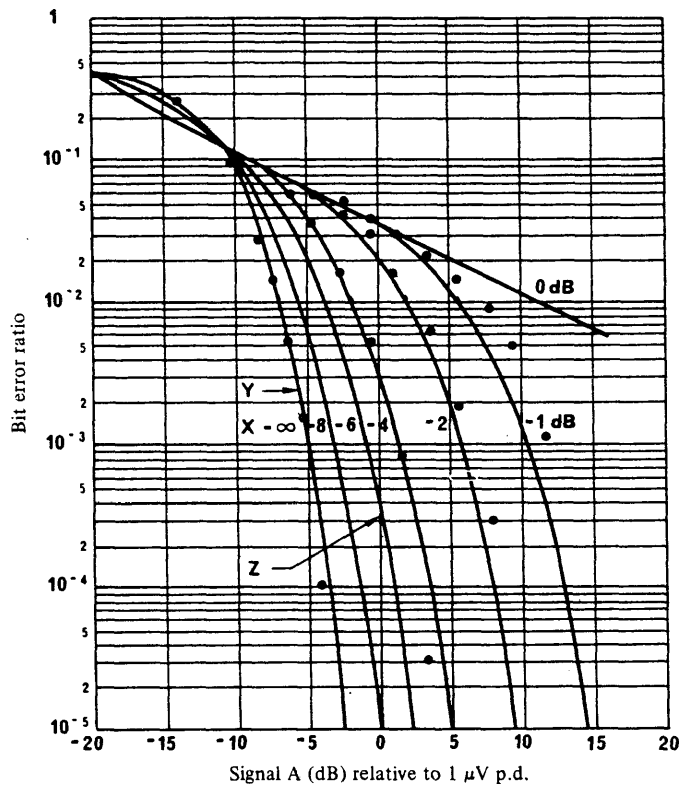


FIGURE 10 – Stationary AM quasi-synchronous error performance

12 dB SINAD at -6.5 dB

$\Delta f_c = 2$ Hz

X: level of signal B relative to A

Y: A alone

Z: theory

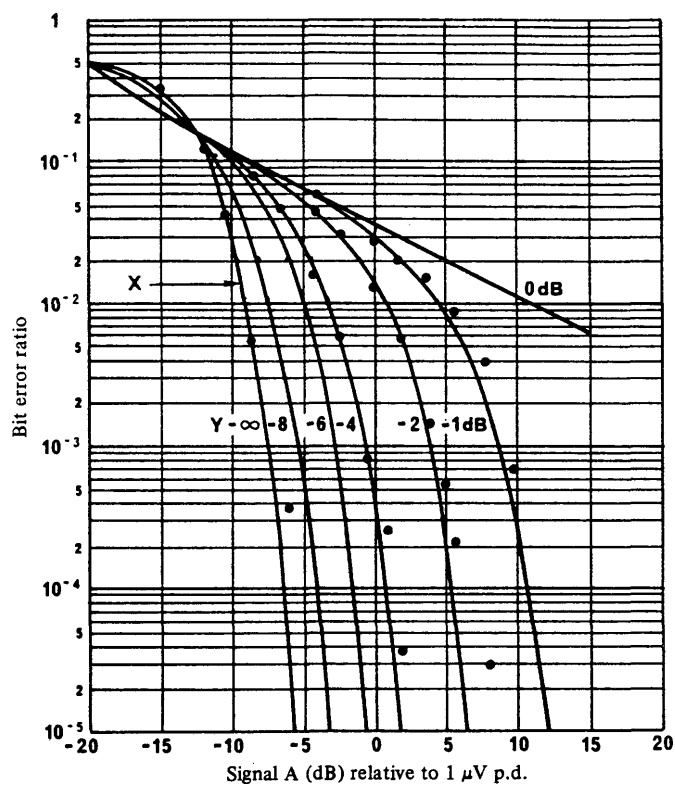


FIGURE 11 – Stationary FM quasi-synchronous error performance

12 dB SINAD at -9 dB

$\Delta f_c = 2$ Hz

X: A alone

Y: level of signal B relative to A

The mean word error ratio, averaged over a large number of stationary receivers in Rayleigh fading conditions is given in Fig. 12, compared with single transmitter operation, for 64 bit code words and a carrier frequency difference of 1.8 Hz. At lower signal levels (< 12 dB relative $1 \mu\text{V}$) performance is improved with the second transmitter ($= 2.5$ dB) but at high signal levels it is degraded by about 4.5 dB. At higher carrier frequency differences of 18 Hz (see Fig. 13) and 180 Hz, the performance is degraded by between 5 and 10 dB [French, 1982].

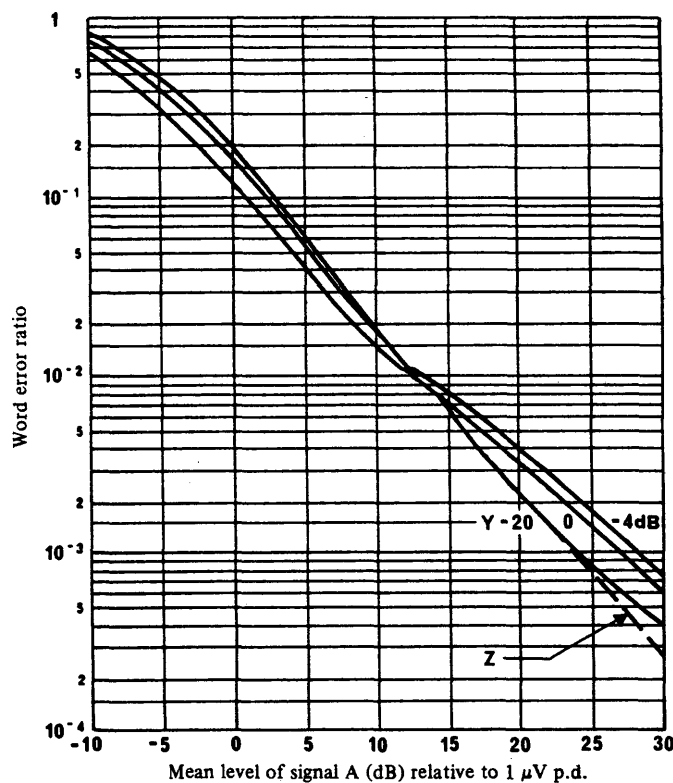


FIGURE 12 – Mean word error ratio, amplitude modulation

$$\Delta f_c = 1.875 \text{ Hz,}$$

$$f_b = 1200 \text{ bit/s,}$$

64-bit words.

Y: relative mean of signal B

Z: A alone

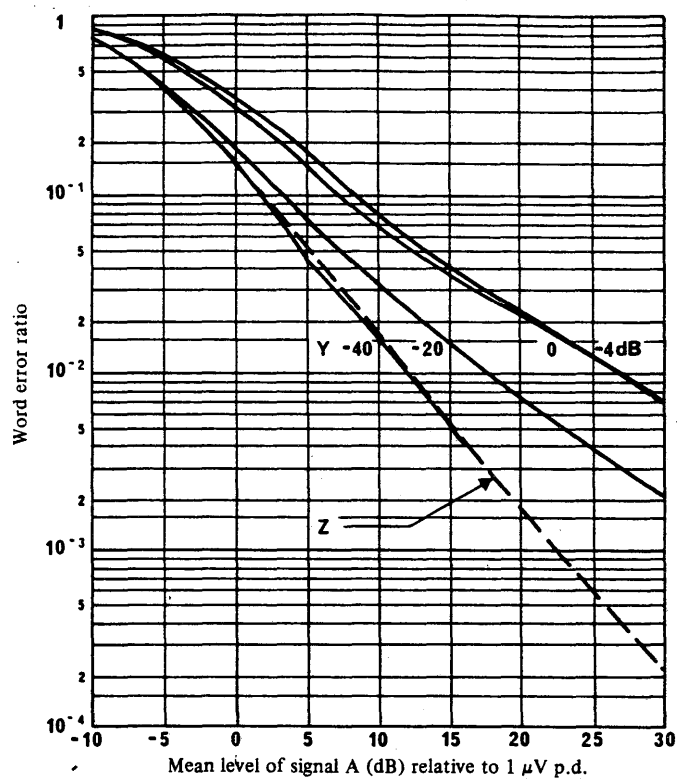


FIGURE 13 - Mean word error ratio, amplitude modulation

$$\Delta f_c = 18.75 \text{ Hz}$$

$$f_b = 1200 \text{ bit/s,}$$

64-bit words

Y: relative mean of signal B

Z: A alone

PART B

APPLICATION OF QUASI-SYNCHRONOUS (SIMULCAST)
TRANSMISSION FOR DIGITAL TRANSMISSION

1. Introduction

Land mobile radio telephone systems using analogue cellular technology are either in place or are being deployed in many parts of the world. There is a growing demand for data communications over these cellular systems. Currently, the cellular system providers are investigating new concepts for long and short term solutions to accommodate data application in their networks.

A single frequency system concept is investigated and simulated for the transmission of data over land mobile radio channels. This single frequency system concept operates on a duplex channel dedicated for data applications. The mobile-to-base link is

shared by all the data users in the system (Hafez et al., 1987) based on a slotted-ALOHA scheme (Kleinrock and Tobagi, 1975). The base-to-mobile link carries the data packets to the mobiles as well as the timing information needed to regulate the traffic flow on the mobile-to-base link. In the base-to-mobile transmission the same data packet is transmitted simultaneously from all cell sites. This scheme, in its simplest form, will not require hand-offs. One potential application of the single frequency concept is as an overlay over the existing cellular systems through an extended protocol.

2. Conclusions

The performance criterion, as expressed in delay versus throughput with the number of cells as a parameter, is illustrated in Fig. 14. As shown, two observations can be made:

- 1) There is a strong correlation in the delay throughput between calculated and simulated results. Moreover, better performance is observed in the simulated data.
- 2) The throughput of the mobile-to-base channel increases approximately linearly with the number of cells in the system provided that saturation has not been reached on the base-to-mobile channel. In the event of saturation, additional base-to-mobile channels or enhanced intelligence can be provided on the system. For example, with an average delay of six (6) time slots, the 37, 61, 91, and 127 cells systems mobile-to-base throughputs are, respectively, 2.7, 4.2, 5.9, and 8.1 times greater than that of an equivalent single cell system.

3. Further Study

Further studies are required to investigate single frequency simulcast systems, including the following:

- (a) Optimum bit rate on the base-to-mobile link.
- (b) Optimum location and packet routing strategy.

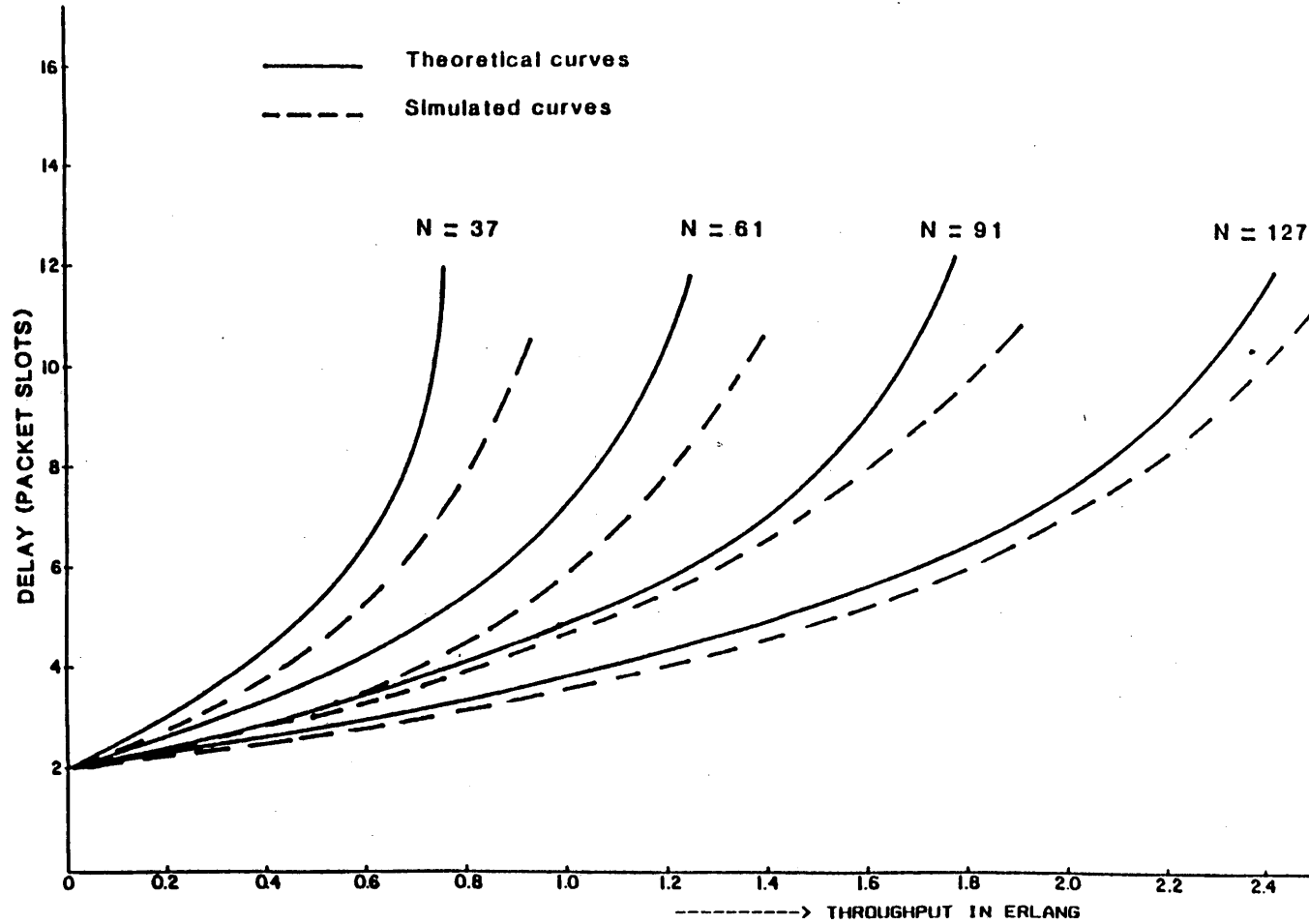


FIGURE 14

Calculated and simulated delays vs throughput as function of number of cells"

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REPORT 1023-1

**FREQUENCY SHARING BETWEEN THE LAND MOBILE SERVICE AND THE
BROADCASTING SERVICE (TELEVISION) BELOW 1 GHz**

(Question 69/8)

(1986-1990)

1. Introduction

The WARC-79 increased the number of frequency bands that might be shared between the broadcasting service and the mobile service below 1 GHz.

If the same portion of the frequency spectrum is allocated to two or more services, sharing may be effected in one of the following ways:

- a) use of the same frequency band by different services at different times;
- b) simultaneous use of different parts of the shared bands by different services;
- c) simultaneous use of the same parts of the shared bands by different services, but in separate geographical areas.

This Report examines the case of sharing in separate geographical areas as in c) above.

2. Characteristics of the two services

Television services typically operate with considerably larger radiated powers than mobile services (typically 40 dB more) and mobile services typically operate with considerably smaller usable field strengths than television (typically 30 dB less). This total difference of 70 dB in the planning criteria of the two services suggests initially that sharing will have very limited application. However, in practice the following criteria assist the mobile services:

- the protection ratio required by the mobile services is typically less than that for the television service (typically 10 dB as opposed to 50 dB);
- the service areas in the mobile service are smaller, so antenna heights are less than broadcasting antenna heights and the height gain correspondingly less;
- the television service employs a much wider bandwidth than the mobile service (typically 8 MHz against 12.5 kHz) so the full power of the television signal will not be present in any one mobile channel. In fact, the power density of the television signal is not constant across the television channel, as discussed below.

3. Power spectral density of a television signal

Figure 1 shows the limit of the power spectral density of a television signal when measured in a 7 kHz bandwidth (appropriate to a 12.5 kHz channelled mobile radio receiver). This mask is a composite for the PAL and SECAM systems in use in Europe. The frequency separation between the various carriers is different for the different systems but the relative power levels do not change greatly. It can be seen that between the carriers the power is at least 30 dB below the total power and for much of the television channel is at least 40 dB below.

Figure 1 refers to normal picture scenes. The power spectral density of electronically generated pictures requires further study.

* This Report should be brought to the attention of Study Groups 1 and 11.

4. Example of sharing

As an example of the sharing possibilities, consider the case of mobile operation around 200 MHz in television Band III (174-230 MHz). From Report 358 for Grade 4 signal quality the requirements of the mobile service are:

- median field strength to be protected: 22 dB(μ V/m);
- protection ratio: 10 dB;
- maximum interfering field strength: 12 dB(μ V/m).

Assume the following typical characteristics for the mobile service:

- base station effective antenna height: 75 m
(height gain relative to 10 m: +9 dB)*;
- mobile station effective antenna height: 3 m
(height gain relative to 10 m: -4.5 dB)*.

Assume the following to be typical for the television service:

- effective antenna height: 300 m;
- effective radiated power: 250 kW.

Then from Recommendation 370 the separation distances required (for 10% time 50% location) are given in Table I.

TABLE I – Required separation distances to protect the mobile service
(television radiated power between carrier frequencies assumed 30 dB
below total radiated power)

	On carrier frequencies (km)	Between carrier frequencies (km)
Base station	560	260
Mobile station	430	180

It can be seen that, whereas the distances are large for operation on the carrier frequencies, the distances are considerably reduced for operation on frequencies between the carriers. Thus, mobile operation may not be practicable on the television carrier frequencies, but a substantial amount of spectrum remains in which operation may be practicable.

5. Receiving antenna discrimination

Mobile services are constrained to use vertical polarization due to the difficulty otherwise of mounting the mobile antennas, particularly at the lower frequencies. Television services may, however, use horizontal or vertical polarization. The use of horizontal polarization by the television service greatly facilitates frequency sharing by the mobile and television services as advantage can be taken of an antenna discrimination factor of around 15 dB. This discrimination applies only to base stations. In the case of mobiles the discrimination will be much less – perhaps 3 dB – but the base station interference is the dominant case as shown in Table I. The separation distance of 260 km reduces to 160 km in the example of Table I if orthogonal polarization is assumed.

* These values require further study.

6. Protection of television services

According to Recommendation 417 the minimum (median) field strengths for which protection may be sought in planning television services are 48, 55, 65, or 70 dB(μ V/m) for bands I (47-68 MHz), III (174-230), IV (470-582 MHz) and V (582-960 MHz), respectively. The use of lower values militates against sharing. It can be seen from Fig. 1, that (apart from in the vicinity of the carriers) moderate increases in television transmitter power to improve the field strength at the edge of the service area can have little effect on mobile usage. Also, since the carrier frequencies are unlikely to be usable by the mobile service, the interference will not be increased if low power "fill-in" transmitters are employed, providing that the carrier frequencies align to the same frequency grid. Sharing is further facilitated if higher protected field strengths can be accepted for the service areas of these fill-in transmitters.

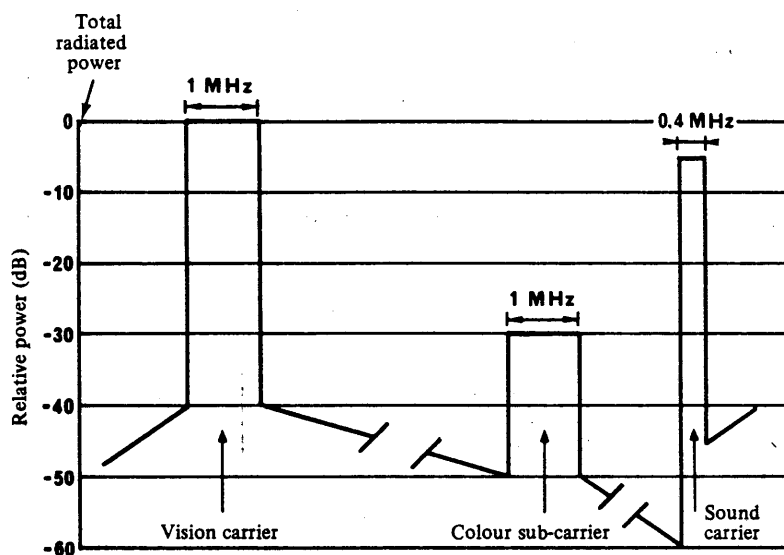


FIGURE 1 – Relative values of peak radiated power (measured in a 7 kHz band)

7. Protection ratios

Protection ratios for television signals are given in Recommendation 418 and Report 306. These references show that the protection ratio is not constant, for narrow-band interference across the television channel, being greatest (around 50 dB) near the vision carrier and least (around 30 dB) near the sound carrier. Mobile services can be planned to take advantage of this by assigning frequencies near the sound carrier to base station sites having the greatest interference potential and frequencies near the vision carrier to those having the least interference potential, or to mobiles.

8. Multiple interference

A new consideration that will arise with sharing between the mobile and television services is the effect on the television signal of multiple interfering sources. This arises because of the large difference in bandwidth used by the two services which implies that there could be 300 or 400 mobile service transmitters operating in a television channel.

It is possible that the simplified multiplication method as detailed in Report 945, is appropriate to this case, but further study is required. Also, it is unlikely that all the mobile service transmitters would be operating simultaneously and further study is required on the statistics of mobile operation noting that sharing will be facilitated if mobile operation can be confined to specific areas.

9. Interference from base and mobile stations

Base stations are likely to cause a greater interfering effect to television than mobile stations. The lower antenna height of mobiles, coupled with their limited radiated power and the general clutter effects caused by buildings, combine to make the interfering field strength from a mobile station to be around 20 dB less than that from the corresponding base station. This factor will not be achieved if the mobile is operating from a high clear site, (such interference may be transitory) or if mobiles operating on a co-channel basis encroach upon the television service area. While interference from mobiles in these situations may only occur for a small percentage of time the interference caused will be unacceptable if it results in severe disruption (as distinct from minor observable degradation) of the television reception.

A typical base-station radiated power in the mobile service is 25 W. By using directional antennas this can be effectively reduced, typically to 2.5 W, in the direction of the television broadcast service area.

Annex I gives a method showing how the effect of multiple interference from narrow-channel land mobile stations may be aggregated within the wider bandwidth of a television channel, and demonstrates the advantages accruing from the use of directional antennas.

10. Conclusions

Sharing between the mobile and broadcasting (television) services is practicable in separate geographical areas, particularly if the television carrier frequencies are avoided.

Sharing becomes easier as the protected field strength of both services is increased. Also, sharing is facilitated if horizontal polarization is used in the television service. However, where polarization discrimination could be employed to facilitate sharing within the broadcasting service, the overall spectrum advantages will need careful study.

The effects of multiple sources of interference on the television service are not fully understood and require further study.

ANNEX I

At sites close to the coast that offer a predominantly sea path, particular care should be taken to reduce field strengths that interfere with the television service. It is possible for a single base station installation to produce an interfering field strength equal to the value of field strength of the television service to be protected. This prevents use of the television channel by the land mobile service.

The interfering, or nuisance, field is calculated by (see Report 945):

$$E_n = \sqrt{\sum_{i=1}^n (a_i + b_i + e_i)^2}$$

where:

a_i : radio-frequency protection ratio (dB) associated with the i th unwanted transmitter;

b_i : receiving antenna discrimination (dB);

e_i : field strength of the unwanted transmitter (dB(μ V/m)).

By using a directional base-station antenna, the interfering field strength can be reduced by at least 10 dB.

It now remains to demonstrate what this means in terms of channel usage.

The simplified multiplication method for the assessment of multiple interference (see Report 945) gives the following formula for calculating usable field strength (field strength to be protected):

$$p_c = 0.5 = \prod_{i=1}^n L(E_u - F_i)$$

These terms are defined in Report 945, but the following should be noted:

$$\begin{aligned} F_i &\equiv E_{si} \text{ (in Report 945)} \\ F_i &= P_i + E_{ni}(50, T) + A_i + B_i + K_i \end{aligned} \quad (1)$$

where:

K_i : directional antenna correction factor associated with the i th unwanted transmitter.

Now, if k interfering fields are introduced such that:

$$F_{(n+1)} = F_{(n+2)} = F_{(n+3)} = \dots F_{(n+k)} = F'$$

and the new value of protected field strength (E_u) = E_p , then:

$$\begin{aligned} p_c &= \prod_{i=1}^{n+k} L(E_p - F_i) \\ &= \left[\prod_{i=1}^n L(E_p - F_i) \right] \times \left[L(E_p - F_{(n+1)}) \times L(E_p - F_{(n+2)}) \times \right. \\ &\quad \left. \times L(E_p - F_{(n+3)}) \times \dots \times L(E_p - F_{(n+k)}) \right] \\ &= \left[\prod_{i=1}^n L(E_p - F_i) \right] \times [L(E_p - F')]^k \end{aligned}$$

therefore:

$$k \log_e [L(E_p - F')] = \log_e \left[\frac{p_c}{\prod_{i=1}^n L(E_p - F_i)} \right] \quad (2)$$

$$\text{Note.} - L(E_p - F_i) = 0.5 + 0.5 \varphi \left(\frac{E_p - F_i}{8.3 \sqrt{2}} \right) \quad (\text{see Report 945})$$

$$\text{and} \quad \varphi(x) = \frac{2}{\sqrt{2\pi}} \int_0^x [\exp(-t^2/2)] dt$$

In this example E_p has been taken to be 70 dB(μ V/m) and F_i to be 60 dB(μ V/m).

$$k \log_e [L(70_p - F')] = \log_e \left[\frac{0.5}{L(70 - 60)} \right]$$

$$= -0.4736002$$

Therefore, various values of interfering field strength, F' , can be equated to k extra interfering base-station transmitters, each giving rise to an interfering field strength F' (F' being calculated using equation (1)) and operating simultaneously with the re-engineered base station giving rise to the interfering field strength F_i .

TABLE II - Number of base-station transmitters giving rise to an interfering field strength F'

Interfering field strength F' (dB(μ V/m))	No. of additional base stations k $\left[k = \frac{-0.4736002}{\log_e L(70 - F')} \right]$
30	1446
40	89
50	10
60	2

The value of k can be increased still further by careful choice of base-station transmit frequency and keeping the antenna heights as low as possible.

REPORT 1024

PERSONAL RADIO SYSTEM

(Question 71/8)

(1986)

1. Introduction

This Report relates to the personal radio system used in Japan as an example of multi-channel access technique without a central controller. Intended as an initial technical response to Question 71/8, this Report introduces the basic characteristics, connecting procedure, and receiver input level versus connection reliability.

2. Basic characteristics

- 2.1 *Frequency*: 903.0125 to 904.9875 MHz
- 2.2 *Channel separation*: 25 kHz
- 2.3 *Number of channels*: 80 channels (one control channel and 79 traffic channels)
- 2.4 *Class of emission*: F2D: control channel
F3E: traffic channels
- 2.5 *Type of operation*: Simplex
- 2.6 *RF power output*: 5 W

3. Connecting procedure

3.1 Configuration of control signal

An automatic transmitter identification system (ATIS) is included in a read-only memory (ROM) obtained from the licensing authority. The ROM is required for the operation of the personal radio system (PRS) transceivers.

Code configuration for ATIS and circuit linkage

Bit synchronization:	50 bits 101010 ...
Word synchronization:	15 bits 111011001010000
Selective calling number:	20 bits, 5 BCD bits
Channel number:	8 bits, binary
Reserved bits:	4 bits, 0000
ATIS code:	48 bits binary: identification codes (for more than ten million stations, licence issue date and scrambling)
Length of Hagelbarger code:	$2 \times \text{data bit length} + 12 = 172 \text{ bits}$
Total:	$172 + 65 + 237 \text{ bits (197.5 ms)}$
Code type:	NRZ
Bit rate:	1200 bit/s
Modulation method:	MSK 1200 Hz mark 1800 Hz space

3.2 Receiver (RX) input versus connection reliability

In Fig. 1, an example of receiver input level versus bit error ratio (BER) is shown. Line C of $\text{BER} = 10^{-2}$ corresponds to 90% connection reliability for the personal radio system.

3.3 Flow chart

Figure 2 is a simplified communication procedure flow chart for the personal radio system. The procedure of connection is as follows. All the radios in the system are in the stand-by state on the control channel. The calling station looks for and finds an idle traffic channel and stores the channel number in its memory. Then it emits the control signal on the control channel. Those radios whose selective call number coincides with the control signal transfer to the specified traffic channel and enter into conversation. The call sign consisting of the ATIS code is automatically transmitted before the start of conversation, every 60 s through conversation and at the end of the conversation. A selective call number is specified before transmission. On the other hand, up to two selective call numbers can be set for receiving.

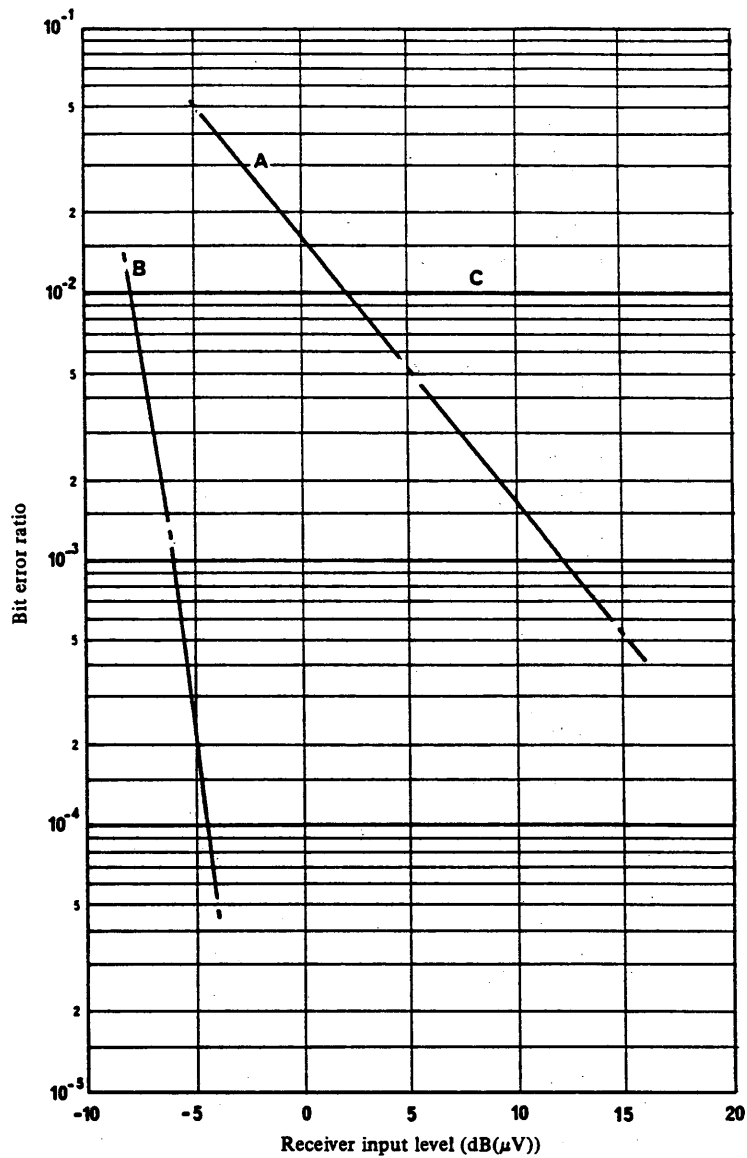


FIGURE 1 – Receiver input level versus bit error ratio performance

Curves A: Rayleigh fading

– Centre frequency: 903.8875 MHz

– Fading frequency: 20 Hz

B: no fading

C: connection reliability: 90% or more

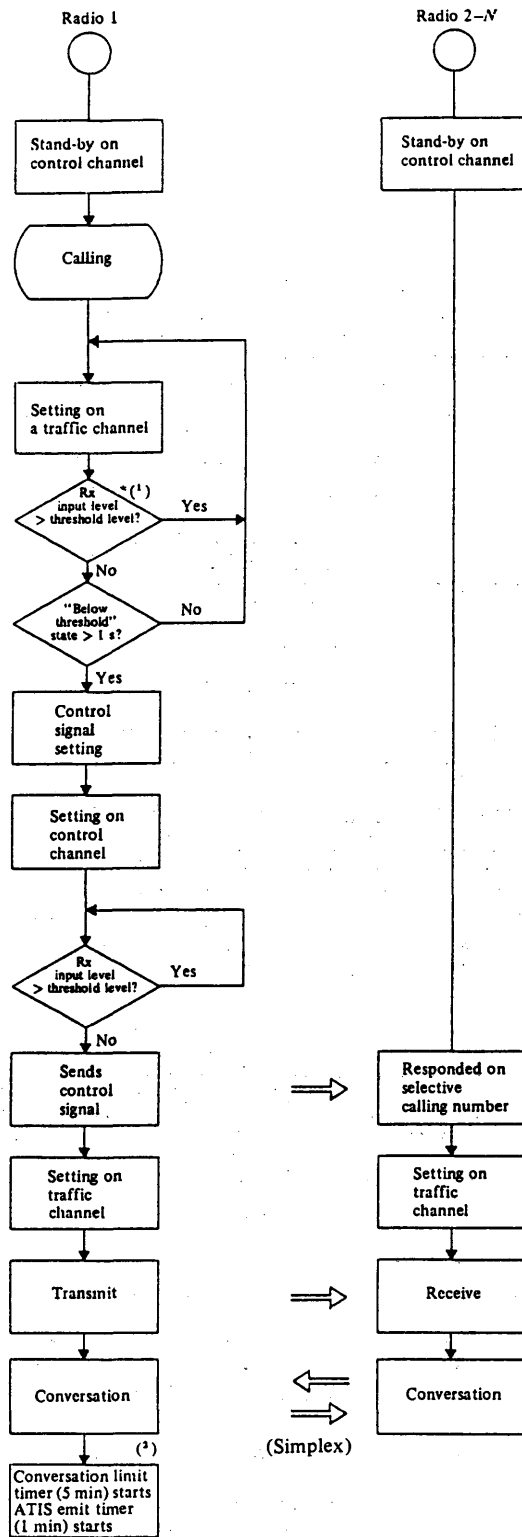


FIGURE 2 – Simplified communication procedure flow chart

(¹) Threshold level.

The standard threshold carrier level for the receiver is set at 1 μ V (open voltage).

(²) In consideration of traffic congestion, a conversation timer is provided.

4. Conclusion

Without a central controller, a personal radio system has been implemented as an economical communication system of relatively high reliability. Thus, this system is popular in use.

REPORT 1025-1

TECHNICAL AND OPERATING CHARACTERISTICS OF CORDLESS TELEPHONES

(Question 71/8)

(1986)-1990)

1. Introduction

In recent times, cordless telephones which can be connected into the public switched telephone network (PSTN) and have a small service area limited to a few hundred metres, have gained widespread popularity. To meet the growing demand for these cordless telephones, it would be advantageous to develop a system using multi-channel access techniques that does not rely on centralized traffic channel selection and provides efficient frequency utilization.

Part A of this Report deals with the general principles of cordless telephone systems using multi-channel access techniques, in particular, with the basic objective and technical characteristics which are important for their operation.

The major characteristics of some cordless telephone systems are given in Part B of this Report together with a brief description and other aspects of system operation.

PART A

1. Objectives

The basic objectives of the application are:

- that the radio spectrum be used efficiently;
- that a system of high subscriber capacity be realized;
- that simple and miniature circuits be used to ensure that the weight and size of the equipment are compatible with ordinary telephones and that it can be provided economically;
- that the system provide good quality for public communication and that a flexible system of operation can be provided that does not require complicated frequency management;
- that the system provide as far as possible normal telephone features;
- that the system be provided security of call charges.

2. Technical and operational aspects

A cordless telephone system consists of two parts:

- a fixed part which is connected to the ordinary telephone network; and
- one or more portable sets.

The simplest version, called "cordless telephone" consists of one portable set and one fixed part connected to a subscriber line. More complex versions may encompass both public base stations ("telepoints") for access to the PSTN/ISDN and wireless PABXs.

Each part uses multi-channel-access techniques and can individually perform the following operations:

- search for idle channels;
- set up speech paths using the selected channel;
- check identification codes in the signals between the fixed part and the portable set in order to ensure that only associated units will lock to each other.

As a consequence, the following operation and management procedures can apply:

- in system operation, it is only necessary to manage the number of cordless telephone systems within the area determined by the distance at which interference occurs, while taking into account the subscriber capacity which is determined by the number of — channels and the channel traffic;
- flexible operation can be provided, so as to permit the use of a greater number of portable sets than the subscriber capacity per zone, if the number of cordless telephones in use in the adjacent zones is small.

PART B

SYSTEMS BEING USED OR PLANNED IN THE NEAR FUTURE

1. Systems features

Cordless telephone systems employing frequency and/or time division multiple access techniques (FDMA/TDMA) have been developed in many countries.

A summary of major system specifications is given in Table I.

In Annexes I to IV the main features of four systems are described.

2. Conclusion

The technical basis and system features of cordless telephone systems using multi-channel access techniques which do not rely on a central system controller have been presented. The techniques described here would provide efficient frequency utilization, high subscriber capacity and flexible system operation.

TABLE I - System characteristics

	System 1 ⁽¹⁾	System 2 ⁽²⁾ [CEPT, 1983]	System 3 ⁽³⁾ [BSI, 1987; DTI, 1987; DTI, 1989]	System 4 ⁽⁴⁾ [CCIR, 1986-90 ESPA, 1987] [STA, 1989]	System 5 ⁽⁵⁾ [ETSI, 1987- 1991]
Class of emission	F1D, F2(A, B, C, D, N, X), F3E	F3E or G3E	F1EJT or G1EJT	F1W and F7W	F1W and F7W
Transmit frequency band (MHz):				Time division duplex:	Multi-carrier time division duplex:
- fixed part	380.2 - 381.3	959 - 960	864.1 - 868.1	862-866	1880-1900
- portable set	253.9 - 255.0	914 - 915	864.1 - 868.1	862-866	1880-1900
Radiofrequency channel spacing (kHz)	12.5	25	100	1000 (TDMA RF channels)	1728 (TDMA RF channels)
Number of speech channels	87 (adding two control channels)	40	40	8 or 16 (duplex time-slots per carrier)	12 (duplex time-slots per carrier)
Transmitted power, e.r.p. (mW):				Peak power over time-slot max. 100	Peak power over time-slot max. 250
- fixed part	max. 10	max. 10	max. 10		
	(Antenna input power)				
- portable set	max. 10	max. 10	max. 10	max. 100, avg. < 6 (6)	max. 250
Typical service range (m):					
- indoor	50	50	40	40	30
- outdoor	100	200	200	200	200

TABLE I (continued)

	System 1 ⁽¹⁾	System 2 ⁽²⁾ [CEPT, 1983]	System 3 ⁽³⁾ [BSI, 1987; DTI, 1987; DTI, 1989]	System 4 ⁽⁴⁾ [CCIR, 1986-90 ESPA, 1987] [STA, 1989]	System 5 ⁽⁵⁾ [ETSI, 1987- 1991]
Voice signals					
- type of modulation - processing	analogue	angular syllabic compandor CCITT Rec. 162 (suggested)	digital ADPCM	digital ADPCM or CVSDM	GMSK ADPCM or CVSDM
Audio signal-to-noise ratio (dB)	min. SINAD 20 dB(7)	(9)	(9)	(8)	(8)
Identification code	3×10^7 combinations	1×10^4 combinations	1.5×10^6 combinations	$> 10^7$ combinations	$> 10^7$ combinations
Minimum acceptable C/I ratio	(9)	(9)	(9)	(9)	(9)

- 1) System 1 is in use in Japan.
- 2) System 2 is used by several European countries.
- 3) System 3 is used in the United Kingdom and Finland.
- 4) System 4 is to be used in Sweden.
- 5) System 5 is the Digital European Cordless Telecommunications (DECT) system.
- 6) This average power assumes eight speech channels (duplex time-slots); for 16 speech channels (duplex time-slots), the power reduces proportionately.
- 7) Suggested value using an IEC Pub. 489-3 standard input-signal level of 10 μ V.
- 8) Mean opinion score (MOS), when tested subjectively, shall not fall below the MOS of 32 kbit/s ADPCM as specified in CCITT Rec. G.721.
- 9) Values are required for this parameter; these may be dependent on the measurement method employed.

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CCIR Documents

[1986-90]: 8/67 Sweden

ANNEX I

GENERAL DESCRIPTION OF SYSTEM 1

1. System configuration [RCR STANDARD, 1988]

The basic configuration of a cordless telephone system 1 employing the analogue multi-channel access technique is shown in Fig. 1.

This system does not use a central controller. In the system, two dedicated control channels are assigned.

Each cordless telephone itself performs the functions of searching for idle channels and setting up a speech path over the selected channel.

Fig. 2 shows the configurations of multi-function cordless telephone [Hattori *et al.*, 1988].

The following types are typical examples.

- 1) Multi-terminal cordless telephone
- 2) Cordless key telephone
- 3) Multi-zone cordless telephone

2. Operational characteristics

2.1 Identification code

- 1) Only when its own identification code coincides between the fixed part and the portable set in the control channel, the fixed part and portable set shall switch to a speech channel. The length of the identification code shall be 25 bits.
- 2) Modulation methods and bit rates for an identification code are as follows:
Subcarrier MSK (Fast FSK): 1200, 2400, bps
Direct modulation: 600, 1200, 2400, 4800, bps

2.2 Idle channel decision;

Carrier sense shall be performed immediately before transmission and if more than $2\mu\text{V}$ is sensed, radio signal shall not be emitted.

2.3 Occupied time of control channels

Considering common use of dedicated control channels and easy introduction of sleep mode in portable sets, for originating calls from portable sets, occupied time shall be 1.3 second or less. For originating calls from fixed parts, occupied time shall be 4 seconds or less.

2.4 Transmitter power-off on disconnection

When disconnection is initiated, or when a radio signal is not received on any speech channel, the transmitter shall be turned off automatically.

2.5 Transmitter power off in case of failure

When a failure in the radio equipment causing continuous transmitter power in the control channel is found, the transmitter shall be automatically turned off within 60 seconds.

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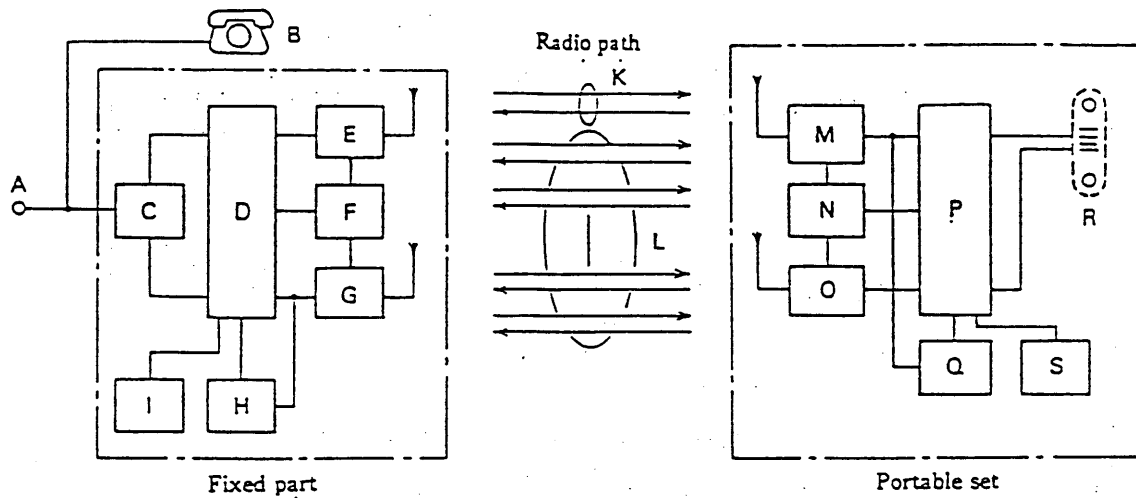


FIGURE 1 — Basic block diagram of a cordless telephone system using multi-channel accessing techniques

A : subscriber line	K : control channels
B : ordinary telephone	L : speech channels
C : hybrid	M : receiver
D : signal controller	N : synthesizer
E : transmitter	O : transmitter
F : synthesizer	P : signal controller
G : receiver	Q : idle channel detector
H : idle channel detector	R : handset
I : identification code ROM	S : identification code ROM

ANNEX II

GENERAL DESCRIPTION OF SYSTEM 2

1. System configuration

The configuration of an analogue cordless telephone of system 2 employing the multiple channel access technique is shown in Fig. 3.

The system configuration which has been adopted, does not use a control channel to set up the RF connection between fixed and portable sets. Occupancy of an RF channel may be initiated by the fixed set or the portable set by following the same procedure.

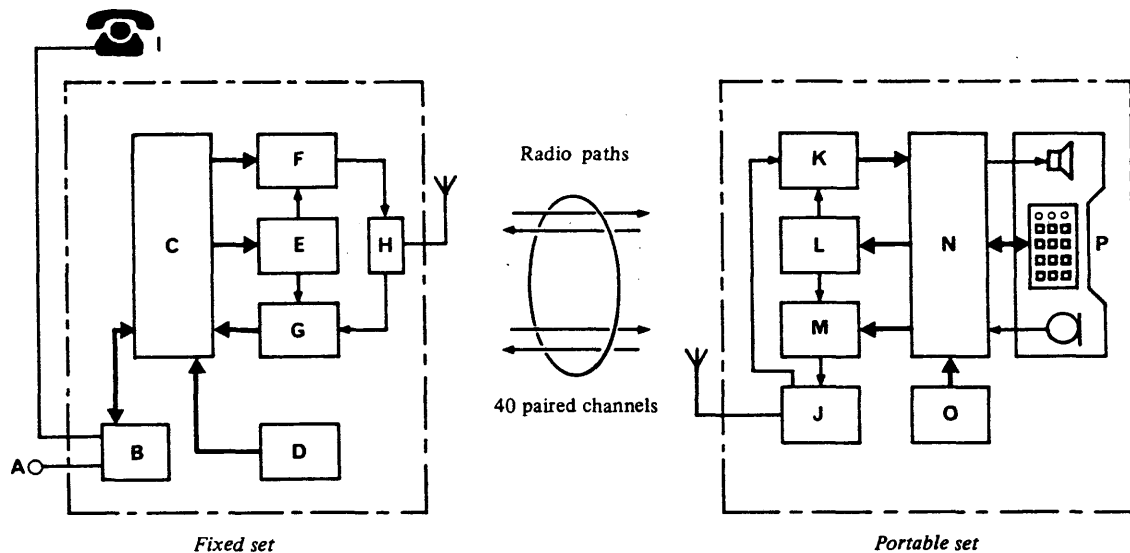


FIGURE 3 – System configuration

A : subscriber line
B : line interface
C : signal controller and CPU
D : identification code PROM
E : frequency synthesizer
F : transmitter
G : receiver
H : duplexer
I : associated telephone

J : duplexer
K : receiver
L : synthesizer
M : transmitter
N : signal controller and CPU
O : identification code PROM
P : handset

2. Call setting-up procedures

The procedures for setting-up a call are given in Fig. 4.

2.1 Waiting mode

In this mode, the receiver of each part of the cordless telephone set is constantly scanning the available RF channels, searching for a signal which contains its matching identification code.

2.2 *Call originating mode*

When the need for a radio-frequency channel arises in either of the two parts of a cordless telephone set, that part searches for an idle duplex channel by sensing the field strength on that channel.

On the idle channel, the initiating part starts transmitting its identification code. Upon detection of this code, the receiver stops scanning and initiates its transmitter to occupy the corresponding return frequency of the duplex channel and to transmit its identification code to the initiating part. As soon as the receiver of the initiating part detects its matching identification code on the return frequency of the selected duplex channel, the initiating transmitter stops transmitting the identification signals and the RF channel becomes available for the transmission of dialling tones and speech.

2.3 *Conversation mode*

To ensure continued locking during a call, the identification procedure is periodically repeated.

2.4 *Call terminating mode*

When the RF connection is to be terminated, the part of the cordless telephone set which initiates the termination procedure transmits a proper termination message including the identification code. The RF circuit is immediately disconnected and the cordless telephone set returns to the idle condition.

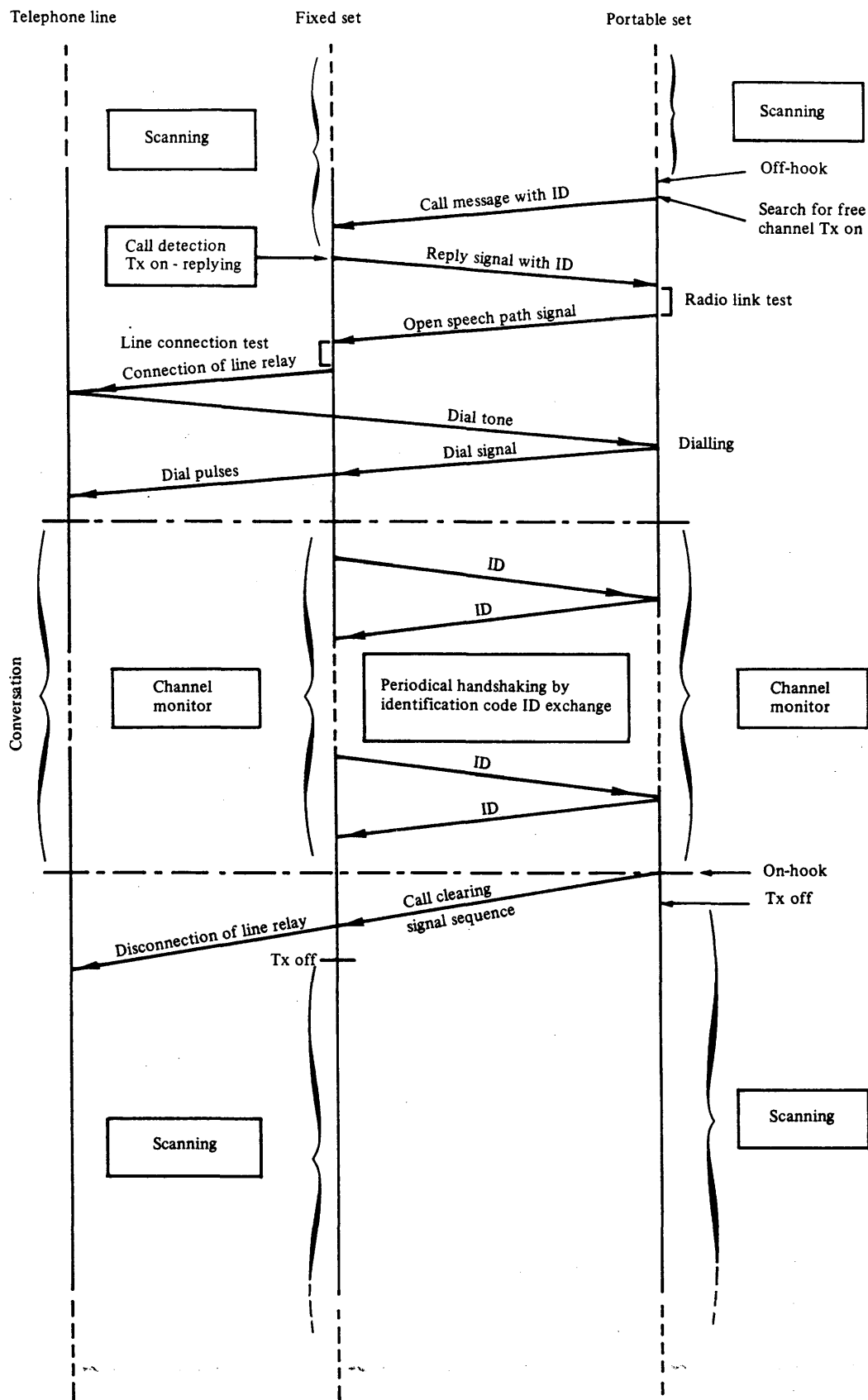


FIGURE 4a — Cordless originated call

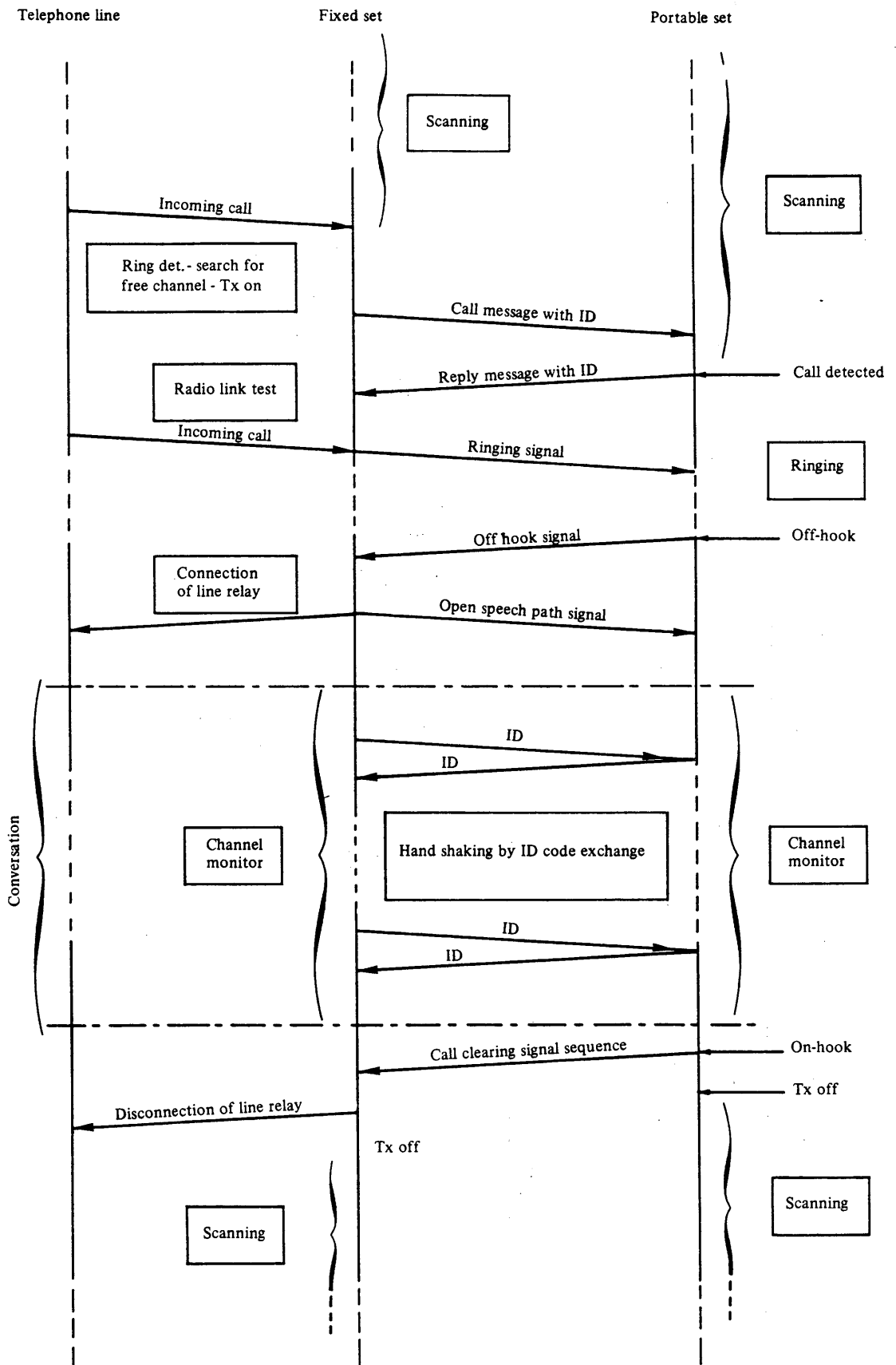


FIGURE 4b — Cordless terminating call

ANNEX III

GENERAL DESCRIPTION OF SYSTEM 3

1. System operation

The digital cordless telephone of system 3 operates with digitally modulated, single channel per carrier equipment using time-division duplex operation and frequency-division multiple access (TDD/FDMA).

The system configuration does not require the use of a control channel to set up the RF link between the fixed and portable sets. Selection of a suitable RF channel may be initiated by either the fixed or portable set, the same procedure being followed in each case.

The basic characteristics of this type of digital transmission link are applicable to cordless telephones operated in residential, small business and telepoint services. In telepoint operation the portable sets are able to initiate calls within 100 - 200m range of a public base station.

This TDD/FDMA approach is well matched to the spectrum availability, and network requirements. The 100 kHz channelling has no need for dispersion equalization, permits easing of frequency tolerances towards 1 part in 10^5 and allows the use of relatively simple filters. There is no necessity for an RF duplexer, less carrier-frequency sources are required and its moderate delay raises no echo control or absolute delay issues. With these attributes, the RF circuitry is simplified and the balance of implementation complexity shifts to baseband which is amenable to very large scale integration.

The above presentation of the digital cordless telephone is as a single base-handset combination. A wide range of situations can be accommodated with a capability of continuous growth.

For example:

- individual, independent, cordless extensions to a pbx;
- local concentrations of cordless base units in nests;
- central pbx-based multi-channel unit using a leaky feeder or discrete, distributed, antennas.

The last two clearly require synchronization between transmit and receive phases of the TDD signals within the nested unit, which is easily arranged. Another consideration is the need to exercise some control of intermodulation between multiple transmissions during transmission and reception. Although the need for intermodulation control is inherent in all such FDMA schemes and special measures may be required in combining RF channels [Johnson *et al.*, 1987], the effects can be contained without incurring significant cost penalties. The low power levels involved also make this aspect somewhat easier. Since pbx applications involve the provision of the pbx interface and control circuitry, any potential intermodulation control can also be incorporated.

In summary, for the digital transmission link there is a very high degree of common circuitry between handset and base hardware and a high level of commonality across all applications (including telepoint).

A common air interface standard has been developed [DTI, 1989] that assures interworking of all portable sets and base stations. Telepoint networks using System 3 will be required to implement this standard. This will allow roaming between the networks, so that any portable set will be able to be used with any network for which the user is a registered subscriber.

REFERENCES

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JOHNSON, A.K. and MYER E. [June 1987] - Linear Amplifier Combiner. IEEE Vehicle Technology Conference, pages 421-423.

ANNEX IV

GENERAL DESCRIPTION OF SYSTEM 4

1. System configuration

Several configurations of digital cordless telephones of system 4 employing the time division multiple access technique (TDMA), are shown in Figure 5. The multiple base station case provides a handover facility.

With one single radio transceiver, each fixed station (FS) can simultaneously communicate on all timeslots that for each moment are allocated to that base station.

2. Call set-up procedure

The system configuration uses a timeslot allocation which is dynamic and decentralized to each base station and each portable. When a traffic channel (timeslot) is wanted, the portable station (PS) communicates with the fixed station (FS) having the highest field strength, and channel is chosen with the least interference at the position of both the portable and the fixed station. For special services, it is possible to allocate temporarily more than one timeslot to a portable station.

3. Interference limited capacity

For system 4 the range can be C/I limited rather than C/N limited. Thus the capacity in an operational area can be increased by denser and denser re-use by installing the fixed stations closer and closer.

4. Dynamic channel allocation and handover

Since all the stations (fixed and portable) have continuous information of the status on all slots due to the TDMA mode of operation, the dynamic channel allocation can be made very efficient and fast. Also handover within and between fixed stations can be made efficient and fast. Furthermore, simulations have shown that this system is stable under heavy traffic conditions.

5. TDMA frame structure

Adjacent systems with differently specified TDMA frames and timeslots can co-exist on the same frequency band as long as the frame cycle times are equal. Timeslot synchronization is not needed between systems, provided that the frame cycle time is reasonably constant (± 5 ppm). Antenna selection at the fixed station provides diversity gain for both directions.

6. Applicable services

The system concept is applicable to large office, small office, telepoint and residential voice and data services.

ANNEX V

General description of system 5

1. System Configuration

System 5 employs the TDMA-time division duplex technique in such a way, that various configurations can share the same spectrum resources.

An overview of the system 5 family is given in fig. 5. One configuration does not have to be synchronized with a neighbouring one, belonging to the same family, in order to operate in the same frequency band, although such a synchronization would increase the capacity of both systems. Each fixed station can communicate simultaneously on all 12 duplex time slots and on any of the carriers, with one single radio transceiver. Handover facilities are provided in multi-base station configurations.

2. Services

The system concept provides for voice and data services in large office environment (wireless PABX 100% replacement of wired telephones by cordless terminals), in small offices, at telepoints and in residential area's. It allows equipment to be connected to PSTN's and for ISDN's.

3. Call set-up procedure

The system configuration uses a timeslot allocation which is dynamic and decentralized to each base station and each portable. When a traffic channel (timeslot) is wanted, the portable station (PS) communicates with the fixed station (FS) having the highest field strength, and the channel is chosen with the least interference at the position of both the portable and the fixed station. In case of incoming calls, the base station chooses a free channel and uses its handshake to establish a communication link with the portable, preferably using the choosen channel, if it is free.

This method of dynamic channel allocation is efficient and fast because all (fixed and portable) stations have continuous information of the status on all slots due to the TDMA mode of operation.

The system concept includes the possibility to allocate temporarily more than one time slot to a portable station, in order to allow for special services, requiring a high capacity communication channel.

4. The radio interface

According to the OSI model, a layered structure is used to describe the radio interface

4.1 Layer 1 (physical layer)

In this layer, the information channel and the control channel are modelled into the radio channel at the transmitter and vice versa at the receiver end.

Radio signals are modulated and demodulated onto one or several carrier-frequencies with a bitstream of 1152 kbit/s. The modulation is Gaussian Minimum Shift Keying (GMSK) with a relative bandwidth $BT = 0.5$. The R.F. channel spacing is 1.728 MHz.

The user's logical channels are multiplexed onto the carrier frequency or frequencies using a Multi Carrier Time Division Multiple Access/Time Division Duplex (MC TDMA/TDD) scheme. An example of the TDMA frame structure is given in fig. 6, where as an example of multiplexing the information and control data in one burst is shown in fig. 7. For ISDN applications, more adequate schemes are available.

4.2 Layer 2 (data link layer)

This layer has to secure reliable datacommunication over the logical channels, as provided by layer one.

It provides for error protection, quality management and a hand-over mechanism. It also supports layer 2 functions of the network to which the system extends.

4.3 Layer 3 (network layer)

The system's layer 3 supports most of the procedures, defined in CCITT Rec. I.450.ff, for the wired D-channel, as well as those necessary in cordless PABX's.

Authentication procedures, including securing access to telepoints and residential fixed stations, are foreseen.

5. Capacity

Due to the fact, that the coverage area of one base station is interference limited, rather than noise limited, increasing the density of base stations will result in a higher capacity of the system.

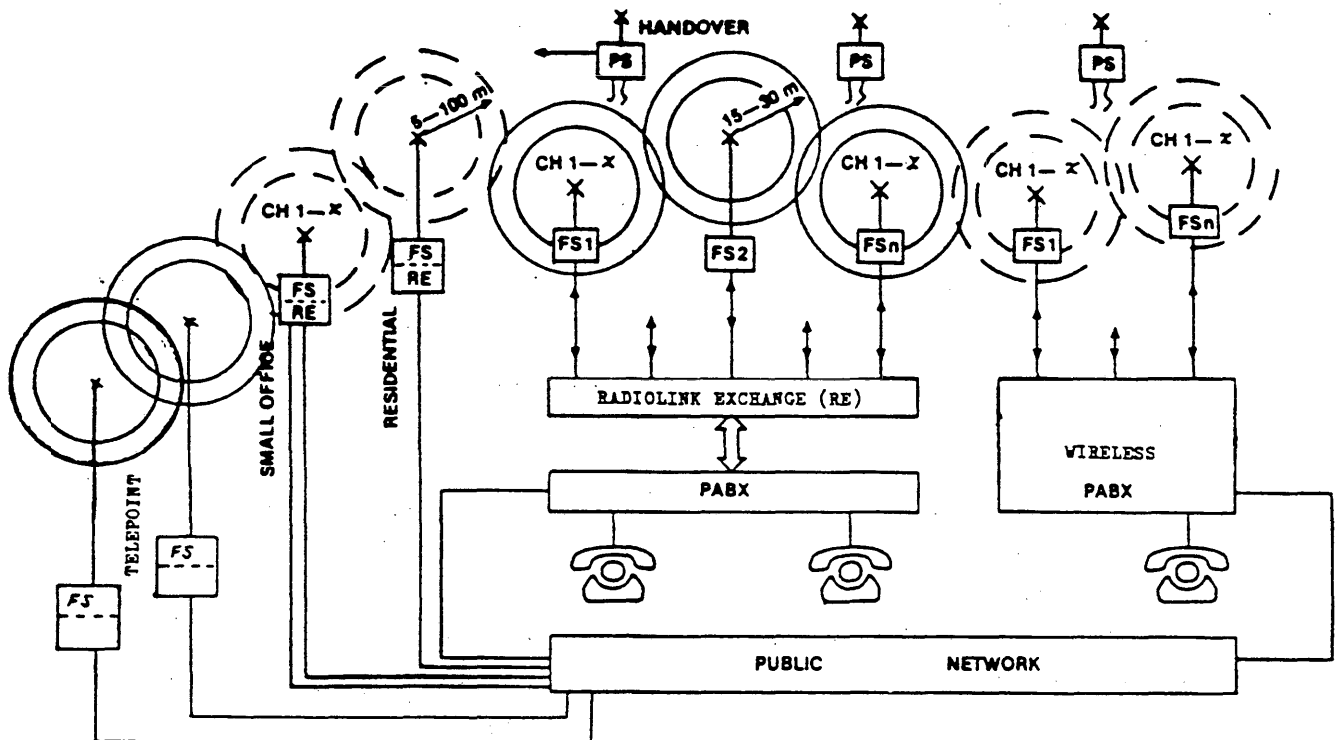


FIGURE 5 - System 4 and System 5 families

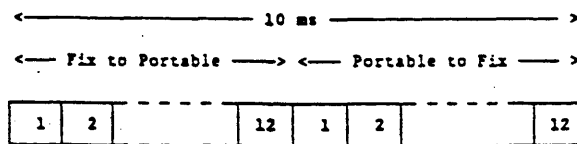


FIGURE 6 - TDMA frame structure example

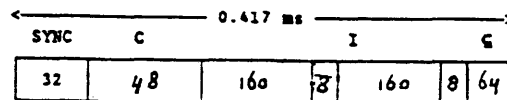


FIGURE 7 - Burst structure

SECTION 8L: AMATEUR SERVICE; AMATEUR-SATELLITE SERVICE

REPORT 1154*

TECHNIQUES AND FREQUENCY USAGE IN THE AMATEUR AND
AMATEUR-SATELLITE SERVICES

(Question 48/8)

(1990)

1. This Report describes, in four appendices, the various techniques employed and uses made of allocated frequencies in the amateur and amateur-satellite services. The technical characteristics of equipment in current use or planned for early application are also described.
2. Appendix 1 describes technical investigations in the amateur service and amateur-satellite service.
3. Appendix 2 describes frequency usage in the amateur service.
4. Appendix 3 is concerned with techniques and frequency usage in the amateur-satellite service.
5. Appendix 4 deals with amateur-satellite station technical characteristics and plans for future systems.
6. Appropriate references are included with these appendices.

* This Report should be brought to the attention of Study Groups 5 and 6.

Appendix 1

Technical investigations in the amateur service
and amateur-satellite service

1. Introduction

This appendix describes examples of technical investigations undertaken by the amateur and amateur-satellite services. These investigations may result in new or improved spectrum conservation techniques and additional knowledge concerning spectrum sharing among telecommunications services. One motivation is a desire to develop reliable low-cost communications systems and techniques.

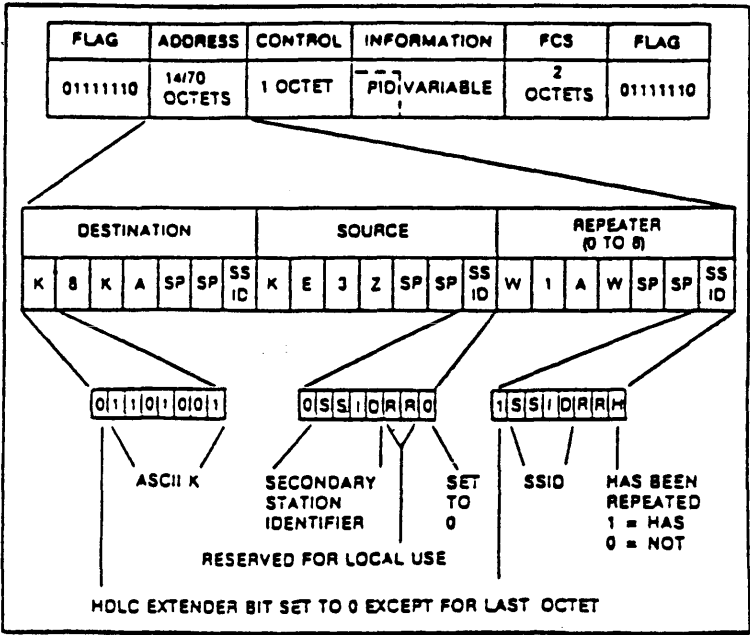
2. Packet radio technology

2.1 General approach

Early amateur packet radio development was conducted in Canada in the late 1970s and in the United States during the early 1980s. Amateur technical investigations are underway in other countries, notably the Federal Republic of Germany, Japan, Sweden, and the United Kingdom [ARRL, 1981; 1983; 1984; 1985; 1986; 1987].

2.2 AX.25 data link layer protocol

The AX.25 amateur packet-radio link-layer protocol [Fox, 1984] is a radio version of the data link layer of the protocol defined in CCITT Recommendation X.25. It differs from CCITT Recommendation X.25 primarily in expansion of the address field to accommodate radio call signs as addresses for the called station, the calling station and optional intermediary data-link-level repeater stations. AX.25 has gained wide acceptance and is now in operational use in numerous countries. It has been adopted by several governmental agencies and commercial organizations, including some developing nations. The AX.25 frame structure is shown below:



Abbreviations used:

ASCII	American national standard code for information interchange
HDLC	high-level data link control
FCS	frame check sequence
PID	protocol identifier
SSID	secondary station identifier

Amateur technical investigations are underway to revise the AX.25 protocol to accommodate call signs varying in length, such as JY1 to GB1AAA/KL7, and to minimize transmission impairments via fading high-frequency and meteor-scatter media.

2.3 Higher layer protocols

Intensive field trials of candidate datagram and virtual-circuit network (packet) and transport layer protocols are underway. Protocols for message-handling systems are being developed.

2.4 Equipment development

Amateurs have developed inexpensive packet assembler/disassemblers which are now being manufactured in several countries. Work is progressing on network switching controllers.

Several designs for inexpensive modems exist and more are being developed for spectrally efficient modulation techniques, such as minimum-shift keying (MSK) at speeds from 1.2 to 56 kBd.

3. Multiple protocol communications processors

Amateurs are currently experimenting with microprocessor based equipment capable of supporting AX.25 packet radio, ITA2 radioteleprinter, AMTOR (a subset of CCIR Recommendation 625 (SITOR)), and weather facsimile.

4. Document facsimile

Amateurs, principally in Japan, are developing the interfacing of CCITT group II and III document facsimile machines to amateur HF and VHF radio equipment. Investigations include personal computer emulation of facsimile transceivers.

5. Bandwidth expansion techniques

After several years of experimentation on special temporary authority, amateur regulations in the United States were amended to permit limited spread-spectrum communications on frequencies above 420 MHz. Objectives of this experimentation are development of inexpensive equipment and operational experience with civil applications of this frequency-utilization technique.

6. Signal processing techniques

Amateurs are investigating digital signal processing (DSP) for voice, data and other applications, using low-cost printed-circuit boards which plug into personal computers.

REFERENCES

ARRL [1981; 1983; 1984; 1985; 1986; 1987] - 1st - 6th ARRL Amateur Radio Computer Networking Conferences, ARRL, Newington, CT, 06111, USA.

FOX [1984] - AX.25 Amateur Packet-Radio Link-Layer Protocol, ARRL, Newington, CT, 06111, USA

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Appendix 2

Frequency usage in the amateur service

1. Modes of emission to be accommodated

In considering the ways in which the various frequency bands allocated to the service are employed by stations and groups, it is convenient to consider the information transfer mode as composed of families, one grouping being:

1.1 Narrow-bandwidth systems, including Morse code, teleprinter and very slow speed data, telemetry and telecommand.

1.1.1 Morse telegraphy (A1A class of emission) owes its popularity to the effectiveness under conditions of low power, weak signals, or transmission impairments; its inherent capability to bridge the language barrier between operators who do not speak the same language; and its narrow-bandwidth characteristics, which permit a very high density of stations in a heavily occupied band.

1.2 Voice-bandwidth systems, including single sideband, narrow-band frequency modulation, data, telemetry, telecommand, slow-scan television and facsimile systems using essentially the bandwidth of a voice signal.

1.3 Extended-bandwidth systems, used for all information transfer modes operating at low multiples of voice-bandwidths, primarily single-channel but with multiplexing coming into use.

1.4 Wide-bandwidth systems, including television, data, pulse and spread-spectrum systems.

2. Frequency usage in the HF bands (1.7 to 30 MHz)

2.1 Because of the relatively low-power levels typically used by amateurs, absorption could be a significant barrier to communication if suitable frequency bands were not available. The essential consideration in determining the bands for the amateur service was that there be allocations at increments of approximately 3 to 4 MHz throughout the HF spectrum. Such increments take into account the rapid variations of the MUF.

2.2 A band around 2 000 kHz provides frequencies the propagation characteristics of which allow short range communications during the daytime hours and medium and long range communications during the night-time hours. This band is particularly useful during sunspot minima, when the MUF is below most of the amateur allocations in band 7 (3 to 30 MHz).

2.3 The amateur service is presently allocated spectrum in the 3.5, 7, 10, 14, 18, 21, 25 and 28 MHz bands. These bands permit working close to the MUF at all times.

3. Frequency usage in the VHF bands allocated to the amateur service

3.1 Bands around 50 MHz

Frequencies around 50 MHz are used for local amateur communication on a round-the-clock basis. Tropospheric scatter and sky-wave propagation (principally sporadic-E and occasional F-layer propagation at sunspot maxima) are used for longer distances. Meteor scatter has been used for Morse code and voice communications primarily during meteor showers. However, newer data communications (packet radio) systems operating at speeds above 1,200 bauds promise to make meteor scatter a routine propagation mode for distances up to 2,000 km.

3.2 144 to 148 MHz

The relatively low variability makes this band technically suitable for all information transfer modes in the amateur service. It is very useful for local, regional and international communication, experimentation and self-training. The balance between natural and equipment noise makes this band suitable for Earth-Moon-Earth communications using narrow-bandwidth techniques. Transequatorial (TE) paths have been found useful over long distances. The discovery of TE propagation can be attributed in part to the increased activity in this band. Sporadic-E investigations have been documented, [Canivenc, 1988].

3.3 220 MHz

Sharing considerations reduce opportunities for amateur utilization of this band, but it is nevertheless used for technical investigation, and for local intercommunication.

During 1987, amateur stations first observed reciprocal sporadic-E propagation [Owen, 1987]. Further investigations of this phenomenon are underway.

4. Frequency usage in UHF and higher bands allocated to the amateur service

4.1 Bands around 430 MHz

Factors relating to equipment limitations and to "percentage of available bandwidth" make these the lowest available bands for local wideband communication and experimentation. The small propagation variability makes these bands operationally suitable for local functions requiring communication reliability (e.g., data transmission, repeater control and on-scene communications in emergencies). The lack of ionospheric effects makes these bands technically suitable for Earth-Moon-Earth communication and technical investigation.

4.2 Bands around 915 MHz

Frequency bands around 915 MHz are useful for reliable local communication and technical investigation including wideband techniques. The sharing with other services and presence of industrial, scientific and medical emissions centered on 915 MHz provide an operational laboratory for study of electromagnetic compatibility techniques.

4.3 Bands around 1 240 MHz and above

Many serious investigations are proceeding in these bands. Access on a secondary basis to bands allocated to the radiolocation service permits operation and investigation in these bands. Interference to the radiolocation service is avoided by operational practices in the amateur service as well as by geographical and frequency separation. There is very significant technical investigation at microwave frequencies, especially around 10.5 GHz, including narrow-band communication (SSB) and propagation via precipitation scatter.

5. Frequency usage in the event of natural disasters

This aspect of the amateur service is recognized in Resolution No. 640 of WARC-79, and in footnote No. 510 in Article 8 of the Radio Regulations.

Emergency communication has been provided, for example after the earthquakes in Nicaragua (1972), Guatemala (1976), Italy (1976 and 1980), Romania (1977), Algeria (1980), Mexico (1985), Armenia (1988) and after a dam disaster in India (1979), after the floods in Honduras (1974) and the United States (1977), after a volcano disaster in the United States (1980) and Colombia (1985), as well as the hurricanes in the Caribbean (1980).

REFERENCES

OWEN [October 1987] OEX - ARRL Experimenters' Exchange and AMSAT Satellite Journal, ARRL, Newington, CT, 06111, USA.

CANIVENC [January 1988] [February 1988] [March 1988] "Propagation via la couche E sporadique", Radio-REF 01, 02, 03/1988.

Appendix 3

Techniques and frequency usage in the
amateur-satellite service1. Introduction

1.1 The first radio amateur satellite, OSCAR 1 (Orbital Satellite Carrying Amateur Radio) was launched on 12 December 1961. Three additional spacecraft in the OSCAR series were launched prior to 1971 when the ITU established the amateur-satellite service as a separate entity.

1.2 As of mid 1988, approximately 26 amateur satellites have been launched; 24 have successfully attained orbit. Ten have been authorized by the United States (US), three by Germany (Federal Republic of), one by Japan and 12 by the USSR (see Annex I).

1.3 To date, authorities responsible for the operation of spacecraft in the amateur-satellite service have endeavoured to make these spacecraft available to authorized amateurs in all countries. As a result, more than 20,000 amateurs in more than 150 countries have been able to participate directly in satellite communication.

1.4 Amateur-satellite service operation has also occurred from the space shuttle involving astronaut-radio-amateurs licensed by Germany (Federal Republic of), Netherlands and the United States. Additionally, amateur-satellite service two-way voice contacts were initiated from the USSR MIR space station in late 1988.

1.5 Flight hardware for AMSAT-OSCAR satellites has been provided by groups in Australia, Canada, the United Kingdom, Germany (Federal Republic of), Hungary, Japan, United States and South Africa.

2. Technical investigations and developments

2.1 The design, construction, utilization and control of amateur satellites have provided a focus for technical, scientific and educational developments by the international radio amateur community. This section outlines developments in communications technology and related applications.

Additional information may be found in Annexes II and III.

2.2 Radio amateurs have used satellite transponders to support real-time two-way communication by Morse, SSB voice, amplitude-companded SSB voice, slow-scan TV, FM voice, and several digital modes. Experiments have demonstrated the feasibility of:

- (1) earth gateway stations to allow low-powered, hand-held transceivers to communicate with spacecraft;
- (2) using a satellite-to-satellite relay as part of a real-time communications link; and
- (3) employing mobile terminals in automobiles, ships, and small aircraft to communicate via satellite.

2.3 A number of significant developments are related to high-efficiency linear transponders for space use. These include EER (Envelope Elimination and Restoration) and HELAPS (High-Efficiency Linear Amplification by Parametric Synthesis).

2.4 The feasibility of using linear transponders to support true random and multiple-access communication via satellite has been demonstrated. Downlink monitoring permits the transponder to support a high density of users through time-sharing techniques. Low-earth-orbit amateur spacecraft have carried more than 1,000 message exchanges during a single 20-minute pass.

2.5 Digital store-and-forward technology has been extensively utilized for communication between satellite command stations.

3. Frequency usage

3.1 The factors involved in frequency planning for AMSAT coordinated spacecraft include:

- availability of allocated frequencies;
- suitability (technical considerations such as path loss, power budget, satellite attitude control and antenna configuration, etc.);
- user needs (equipment available to ground stations, transponder bandwidth requirements);
- potential interference (band occupancy by the terrestrial amateur service, use of shared bands (see section 4)).

These factors, and the tradeoffs involved, have been documented [Soifer (1973), Meinzer (1973) and King (1986)].

3.2 Narrow-band communications modes such as SSB have been employed for two-way voice communications despite complications due to the large Doppler shifts associated with low-earth-orbit spacecraft and the perigee of elliptical orbit spacecraft. Transponders designed to accommodate digitally compressed video signs are being developed [King, 1986].

3.3 Despite the use of spectrum efficient techniques the amateur-satellite service is moving to higher frequencies where allocations and band occupancy permit the required bandwidth. See Table I. Secondary allocations have been valuable to the amateur-satellite service in moving upward in frequency.

TABLE I

Satellite transponders: maximum frequency and bandwidth

Spacecraft	OSCAR 3	OSCAR 4	AMSAT OSCAR 6	AMSAT OSCAR 7	AMSAT Phase 3A	AMSAT OSCAR 10	AMSAT OSCAR 13
Launch date	1965	1965	1972	1974	1980	1983	1988
Maximum bandwidth employed on single transponder	40 kHz	10 kHz	100 kHz	100 kHz	200 kHz	800 kHz	250 kHz
Maximum frequency employed on transponder	144 MHz	432 MHz	148 MHz	432 MHz	435 MHz	1270 MHz	2.4 GHz

4. Potential for interference in shared bands

4.1 Satellite OSCARs 6, 7, 8, 9, 10, 11, 12 and 13 have employed communications links (beacons and/or transponders) operating in shared bands. No reports of interference attributable to these spacecraft have ever been received. As a result of conservative design, all spacecraft have low levels of spurious and out-of-band emissions.

4.2 Frequencies considered for a satellite link are carefully surveyed before adoption, using information contained in: documents listing international frequency assignments; observations by radio amateurs around the world; and experience with previous amateur satellites. With a world-wide population of more than one million radio amateurs the amateur-satellite service has a widely distributed group of competent observers capable of obtaining, through direct observation, accurate and immediate band occupancy data. Transponder frequency translation permits location of clear up-link frequencies prior to transmission.

4.3 Telecommand systems enable command stations speedy adjustment of satellite parameters for the resolution of interference problems. (See section 5.1).

5. Telecommand

5.1 To ensure immediate cessation of any interference, Radio Regulation No. 664 requires that amateur satellites must include a command link. Command options and capabilities have increased as new satellites have become operational. Currently, command stations have several of the following options:

- 1) stop all spacecraft RF emissions;
- 2) deactivate specific RF links;
- 3) reduce radiated RF power on specific links;
- 4) change operating schedule of specific links;
- 5) change satellite antenna or attitude to modify coverage area.

5.2 Recent satellites are managed by on-board computers which handle scheduling, telemetry, command decoding, etc. Programming of the computer over the command link permits fewer command stations to issue commands less frequently than was the case with earlier satellites.

5.3 While command stations have never had to respond to an emergency involving interference, they have on several occasions faced other serious satellite emergencies. In each instance they have demonstrated their capability of responding appropriately [Meinzer, 1986].

ANNEX I
OF APPENDIX 3
Amateur-satellite operations: 1961-1987

Satellite Launch date Operating life	Launch agency, Licensing authority	Linear transponders input/output frequency (bandwidth)	Beacon frequencies	Transmitting power (peak)	Apogee height
OSCAR I Dec. 12, 1961 21 days	U.S., U.S.	—	144.983 MHz	0.1 W	430 km
OSCAR II June 2, 1962 19 days	U.S., U.S.	—	145.00 MHz	0.1 W	390 km
OSCAR III March 9, 1965 transponder: 18 days beacon: several months	U.S., U.S.	144.10/145.90 MHz (50 kHz)	145.85 MHz 145.95 MHz (*1)	1.0 W	840 km
OSCAR IV Dec. 21, 1965 85 days	U.S., U.S.	144.100/431.838 MHz (10 kHz)	431.828	3.0 W	33,800 km
Australia-OSCAR 5 Jan. 23, 1970 52 days	U.S.-NASA, U.S.	—	144.05 MHz 29.450 MHz	0.2 W	1,480 km
AMSAT-OSCAR 6 Oct. 15, 1972 4.5 years	U.S.-NASA, U.S.	145.85/29.50 MHz (100 kHz)	29.450 MHz 435.10 MHz	1.5 W	1,480 km
AMSAT-OSCAR 7 Nov. 15, 1974 6.5 years	U.S.-NASA, U.S.	145.90/29.45 MHz (100 kHz) 432.15/145.95 MHz (40 kHz)	29.502 MHz 145.972 MHz 435.100 MHz 2304.1 MHz (*2)	8.0 W	1,480 km
AMSAT-OSCAR 8 March 5, 1978 5.3 years	U.S.-NASA, U.S.	145.90/29.45 MHz (100 kHz) 145.95/435.15 MHz (100 kHz)	29.402 MHz 435.095 MHz	1.5 W	810 km
RS-1, RS-2 Oct. 28, 1978 several months	USSR, USSR	145.89/29.37 MHz (40 kHz) 145.89/29.37 MHz (40 kHz)	29.401 MHz 29.401 MHz	1.5 W 1.5 W	1,700 km 1,700 km

ANNEX I
OF APPENDIX 3 (cont.)

Satellite Launch date Operating life	Launch agency, Licensing authority	Linear transponders, input/output frequency (bandwidth)	Beacon frequencies	Transmitting power [peak]	Apogee height
AMSAT-Phase 3A May 23, 1980 Launch failure	ESA, FRG	435.22/145.90 MHz (180 kHz) 435.22/145.90 MHz (180 kHz)	145.99 MHz 145.81 MHz	50 W	35,800 km [target]
UoSAT-OSCAR 9 Oct. 6, 1981 operating [11/87]	U.S.-NASA, U.S.	—	145.825 MHz 435.025 MHz 7.050, 14.002 MHz 21.002, 29.510 MHz 2.401, 10.47 GHz	0.8 W	540 km
RS-3, RS-4, RS-5,	USSR, USSR	— — 145.93/29.43 MHz (40 kHz) 145.828 MHz (*3)	29.321, 29.401 MHz 29.380, 29.403 MHz 29.331, 29.452 MHz	1.5 W 1.5 W 1.5 W	1690 km 1690 km 1690 km
RS-6, Dec. 17, 1981 RS-7, several years		145.93/29.43 MHz (40 kHz) 145.98/29.48 MHz (40 kHz) 145.836 MHz (*3)	29.411, 29.453 MHz 29.341, 29.501 MHz	1.5 W 1.5 W	1690 km 1690 km
RS-8		145.98/29.48 MHz (40 kHz)	29.481, 29.502 MHz	1.5 W	1690 km
Iakra 2 May 17, 1982 53 days	USSR, USSR	21.25/29.60 MHz (40 kHz)	29.578 MHz	1.0 W	335 km
Iakra 3 Nov. 18, 1982 37 days	USSR, USSR	21.25/29.60 MHz (40 kHz)	29.583 MHz	1.0 W	335 km
AMSAT-OSCAR 10 June 18, 1983 operating [11/87]	ESA, FRG	435.103/145.901 MHz (152 kHz) 1288.45 MHz/436.55 MHz (800 kHz)	145.810 MHz 145.987 MHz 436.02 MHz 436.04 MHz	50 W	5,500 km
UoSAT-OSCAR 11 March 1, 1984 operating [11/87]	U.S.-NASA, U.S.	—	145.828 MHz 435.025 MHz 2401. MHz	1.0 W	690 km
Fuji-OSCAR 12 Aug. 12, 1986 operating [11/87]	NASDA, Japan	145.95/435.85 MHz (100 kHz) 145.85, .87, .89, .81 MHz (*4)	435.787 MHz	2 W	1510 km

ANNEX I
of APPENDIX 3 (cont.)

Satellite Launch date Operating Life	Launch agency, Licensing authority	Linear transponders input/output frequency [bandwidth]	Beacon frequencies	Transmitting power (peak)	Apogee height
RS-10/11 June 23, 1987 operating [11/87]	USSR, USSR	21.18/29.38 MHz [40 kHz] 21.18/145.88 MHz [40 kHz] 145.88/29.38 MHz [40 kHz] 21.18/29.38 & 145.88 MHz [40 kHz] 21.18 & 145.88/29.38 MHz [40 kHz] 21.120 & 145.820 MHz [*3] 21.23/29.43 MHz [40 kHz] 21.23/145.93 MHz [40 kHz] 145.93/29.43 MHz [40 kHz] 21.23/29.43 & 145.93 MHz [40 kHz] 21.23 & 145.93/29.43 MHz [40 kHz] 21.130 & 145.830 MHz [*3]	29.357, 29.403 MHz 145.857, 145.903 MHz 29.407, 29.453 MHz 145.907, 145.953 MHz	5 W 5 W	1000 km
AMSAT-Phase 3C launch due early 1988	ESA, FRG	435.50/145.90 MHz [150 kHz] 1269.45 & 145.84/435.85 MHz [250 kHz, 40 kHz at 146 MHz] 435.625/2400.71 MHz [30 kHz] 1269.675/435.675 MHz [*4]	145.813, 145.975 MHz 435.650, 435.675 MHz 2400.640 MHz	50 W	35,800 km

[*1] Never activated due to technical problems.

[*2] Never activated due to regulatory constraints.

[*3] Autotransponder (ROBOT), downlink on beacon frequency.

[*4] Digital transponder.

ANNEX II
OF APPENDIX 3
Technical characteristics of amateur satellites

1. Early experimental satellites

The early OSCAR satellites (OSCARs I, II, III, IV and 5) enabled amateurs to experiment with satellite construction, space communications, and telecommand techniques. OSCAR III (1965) was the first in the series to contain a linear transponder. (See Report 542-1, 1974-78.)

2. AMSAT low earth orbit communication satellites

AMSAT-OSCARs 6, 7, 8 and Fuji-OSCAR 12 were low-earth-orbit satellites designed to provide communications capabilities over extended time periods to all radio amateurs. All contained at least one open access linear transponder. Store-and-forward communications capabilities were included on OSCARs 6 and 7 (limited access) and Fuji-OSCAR 12 (open access). All satellites in this series used simple passive magnetic attitude stabilization and were solar powered. Fuji-OSCAR 12 (launched August 1986) is controlled by an on-board computer which handles command decoding, telemetry and scheduling, and manages the digital transponder [Morimoto *et al.*, 1984]. OSCARs 6, 7 and 8 had an average operational lifetime of more than 5.4 years (Fuji-OSCAR 12 remains operational in 1988).

3. Advanced experimental satellites: UoSAT-I (AMSAT-OSCAR 9) and UoSAT-II (AMSAT-OSCAR 11)

UoSAT-I and UoSAT-II, constructed primarily at the University of Surrey (United Kingdom), are solar powered, long lifetime, low-earth-orbit satellite. Primary mission objectives were:

- (1) technical investigations of propagation, intercommunication and the development of amateur-satellite equipment, and
- (2) to assist amateurs involved in self-training in space communications.

These satellites carry a complex array of scientific experiments to investigate the Earth's magnetic field and the radiation environment; to capture Earth images using a Charge Coupled Device camera; to measure the mean time to, and mode of, failure of various digital ICs in the radiation environment; and to study propagation using a set of phase locked beacons spanning the HF to microwave spectrum. They also contain limited access digital communications facilities and a digital voice telemetry. To accommodate the scientific aspects of the mission both satellites use active attitude stabilization employing a gravity gradient boom and magnetorquing. Redundant microprocessors are included for the security of satellite management.

4. AMSAT-OSCAR 13 series (high apogee elliptical orbit satellites)

The Phase-3 designation includes Phase 3A (launch failure 1980), AMSAT-OSCAR 10 (launched in 1983), AMSAT OSCAR 13 (launched in 1988). These satellites are designed for highly elliptical orbits. Phase-3 satellites carry multiple wide-bandwidth linear transponders. OSCAR 13 also carries an open access digital transponder. Each of these satellites is controlled by an on-board computer which oversees all telecommand, scheduling and telemetry functions. The computer also manages an active spin stabilization attitude control. Thermal design was based on a 121-node model [Jansson, 1984]. Each spacecraft has an on-board propulsion system (solid-fuel apogee kick motor on Phase 3A, liquid fuel, multiple fire, kick motors on OSCAR 10 and OSCAR 13) needed to attain the desired final orbit and handle all orbit transfer operations.

5. RS-series (USSR)

RS-1 through RS-8 and RS-10/11 are low earth orbit, solar powered, long-lifetime, ground commandable communications satellites. Most carry one or more open-access linear transponders, RS-5 and RS-7 also employed a regenerative ROBOT transponder, capable of automatically receiving, logging, and acknowledging up-link signals. [Labutin, 1979]

6. ISKRA series (USSR)

ISKRA 2 and 3 are communications satellites ejected from manned spacecraft directly into very low earth orbit.

ANNEX III
OF APPENDIX 3
Results of experiments using amateur satellites

1. Communications Technology Experiments

1.1 A number of innovations in linear-transponder technology have resulted from amateur-satellite service development. These include EER (Envelope Elimination and Restoration) and HELAPS (High-Efficiency Linear Amplification by Parametric Synthesis) [Meinzer, 1974; Hardman, 1986].

1.2 Amateurs have demonstrated that open access linear transponders, operated FDMA, can provide extremely efficient spectrum utilization through time and frequency sharing [Klein and King, 1974]. Linear transponders have been used to support two-way communication via Morse, SSB voice, amplitude-companded SSB voice [Eagleson, 1986; Rinaldo, 1986], slow-scan TV, FM voice, and several digital modes.

1.3 After extensive testing with limited access systems, open access digital store-and-forward transponders have been placed on low altitude (Fuji-OSCAR 12) and high altitude (Phase 3C) [Brenndoerfer, 1987] satellites.

1.4 Mobile terminals on ships, planes, and land vehicles have been used to communicate directly through high and low altitude satellites.

1.5 Experiments have demonstrated the feasibility of:

- 1) using a gateway station to connect a group of low powered earth stations to a satellite;
- 2) employing a satellite-to-satellite relay in real time communications [Klein and Soifer, 1975].

1.6 OSCARs 6 and 7 were used to test the viability of the search and rescue satellite concept in preparation for the COSPAS-SARSAT program [Brandel, Schmid, and Trudell, 1976, Wright and Winter, 1976].

1.7 OSCAR satellites were used to test the feasibility of transmitting medical telemetry from isolated locations via satellite [Kleinman, 1976; Nelson, 1977].

1.8 Telecommand systems on recent satellites use microcomputers to control all on-board telemetry, scheduling, attitude control, and telecommand decoding. Extensive work has been done in experimenting with recovery from radiation-induced errors [Meinzer, 1979; Ward, 1987; Jeans and Traynar, 1982; Haynes, 1982].

2. Self-training

2.1 Tracking procedures suitable for locating spacecraft in circular low-earth-orbits, and high altitude elliptical orbits, have been developed and documented. These include various nomograph techniques [for a summary see Davidoff, 1984] and programs for microcomputers [Clark, 1981]. The nomograph techniques and the computer programs have been widely adopted by workers in science and education.

2.2 UoSAT-I and UoSAT-II have provided amateur satellite users with the opportunity to undertake technical investigations on through-the-ionosphere propagation (HF to microwave frequencies). They have also provided an opportunity to develop special hardware, software, and telecommand procedures [Sweeting, 1982].

2.3 Recent satellites have employed specially designed microcomputers to control on-board activity including telecommand decoding, telemetry, active attitude control, scheduling, and store-and-forward digital communications. Significant advances have been made in:

- 1) high reliability telecommand techniques and protocols; and
- 2) error correcting codes to correct soft errors resulting from a high radiation environment [Meinzer, 1979; Haynes, 1982; Jeans and Traynar, 1982; Ward, 1987].

2.4 Amateur satellites have liquid and solid rockets and orientation sensor systems so that they could be relocated from geostationary transfer orbit to highly elliptical orbits. Telecommand stations have all ancillary systems, including software, needed to accomplish orbit transfer. This includes accurate ranging equipment; and software for:

- 1) orbit determination;
- 2) rapid telemetry reduction;
- 3) attitude determination;
- 4) evaluation of firing options [Meinzer, 1986, Miller 1986].

2.5 Educational materials developed for amateur-satellite users have been expanded and adapted to serve the needs of educators. These activities have been supported by grants from various sources. [Space Science Involvement, 1974; Davidoff, 1978.]

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Appendix 4

Amateur-satellite station technical characteristics
and plans for future systems1. Radio frequency characteristics of amateur-satellite earth stations

1.1 Earth stations in the amateur-satellite service vary in characteristics more than those in most space services. The experimental nature of the service results in variations of e.i.r.p. and receive system figure of merit (G/T). These differences have decreased in recent years due to wider availability of suitable equipment. Given the combination of antennas, high power amplifiers and low-noise amplifiers (LNAs) available up-link e.i.r.p. typically varies by approximately 10 dB with G/T variations of about 6 dB/K. Table I-1 gives the typical operating characteristics of earth stations using current amateur satellites.

1.2 Modulation methods employed by amateur-satellite earth stations tend to be primarily SSB (J3E) with some use of Morse telegraphy. There is an increased use of slow-scan television and angle modulated packet communications via satellite. New modems employing digital signal processors are used. Error corrective coding has been the subject of some experiments and is expected to see more extensive use in the future. Digital voice and amplitude-compandored single sideband are also subjects of investigation.

2. Radio frequency characteristics of amateur-satellite space stations

2.1 Three factors have significantly contributed to improvements in satellite performance and thus reduced requirements for earth stations: improvements in receiver design (primarily reduction in LNA temperature), increases in e.i.r.p. attributable to increased transponder efficiency and linearity, and the use of directional antennas. Table II-1 gives the characteristics of typical satellites in low earth orbit. Table II-2 gives the characteristics of the current satellite AMSAT-OSCAR 13. These satellites use linear amplification in order to support frequency-division multiple access (FDMA).

3. Anticipated future station characteristics and frequency usage

3.1 Design objectives are to increase the S/N ratio to the user and to provide service at the times most useful to operators in different parts of the world; for emergency purposes continuous coverage of the Earth is desirable.

3.2 The use of higher frequency bands is essential to provide for continued growth of the service. The band 435 to 438 MHz is already in considerable use and will be the primary down-link band for high earth orbiting amateur satellites to be launched up until 1998. Significant new use of the 1 260 to 1 270 MHz band is being made by Phase 3 satellites. This band is planned as the primary Earth-to-space band for these satellites. Multi-mode earth station equipment for this band is now readily available.

3.3 The first use of the geostationary orbit by the amateur-satellite service is now practical and is under consideration. The geostationary satellite will be supplemented by others in highly elliptical orbits. Table III-1 gives proposed characteristics for this geostationary satellite with a launch data expected not earlier than 1992. Table III-2 gives the characteristics of associated earth stations.

TABLE I-1

Typical operating characteristics of amateur-satellite
earth stations

		Using Satellites Low Earth Orbiting		Using Phase 3 Satellites High Earth Orbiting		D O W N L I N K
Receiver Characteristics	Frequency	435 MHz	29.5 MHz	436 MHz	145 MHz	
	Figure of Merit (G/T)	-7 dB/K	-22 dB/K	-6 dB/K	-6 dB/K	
	Antenna Gain	13 dBic	0 dBic	14 dBic	13 dBic	
	LNA Noise Temperature	70 K	120 K	70 K	80 K	
Transmitter Characteristics	Frequency	145 MHz	145 MHz	1296 MHz	435 MHz	U P L I N K
	EIRP	+20 dBW	+20 dBW	+25 dBW	+30 dBW	
	Antenna Gain	10 dBic	10 dBic	20 dBic	13 dBic	
	Transmitter Power	+10 dBW	+10 dBW	+5 dBW	+17 dBW	
Signal- to- noise ratios	$\left(\frac{C}{N}\right)_T$ or $\left(\frac{S}{N}\right)_T$	5 to 15 dB	8 to 15 dB	15 to 20 dB	10 to 18 dB	
	$\left(\frac{C}{N_0}\right)_T$	39 to 48 dBHz	42 to 49 dBHz	49 to 54 dBHz	44 to 52 dBHz	

Notes:

- 1 The symbol dBic represents decibels relative to isotropic, circularly polarized source.
- 2 $\left(\frac{C}{N}\right)_T$ and $\left(\frac{S}{N}\right)_T$ respectively indicate total end-to-end carrier-to-noise power ratio, and total end-to-end signal-to-noise power ratio.
- 3
$$\left(\frac{C}{N_0}\right)_T = \left\{ \left(\frac{C}{N_0}\right)_{\text{uplink}}^{-1} + \left(\frac{C}{N_0}\right)_{\text{downlink}}^{-1} \right\}^{-1}$$

where $\left(\frac{C}{N_0}\right)$ represents the ratio of overall carrier power to noise power density.

(The $\left(\frac{C}{N_0}\right)$ values in the above formula should be expressed as a ratio of powers in watts.)"

TABLE II-1
Characteristics of typical low-earth-orbit
amateur-satellite space stations

		Frequency	435 MHz	145 MHz	29.5 MHz	DOWNLINK
Satellite Transmitter Characteristics	TX Power					
	Output (PEP)		0-5 dBW	2 to 9 dBW	0-5 dBW	
	Transponder					
	Efficiency (DC to RF)		20 %	35 to 40 %	20 %	
	Intermodulation (NPR)		-20 dBr	-24 to -26 dBr	-20 dBr	
		Antenna Gain	0 to +3 dBic	0 to +3 dBic	0 to +2 dBic	UPLINK
		EIRP	0 to 8 dBW	2 to 12 dBW	0 to 7 dBW	
		Frequency	145 MHz	435 MHz	145 MHz	
Satellite Receiver Characteristics	System Noise					
	Temperature		300 K	360 K	300 K	
	Antenna Gain		0 to +3 dBic	0 to +3 dBic	0 to +3 dBic	
		Figure of Merit (G/T) (Worst Case)	-25 dB/K	-26 dB/K	-25 dB/K	

Note - The symbol dBr represents decibels relative to the reference level, which is the highest amplitude carrier.

TABLE II-2

Characteristics of AMSAT-OSCAR 13 satellite

=====	
AMSAT Phase 3C	
=====	
SATELLITE BUS:	
S/C Bus Power	35 W EOL
S/C Bus Mass	150 kg
S/C Power Available for High Power Amplifier	30 W
=====	
TRANSPONDER:	
Mode:	Simplex (no feeder link frequencies employed)
Operating Frequencies:	
Uplink:	1269.5 MHz
Downlink:	435.8 MHz
Bandwidth: (Per link direction)	250 kHz
Total Spectrum Occupied (all links)	500 kHz
Transponder HPA Power Output	12.5 W Avg (50 W peak)
Transponder DC to RF Efficiency (Rayleigh Weighted Avg)	40 % [note 1]
Transponder Peak/Avg Ratio (output power backoff)	6 dB
Intermodulation Ratio (NPR method)	-23 dBc
Antenna Gains:	
Uplink	10 dBic
Downlink	9.5 dBic
S/C EIRP	20.5 dBW
S/C G/T	-14 dB/K
S/C System Effective Noise Temperature	260 K
=====	

TABLE II-2 (cont'd)

AMSAT Phase 3C	
SYSTEM CHARACTERISTICS:	
Voice Modulation Method	SSB [J3E]
Effective Voice Channel BW	5 kHz
Spectrum Limited Channels (per satellite)	50
User G/T Standard	-6 dB/K
User C/N0 Requirement	49 dBHz [corresponds to 26 dB peak S/N in 2800 Hz bandwidth]
S/C EIRP per Standard Channel	5 dBW [note 2]
Power Limited Channels Available per Satellite [No Channel Activity Factor Applied]	35 channels simplex [15.5 dB-Channels]
Spacecraft Cost (in orbit) and Insurance	< US \$1,000,000 [note 3]
Total Bandwidth Occupied Per Supportable Voice Channel	14.2 kHz/channel
Gross Cost/Channel	< US \$29,000/channel [note 3]

Note 1 - Transponder uses a new HELAPS-C amplifier technology (see § 1.1).

Note 2 - Based on spacecraft slant range at apogee of 40,000 km, which is equivalent to the slant range for a geosynchronous global beam system.

Note 3 - Cost represents actual expenses paid by AMSAT but does not include donated labour. Commercial value of spacecraft would be approximately US \$ 12,000,000.

TABLE III-1

Proposed geostationary amateur-satellite characteristics

S/C Mass (at Launch):	400 kg
S/C DC Power (End of Life):	205 W
S/C Lifetime:	5 Years

S/C Transponders:	<-----Transponder----->					
Transponder Designation	Uplink	Downlink	Bandwidth	RF Power	EIRP	G/T

Mode J	145 MHz	436 MHz	175 kHz	10 W	23 dBW	-13 dB/K
Mode L	1268 MHz	436 MHz	325 kHz	20 W	26 dBW	-9 dB/K
Mode S	1268 MHz	2401 MHz	250 kHz	12 W	27 dBW	-9 dB/K
Mode S (Video)	6-cm band	2402 MHz	2.0 MHz	10 W	26 dBW	-10 dB/K

TABLE III-2

Characteristics of earth stations for geostationary satellites

Earth station up-link requirements are under study.

Transponder Designation				
<-----Earth Station----->				
	Transmit Frequency	Receive Frequency	EIRP	G/T
Mode J	145.9 MHz	436 MHz	20 dBW	-10 dB/K
Mode L	1268 MHz	436 MHz	29 dBW	-10 dB/K
Mode S	1267 MHz	2401 MHz	30 dBW	+5 dB/K
Mode S (Video)	[note 1]	2401 MHz	[note 1]	+11 to +13 dB/K

Note 1 - Earth station uplink requirements are under study.

DECISIONS

DECISION 69-2

FUTURE PUBLIC LAND MOBILE TELECOMMUNICATION SYSTEMS (FPLMTS)

(1985-1988-1989)

CCIR Study Group 8,

CONSIDERING

- (a) Recommendation No. 205 (Rev. MOB-87) of the World Administrative Radio Conference for the Mobile Services, Geneva, 1987;
- (b) that the ITU Plenipotentiary Conference (Nice, 1989) has resolved that a World Administrative Radio Conference for dealing with frequency allocations in certain parts of the spectrum be convened early in 1992, and that a key item of its agenda will concern spectrum for the various mobile services and mobile satellite below 3 GHz;
- (c) Recommendation 689 on Future Public Land Mobile Telecommunication Systems (FPLMTS);
- (d) that different future systems are under study;
- (e) that system compatibility is necessary for international operation, and that commonality is desirable to ensure that the overall system cost per mobile user is significantly less than it is with present systems;
- (f) the need for a flexible system structure able to match network investment to revenue growth, to adapt readily to environmental factors, and to respond to new developments without restricting innovation;
- (g) the possible need for common channels or frequency bands to allow regional and/or world-wide operation, particularly with the increasing use of personal (hand-held, portable) terminals;
- (h) that the use of internationally agreed frequency bands also facilitates the planning of national networks and reduces the risk of harmful interference with other radio services;
- (j) the increasing importance of the various types of non-voice telecommunication services;
- (k) that Personal Telecommunication services are expected to be introduced throughout the public telecommunication networks based on evolving intelligent network capability;
- (l) that the cost of radio and VLSI technology is continually decreasing, thus making, in a number of cases, the radio approach a competitive alternative access option to the voice and non-voice telecommunication services;

- (m) that some mobile terminals of the FPLMTS may require to access mobile satellite systems for use on land, ships and aircraft;
- (n) the allocation of radio spectrum in the 1.5/1.6 GHz band by the WARC MOB-87 to the land mobile-satellite service;
- (o) that there is a need for mobile terminals to roam between public land mobile telecommunication networks in different countries;
- (p) that a standardized radio interface would facilitate the roaming of mobile units between networks;
- (q) that users may want to be able to use the same terminals and procedures as in the fixed network ISDN to access similar telecommunication services in FPLMTS;
- (r) Questions 77/8 and 38/9, Study Programme 39A/8, and Decisions 81, 103, and [JIWP-WARC-92];
- (s) the relevant CCITT Recommendations and ongoing studies,

DECIDES

1. that Interim Working Party 8/13 should continue its work;
2. that the Interim Working Party shall:
 - 2.1 study the following spectrum matters in order to provide timely guidance to IWP 8/15 as part of the preparations by CCIR for the 1992 World Administrative Radio Conference on frequency allocations:
 - identification of spectrum bandwidth needs, based on relevant technical and operational parameters together with FPLMTS services and traffic estimates, in the frequency range below 3 GHz, noting that there will be pressures from other radio services for spectrum in the same frequency range;
 - special criteria for spectrum sharing between FPLMTS and other radio services;
 - examination of any special spectrum needs for FPLMTS, for example that to enable international/world-wide roaming;
 - evaluation of the technical factors affecting choice of frequency bands for FPLMTS;
 - examination of the system interworking and spectrum requirements of allied mobile and mobile satellite technologies through liaison with IWP 8/14, as appropriate;

2.2 identify and study appropriate FPLMTS matters corresponding to CCITT studies of Personal Telecommunications (PT) and other relevant areas, in particular:

- mobility-based service definitions;
- requirements for CCITT concerning numbering, routing and charging matters in PT;
- the requirements for signalling over the radio interface and within the fixed network;
- coding/transcoding of information, especially speech coding, by methods suitable for radio systems;
- requirements for interworking and integration between FPLMTS and other telecommunication networks;
- appropriate working relationships with CCITT Study Groups for the above subjects;

2.3 determine the degree of compatibility and commonality which is desirable or achievable, bearing in mind the application of FPLMTS technology for the provision of fixed services, particularly for the needs of the developing countries*;

2.4 determine the services and relevant technical and operational characteristics for FPLMTS, in particular:

- personal telecommunication services;
- mobility-based services;
- basic structure of information and control channels;
- overall performance objectives;

3. that the IWP should provide appropriate outputs in the above areas according to the following time scales:

- DECIDES 2.1 in time to meet the commitments of IWP 8/15;
- DECIDES 2.2 in a timely manner for coordination with the various CCITT Working Groups;

4. that the IWP should aim to complete its work in time for consideration by Study Group 8 at the Interim Meeting of the 1990-1994 study period;

5. that the Chairman and composition of IWP 8/13 will be as shown in Annex I.

* Study Group 9 should be informed of this activity.

ANNEX I

At the end of the interim meeting, 1988, the following Administrations and Organizations have indicated their participation in the work of Interim Working Party 8/13:

Administrations:

Germany (Federal Republic of)
 Saudi Arabia
 Australia
 Brazil (Federative Republic of)
 Canada
 Costa Rica
 Denmark
 Spain
 United States of America
 Finland
 France
 Greece
 Hungary
 India
 Iran
 Italy
 Japan
 Mexico
 Netherlands
 New Zealand
 Norway
 Oman
 United Kingdom
 Singapore
 Sweden
 Switzerland
 USSR

International Organizations, Recognized Private Operating Agencies and Scientific or Industrial Organizations:

International Maritime Satellite Organization (INMARSAT)
 European Space Agency (ESA)
 EUTELSAT
 International Electrotechnical Commission (IEC)
 Compañía Telefónica Nacional de España (CTNE)
 Rhode & Schwarz
 Bell Northern Research
 AT&T
 GTE, SC, Telephone Operations
 Telefonaktiebolaget LM Ericsson
 S.I.T.A. Société internationale de télécommunications
 aéronautiques

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DECISION 97

SHARING CRITERIA BETWEEN BROADCASTING
AND FIXED AND MOBILE SERVICES IN THE BAND 2-30MHz

(1989)

This text may be found in the Annex to Volume X-1.

