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DIGITAL MOBILE TELEPHONY - PERFORMANCE OBJECTIVES AND NEW POSSIBILITIES

Jan Uddenfeldt and Jan-Erik Stjernvall
Ericsson Radio Systems AB
S-163 80 Stockholm, Sweden

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1. OBJECTIVES OF A PAN-EUROPEAN MOBILE TELEPHONE SYSTEM

The Pan-European 900 MHz mobile telephone system to be used in the 1990'ies shall meet the requirements of a mass market. A considerable increase of the capacity compared to the telephone systems used today, and the possibility to use low-cost hand-held radio telephones, will be essential in a future system.

The system shall be able to handle a very large number of subscribers in an economic and technical efficient way. The system must allow high frequency re-use and possibilities to adapt the system to mini-cells. It shall be designed to allow a good indoor service. The limited availability of radio spectrum introduces powerful demands on increased spectrum efficiency.

The future system shall be specified in such a way that the cost and size of hand-held and mobile equipment will allow exploitation of the full market for hand-held equipment.

There is a strong demand for new services, in particular, new and and improved data services including connection to ISDN will be of great importance.

However, a major part of the communication will still be speech. It shall be possible to include a speech privacy function with a satisfactory degree of security.

The system shall allow low cost multi-user terminals (PBX-type) to be used e. g. in trains and buses.

2. DIGITAL SYSTEM SOLUTION

A well designed digital system will meet these performance objectives better than any analog system. Many new services require a digital system. A digital radio system will also give better capacity than the present analog systems. The digital FDMA system, presented in reference (1) is one example.

With digital radio, it is possible to use TDMA-technology. This will facilitate the introduction of several new services. Furthermore, TDMA makes it possible to reduce the cost and size of the radio telephones as well as of the base stations.

Several digital system solutions are being studied by Ericsson. The features of TDMA systems are discussed in the following sections. Narrow-band TDMA with a channel spacing of around 300 kHz is considered to be a promising system solution, which is discussed in some detail below.

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3. HIGH CAPACITY FEATURES

High frequency re-use

In a presentation of a digital FDMA system (ref. 1), it was shown that in a digital system the spectrum efficiency was increased by a factor of 3 - 4 compared to analog systems used today. This basic conclusion can be carried over to a narrow-band TDMA system as well.

The fading margin can be reduced in a narrow-band TDMA system by means of introducing different kinds of diversity functions. Channel coding used together with interleaving and frequency hopping (one hop every burst) will give such a function. This arrangement will lead to a decreased C/I criterion and hence the spectrum efficiency will increase.

The time dispersion can be taken care of by an adaptive equalizer in which the taps are adaptively set every TDMA-burst. The adaptive equalizer can take advantage of the multi-path delay-spread, as explained in reference (3). The main attribute of an adaptive equalizer is that it can perform this function without any band-width expansion. Thus, the band-width requirement for a narrow-band TDMA system is virtually the same as for FDMA systems. In a 300 kHz radio channel, it will be possible to accomodate ten simultaneous users (i. e. 10 time slots). This is equivalent to 30 kHz per user channel, which is substantially better than what can be obtained in a wide-band TDMA system, where spread spectrum is used.

Macro diversity

The C/I criterion can be further reduced by introducing a fast hand-over function, based on macro diversity. This can easily be achieved in a TDMA system, since the received signal strength and the signal quality (BER) can be supervised both in the radio telephone and in the base station (e. g. in the channel decoder). This function will perform fast detection of interference. In addition, hand-over and all the necessary signalling can be achieved without any (or with very small) communication interruption.

Finally, adaptive hand-over to a new channel without interference, can be performed, since the radio telephone can monitor signal levels from nearby cells by momentarily switching to other time slots and frequency channels.

Mini-cells

To achieve high capacity in the mobile telephone system, mini-cells together with a common calling channel can be introduced. The fast TDMA hand-over function forms the basis for this concept.

Low-cost, one-frequency base stations can easy be built using TDMA, since one 300 kHz radio channel will be sufficient for the base station. This would make mini-cells (radius: 100 - 200 meter) with low elevated base station antennas economically feasible. The one-channel base station would be less complicated than a multi-channel FDMA base station.

A narrow-band TDMA system, consisting of mini-cells with a radius of 100 meter, each served by a one-frequency base station with ten time slots (using one 300 kHz radio channel), will have a capacity of around 8000 subscribers per square-kilometer.

4. NEW POSSIBILITIES FOR HAND-HELD RADIO TELEPHONES

Change of technology and regulations

New technology will be required in order to meet the market demands of small hand-held equipment in the 1990'ies. The major short-coming of analog radio telephones is that the cost and size of the high-frequency radio parts will remain high in the future. In a typical 900 MHz hand-held radio telephone, around 50 % of the manufacture cost is due to the radio part.

Furthermore, the radio part can not take advantage of the rapid technology development towards VLSI. Thus, the cost of future analog radio telephones will be completely dominated by the radio part.

A digital mobile telephone system, based on TDMA, will reduce both cost and size of the radio part drastically. The reason for this is that the channel spacing is increased in TDMA and also that the selectivity requirements can be reduced.

Reduced cost

To make low cost equipment feasible, it is necessary to reduce the selectivity requirements as compared to present mobile telephone systems. In these systems it is possible to reduce adjacent channel selectivity to 20 - 30 dB by proper frequency planning. This will also increase the capacity, since a higher bit rate then can be transmitted for a given channel spacing.

Using a fast hand-over functin, as described earlier, a TDMA system can quickly detect any kind of interference (such as intermodulation, spurious or co-channel interference) by BER-measurements and allocate a new time-slot with low interference.

Thus, the selectivity requirements of a TDMA system can be decreased to 40-50 dB for intermodulation, spurious, etc.

The combination of reduced selectivity requirements and increased channel spacing will eliminate the major costs of the radio part. The duplex filter can be eliminated, since time duplex can be used.

The receiver can possibly be of the direct-conversion type. This allows integration of the receiver into a low cost analog LSI circuit, which will replace the double or triple super heterodyne receiver with all it's mixers, amplifiers and high-selective filters. The synthesizer cost will be significantly reduced, since the dominating elements (VCO and TCXO) can be simplified. Even a moderate increase in channel spacing of ten times is sufficient, both to replace the TCXO by a low cost (20 ppm) crystal and to reduce the VCO to a low Q circuit.

The total cost reduction for a hand-held radio telephone can be substantial in a short-term perspective. Even more important is that the cost reduction of a digital radio will continue as long as the VLSI technology continuous to improve.

Reduced size

Monolithic integration of virtually the whole radio part (except for the power transmitter) is a possibility since the receiver and the synthesizer are well suited for integration and the duplexer can be eliminated.

Also the power consumption can be reduced. In a digital TDMA system, the average transmitter power can be reduced by up to 10 dB. This is partly because the fading and shadow margins can be decreased. Another factor is that voice operated transmission (VOX) can be used, since TDMA uses fast burst-syncronization.

Power saving methods can easily be introduced without increasing the time for calling, hence the stand-by current can be reduced by roughly 10 dB. Even though some power consumption will be required for the adaptive equalizer in the demodulator, this will not be a dominating factor.

Altogether, substantially less power consumption (around five to ten times) is the result which also forms the basis for pocket-sized equipment.

Transmitter peak power

A short-coming of TDMA is that high peak power may be required in the transmitter, in order to compensate for the wider receiver noise band-width. To some extent, this high peak power can be decreased by the reduced fading and shadow margins. For a wide-band TDMA system, however, this is not sufficient.

For example, a 10 MHz wide-band TDMA-system has a receiver thermal noise power, which is around 26 dB higher than in the present FM systems. Even if TDMA can operate with 10 dB less fading margin, the transmitter peak power has to be increased by 16 dB, which would require 20 - 40 W peak power, in order to get the same radio range as with portable FM telephones. This ruins the possibility to introduce low cost hand-held equipment.

However, for a narrow-band TDMA-system with a band-width of, say, 300 kHz, the peak power will be reduced to 1 -2 W.

5. CONCLUSIONS

The requirements of a future system are probably best handled by a digital system solution. In particular, low cost hand-held equipment is one of the basic requirements from which the system shall be designed. Good indoor service and high capacity are other prime requirements. It is necessary that a digital system is designed to meet these requirements.

In fact, a narrowband TDMA system appears most promising. The cost advantages concerning the radio part are equally good for narrow-band and wide-band TDMA. The disadvantage of high transmitter peak power is avoided in a narrow-band TDMA system.

High capacity is obtained in narrow-band TDMA both through reduced fading margins and mini-cell operation. The fading margin can be reduced without sacrificing band-width by using a combination of adaptive equalization, frequency hopping and channel coding.

Furthermore, a single 300 kHz radio channel will be sufficient for many rural base stations.

Also the compatibility with existing FDMA systems in different parts of the world will make it difficult to introduce too wide radio channels. A world standard is an obvious extension of the Pan-European system.

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DIGITAL MOBILE TELEPHONY WITH IMPROVED SPECTRUM EFFICIENCY

Jan Uddenfeldt
Ericsson Radio Systems AB
S-163 80 Stockholm
Sweden

ABSTRACT

A digital FDMA feasibility system is presented. It is based on available technology. The system offers 3-4 times better capacity (measured in Erlangs per cell and MHz) than a companded FM system. The system is using 16 kb/s RELP voice coding and 27 kb/s GMSK transmission. The channel spacing is 25 kHz. The improved capacity is obtained by using a combination of channel coding and slow frequency hopping, which allows a much higher frequency re-use than in analog systems.

1. INTRODUCTION

The mobile telephone service has recently been established and is now developing rapidly. New technical solutions will be required in order to comply with the market demand in the next decade.

One severe short-coming of the first generation of mobile telephone systems is that the capacity per MHz bandwidth will be insufficient. Another problem is that the cost and the size of mobile and portable units probably will not decrease to a level, where the full market can be exploited.

Digital transmission gives new possibilities for the next generation of mobile telephone systems. The main advantages by going digital are:

(1) Improved spectrum efficiency, i. e. increased number of subscribers per MHz bandwidth;

(2) Reduced cost of mobiles/portables as well as of base stations;

(3) New system services e. g. secure voice and integrated voice/data transmission.

In the following, a feasibility system for 900 MHz digital transmission will be described. Primarily, the issue of spectrum efficiency will be adressed, but aspects on choice of technology will also be given.

It should be noticed, that the aim with the feasibility system is to show what can be obtained with available technology, rather than propose a final solution to a digital mobile telephone system.

The system can be regarded as a baseline system and is not limited to FDMA operation. The results concerning spectrum efficiency are carried over to a combined TDMA/FDMA system as well.

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2. SYSTEM OUTLINE

The technology for digital speech transmission is developing rapidly. With present technology, 16 kb/s digital voice gives full telephony quality. Simultaneously, spectral efficient digital modulation methods (e. g. TFM, GMSK) have envolved. These schemes provide a modulation efficiency above 1 bps/Hz with sufficient adjacent channel protection for cellular systems. A channel spacing of 15 kHz is therefore sufficient for a digital FDMA system.

From a spectrum efficiency point of view, it is however more advantageous to introduce channel coding and use 25 kHz channel spacing than to use 15 kHz channel spacing without any coding. This will be explained in more detail in subsequent sections.

Channel coding reduces the impact of Rayleigh fading in a way that is similar to diversity schemes. The main difference is that diversity requires multiple antennas and/or multiple receivers, which substantially increases the cost and the size of portable/mobile units. Coding, on the other hand, requires a bandwidth expansion, which, however, is well motivated from a spectrum efficiency point of view.

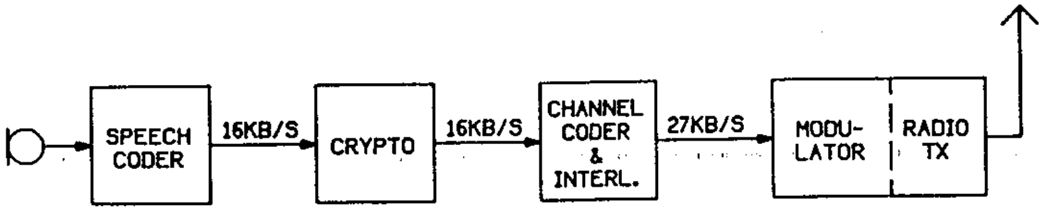
For stationary use, coding will not provide any diversity gain unless it is combined with other digital methods. One such method is frequency hopping (of the order 100 hops/sec) which will destroy excessive long fades. The coding gain will be retained for arbitrarily slow fading as long as the interleaving depth exceeds the hop duration. Thus, the combination of channel coding and frequency hopping provides a diversity function for stationary as well as for mobile users.

3. SYSTEM DESCRIPTION

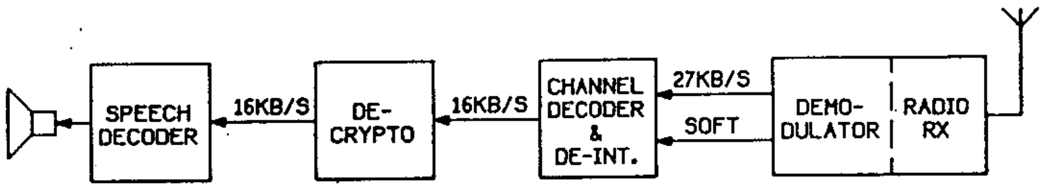
3.1 General

A block diagram for the digital 25 kHz FDMA feasibility system is shown in Figure 1.

The systems consists of a 16 kb/s speech codec, a digital crypto, a rate 3/5 interleaved channel codec and a coherent 27 kb/s GMSK radio modem. The IF-bandwidth is 20 kHz and the adjacent channel protection is similar to the U.K.'s 25 kHz TACS-system. Further description is given in Table 1.



(A) TRANSMITTER PART



(B) RECEIVER PART

Fig. 1. Block diagram of the coded digital system for 25 kHz FDMA channels.

Transmission rate:	27 kb/s
Speech codec	
codec rate:	16 kb/s
codec:	RELP
operating BER:	1 %
Channel codec	
rate:	(10,6) block code
interleaving:	9.5 ms (256 bits)
decoder:	soft Chase decoder
Synchronization:	32 bits every 96 ms
Modulation	
modulator:	GMSK 0.25
demodulator:	coherent
IF-bandwidth:	20 kHz
adjacent channel protection:	25 dB
Required E_s/N_0 for 1 % BER at speech decoder	
without fading	6 dB (-118.3 dBm)
with fading (20 Hz)	12 dB (-112.3 dBm)

Table 1. Some characteristics of the digital 25 kHz FDMA feasibility system.

3.2 Speech codec

The speech coder rate, 16 kb/s, is chosen to assure toll quality speech. Several investigations have shown that toll quality speech can be obtained at this rate. One example is (Daumer, 1982), where it was found that e. g. a 16 kb/s adaptive predictive coder (APC) gives a mean opinion score (MOS) of 4 on a 1-5 grade scale. Another example is 16 kb/s APC-AB (Honda, 1984) which was found to give the same MOS as 7 bit PCM.

In the feasibility system, a 16 kb/s standard RELP coder is used at present. A block diagram is shown in Fig. 2. The speech quality of the 16 kb/s RELP is somewhat below toll quality. However, it was selected for the following reasons:

- the speech quality is comparable to 32 kb/s adaptive deltamodulation and it was judged that this quality was good enough to make comparisons with FM in a fading environment,

- it is robust to channel errors and can accept an input BER of 1 %,

- it gives a fairly simple codec and can be implemented with a few standard signal processors and is well suited for VLSI.

The RELP-coder is a hybrid waveform/parametric coder, where LPC is used both to represent the spectral parameters and to produce a residual waveform. By low-pass filtering the residual at 1300 Hz, the essential voice character is preserved. At the decoder end a high frequency regenerator (consisting of a rectifier and a three-band spectral flattener) is used in order to regenerate the signal.

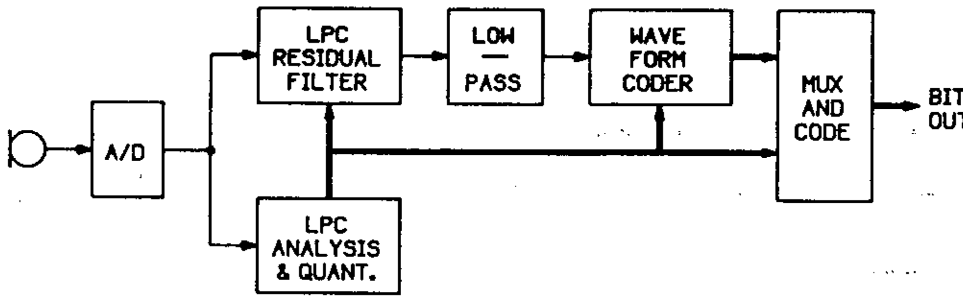
The 16 kb/s RELP has been implemented in full duplex by using one standard PCM codec chip, three NEC 7720 signal processor chips and one 8 bit microprocessorcontroller. Two of the 7720 chips are used for the coding and one for the decoding.

Parameters for the 16 kb/s RELP are given in Table 2. Some error protection coding bits are included in the 16 kb/s bit rate. The proposed RELP operates in two modes. In mode A, error protection is only applied to the LPC-parameters, whereas in mode B some error protection is applied to the residual waveform as well. Mode B is embedded in mode A.

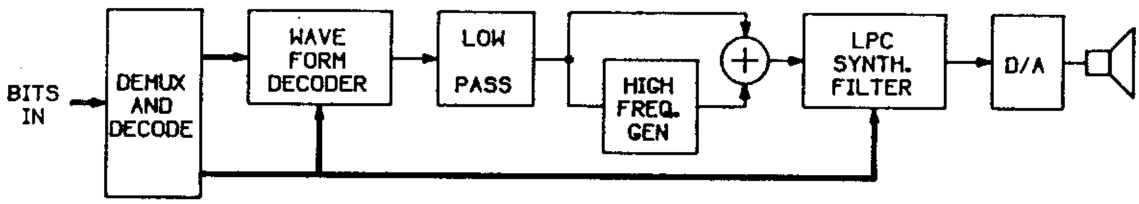
The LPC-parameters are more sensitive to bit errors than the residual waveform. On the other hand, the parameters are easy to protect. When an error is detected, the parameters from the previous frame are used. Error protection coding is applied to the 3 most significant bits of every LPC-parameter and to all 6 bits of the gain factor by using multiplies of a (11,6) single error correcting, double error detecting code.

The residual waveform is more robust to bit errors. Many of the bit errors can be detected and smoothed out by using signal processing such as peak detection. In mode B error protection is added at the expense of 1 bit in the APCM coder. For each block of 4 residual samples, the two most significant bits of each sample are coded by a (12,8) single error correction code.

Further developments of the RELP are being considered. A promising approach is the residual coder with adaptive sampling AS-RELP presented by (Hedelin, 1983). At 16 kb/s this coder gives a quality which can not be distinguished from 64 kb/s PCM.



(A) CODER PART



(B) DECODER PART

Fig. 2. Block diagram of 16 kb/s RELP codec.

TYPE OF RELP	A	B
Coder bit rate	16 kb/s	16 kb/s
Sampling rate	8 kHz	8 kHz
Frame size	24 ms	24 ms
Downsampling factor	3	3
Waveform quantizer	5 bit APCM	4 bit APCM
No. of LPC-coeff.	8	8
Allocation of bit rate		
Residual waveform		
Information	13.33 kb/s	10.67 kb/s
Error correction	-	2.67 kb/s
LPC-parameters		
Information	1.63 kb/s	1.63 kb/s
Error correction	1.04 kb/s	1.04 kb/s

Table 2. Some parameters for the 16 kb/s RELP. It can be operated in two modes (A and B), with different degree of error protection. The total rate is fixed at 16 kb/s in both cases.

In fact, these results are so attractive that lower rate codecs, e. g. 8-10 kb/s could be considered. This has also been shown by other researchers (Sluyters, 1984). At present, however, it seems likely that although good speech quality can be obtained at 8-10 kb/s the codec complexity will be too high and the robustness to channel errors will not be good enough. In any case it is clear that 16 kb/s is a conservative choice for digital voice.

3.3 Channel codec

In a secure voice system with a digital crypto, it is important to distinguish between two aspects of coding. Error protection coding that are related to the speech source has to be inserted before the enciphering unit. On the other hand, channel coding that are related to the radio channel should be inserted after the enciphering unit. This gives the solution shown in Fig. 1, where the speech-related coding is included in the speech coder.

Notice, that the channel coding is only applied on the radio path. This will reduce the required bit rate for communication between base stations and the MTX. Another aspect is that the BER for the deciphering unit is decreased to a level (around 1 %) where a highly secure crypto can be used. Notice that the BER for the 27 kb/s demodulator is 5 - 10 % when the deciphering unit operates at 1 % BER.

The channel coder in Fig. 1 is used to reduce the impact of fading and to improve the spectrum efficiency. Important aspects of coding are:

- Soft decision is essential to obtain improved spectrum efficiency (see e. g. (Stjernvall, 1984));
- The bandwidth expansion due to coding should not be too large. For example (Goodman, 1984), found that the bandwidth expansion should be below 2 for optimum spectrum efficiency;
- Coding delay must not be excessive.

In the feasibility system a short (10,6) block code with interleaving and soft decision is used (see Table 1). Soft decision is accomplished by a Chase decoder. The required bit rate is 27 kb/s, when some synchronization bits have been added. A more detailed description as well as measurement results are given in an accompanying paper (Ekemark, 1984).

3.4 Radio modem

It has been shown that in cellular systems, the adjacent channel protection does not have to be better than 20-30 dB. This has been taken into account in the design of the FCC-system in U.S.A. and in the U.K.'s TACS-system.

For digital cellular radio, the modulation efficiency drastically drops with an increased requirement on the adjacent channel protection. See Fig. 3. With 20-30 dB protection a modulation efficiency above 1 bps/Hz can be reached, whereas with traditional requirements on protection (70 dB), modulation efficiency above 0.7 bps/Hz can hardly be reached.

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In the feasibility system 27 kb/s GMSK modulation (0.25) is used for 25 kHz channel spacing. A coherent demodulator is used, since it gives superior performance for portable use. In a fading environment, there is no disadvantage to use the coherent demodulator since the influence of random FM is taken care of by the channel decoder.

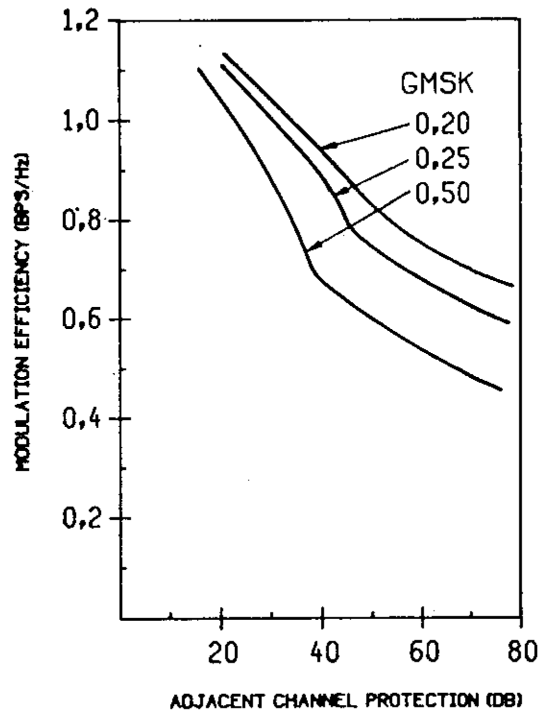

Fig. 3. Modulation efficiency vs. adjacent channel protection for digital GMSK. Modulation efficiency is measured in bps/Hz and is defined as the ratio between bit rate (bps) and channel spacing (Hz). Ideal IF-filter with bandwidth equal to bit rate is assumed.

3.5 Frequency hopping

By adopting frequency hopping (FH) with one hop per interleaving depth, the coding gain of the channel coder is obtained also for stationary use. With the interleaving depth given in Table 1, the required hopping rate is around 100 hops/sec, which is quite feasible with present synthesizer technology.

In a cellular system, this is achieved by co-ordinated, orthogonal frequency hopping at every basestation so that no collisions will occur between users at the same site.

FH has the additional advantage that co-channel interference will be further reduced by using different hopping frequences at different co-channel sites. This has been called mixed FH (Verhulst, 1984).

From these investigations, spectrum efficiency can be estimated. Table 4 gives a comparison of analog and digital transmission. We are considering a high capacity system with small cells. Large cities will be the dimensioning case. A uniform re-use pattern with minimum cell size is used. The required number of frequency groups in a re-use pattern for a small cell system with 1 km cell radius has been estimated for 90 % reliability based on typical mobile radio propagation. Details are given in an accompanying paper (Stjernvall, 1984a). Important conclusions are:

- The coded digital system gives approximately a **three-folded** increase in spectrum efficiency when compared to analog FM (35.1 and 12.4 Erlangs per cell for a 10 MHz system, respectively).

- The uncoded digital system is only around 50 % more efficient than the FM-system, despite the reduced channel spacing.

- The capacity improvement in a digital system is obtained by a **higher frequency re-use** rather than by a reduced channel spacing. Channel coding is vital for the increase in frequency re-use.

	ANALOG SYSTEM	DIGITAL SYSTEM	CODED DIGITAL SYSTEM
Description	Comp. FM	RELP/GMSK	RELP/GMSK
Speech coder rate	-	16 kb/s	16 kb/s
Transmission rate	-	16 kb/s	27 kb/s
Channel spacing	25 kHz	15 kHz	25 kHz
Minimum C/I in fading	18 dB	20 dB	13 dB
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	12.4 E	17.2 E	35.1 E

Table 4. Spectrum efficiency for companded FM system and two digital narrowband FDMA-systems. Sectorized cells with 3 cells per site is used.

5. DISCUSSION

It has been shown that with available technology, a digital cellular system is substantially more spectrum efficient than present analog FM systems. Still, the digital system is not optimized. Future developments may also include TDMA-operation.

Furthermore, the spectrum efficiency of the digital system can be improved beyond what is shown in Table 4. With digital techniques, a system which allows very frequent hand-overs between base stations, can be implemented. This will allow that the shadow margin for C/I can be reduced. Accordingly, it is not necessary to require 90 % reliability against C/I. For example, the required C/I = 13 dB can be obtained with 50 % reliability in a 3 cell system. With a 3 cell system, the capacity rises to 120 Erlang per cell for a 10 MHz system, which is almost 10 times better than for analog FM (see Table 4). However, a more realistic system is to use a combined 3/9 cell system. Such a system will give around 50 Erlang per cell for a 10 MHz system, see (Stjernvall, 1984).

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TMS 17/85

CEPT/GSM

GSM Doc 73/85

Source: GSM

BASIC REQUIREMENTS FOR A HARMONIZED MOBILE COMMUNICATION SYSTEM

The following text replaces the text in Section 3, GSM Doc 2/82 (Doc T/CCH(82)21R)

a. **The 900 MHz CEPT mobile communications system must co-exist with earlier systems in the same frequency band.**

CEPT recommends the frequency bands 890-915/935-960 MHz to be used for public mobile services.

Furthermore, CEPT recommends the use of 890-905/935-950 MHz for earlier (1st generation) public, mobile systems, and that the bands 905-915/950-960 MHz are reserved for the 900 MHz CEPT mobile communications system. The attention is drawn to the fact that several countries use 914-915/959-960 MHz for cordless telephones according to CEPT recommendations.

It shall be possible for each individual country to choose the point of time for transition from earlier systems to the new system and also to have these systems operating simultaneously, on the same base station sites.

b. **The system shall be designed such that mobile stations can be used in all participating countries.**

The system shall be designed for automatic roaming within each network. The degree to which the roaming facility between networks (or countries) shall be automatic has yet to be decided, but the system should be capable of providing fully automatic roaming between networks which so desire.

It must be kept in mind that there will be a need to bar calls to and from undesirable mobile stations, i e in the case of blacklisted subscribers, stations from non-participating networks with systems built to the same specifications, etc.

The system should be designed in such a way that the fixed subsriber does not have to know the location of the mobile. (It must also be kept in mind that many roaming subscribers do not want their location to be disclosed to a calling subscriber.) However, where the distance of the mobile from the fixed subscriber would lead to a higher tariff than the minimum, the fixed subscriber should have the choice of whether to incur such a higher tariff.

c. **In addition to telephone traffic, the system must allow maximum flexibility for other types of services, e g ISDN related services.**

It is to be expected that in most countries, the dominating type of traffic will be speech traffic via the PSTN for a number of years after system start. However, other types of traffic will take on an increasing importance, in particular the ISDN related services. Hence, the system must be capable of, but not dependent on, interworking with the ISDN (within the limitations set by the mobile radio environment) by the time the ISDN is implemented in each particular country.

The ISDN-services which could be mapped to and from the GSM system will be subject to further study bearing in mind the differences in transmission data rates.

d. **The system concept to be chosen shall permit a high level of spectrum efficiency and state-of-the-art subscriber facilities at a reasonable cost, taking into account both urban and rural areas and also development of new services.**

A suitable measure of the spectrum efficiency should be decided. One approach could be to set figures for the characteristic system parameters, such as the traffic handling capability, expressed in Erlangs or information transfer rate per square kilometer.

It must also be kept in mind that the spectrum efficiency question is likely to be the most important limit to growth in large cities. In sparsely populated areas, the cost per Erlang of carried traffic will probably be a more important problem than spectrum efficiency. It is necessary for the success of the system that both applications are considered when choosing system concept.

e. **The services and facilities offered in the PSTN/ISDN and other public networks should as far as possible be available in the mobile system. The system shall also offer additional facilities, taking into account the special nature of mobile communications.**

The services and facilities offered in the GSM system may not always be the same all over the system. This may be due to various causes, e g when a GSM service requires a certain, not yet achieved, degree of modernization of the PSTN in order to be offered. Another case may occur when an Administration wishes to offer its subscribers additional services, not offered by all Administrations. In the latter case, such additional functions and facilities should have no adverse influence on the operation of foreign mobile stations in that network.

The selection of a minimum set of services and facilities to be provided in all networks (including user procedures) is yet to be made. This set should include even services and facilities beyond those offered in the fixed networks, since there are a number of services that are particular to a mobile system and that have no meaning in a fixed network.

f. **The system shall allow for operation in the entire frequency band 890-915 MHz and 935-960 MHz.**

Although the system shall be designed for the entire frequency band, use of only parts of the band shall also be possible e g for frequency management purposes.

g. **It should be possible for mobile stations belonging to the system to be used on board ships, as an extension of the land mobile service. Aeronautical use of GSM mobile stations should be prohibited.**

The use of mobile stations in the GSM system should be permitted in coastal waters, inland waterways etc for public correspondence only. The system should not be confused with the maritime radio services, however.

h. **The identification plan shall be based on the relevant CCITT Recommendation.**

CCITT Rec. E.212 concerns an international identification plan for mobile stations of public land mobile networks in different countries. According to the Recommendation, even countries not participating in the GSM system may be included in the identification plan, thus enabling participating at a later stage.

The implications for the system to accommodate a personal user identity which is independent of terminals is under study.

i. **The numbering plan shall be based on the relevant CCITT Recommendation.**

CCITT Rec. E.213 concerns a numbering plan for land mobile stations in public land mobile networks. The numbering plan for GSM mobile stations shall be independent of the identification plan. It shall also allow for national differences in numbering and routing.

j. **In addition to vehicle-mounted stations, the system shall be capable of providing for hand-held stations and other categories of mobile stations.**

With a view to the growing interest in the use of hand-held stations, it is necessary that the system is technically capable of handling low-power hand-held stations, facilitating battery economy and intrinsically not demanding battery consumption in excess of that for any hand-held stations associated with existing earlier public mobile telephone systems working in the 900 MHz band. In addition, a number of other categories of mobile stations can be envisaged.

k. **The system shall be capable of offering encryption of user information but any such facility should not have a significant influence on the costs of those parts of the system used by mobile subscribers who do not require such facility**

The demand for special protection could be considerable among some users. When an Administration offers foreign subscribers an encryption facility, the means must be provided of conveying the key held in the home network of the subscriber to the network of that Administration. The consequences of this are for further study.

It will be possible for the subscribers themselves to provide some kind of end-to-end encryption, subject to the technical limitations of the network.

l. **No significant modification of the fixed public networks must be required.**

Because of the immense size of the fixed public networks, it is not economically possible to modify them in order to introduce the GSM system, except for minor modifications such as reprogramming of registers, etc.

m. **The system parameters shall be chosen with a view to limit the cost of the complete system, in particular the mobile units.**

The cost of the system needs to be considered in terms of the cost of the fixed infrastructure to be met by the telecommunication operators, and the mobile equipment, usually met by the mobile subscribers. Both need to be within affordable limits, which may be stated as not in excess of that for existing earlier public mobile telephone systems working in the 900 MHz band. Since the cost of the mobile will constitute the main portion of the total system cost, it is preferable for the mobile equipment cost to be lower than that for existing earlier public mobile telephone systems working in the 900 MHz band.

n. **The system design must permit different charging structures and rates to be used in different networks.**

The charging structures, as well as the rates used in the present telecommunication networks in Europe are very different, and this must be accepted even when the GSM system is introduced. It is therefore necessary to design the system in such a way that each Administration is free to choose its own tariff and charging policy.

o. **For the interconnection of the mobile switching centres and location registers, an internationally standardized signalling system shall be used.**

The only internationally standardized signalling system with an information transfer capacity suitable for the GSM system, is the CCITT signalling system No. 7. The necessary adaptation for use in mobile communication systems must be done in co-operation with the relevant CEPT groups.

p. **From the subscriber's point of view, the quality for voice telephony in the GSM system shall be at least as good as that achieved by the first generation 900 MHz analogue systems over the range of practical operating conditions.**

q. **The GSM system shall enable implementation of common coverage PLMN's.**

r. **Protection of signalling information and network control information must be provided for in the system.**

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B/MOO Mats Dahlin

Status of Pan-European Digital Cellular Radio System

In 1983, CEPT formed a special group, GSM (Groupe Speciale Mobile) with the task to define a Pan-European Public Mobile Radio Network.

The time schedule is to:

- Establish an outline specification by December 1986. (Basic parameters, e. g. analog/digital, multiple access scheme, bit rates, etc.).

- Provide detailed system specification by December 1988 (complete air-interface, network interface).

- Make the system ready for service by December 1990.

Further details are given in attachment 1.

At this stage CEPT/GSM has provided the basic requirements for the harmonized 900 MHz mobile system, see attachment 2.

The CEPT/GSM work is directed towards a digital cellular system, i. e. a system where digital speech transmission is used. The work is currently concentrated on the design of the digital radio system.

Several modulation and multiple access techniques are being considered, e. g.:

- FDMA transmission system
  e. g. 25 kHz radio channels with 16 kbps digital voice;

- Narrow-band TDMA transmission system
  e. g. 300 kHz radio carriers with 10 user time slots per carrier;

- Wide-band TDMA transmission system
  e. g. up to 10 MHz radio carriers combined with spread-spectrum technology.

ERICSSON is engaged in technical experiments with different technical solutions. The position of Ericsson is currently to obtain the required know-how in order to make a good system solution which is not only suitable for a Pan-European system, but can also be adopted as a world wide standard.

At present, an experimental system for FDMA has been developed by Ericsson, and radio field tests has been performed. The system uses a 16 kbps voice codec (RELP), which is designed to withstand high bit error rate. Furthermore, channel coding with bit interleaving and soft decision is used. The transmission rate after channel coding is 27 kbps, which is transmitted on a 25 kHz radio channel by the use of narrow-band, coherent GMSK modulation. Further details are given in attachment 3.

An experimental system for narrow-band TDMA is also under development. Transmisson rate is 300 kbps and 10 time slots of 1 ms length are available on each radio channel. Speech coder rate is 16 kbps. Modulation and channel coding is similar to the FDMA design. However, in order to operate in the time-dispersive multi-path environment, an adaptive equalizer is also incorporated. An overview presentaton is given in attachment 4.

Attachment 1.	GSM Action Plan.
Attachment 2.	Basic requirements for a harmonized mobile communication system (GSM 73/85).
Attachment 3.	Digital mobile telephony with improved spectrum efficiency, ERA Report T/UT 84:32. Presented at the Digital Mobile Seminar in Helsinki, February 1985.
Attachment 4.	Digital mobile telephony - performance, objectives and new possibilities, ERA Report T/UT 85:25. Presented at the Digital Land Mobile Workshop in Bologna, September 1985.

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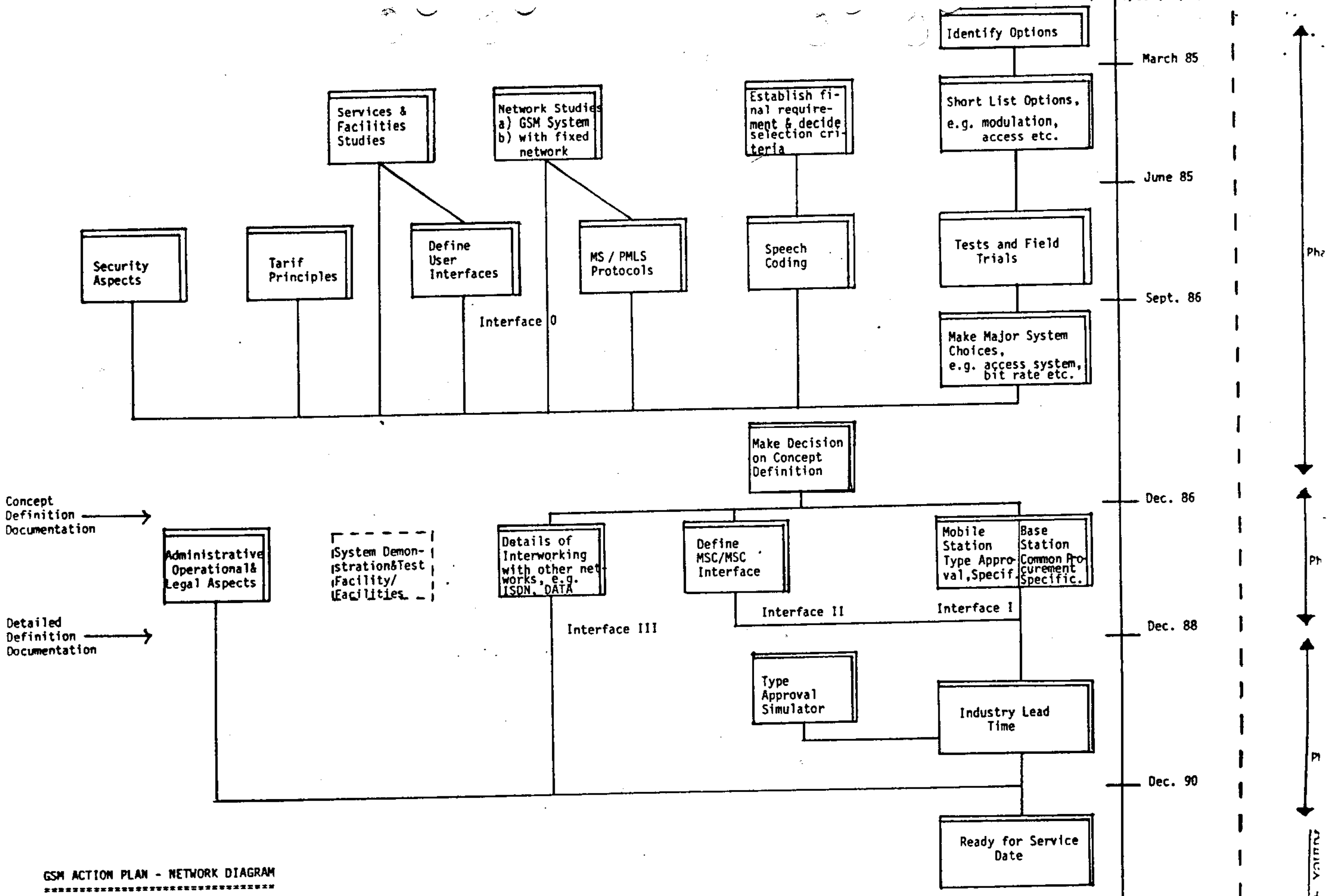

GSM ACTION PLAN - NETWORK DIAGRAM

CFPT CCH/GSM Meeting 25 Feb - 1 March 1985