Capacity of Digital Cellular TDMA Systems

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Abstract—**The market for cellular radio telephony is expected to increase dramatically during the 1990's. Service may be needed for 50% of the population. This is beyond what can be achieved with the present generation analog cellular systems. The evolving digital time division multiple access (TDMA) cellular standards in Europe, North America, and Japan will give important capacity improvements and may satisfy much of the improvements needed for personal communication. The capacity of digital TDMA systems is addressed in this paper. Capacity improvement will be of the order 5–10 times that of analog FM without adding any cell sites. For example, the North American TIA standard offers around 50 Erlang/km^2 with a 3-km site-to-site distance. However, in addition, the TDMA principle allows a faster handoff mechanism (mobile assisted handoff, MAHO), which makes it easier to introduce microcells with cell radius of, say, 200 m. This gives substantial additional capacity gain beyond the 5–10 factor given above. Furthermore, TDMA makes it possible to introduce adaptive channel allocation (ACA) methods. ACA is a vital mechanism to provide efficient microcellular capacity. ACA also eliminates the need to plan frequencies for cells. A conclusion is that the air-interface of digital TDMA cellular may be used to build personal communication networks. The TDMA technology is the key to providing efficient handover and channel allocation methods.**

I. Introduction

High capacity radio access technology is vital for cellular radio. Digital time division multiple access (TDMA) is becoming a standard in the major geographical areas (Europe, North America, and Japan). Digital technology is capable of giving higher capacity than in the present analog FM systems. For example, the demonstration performed by Ericsson in 1988 [1] showed that multiple conversations could be carried out on a 30-kHz radio channel without degradation in radio range or carrier-to-interference (C/I) radio performance.

This paper gives capacity estimates for all digital TDMA standards and compares this with analog FM. First the different standards are described in Section II. Section III deals with a capacity comparison between different systems. In Section IV the benefits of digital for microcellular operation is discussed.

II. Digital Cellular Systems

Digital cellular technology will be introduced around 1991. There are three emerging standards, the pan-European GSM system specified by European Telecommunications Standards Institute (ETSI), the American Digital Cellular (ADC) specified by Telecommunications Industry Association (TIA), and the Japanese Digital Cellular (JDC) specified by the Ministry of Post and Telegraph (MPT).

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The standardization bodies have had different driving forces, time plans and scopes of work but all three have had in common that they address the lack of capacity in existing analog systems and that the new systems are digital and use TDMA as access method.

The GSM and ADC systems will be described in more detail in the following text. The JDC standardization work started recently, and detailed decisions have not yet been made. In short, the JDC system can be described as being very similar to the ADC system regarding the digital traffic channels whereas it has more in common with the GSM system in the lack of backward compatibility and in that the scope of the work is a single phase standardization process. Table I describes some of the characteristics regarding the air-interface for all the three systems.

A. The GSM System

The GSM system specifies many interfaces but only a part of the air-interface will be considered here. The frame and slot structure is shown in Fig. 1. There are also a super- and hyperframe (not shown in the figure) for various purposes, e.g., synchronization of crypto and provision for mobiles to identify surrounding base stations.

There are eight voice channels on one carrier (full rate) with the capability to introduce half rate speech codecs in the future. The carrier spacing is 200 kHz. Thus 25 kHz (200/8) is allocated to a full rate user. In all, there is a bandwidth of 25 MHz giving 125 radio channels i.e., 1000 traffic channels.

The gross bit rate is 270.8 kb/s. The modulation scheme is GMSK with the normalized pre-Gaussian filter bandwidth equal to 0.30 e.g., constant envelope allowing a class-C amplifier. The 33.85 kb/s per user are divided into

speech codec	13.0 kb/s
error protection of speech	9.8 kb/s
SACCH (gross rate)	0.95 kb/s
guard time, ramp up, synch.	10.1 kb/s

The overhead part could be defined to be 10.1/33.85 = 30%.

The bits in a speech block (20 ms) consist of two main classes according to sensitivity to bit errors. The most sensitive bits (class 1) are protected by a cyclic redundant check (CRC) code and a rate = 1/2 convolutional code with constraint length equal to 5. The coded speech block is interleaved over eight TDMA frames to combat burst errors. To further enhance the performance, frequency hopping, where each slot is transmitted on different carriers, can be used by

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TABLE 1
AIR-INTERFACE CHARACTERISTICS OF THREE DIGITAL CELLULAR STANDARDS

	Europe (ETSI)	North America (TIA)	Japan (MPT)
Access method	TDMA	TDMA	TDMA
Carrier spacing	200 kHz	30 kHz	25 kHz
Users per carrier	8 (16)	3 (6)	3 (tbd)
Modulation	GMSK	$\pi/4$ DQPSK	$\pi/4$ DQPSK
Voice codec	RPE 13 kb/s	VSELP 8 kb/s	tbd
Voice frame	20 ms	20 ms	20 ms*
Channel code	convolutional	convolutional	convolutional*
Coded bit rate	22.8 kb/s	13 kb/s	11.2 kb/s
TDMA frame duration	4.6 ms	20 ms	20 ms*
Interleaving	≈ 40 ms	27 ms	27 ms*
Associated control channel	extra slot	in slot	in slot*
Handoff method	MAHO	MAHO	MAHO*

*Ericcson proposal

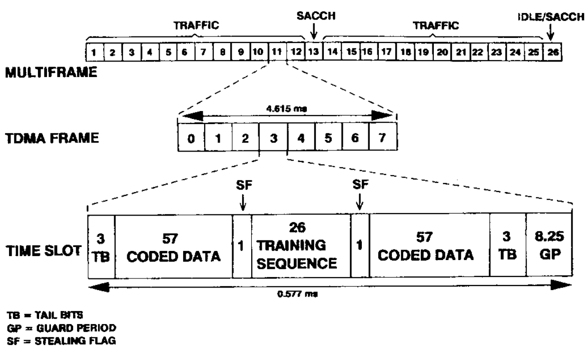

Fig. 1. GSM slot and frame structure showing 130.25 b per time slot (0.577 ms), eight time slots/TDMA frame (full rate), and 13 TDMA frames/multiframe.

the system. This is a mandatory function for the mobile but is optional for the system operator to use.

There are two control channels associated with the traffic channels, the slow and the fast ACCH. The FACCH is a blank-and-burst channel and replaces a speech block whenever it is to be used. Two frames in the multiframe (see Fig. 1) are allocated for the slow associated control channel (SACCH). With full rate users the second SACCH frame is idle. In a SACCH frame the slots are assigned in the same way as for traffic frames. The gross bit rate on this channel is 950 b/s and the net rate is 383 b/s. A SACCH message is interleaved over four multiframes.

With the fast growing number of subscribers anticipated in conjunction with smaller cell sizes it becomes increasingly important that the locating of mobiles can be done more accurately and faster than in the present analog systems. The method chosen by GSM is that the mobiles shall measure the signal strengths on channels from neighboring base stations and report the measurements to their current base station (Mobile Assisted Handoff (MAHO)). The land system evaluates these measurements and determines to which base station the mobile shall be transferred (handoff) if the mobile is about to leave its present cell or for other reasons would gain in radiolink quality by a handoff. The number of handoffs increases with the amount of traffic carried in a cell and the reduction of cell size. In analog systems where neighboring base stations measure the signal transmitted from a mobile, a very high signaling load is introduced on the links between base stations and the switch and also higher processing requirement in the switch. Thus a decentralized location procedure where each mobile is a measurement point will reduce the burden on the network.

Of the eight time slots in a TDMA frame, two are used on different frequencies for transmission and reception. In the remaining time the mobile can measure the received signal strength on a broadcast control channel (BCCH) from its own and surrounding base stations. These measurements are averaged over a SACCH block (480 ms) before they are transmitted to the base station using the SACCH. The maximum number of surrounding base stations contained in the measurement list is 32 but only the result from the six strongest ones is reported back to the land system. Thus the mobiles preprocess the measurements and the reports contain results from different base stations for every SACCH block. Since there is a possibility that the signal strength measurements can be affected by a strong cochannel, and thereby be highly unreliable, the mobile is required to identify the associated base stations on a regular time basis. Therefore, it is necessary for the mobile to synchronize to and demodulate data on the BCCH in order to extract the base station identity code. This code is included in the measurement report informing the land system which base station is measured.

The mobile performs this identification process in its idle TDMA frame. There is one of these per multiframe, see Fig. 1. For half-rate, this idle frame is used for SACCH for the new traffic channels created. The mobile measurement report also contains an estimate of the bit error rate on the traffic channel used. This additional information is useful to determine the radio link quality since the received signal strength measurement cannot indicate a cochannel interferer or severe time dispersion.

B. The ADC System

This standard covers only the air-interface. Another subgroup of TIA is currently dealing with the intersystem connection. Since there is a single analog standard in North America and roaming is already possible, it has been decided that the first mobiles shall be dual mode, i.e., they should be capable of operating on both analog and digital voice channels. This makes it possible for the operators to introduce

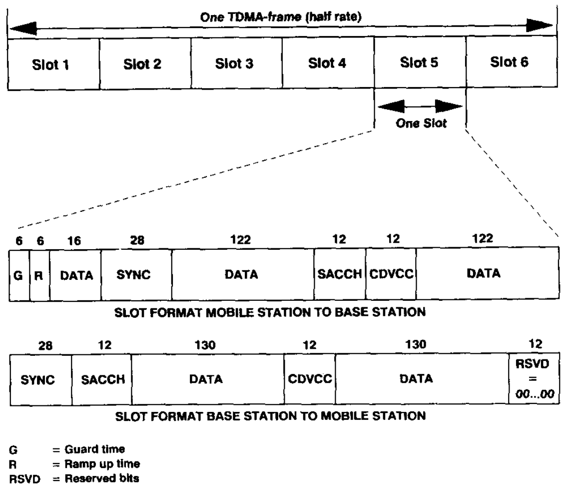

Fig. 2. ADC slot and frame structure for down- and uplink with 324 bits per time slot (6.67 ms) and 3(6) time slots/TDMA frame for full rate (half-rate).

digital radio channels according to capacity needs. In this first phase of digital technology the current analog control channels are used. Later on, provision for digital mode only mobiles will be made by introducing digital control channels.

With the dual mode requirement, it was natural to select a 30-kHz TDMA radio format. Each burst is 6.7 ms and for full rate users the TDMA frame length is 20 ms, see Fig. 2. Thus 10 kHz are allocated to a full rate user. In all, this gives 2500 traffic channels over a 25-MHz bandwidth.

The gross bit rate is 48.6 kb/s. The modulation scheme is differentially encoded $\pi/4$ QPSK with root-raised cosine pulse shaping and a roll off factor equal to 0.35. The 16.2 kb/s per user are divided into

speech codec	7.95 kb/s
error protection of speech	5.05 kb/s
SACCH (gross rate)	0.6 kb/s
guard time, ramp up, synch., color code	2.6 kb/s.

The overhead part could be defined to be $2.6/16.2 = 16\%$ (compare corresponding calculation for the GSM system). The color code is an 8-bit signature to provide the capability to distinguish between connections using the same physical channel i.e., cochannels. This signature is transmitted in each burst and is protected by a shortened Hamming code to form the 12-bit field CDVCC.

The 20-ms speech block consisting of 159 b has two classes of bits with different sensitivity to bit errors. The most sensitive class of bits is protected by a CRC code and then coded with rate $= 1/2$. The other part (class 2 b) is not protected at all. The channel coding method used for speech and signaling is a convolutional code with constraint length equal to six. The coding rate for speech and SACCH is 1/2 and for FACCH 1/4. The interleaving for speech (and FACCH) is diagonal over two slots. A SACCH message is distributed over 22 slots by means of a self-synchronized interleaving process. The net rate on SACCH is 227 b/s.

Mobile Assisted Handoff is also used in the ADC system. Perhaps the major difference in comparison to the GSM system is that the mobiles are not required to extract the base station identity code. In the dual mode phase of the ADC system there are no digital control channels on which to perform this task. There are only three time slots in a TDMA frame, and there is no idle frame as for GSM. Thus there is not enough remaining time to synchronize and demodulate data on another carrier without introducing high complexity in the mobile. Instead, there is the capability for a neighboring base station to identify a mobile, using the unique synch. word to identify a time slot and the CDVCC to distinguish the intended user from a cochannel.

Thus an implementation of the handoff process in an ADC system is that the land system evaluates the measurements from the mobile and lets the candidate base station verify that it can take over the call, before ordering the intended handoff. The MAHO is a mandatory function in the mobile but can optionally be turned on or off by the system. Thus, a traditional handoff implementation is also a possible method in which only information related to the traffic channel in use is considered.

The measurement reports contain the same information as in GSM (signal strength and estimated bit error rate) with the difference that in ADC the measurements from all base stations are reported, rather than only the six strongest. The list may contain up to 12 channels including the current traffic channel. For the same number of channels in the channel list, the GSM measurement reports are somewhat more accurate because of better averaging out the Rayleigh fading. The total number of samples with a certain time period is dependent on the number of TDMA frames within that time. There are 50 TDMA frames per second in the ADC system and approximately 216 per second in the GSM system. The reporting interval is once every second in the ADC system and once every 0.48 s in the GSM system.

C. The JDC System

As stated earlier, the JDC system is very similar to the ADC system i.e., it has a three-split TDMA air-interface. The main difference lies in the narrower channel bandwidth of 25 kHz compared to the 30-kHz bandwidth selected for the ADC system. The same type of modulation, $\pi/4$ DQPSK, as for the ADC system has been selected. To avoid extreme complexity in the power amplifier the gross bit rate has to be lower than in the ADC system (48.6 kb/s) and has been chosen to be 42.0 kb/s. The pulse shaping in the modulation scheme is root-raised cosine with a roll off factor equal to 0.5.

As was the case in North America, the speech and channel coding algorithm will be selected by testing candidates implemented in hardware. 11.2 kb/s has been selected for the total bit rate of the test. The difference between the gross bit rate per user (14 kb/s) and the protected speech rate (11.2 kb/s) is 2.8 kb/s and it will be allocated to the same functions as in the ADC system (see Section II-B) but the details will be different. Since the JDC system does not have any backward compatibility, all the control channels have to be specified within the first specification.

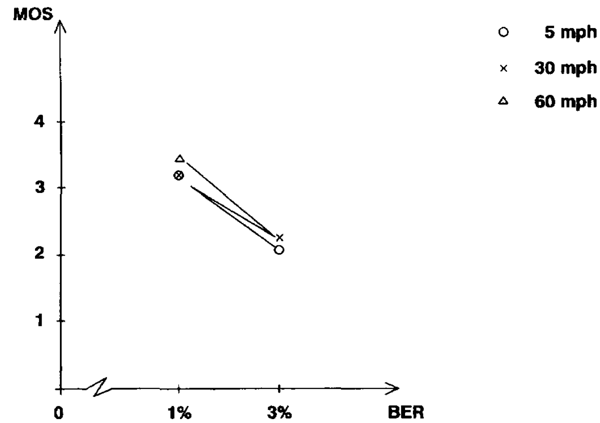

Fig. 3. Results from subjective listening tests (formal TIA tests) with a DCELP 8.7 kbs voice codec at different vehicle speeds.

III. Capacity of Second Generation Cellular Systems

To accurately evaluate the capacity for the emerging digital systems, field trials are needed. Until these systems are implemented and operated in a multicell environment under high traffic conditions, the capacity can only be estimated. Nevertheless, it is possible to evaluate the performance of the system since there is experience from speech codec evaluations, computer simulations and limited field trials e.g., [1]-[4]. This is also necessary because the system operators have to make cell planning assumptions and system suppliers have to design handoff algorithms and do traffic analysis.

A. C/I Performance of the ADC System

In the selection process of the speech codec for the ADC system, subjective listening tests of hardware implemented candidate speech and channel-coding algorithms under a channel imperfection environment was used. The result from this test can be used as a starting point to estimate the capacity of the ADC system.

Fig. 3 shows the result with the Ericsson candidate 8.7 kb/s DCELP speech codec [4]. This is a deterministic code excited linear predictive codec running at 8.7 kb/s and error protected with a convolution code at a total of 13 kb/s. The mean opinion score (MOS) is the relative subjective quality perceived by the listening group. In the channel between the coder and the decoder bit errors were introduced. The bit error sequence was generated with a flat Rayleigh fading model using three-split TDMA and QPSK modulation. Soft information was not included in the test procedure. The channel model was tuned to give a bit error rate (BER) of 1% and 3% at a vehicular speed of 5, 30, and 60 mi/h. The 0% BER case is the basic voice quality, i.e., with an undisorted channel. The conclusion which can be drawn from the figure is that the speech quality is not sensitive to vehicular speed. At high vehicular speeds the channel coding and interleaving can combat the burst errors, whereas the interleaving (27 ms) is not long enough at low vehicular speeds. On the other hand, the speech frame repetition mechanism is more effective at lower speeds because incorrect speech blocks generally are followed by a number of correct ones.

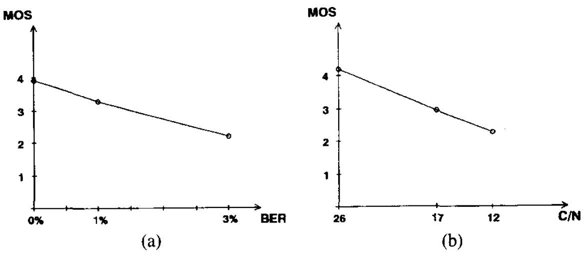

Fig. 4. Comparison of MOS results between (a) digital and (b) analog.

In Fig. 4 the effect of vehicular speed has been taken out by averaging over the three different vehicular speeds. In the listening test there was an analog FM 30-kHz reference so that the results for the candidates could be evaluated against the quality perceived in the present analog system.

A design criterion used in cell planning of present analog FM systems is to achieve a C/I power ratio not less than 18 dB over most of the cell area (e.g., 90%). The speech quality at 18 dB C/I is about equal to a carrier-to-noise power ratio (C/N) measured over 30 kHz equal to 17 dB. As seen in Fig. 4(b) this subjective speech quality for analog is an MOS of 3.0. The BER before channel decoding giving the same subjective quality for digital is about 1.5% (see Fig. 4(a)).

We can now translate the BER figure to derive the minimum required C/N and C/I for digital. To this end, a computer implemented ADC radio format for performance evaluation is used, see Figs. 5 and 6. From Fig. 5, a BER at the demodulator output (class 2) of about 1.5% corresponds to about 18-dB C/I. From Fig. 6 we derive the peak E_b/N_b to be 15 dB. The peak E_b/N_o is converted to peak C/N over 30 kHz by 48/30 (+2 dB), e.g., the required peak C/N for digital is about 17 dB. (Here we define C/N as the carrier-to-noise power ratio measured in a 30-kHz bandwidth and $E_b N_o$ as the bit energy divided by the noise spectral density. $E_b N_o$ can also be viewed as the ratio between carrier power and noise power measured in a bandwidth equal to the bit rate.) This means that a 0.6-W peak power digital mobile will have the same coverage as a 0.6-W analog mobile. Thus the power consumption needed for the same coverage is only a third for the digital compared to the analog because of the three-split TDMA. In summary, the speech quality perceived in analog with 18-dB C/I is about the same as for digital at 18-dB C/I. Defining coverage with the same quality, both systems need about 17-dB C/N.

So far, the enhancement of using soft information in the channel decoding process has not been considered because the speech test did not provide any soft information. At a C/I of 18 dB the gain of using soft information is about 3 dB, see Fig. 5. Since only the performance of the coded bits (class 1) is affected by the use of soft information, we assume here for further calculation that the overall gain is around 2 dB. This means that the minimum required C/I for digital is 16 dB.

B. C/I Performance of the GSM System

We can now make use of the information derived for the ADC system to determine the necessary C/I for the GSM system by comparing BER curves for the two systems. Fig. 7 shows simulated BER performance for the GSM system for

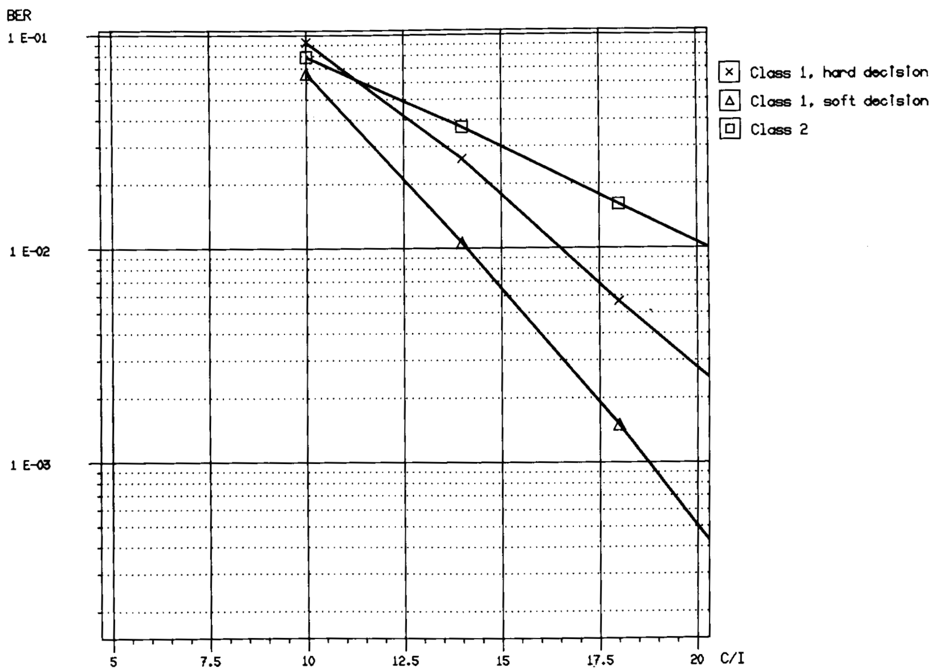

Fig. 5. Performance of ADC in terms of BER versus C/I for a Rayleigh fading channel at 55 mi/h.

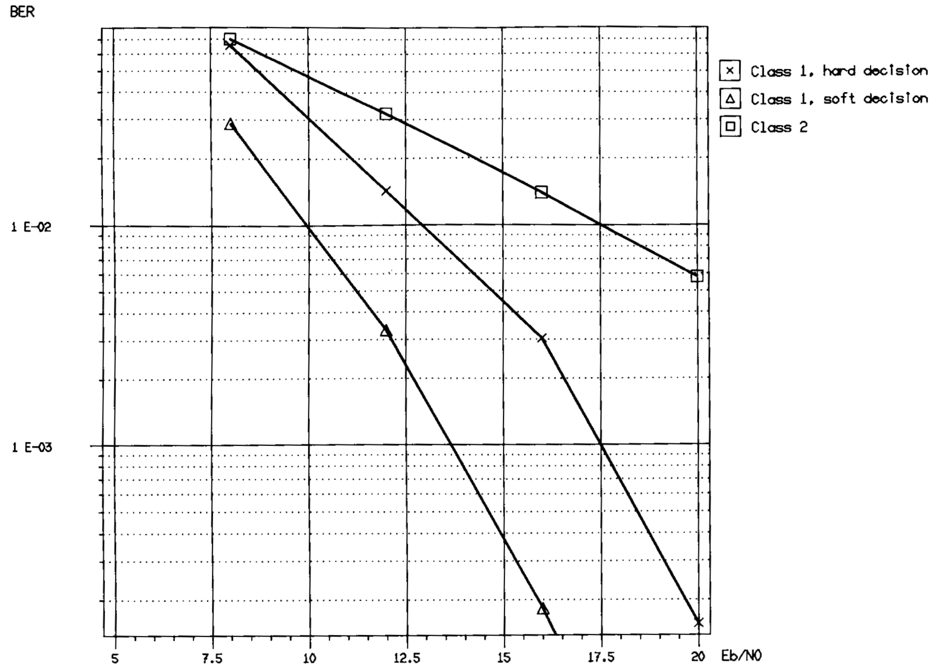

Fig. 6. Performance of ADC in terms of BER versus C/I for a Rayleigh fading channel at 5 mi/h.

class 1 and class 2, respectively. The vehicle speed is 25 mi/h and frequency hopping is used. The channel contains a small amount of time dispersion with a delay spread of about 1 μs which represent a typical case in urban environment. All paths are within the required capability of the equalizer.

One method to compare C/I performance is to use the BER of class 1. At the required C/I = 16 dB the BER in class 1 is 0.4% for ADC, see Fig. 5 (soft information). From Fig. 7 it is seen that the C/I value for GSM corresponding to a 0.4% BER in class 1 is C/I = 9 dB. It can be questioned if

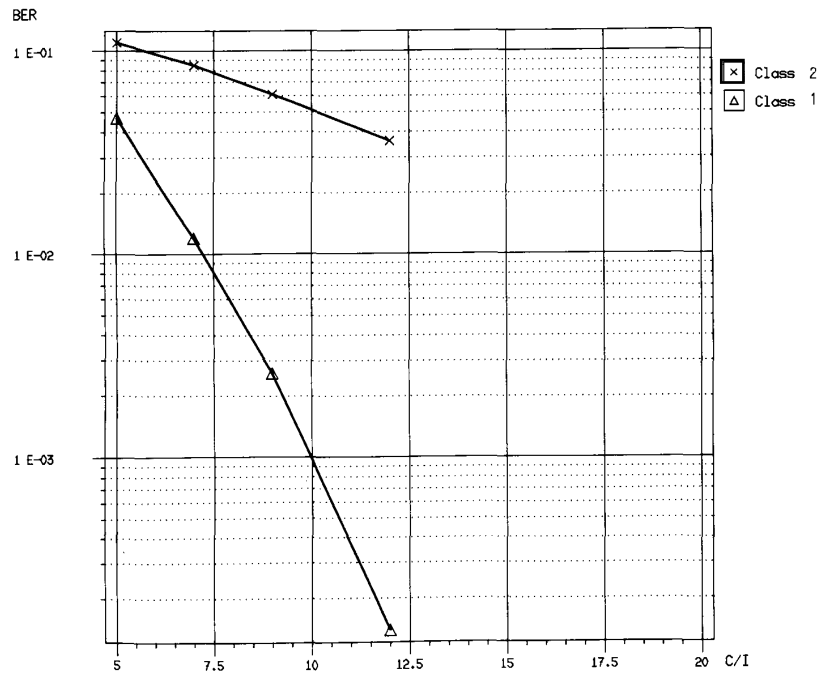

Fig. 7. Performance of GSM in terms of BER versus C/I for a typical urban fading channel at 25 mi/h.

this represents the same speech quality because we made the comparison with equal BER in class 1 only, neglecting the much higher BER in class 2 (8%) in the GSM system whereas the ADC class 2 b only exhibits 2.3% BER. However, this comparison is also confirmed by a GSM voice codec MOS-test. Fig. 8 shows test for the GSM voice codec [6]. It is seen that the MOS is only marginally affected at 8% BER (class 2). Qualitatively this difference between GSM and ADC codecs is due to the following.

1) The most important factor why the ADC codec can operate at 8 kb/s whereas the GSM codec operates at 13 kb/s, and giving about the same basic voice quality, is the reduction of the bit rate in representing the so called residual signal, see [4]. Thus the fewer bits describing the residual signal in the ADC codec are more sensitive to bit errors. For the same speech quality degradation, the GSM system can handle a higher BER in the class 2 bits because the bits for describing the residual belong to class 2.
2) With frequency hopping, the distribution of bit errors will be more random. This affects the class 2 bits, although unprotected, because they are interleaved over 8 different carriers, i.e., the interleaving in GSM is more powerful than in ADC.

C. C/I Performance of the JDC System

At the time of writing this paper, the selection for the JDC system has not yet taken place. Because of the similarities with the ADC system, at least the same minimum C/I is required. Since the total bit rate (11.2 kb/s) is less than for the ADC system we add, somewhat arbitrarily, 1 dB to give

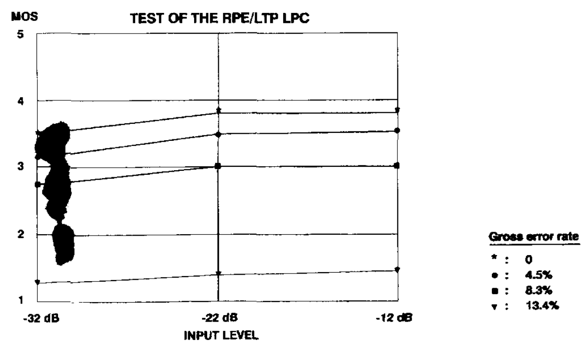

Fig. 8. Subjective MOS tests of GSM voice codec RPE/LTP (from [6]) as measured in the characterization tests based on channel simulation by GSM/SCEG in 1988. MOS is given for different error rates on the channel (Class 2). The relation between Class 1 and Class 2 bit errors is shown in Fig. 7.

17 dB. Antenna diversity mandatory for the mobiles is considered in the Japanese standardization process. Depending on the implementation, this can give a gain of 3-7 dB. For the capacity calculation, we assume 5 dB, thus the required C/I would be 12 dB.

D. Influence on C/I of Diversity and Frequency Hopping

Antenna diversity can also be implemented in the mobiles in the ADC system. In fact, the specification of the radio format has been derived so that a simple antenna diversity method, only using one RF receiver, can be implemented. With this simple antenna selection scheme, the gain is about 4 dB, thus an ADC system with mobiles using this type of diversity the minimum required C/I is also about 12 dB.

TABLE II
MINIMUM REQUIRED C/I IN DIFFERENT DIGITAL SYSTEMS (IN FADING CHANNEL CONDITIONS)

GSM		ADC		JDC	
With frequency hopping	Without frequency hopping	With antenna diversity	Without antenna diversity	With antenna diversity	Without antenna diversity
9 dB	11 dB	12 dB	16 dB	13 dB	17 dB

TABLE III
APPROXIMATE CAPACITY IN ERLANG PER KM^2 FOR ONE FIRST GENERATION ANALOG SYSTEM (AMPS) AND SECOND GENERATION DIGITAL SYSTEMS IN EUROPE, NORTH AMERICA, AND JAPAN ASSUMING A CELL RADIUS OF 1 KM (SITE DISTANCE OF 3 KM) IN ALL CASES AND 3 SECTOR PER SITE. THE LEE-MERIT IS THE NUMBER OF CHANNELS PER SITE ASSUMING AN OPTIMAL REUSE PLAN

	ANALOG FM	GSM pessimistic	GSM optimistic	ADC pessimistic	ADC optimistic	JDC pessimistic	JDC optimistic
Bandwidth	25 MHz	25 MHz		25 MHz		25 MHz	
Number of voice channels	833	1000		2500		3000	
Reuse plan	7	4	3	7	4	7	4
Channels per site	119	250	333	357	625	429	750
Erlang/km^2	11.9	27.7	40.0	41.0	74.8	50.0	90.8
Capacity gain	1.0	2.3	3.4	3.5	6.3	4.2	7.6
(Lee merit gain)	(1.0)	(2.7)	(3.4)	(3.8)	(6.0)	(4.0)	(7.2)

In some countries in Europe the frequency band allocated for the GSM system is presently used for current analog systems (NMT 900 and TACS). When introducing the GSM system in those countries, not all of the band can be used. The remaining band has to be divided by the number of operators. In most countries the number of operators will be at least two. This will make the use of frequency hopping more difficult, or the available band is not large enough to provide carriers with independent fading resulting in performance degradation to ideal conditions. The degradation if no frequency hopping is used is about 2dB. If frequency hopping is used but the available band is limited, the degradation will be in the range from almost no degradation to 2 dB.

E. Importance of Handoff Algorithm

A handoff margin could be defined as the difference between the C/I used for frequency planning and the C/I where the speech quality is just about acceptable. A tight frequency plan will reduce the handoff margin. The more diversity used to lower the required C/I, the less the handoff margin will be. This effect arises because the BER curves for the coded bits will be more steep. Only a little reduction of the C/I will result in unintelligible speech. This is also valid for the signaling system, e.g., the mobile measurement reports.

A system designed for a very low C/I and operated near the C/I limit needs a fast and reliable handoff procedure and adequate error protection of the signaling messages. Thus, it is not only the static C/I figure that determines the possible frequency plan but also the dynamics (rate of C/I changes) in the system.

F. Capacity Comparison

We can now summarize the minimum required C/I, see Table II. Before a capacity comparison can be made we need to assume a channel plan for each system. Based on [7] we can convert the C/I figures in Table II to channel plans. We assume a sectorized plan with three sectors/site. The values in Table II will correspond to the following reuse plan:

GSM: 3 or 4 site reuse plan corresponding to optimistic or pessimistic case;
ADC: 4 or 7 site reuse plan corresponding to optimistic or pessimistic case;
JDC: as ADC above.

The capacity result is shown in Table III where 25 MHz has been used as the total bandwidth for all systems. A blocking figure of 2% has been used. The capacity gain figure is obtained by dividing the Erlang per square kilometer figure relative to the analog FM. In parenthesis an additional row is showing the capacity gain obtained by using the Lee formula given in [5].

As seen in Table III the digital systems can achieve a capacity increase of up to 7.6 times that of analog FM for the same number of sites in the system. If antenna diversity is used, 90.8 Erlang/km^2 can be achieved with a 3-km site to site distance. Even without diversity substantial capacity improvement can be obtained. For ADC a pessimistic capacity gain factor is 3.5. For GSM a similar pessimistic capacity gain factor is 2.3. This indicates that the GSM system has lower capacity than the ADC system. However, on the other hand, the GSM system will be the first system to make it possible to introduce the half-rate channels. The voice-coding technology already exists, although not mature enough to meet all the requirements of the GSM, because the gross bit rate for the GSM half-rate is equal to the JDC full-rate. This will improve the GSM system regarding capacity by almost a factor of two.

The next step would be the introduction of the half-rate in the ADC system giving a capacity gain factor in the range of 7–12 (depending on whether an optimistic or pessimistic frequency plan is used). In summary, we can conclude that digital TDMA can achieve a capacity gain of up to around 10 times as compared to analog systems. This is without introducing microcells. On top of this the TDMA principal makes it more attractive to introduce microcells which will be addressed in the next section.

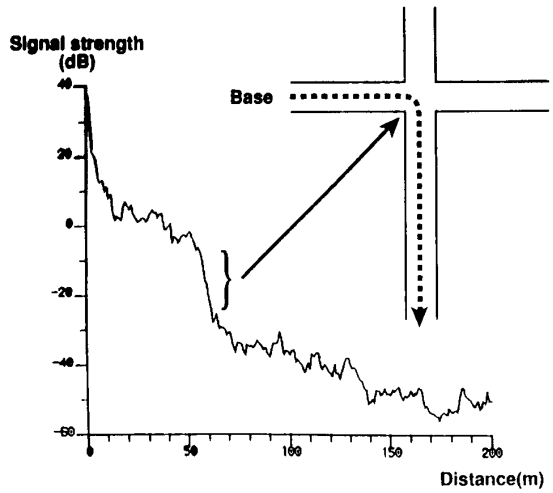

Fig. 9. Example of radio field strength drop in a street corner.

TABLE IV
ILLUSTRATIVE (PESSIMISTIC) EXAMPLE OF CHANNEL PLAN AND CAPACITY WITH MICROCELLS AND FIXED CHANNEL ALLOCATION GROUPS (WITH d_0 ASSUMED TO BE 400 M).

	With Microcells 10 MHz Umbrellas	15 MHz Street Microcells			Without Microcells 25 MHz Umbrellas only
Site–site distance	3 km	0.8 km	0.4 km	0.2 km	3 km
Reuse plan	7-cell sector hex	25 cell omni (rect)	49 cell omni (rect)	100 cell omni (rect)	7-cell sector hex
Channels per site	140	60	30	15	357
Channels per km^2	20	97	195	390	45
Total channels/km^2		117	215	410	45
Total Erlang/km^2		107	174	280	41

IV. MICROCELLULAR OPERATION

A. Microcells and Handoff

The digital TDMA systems are using the MAHO method, which is a more rapid handover method enabling the call to be handed over between base stations at much faster rate than in first generation analog cellular. This is important since it makes the introduction of microcells much easier.

For street microcells the street corners will be the dimensioning case since radio field strength may drop 20 dB in a street corner over a short distance, see Fig. 9. A total reaction time of 1–2 s is needed.

MAHO means that the mobile stations (MS) makes measurements of the radio field strength from surrounding base stations (BS) and report these to the BS in use. It is possible to do these measurements in the MS due to the TDMA structure since there is idle time between received and transmitted time slots as explained in Section II.

The cochannel interference situation in microcells is difficult to predict due to very different radio propagation situation at different sites. This makes traditional frequency planning with fixed channel allocation to every site difficult to pursue. A very conservative reuse frequency plan must be employed. In addition, street microcells will not provide full coverage, i.e., overlaid umbrella cells must also be used. For both these reasons capacity gain will not be proportional to the inverse of the squared cell radius.

An example may illustrate this. Street microcell measurements show propagation results of the following kind. Close to the BS propagation will be close to free space, i.e., a loss of 6 dB per octave in distance. Beyond a certain breakpoint d_0 the propagation loss will be higher. We assume a 12-dB per octave loss beyond a breakpoint $d_0 = 400$ m which is a typically figure encountered at 900 MHz. When the cell radius is decreased below the breakpoint d_0 there will be a C/I penalty for a given distance ratio between wanted and unwanted signals. This loss is of the order 6 dB per octave in cell radius for this example when the cell radius is below d_0. This 6-dB penalty roughly means that the number of channel groups N has to be doubled. In other words decreasing the cell radius by a factor of 2 will increase the capacity per square kilometer by a factor of 2 (instead of the expected factor 4).

In practice there will be some additional loss both due to less trunking efficiency and due to the need to use umbrella cells. Table IV gives an example where it is seen that the total capacity is not increased beyond 280 Erlang per km^2 even with very small street microcells (100-m radius). Even if 280 Erlang per km^2 is a large figure it must be noticed that

TABLE V
COMPARISON OF ADAPTIVE AND FIXED CHANNEL ALLOCATION IN INDOOR PROPAGATION ENVIRONMENT

With adaptive channel allocation		With fixed channel allocation		
Total number of traffic channels in the system	Simulated capacity per site	Required number of channels per site	Required reuse cell plan	Total number of required traffic channels in the system
16	1.1E	5	27	135
32	2.5 E	8	27	216

with a 100-m radius there will be a substantial number of handoffs per call, i.e., the handoff failure rate per call may be too high.

The above indicates that microcells alone do not solve all problems. Capacity may not increase dramatically when cell radius is decreased unless the investment in new cell sites is dramatically increased. Table IV is only an example but it illustrates the phenomenon with microcells. In practice it may be possible to plan frequencies in a more clever way and thus obtain higher capacity than indicated in Table IV.

B. Adaptive Channel Allocation

With adaptive channel allocation (ACA) there is no fixed frequency plan. Instead each BS is allowed to use any channel in the system. The ACA procedure is very attractive in a TDMA frame structure. The basic idea is to allocate channels depending on 1) the actual traffic situation and 2) the actual interference situation.

Early work only took 1) into account with relatively poor capacity gain as a result. In more recent proposals [8]-[10] it has been proposed to dynamically assign channels to every *call* instead of assigning channels to every *cell*.

Channel allocation is done both at call setup and at handoff. Due to the TDMA structure it is possible for the MS to continuously search for new better channels also during the call. An efficient ACA procedure requires that interference measurements of unused channels are performed both in the BS and in the MS. The TDMA format allows that interference measurements are done in the MS during conversation by measuring in idle time slot(s).

The ACA method is built into the standard for the Digital European Cordless Telecommunication (DECT) system specified by ETSI. DECT is a 24-slot TDMA system operating with a carrier spacing of 1.73 MHz in the 1880-1900-MHz band [11]. Similar ACA methods can be built into modification of both the GSM and the ADC standards. Simulations of ACA have revealed large capacity gain. This is particularly true when the number of available channels is relatively few. See Table V.

In a large system the capacity gain is 50% or more [8], [10]. In small systems capacity gains of 3-4 times has been found in [9]. An example is given in Table V from [9]. This shows that separate channel sets can be set aside for use by ACA for indoor picocells and outdoor microcell.

This means that ACA is an extremely powerful method for the following reasons: 1) Microcells and picocells can be deployed without frequency planning and 2) capacity gain in micro- and picocells operation is significant.

The advantages of ACA are from a capacity point of view:

i) Trunking efficiency loss is reduced almost entirely since there is no fixed allocation of channels to each BS.
ii) It is not necessary to take worst case propagation condition into account. With ACA one can design against the average interference situation as opposed to the case with fixed allocation where the worst possible interference situation must be used as a design criteria to choose the size of the channel group N.

ACA gives significant capacity improvement varying from a factor 1.5-2 in large systems to a factor of 4-6 in small systems. The large capacity gain in small systems (i.e., small allocated bandwidth) is due to an extra trunking efficiency.

V. CONCLUSION

The evolving digital TDMA cellular standards will provide a substantial capacity increase in short term perspective. In the medium term perspective half rate voice coding technology will provide a further improvement yielding a capacity gain of around 10 times that of the present analog cellular technology. In the long range, microcellular technology combined with ACA will move this into the personal communication capacity range.

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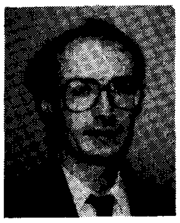

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