MASTER

# ATM in an indoor radio network : a feasibility study

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# ATM IN AN INDOOR RADIO NETWORK; A FEASIBILITY STUDY.

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ABSTRACT.

The subject of this thesis is to study if the 'Asynchronous Transfer Mode (ATM)' can be employed in indoor radio networks.

ATM can be applied to an indoor radio network, using a central station with line-of-sight link to all other terminals. The ATM system complexity increases when bitrates above 1 to 2 Mbit/s are applied.

Assuming low error probability on the radio channels and no constraints for delay, the system efficiency is likely to be somewhere in between 60% and 90%, depending upon traffic statistics. Stringent delay constraints cause drastic reduction of system performance.

CONTENTS.

# 1. INTRODUCTION.

Since 1988 the research programme of the Telecommunications Division of Eindhoven University of Technology contains a project in which various items in the field of inhouse radio are studied. The objective of this project is to examine the feasibility of a high-capacity indoor wireless network. The project is conducted by Peter Smulders. Research is carried out in cooperation with many institutes by means of participation within COST (European Cooperation in the field of Scientific and Technical Research), i.e. COST 231/WG3.

The use of radio networks within an office building, a factory, a warehouse, a hospital etc. implies several advantages over cable based networks. It would free the users from cords tying them to particular locations. In addition the mobility would offer the flexibility of changing or creating various communication services without the need of expensive rewiring.

The present generation of indoor radio systems is primarily designed for speech and low data services, and cannot satisfy the demands in future broadband services. The appearance of powerfull note-book laptop computers for video, voice and data for instance, gives rise to anticipate for the need of flexible broadband radio interfaces in the near future. Therefore current research and development activities in the field of indoor radio are directed towards a large increase of capacity per user.

The main topic of broadband networks is how to join data traffic of different sources. For the future (fiber based) 'Broadband Integrated Service Digital Network (BISDN)', the pros and cons of classical STM (Synchronous Transfer Mode) and ATM (Asynchronous Transfer Mode) have been analysed. As a result the CCITT SG XVIII has decided to choose ATM.

The subject of this report is to study if ATM can be employed in indoor radio networks. Because radio networks are essentially different from wired networks, some extra problems arise. The report starts with a description of the ATM principle and the general features of an indoor radio network. In chapter 3 the performance of signalling over radio channels is studied, to investigate if radio channels can meet the stringent BER-objectives for ATM. Next we study the performance of other multiaccess schemes for both comparative reasons and the fact that ATM needs an access scheme to control the network. Finally we present an radio-based ATM protocol and discuss its efficiency and delay constraints.

An appendix is included which contains a description of a simulation program.

## 2. The Asynchronous Transfer Mode.

In this section we introduce ATM [1] as a multiplexing method in (fiber-based) networks and look at the general properties of an indoor radionetwork based on ATM.

### 2.1 What is ATM ?

In the case of ATM, information transfer between terminals A and B happens as follows (Fig. 2.1): Terminal A sends a certain amount of information - called a data block or cell - intended for terminal B into the network. The nearest switching point puts the cell in a buffer. On account of a completed signalling procedure (the 'call setup'), the switching node knows the

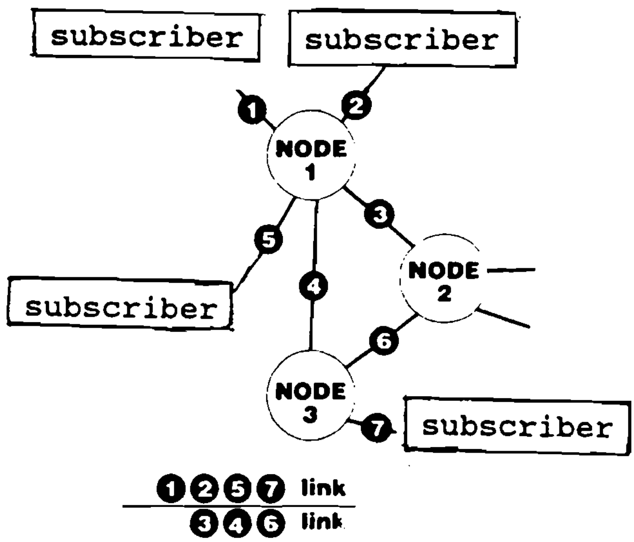


*Fig. 2.1: Concept of an ATM network.*

direction to the next switching node. In such a switching node the cell has to join a queue of earlier arrived cells which are also being transported via this switching node. This queuing causes a varying delay. A served cell will be transported to the next switching node. By repeating this procedure, the cell travels from switching node to switching node and will finally reach terminal B.

A cell (fig. 2.2) consists of a header and an information field. In the latest CCITT SG XVIII meeting the header size is fixed on 5 octets and the information field is fixed on 48 octets (as a compromise between the European 32 octets proposal and the American 64 octets proposal). Routing information is derived from information contained in the header. The information in the header depends on the interface used (UNI: User to Network Interface, or NNI: Network Node Interface).

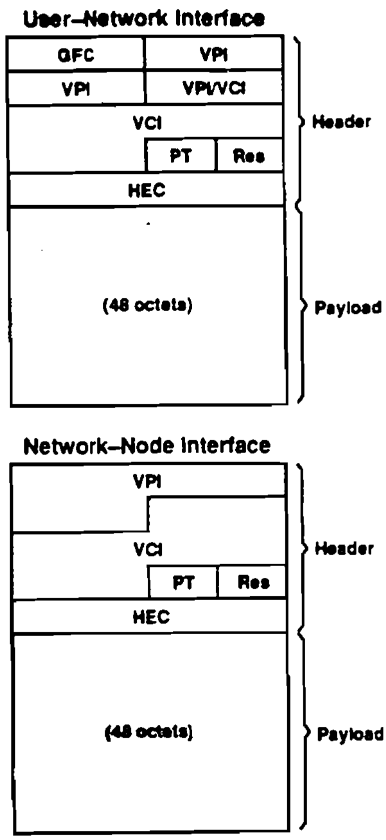


*Fig 2.2: CCITT ATM cell structure recommendations.*

All the messages that terminals and switching points exchange for the purpose of setting up, maintaining en closing a connection, is called signalling. An ATM connection between terminals and switching points or between switching points themselves, will probably have a permanent channel which is exclusively reservated for signalling purposes.

Signalling enables terminal A and B to exchange more than one cell during one session. It is said that A and B have a 'virtual connection' during the session. The network as a whole reserves a certain amount of bandwidth or transmission capacity to support the connection. This is done because the network has no a priori knowledge of the moment at which the terminals want to communicate. The individual switching points distinguish a connection by identifying the label of each incoming cell (fig 2.3). On the basis of this label the switching point recognizes the cell as belonging to the connection A-B and sends it in the correct direction. On behalf of the next switching point the cell can get a new label.

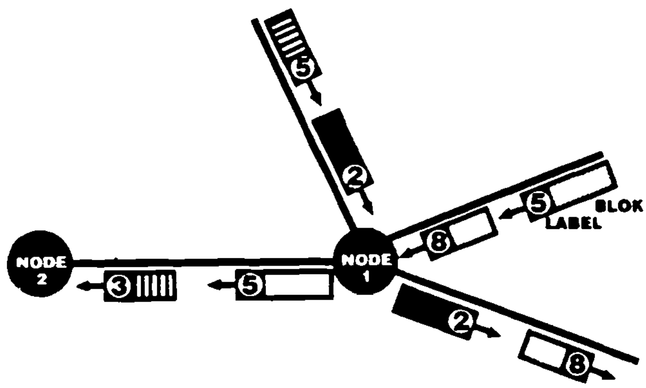


*Fig 2.3: Routing concept of an ATM switching point.*

As mentioned before, 'A' stands for 'Asynchronous'. This seems to implicate that ATM uses asynchronous transmission techniques, which is wrong. ATM is meant for data transmission over synchronous transmission links of the future ISDN. The transfer mode is called 'asynchronous' since there is no time-relation between the source or destination and the network; they may operate at different bitrates. Cells allocated to the same connection may exhibit an irregular pattern, as cells are filled according to the actual demand. This is different from STM, which maintaince a fixed sequence (fig. 2.4). The buffers take care of different bitrates between terminals and the network.

The bitrates used for ATM interfaces are dominated by the perspective of broadband communication. BISDN should make digital television (HDTV) possible and therefore bitrates of

150 Mbit/s or even higher values have been proposed for ATM user interfaces. If ATM will become popular as a method of transporting both data and voice signals, it has to be compatible with the existing PCM networks. This because there has already been made an enormous investment in PCM networks.

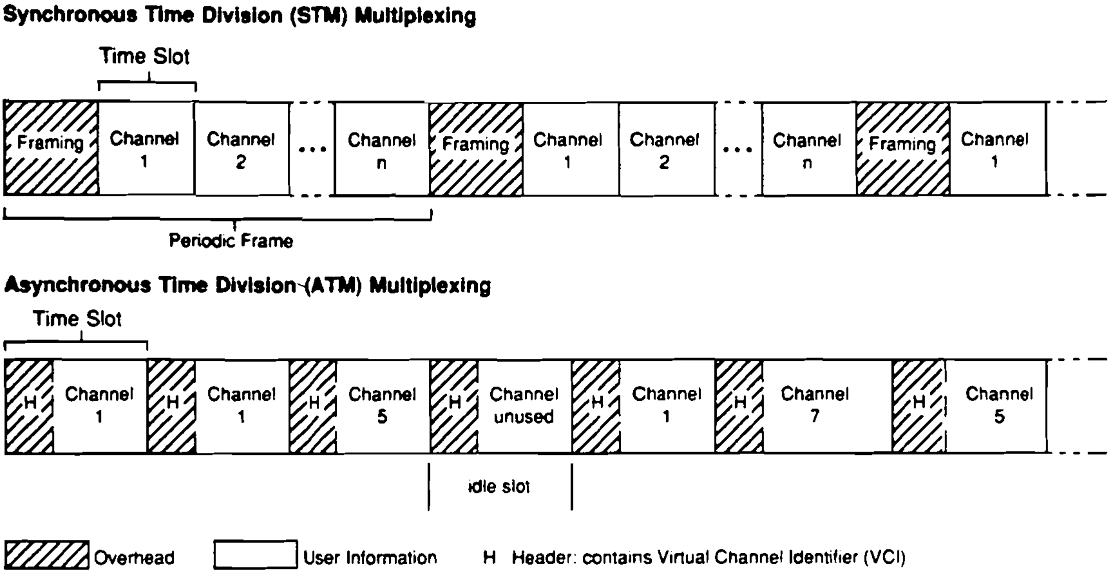


*Fig 2.4: STM and ATM principles.*

The existing data networks meet high requirements. The most stringent requirements relate to the acceptable errorrate, the type of communication service ('connectionless' or 'connection oriented') and the permitted delay [2]. Error control, especially when it concerns the overflow of a queue, is very important in a ATM network and it depends on how the protocol combats congestion. From this point of view the ATM network should be 'connection oriented' because the principle is based on a high transmission quality. Therefore it doesn't need 'link-by-link error control' (i.e. rectifying errors between network nodes). 'Connection oriented' networks can regulate the length of the queues and therefore the errors as a result of cell loss. Would the network be 'connectionless' then it has no control over the network load. This leads to a low network performance with respect to errors. The use of 'end-to-end error control' (i.e the terminals rectify the errors) as recorded in the ATM definition, would then be impossible.

Although much more can be told about ATM and its implementation, we restrict ourselves to this brief introduction. More information can be found in [2 t/m 7] but is not relevant for understanding the remaining part of this report. From now on we will focus on the properties of an indoor radio network and its implications on ATM.

**2.2 ATM and the indoor radio network; general properties.**

In this section we will give a description of the indoor radio network and the impact of radio on network properties, using the basic ATM principle.

The radio network can be described as follows: within a room (30x30m.) there are N terminals which all have been equiped with an omnidirectional antenna. In principle these antennas provide a direct radio link between any of the N terminals. As a result of fading and other reasons an unknown part of these links will not be reliable for a certain time. The reliable links, connecting the terminals, can be represented as a graph which varies in time.

The communication medium can be seperated in M frequency bands. All frequency bands will be above 30 GHz. The reason for this is that frequencies below 30 GHz are already being used. In the case of radio all terminals share the same medium, so all terminals share the same channels. However, the properties of such a channel are different for each link, e.g. a channel can be reliable for the link between terminals A and B but unreliable for the link between terminals A and C. Some sort of test should indicate whether a channel is reliable for communication between two terminals or is not. Such a test could be a (periodically transmitted) string of bits between any of two terminals. On behalve of these bits the receiving terminal decides whether the channel is reliable or is not. In the case of ATM such a string can be a whole data cell to which 'end-to-end error control' is applied by counting the errors in the packet (thus we need some sort of coding). Depending on the number of errors found, we can decide to retransmit the packet via another channel.

In wired networks, the direction in which the information is transported, is known. In radio networks the information is radiated in more than one direction (broadcast). Every station will therefore receive (a part of) all transmitted messages, so we have to indicate which information is for whom. This problem demands the use of adresses. At first glance we could use the header of the ATM cell for this purpose.

A second probem arises when two or more terminals decide to transmit via one channel at the same time. The different messages will interfere and in the end no information will have been transported at all. Therefore no terminal should ever transmit on a channel when this channel is already being used.

One way of combatting unreliable data transfer is by routing the data over several reliable links. Although this routing makes the operation of the network more complex there is another reason why we should not use routing. Consider a terminal A which wants to communicate with terminal B by routing its information over terminal C. Both terminals A and C have to transmit the same message at different times or via different channels, thereby doubling the load (read: bandwidth) of the network. Each terminal added to the route will increase this load by the same amount. Therefore excessive routing should be avoided to limit the required bandwidth.

A communication protocol should take care of the problems mentioned and will be the main topic of this report. Besides the protocol has to garantee reliable data transfer, there are more requirements among which:

- The protocol should be simple, as it has to operate fast. Bitrates of up to 150 Mbit/s have to be provided for.

- The delay - the time between the moment a packet is available for transmission and the moment it is succesfully received - should be below service dependent limits.

- The efficiëncy of the network should be as high as possible.

- The sequence in which a terminal receives information from the network should be the same as it is offered to the network by the transmitting terminal.

As yet, we will restrict ourselves to these requirements and will focus on the properties of the radio medium in the next chapter.

## 2.3 Conclusions.

* ATM is a 'connection oriented' communication technique which can support many services using different bitrates. ATM takes advantage of multiplexing a high number of users and low error probability during transmission.

* The main problems of an indoor radion network are the broadcast effect of each user and the time varying connectiviy. Therefore, an ATM-based protocol for radio networks should:
- use a deviating method of addressing,
- take precautions to limit the effects of bad connectivity,
- include a multiacces scheme.

## 3. The Radio Environment.

We will consider an indoor radio environment as a mobile radio environment, because of possible movements of the terminals and objects in the environment (people,doors,machines etc.). Conceptually, we have to understand three things [8]:

1. The way in which the radio signal may be altered or degraded in the mobile channel;

2. The effect that these alterations can have on link quality and system performance, given a specific radio-link technology;

3. The countermeasures that are available to combat these effects.

In this chapter we try to give an answer to these questions.

### 3.1 Characteristics of the indoor radio environment.

Several things happen to a radio wave, transmitted within a mobile environment.

#### 3.1.1 Free space loss.

First, as the signal radiates in all directions, assuming for the moment an omnidirectional antenna in free space, the power of the signal would diminish as the inverse of the square of the distance ($\sim 1/r^2$). In practice the path loss is different from the free space loss because the environment affects the signal. Recently performed measurements in the 900 MHz region [9] indicate different exponents, varying from $1/r^{1.2}$ up to $1/r^{6.5}$ for common indoor environments.

The difference in path loss is highly dependent upon frequency and upon assumptions about antenna design and other aspects of the physical locale. The general rule is that the higher the frequency, the greater the attenuation, the shorter the radius of effective transmission.

### 3.1.2 Shadowing.

The second hazard facing the radio wave in the transmission path is the possibility that it may be partially blocked by shadowing. For the high frequencies used for indoor radio networks (>30 GHz) even small obstacles within the room cause shadows. The fading effects produced by shadowing are often referred to as slow fading, because from the perspective of a (moving) terminal the entrance and exit to and from a shadow takes a fair amount of time.

### 3.1.3 Multipath.

The radio wave may also be reflected, either by the walls or by different objects within the room. In some cases the reflected signal is significantly attenuated, while in others (e.g. metal surfaces) almost all the radio energy is reflected. The effect is to produce not one but many different paths between transmitter and receiver. This is known as multipath propagation and creates some of the most difficult problems associated with the mobile environment. The three most important multipath issues are:

1. The delay spread of the received signal.

2. Random phase shift which creates rapid fluctuations in signal strength known as Rayleigh fading.

3. Random frequency modulation due to different Doppler shifts on different paths.

ad 1): Because the signal follows several paths, and because the reflected paths are longer than the direct paths - if there is one - the multiple signals arrive with slight additional delay. The effect is a signal which is spread out in time: for example, a single sharp transmitted pulse will appear to a receiver as indicated in fig. 3.1. The mean delay time $T_d$ en the rms delay spread $\Delta$ can be calculated as [10]:

$$T_d = \frac{\int_0^\infty t e(t)\, dt}{\int_0^\infty e(t)\, dt} \qquad (3.1)$$

and

$$\Delta^2 = \frac{\int_0^\infty t^2 e(t)\, dt}{\int_0^\infty e(t)\, dt} - T_d^{\,2} \qquad (3.2)$$

respectively, where e(t) is the resultant power impuls response received from an impuls signal $s_o(t) = a_o\ \delta(t)$.

In a digital system, particularly one which operates at a high bitrate, the delay spread causes each symbol to overlap with preceding and following symbols, producing intersymbol interference (ISI). The delay spread is fixed, depending upon frequency. In effect, delay spread sets a limit on the transmission symbol rate in a digital mobile channel. Typical rms values for the indoor delay spread range from 50 ns in case of a little room (i.e. 10x10x3 m. [11]) up to 250 ns in case of a big room (i.e. 40x60x4 m. [12]).

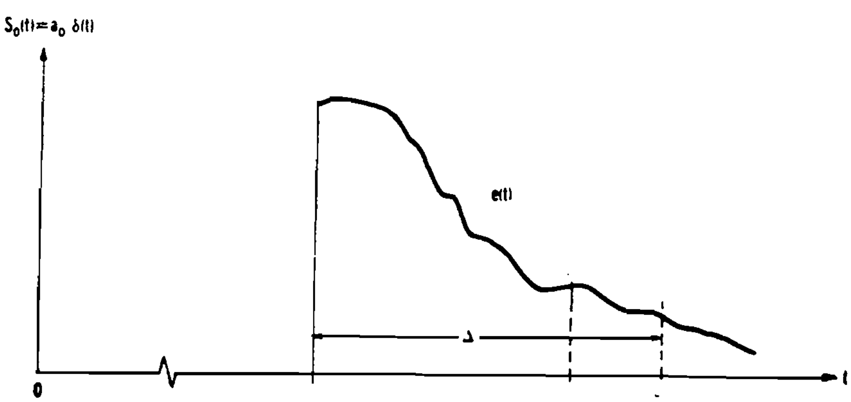


*Fig 3.1: Power Delay Spread.*

ad 2): The second chief effect of multipath propagation is that the reflected radio wave may undergo drastic alteration in some of its fundamental characteristics, particularly phase and amplitude. Assuming the transmitter is stationary, at any given point occupied by the receiver the sum of all direct and reflected paths from a transmitter to a receiver produces an alternation in signal strength related to the degree to which the multipath signals are in phase or out of phase.

When no direct (or other dominating rays) exists, the alternations of the received signal fall within a statistical distribution known as Rayleigh distribution, and for this reason the phenomenon is often refered to as Rayleigh fading. When there is a direct path the distribution is Ricean. The mobile and indoor environment are often called from this perspective, the Rayleigh or Ricean environment.

ad 3): Another aspect of the mobile channel is caused by the movement of the receiver, relative to the transmitter. This is the variation in the frequency of the received signal known as the Doppler shift. Much as the sound of a horn on a moving car appears to the stationary observer to be slightly higher in pitch when the car is approaching rapidly and slightly lower when the car is receding, so radio transmissions are frequency shifted due to the relative motion of the terminals and their environment. The frequency shift can be expressed as:

$$\Delta f_d = \frac{v}{\lambda} \cos \alpha \leq \frac{v}{\lambda} \tag{3.3}$$

where v: is the velocity of the movement relative to the receiver,

$\lambda$: is the wavelength of the radio wave,

$\alpha$: is the angle of the relative movement, measured with regard to the straight line connecting transmitter and receiver.

This frequency shift varies as the mobile unit changes direction and speed, and it introduces random frequency modulation in the radio signal.

## 3.2 Effects on system performance.

In digital systems, all forms of signal degradation are converted into a common measure: the bit error rate (BER). Another measure, derived from the BER is the number of data packets succesfully transported by the network. The influence of the environment has two aspects:

1. the effects on the quality of the individual links;

2. the effect on the overal system performance.

The first aspect requires a full description of environment properties. As already mentioned the indoor radio environment is complex and can vary a lot, as well as in space (3 dimensions) as in time. Therefore we use a statistical treatment of the problem. As indoor radio is a relatively new field of science, we do not have much knowledge of indoor propagation properties at 30-60 GHz. What we do have are some basic (statistical) parameters, defined for mobile radio. With these parameters it is possible to estimate limitations for some basic radio network features. It is possible to choose the best (lowest BER) modulation scheme dealing with the various radio propagation properties. The choice of this modulation scheme is subject to many studies about indoor radio networks, but will only be briefly considered in section 3.2.6.

The overal network performance depends strongly upon the assumptions made about the characteristics of the individual radio links. Because little can be said about the fading effects produced by shadowing, we will only consider the effects of the Rayleigh fading. In the following sections we will develop a model for Rayleigh fading channels, providing the average probability a packet will be received incorrect when having been sent over such a channel.

### 3.2.1 Characterization of fading multipath channels.

Let us examine the effects of the channel on a transmitted signal that is represented in general as [13,14]:

$$s(t) = \mathrm{Re}[\ u(t)\ e^{j2\pi f_c t}\ ] \tag{3.4}$$

where u(t) is the transmitted baseband signal. The channel is represented by multiple paths or rays having real positive gains $\{\alpha_n(t)\}$, propagation delays $\{\tau_n(t)\}$ and associated phase shifts $\{\theta_n(t)\}$, where n is the path index; in principle, n extends from 0 to $\infty$. Thus the complex, low-pass channel impuls response is given by:

$$h(t,\tau) = \sum_n \alpha_n(t)\ e^{-2\pi f_c \tau_n(t)}\ \delta(t-\tau_n(t)) \tag{3.5}$$

If we view the received signal as consisting of a continuum of multipath components, the low-pass time-variant impuls response becomes:

$$h(t,\tau) = \alpha(t,\tau)\ e^{-2\pi f_c \tau} \tag{3.6}$$

where $h(t,\tau)$ represents the response of the channel at time t due to an impulse applied at time $t-\tau$. The received low-pass signal is represented by the convolution of u(t) and $h(t,\tau)$:

$$r(t) = \int_{-\infty}^{\infty} h(t,\tau)\ u(t-\tau)\ d\tau \tag{3.7}$$

and the bandpass signal is obtained by taking the real part of $r(t)\ e^{j2\pi f_c t}$.

The received signal consists of the sum of a number of time-variant phasors having amplitudes $\alpha(t,\tau)$ and phases $2\pi f_c \tau$. Large dynamic changes in the medium are required to change $\alpha(t,\tau)$ significantly. On the other hand, the phases will change by $2\pi$ rad whenever $\tau$ changes by $1/f_c$. But $1/f_c$ is a small number and, hence, $h(t,\tau)$ will vary over its total range with relatively small motions of the medium.

We expect the delays $\tau$ to change in an unpredictable (random) manner. This implies that the received signal can be modeled as a random proces. Consider the transmission of an unmodulated carrier at frequency $f_c$ (then $u(t)=1$ in (3.4)), and the received signal reduces to:

$$r(t) = \sum_n \alpha_n(t)\ e^{-2\pi f_c \tau_n(t)} \tag{3.8}$$

When there are a large number of paths, the central limit theorem can be applied. That is, $r(t)$ can be modeled as a complex-valued gaussian random process. This means that the time-variant impulse response $h(t,\tau)$ is a complex valued gaussian random process in the t variable.

When the impulse response $h(t,\tau)$ is modeled as a zero-mean complex valued gaussian process, the envelope $|h(t,\tau)|$ at any instant t is Rayleigh distributed. In this case the channel is said to be a <u>Rayleigh fading channel</u>. In the case of a line-of-sight connection between transmitter and receiver, $h(t,\tau)$ can no longer be modeled as having zero-mean because of the strong steady signal component. In this case the envelope $|h(t,\tau)|$ has a Rice distribution and the channel is said to be a <u>Ricean fading channel</u>. In our treatment of fading channel, we consider only the Rayleigh fading channel because this model appears to be a realistic one and accounts for the worst case situation within the indoor environment.

The characteristics of a fading multipath channel can be described by a number of correlation functions and power spectral density functions. We define:

$$\phi_h(\tau_1,\tau_2;\ \Delta t) = 1/2\ E[h^*(t,\tau_1)\ h(t+\Delta t,\tau_2)] \tag{3.9}$$

If we assume that the attenuation and phase shift of the channel associated with path delay $\tau_1$, is uncorrelated with the attenuation and phase shift associated with path delay $\tau_2$, then $\phi_h(\tau,0) = \phi_h(\tau)$ is simply the average power output of the channel as a function of the time delay $\tau$. $\phi_h(\tau)$ is called the delay power spectrum of the channel. The range of values of $\tau$ over which $\phi_h(\tau)$ is essentially nonzero is called the multipath spread $T_m$. The rms value of $T_m$, denoted by $\Delta$, has already been defined in 3.1.3.

A complete analogous characterization of the time-variant multipath channel begins in the frequency domain. We define:

$$H(t,f) = \int_{-\infty}^{\infty} h(t,\tau)\ e^{-j2\pi f\tau}\ d\tau \tag{3.10}$$

$$\phi_h(f_1,f_2;\ \Delta t) = 1/2\ E[C^*(f_1,t)C(f_2,t+\Delta t)] \tag{3.11}$$

Under the assumption that the channel is wide-sense-stationary, it can be shown that $\phi_h(\Delta f,\Delta t)$ is the Fourier transform of $\phi_h(\Delta t,\tau)$. If we let $\Delta t=0$ then $\phi_h(\Delta f)$ is an autocorrelation function in the frequency variable and provides us with a measure of the frequency coherence of the channel. As a result of the Fourier transform relationship between $\phi_h(\Delta f)$ and $\phi_h(\tau)$, the reprocial of the mulipath spread is a measure of the coherence bandwidth of the channel, that is:

$$(\Delta f)_c \approx 1/T_m \tag{3.12}$$

We now focus on the time variations of the channel as measured by the parameter $\Delta t$ in $\phi_h(\Delta f,\Delta t)$. We define the Fourier transform:

$$S_h(\Delta f;\Omega) = \int_{-\infty}^{\infty} \phi_h(\Delta f;\Delta t)\ e^{-j2\pi\Omega\Delta t} d\Delta t \tag{3.13}$$

The function $S_h(0;\Omega)=S_h(\Omega)$ is a power spectrum that gives the signal intensity as a function of the time variations of the channel expressed as the Doppler frequency $\Omega$. The range over which $S_h(\Omega)$ is essentially nonzero is called the Doppler spread $B_d$ of the channel. Since $S_h(\Omega)$ is related to $\phi_h(\Delta t)$ by the Fourier transform, the reprocial of $B_d$ is a measure of the coherence time of the channel, that is:

$$(\Delta t)_c \approx 1/B_d \tag{3.14}$$

Refering to eq. 3.3 we note that $B_d$ can be expressed as $B_d \approx 2(\Delta f)_d = 2v/\lambda$.

### 3.2.2 Signal characteristics and channel models.

The signal characteristics have a profound influence on the choice of a channel model. This can qualitatively be explained as follows:

Assume two sinusoids with frequency separation greater than $(\Delta f)_c$ and each sinusoid will be affected essentially different by the channel. When an information-bearing signal is transmitted through the channel and $(\Delta f)_c$ is small in comparison to the signal bandwidth W, the channel is said to be _frequency-selective_. In this case the signal is severely distorted by the channel. On the other hand, if $(\Delta f)_c$ is large in comparison to the bandwidth W, the channel is said to be _frequency non-selective_.

Additional distortion is caused by the time variations of the channel characteristics. These time variations in $h(t,\tau)$ cause variations in the received signal strength and has been termed _fading_. During the coherence time $(\Delta t)_c$ the instantaneous transfer function of the channel $H(f;t)$ is assumed not to change. If we choose a signalling interval T to satisfy the condition $T<<(\Delta t)_c$, we call the channel a _slowly fading channel_. This because channel characteristics are essentially fixed for at least one signalling interval.

The effect of the channel on the transmitted signal is a function of our choice of signal bandwidth and signal duration. These parameters have respectivily to be compared with the coherence bandwidth and coherence time of the indoor radio channels. In the next two subsections we will try to estimate the delay spread and the coherence time of indoor radio channels.

#### 3.2.2.1 Delay spread and intersymbol interference.

As mentioned in section 3.1.3 the delay spread causes symbols to overlap, introducing intersymbol interference (ISI). In effect, delay spread sets a limit on the transmission symbolrate in a digital mobile radio channel. The results of [15] highlight the importance of the ratio $\Delta/T_s$, where $T_s$ is the symbol time. They show that the delay spectrum shape is of no importance for $\Delta/T_s \le 0.2$, but can have a profound influence for $\Delta/T_s \ge 0.3$. With these facts we derive the upper limitation of the bitrate as follows: Given $\Delta/T_s \le 0.2$, where $T_s = 1/r_s$ and $r_s$ is the symbol rate, the bitrate $r_b$ is simply $r_b = kr_s$ where k is a constant expressed in bit/symbol. Therefore the upperlimit for the bitrate is given by:

$$r_{b,max} = \frac{0.2\ k}{\Delta} \tag{3.15}$$

Equation 3.15 states that high bitrates can be achieved by either keeping $\Delta$ small or choosing a modulation method with a high value for k (for example an M-ary signalling scheme). Typical values for $r_{b,max}$ range from 1 to 2 Mbit/s [15].

A method for keeping $\Delta$ small is proposed in [16]. By using a distributed antenna system or a 'leaky feader', dramatic reduction in multipath delay spread and signal attenuation (compared to a centralized system) can be achieved. The signal attenuation can be reduced by as much as a few tens of decibels and the rms delay spread becomes limited to 20 to 50 ns, even in large buildings.

It is mentioned that the influence of the delay spectrum is profound for $\Delta/T_s \ge 0.3$. The intersymbol interference can therefore be expressed as $ISI = r_sT_m$, where $T_m$ is the delay spread defined in eq.(3.12). As we have only information about the mean rms delay spread it is reasonable to assume [10, eq.1.5-60] that $T_m < 2\Delta$. So the rate of intersymbol interference is given by:

$$ISI = \frac{2\ r_b\ \Delta}{k} \tag{3.16}$$

**3.2.2.2 Doppler shift and channel stationarity.**

The Doppler shift (eq. 3.3) affects all multiple propagation paths, some of with may exhibit a positive shift, and some a negative shift. Using eq. 3.3 and assuming that all objects in the room do not move faster as 10 m/s (i.e. the world record for sprinting 100 m) the Doppler shift is 2 kHz maximum (f=60 GHz). This value has to be compared to the bandwidth used. For a bandwidth of 150 MHz the deviation of the original signal spectrum is neglectable. We may therefore assume that the influence on the signal is neglectable also. However, the frequency shift may affect the demodulation process, because the receiver is locked on a certain frequency or phase which suffers from Doppler shift. This effect will not be further studied in this report.

From eq. 3.14 we derive $(\Delta t)_c = 0.25$ ms.

### 3.2.3 Choice of a channel model.

As we have assumed in the previous section that $(\Delta t)_c = 0.25$ ms. The channel may be assumed to be slowly fading if $T_s \ll 0.25$ ms, or, if we use a 1 bit/Hz modulation scheme, the signal bandwidth should obey $W \gg 4$ kHz.

The delay spread $T_m$ is approximated by $2\Delta$, where the rms delay spread $\Delta$ varies from 50 ns upto 250 ns. Consequently the channel may be considered non-selective if $W \ll 2$ Mhz, using $T_m$=500 ns. Actual measurements indicate that intersymbol interference, as a consequence of large $T_m$, is neglectible for $W < 1$MHz.

As a consequence of these results we assume the indoor radio channel to be:

| | |
|---|---|
| selective/slow | $W \gg 1$ MHz |
| selective/fast | never |
| non-selective/slow | 4 kHz $\ll W \ll$ 1 MHz |
| non-selective/fast | $W \ll 4$ kHz. |

using a 1 bit/Hz modulation scheme.

As we are interested in an indoor radio network which can handle large bitrates (upto 150 MHz) over a few channels, we are primarely interested in signalling over frequency selective, slow fading channels. However, to yield insight in the digital signalling on such a channel it is important to studie signalling on a slow, non-selective channel first.

### 3.2.4 Binary signalling over a slow, non-selective channel.

We will first consider the effect of signal characteristics on the statistical characterization of the channel as given in section 3.2.1.

In a frequency non-selective model we select the signalling interval to satisfy the condition $T>>T_m$, i.e. the channel introduces a neglectible amount of intersymbol interference. Consequently, the condition $T>>T_m$ implies $W<<(\Delta f)_c$. But this implies that, within the bandwidth occupied by the signal, the time-variant transfer function $H(f,t)$ of the channel is a complex-valued constant in the frequency variable. Thus, let $u(t)$ be the equivalent low-pass signal transmitted over the channel. Then the equivalent low-pass received signal may be expressed as:

$$r(t) = \int_{-\infty}^{\infty} h(t,\tau)\ u(t-\tau)\ d\tau = \int_{-\infty}^{\infty} H(f,t)\ U(f)\ e^{j2\pi ft}df$$

$$= H(0,t) \int_{-\infty}^{\infty} U(f)\ e^{j2\pi ft}df = H(0,t)\ u(t) \tag{3.17}$$

where $H(0,t)$ may be expressed in the form:

$$H(0,t) = \alpha(t)\ e^{-\phi(t)} \tag{3.18}$$

When $H(0,t)$ is modeled as a zero mean-complex valued gaussain random proces, the envelope $\alpha(t)$ is Rayleigh distributed for any fixed value of t and $\phi(t)$ is uniformly distributed over the interval $(-\pi,\pi)$.

We further assume:

- The change of the phase shift $\phi$ is sufficiently slow it can be estimated from the signal without error. In this case we can achieve ideal (coherent) detection.

- the fading is constant over at least one signalling interval.

In a fading environment, communication system performance is usually characterized by the average probability of bit error $\varepsilon(\rho)$ where $\rho$ is the bit energy-to-noise density ratio. Expressions for $\varepsilon(\rho)$ can be found in various communication literature (13,14). In a ATM radio system however the performance can better be characterized by the probability of average cell error $P_e$.

Consider a block of N bits [17,18] corresponding to a cell. We assume that the communication system employs a digital modulation/demudulation scheme which produces independent bit errors (so this is not applicable to DPSK) with propability $\varepsilon(\rho)$. The probability of block error $P_{be}$ is the probability of at least one bit error in the block:

$$P_e = \text{Prob [ at least one error]} = 1 - \text{Prob [ all bit received correctly ]} = 1 - [1 - \varepsilon(\rho)]^N \tag{3.19}$$

Assuming constant $\rho$ over the block lenght (i.e. $(\Delta t)_c \gg$ block transmission time), the probability of block error averaged over the signal level is given by:

$$P_e = \int_0^{\infty} \{1 - [1 - \varepsilon(\rho)]^N\} \, f(\rho) \, d\rho \tag{3.20}$$

where $f(\rho)$ is the probability density of $\rho$. For the case of Rayleigh fading, the signal power has an exponential probability density function, and:

$$f(\rho) = 1/\bar{\rho} \, e^{-\rho/\bar{\rho}} \qquad \rho \geq 0 \tag{3.21}$$

where $\bar{\rho}$ is the average value of $\rho$.

This model can be extended to the case of the average probability of having M bit errors in a block of N bits under conditions of Rayleigh fading [17]. In the case of other fading characteristics we apply another $f(\rho)$ in equation 3.20. Figures 3.2 and 3.3 [17] show the dependence of probability of block error on average signal-to-noise ratio for NCFSK and CFSK over a wide range of block lengths N.

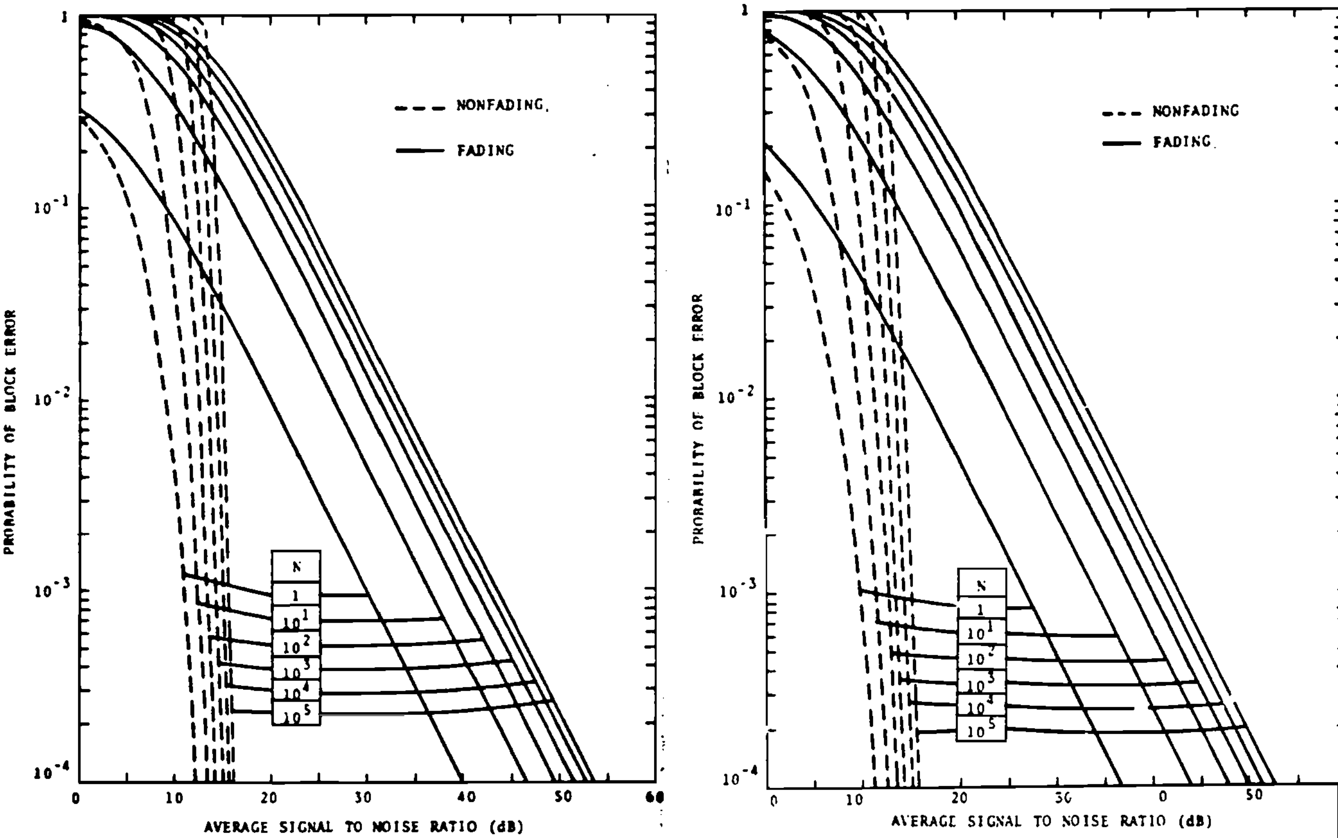


*Fig. 3.2 and 3.3: block transmission performance for NFSK and CFSK.*

Note that for error probabilities approaching 1, the performance of the fading channel exceeds that of the nonfading channel for $N>10^3$. It is assumed that this is a mistake made by the authors of [17]. The performance of PSK can be obtained from figures 3.2 and 3.3 by translating the curves by 3 dB to the left. For future broadband services a BER of $10^{-6}$ or higher has been specified. Equation 3.19 then gives $P_e$=4.23 $10^{-4}$ and therefore the SNR has be approximately 47 dB for cells between a 100 and 1000 bits. More realistic values for radio links are BER=$10^{-4}$, $P_e$=4.15 $10^{-2}$ and SNR=30 dB, introducing a high number of packet retransmission.

### 3.2.5 Binary signalling over a slow, selective channel.

When the channel is assumed to be a slow, but selective fading channel the transfer function H(f,t) is assumed not to change of form over the duration of a message (bit/block) but where its spectrum will exhibit variation over the band occupied by the transmitted signal. The selectivity is a result of $W \gg (\Delta f)_c$. Physically, this is a result of having frequencies within the signal band which, due to interference, weaken or attenuate the received signal strength for that particular frequency.

In view of the close relationship between selective fading and multipath spread, proper evaluation of error rate must not only include the distortion of single pulses by selective fading but also the intersymbol interference produced by the time dispersion of pulse energie. Because of multipath a puls whose energy is normally confined to an interval $(0,T_s)$ will now have associated energy over an interval roughly described by $(-\Delta/2, \Delta/2)$ where $\Delta$ is the rms delay spread. Thus in calculating probability of error for a particular symbol waveform, one has to account for the preceding and following waveforms. These are in turn determined by the message content. Hence one has to solve for probability of error per symbol by averaging over the probability of error associated with all possible message sequences containing that symbol [13], which is very difficult. The message sequences are often scrambled to cut off the association between message sequences and probability of error.

To simplify the problem we assume to have an ideal equalizer (see section 3.3.2), which removes any intersymbol interference (this is clearly a very optimistic assumption). The operation of such an equalizer can be viewed in two ways. In the time domain it weights the signal sample and intersymbol interference with neighbour symbols to achieve an optimal SNR of the combined signal. In the frequency domain the spectra of the various symbols are combined to approximate the original spectrum of the transmitted signal, removing deep fades in the former selective spectrum.

In the case of the ideal equalizer we obtain approximately $2\Delta W$ samples of the same signal, thus providing the receiver with several, independently (Rayleigh) fading signal paths. Since $T_m \approx 2\Delta$ and $(\Delta f)_c \approx 1/T_m$, the use of a wideband signal may be viewed as an option for obtaining frequency diversity of order $L = W/(\Delta f)_c$ [16,pg.479], which improves the SNR. The power delay profile is expected to be exponential in the case of Rayleigh fading, thus not all signal components will contribute to the diversity gain to the same amount. If the power delay profile is known, only those components, say K, contribute which have a sufficiently large SNR. When we use maximal-ratio combining, the achieved diversity will therefore be only (L-K)-fold at this level of SNR [15,pg.450]. At this point the same theory of section 3.2.4 can be applied with the difference that $f(\rho)$ in equation 4.16 will be a Gamma function of order L-K. In the case of an non-ideal equalizer a correction has to be made on $\varepsilon(\rho)$ and $f(\rho)$. These corrections could be a topic for further studie.

For the time being we assume having an ideal equalizer, which removes all intersymbol interference and thereby turning the selective channel into a non-selective one. In [19] it is pointed out that, whithin any frequency band, the distribution of $|H(f,t)|$ over a large number of fades, i.e. averaged over the t variable, is independent of f. This is known from both physical considerations and previous data reductions. In our assumptions, we will use therefore $|H(0,t)|$ explicitly, which will have an exponential distribution and enables us to use the results of fig. 3.2 and fig. 3.3.

### 3.2.6 Modulation techniques.

The radio links between the base station and the users and the possible links between users themselves, are the heart of the system. The choice of radio technology will determine to a great extend the economic and performance characteristics of the communication system. The most popular approaches [21] are based upon modulating the phase of the carrier wave. Phase shift keying (PSK) modulation defines various discrete phase states which are used to carry the digital information from the coder. The PSK-family includes high level PSK like 4-level PSK (QPSK). A very popular form of a mixed system is quadrature amplitude modulation (QAM), because it is capable of yielding high spectral efficiencies. QAM utilizes both phase and amplitude information to define the modulation levels.

High level modulation schemes however, are very sensitive to channel impairments (like: lineair/nonlineair distortion, synchronization errors, interference, multipath fading) so various associated techniques must be deployed (like: diversity, equalization and forward error correcting coding (FEC)). So far, the use of higher level modulation has always led to lower cost by channel.

## 3.3 Countermeasures.

As indicated in section 3.1 and 3.2 there are various ways of improving the signal-to-noise ratio of the received signal(s), thereby improving the bit error rate and the average packet error rate. The strategies for dealing with the problems of the mobile channel may be divided into two groups: those that are, so to speak, conventional, and those that are based upon robust digital signal-processing techniques. The last require a digital architecture and have not been as widely applied.

### 3.3.1 Conventional countermeasures.

**Fade margins.**

The concept of fade margins is straightforward: extra power is added to the transmission to overcome potential fading. The fade margin is normally equal to the maximum expected fade. In indoor radio systems however, the use of large fade margines is limited by the maximum power that may be radiated by indoor equipment. Large fade margins also complicate the implementation of frequency reuse, since large radiation power enlarges the distance at which the frequency is available for reuse.

**Diversity.**

Diversity refers to any of several techniques for sampling the signal more than once, by either combining these signals or selecting the best of them, improving the signal-to-noise ratio at the receiver. The samples must be taken in such a way that the fading characteristics of the different samples are uncorrelated.

In [22] it is shown that the diversity gain factor, assuming independent Rayleigh fading, is approximated by:

$$g_p = \frac{n-1}{n}\, 10\ {}^{10}\log 1/P \qquad (3.22)$$

Where n: is the level of diversity,
P: is the propability that the signal strength is below a certain threshold level.

The gain accomplished by diversity increases with the number of created channels (n) but the increase of the gain itself decreases.

The most common forms of diversity are:

- space diversity,
- polarization diversity,
- time diversity,
- frequency diversity.

For our ATM based indoor radio channels, space diversity is the best option of the four diversity methods because:

- Most indoor media are dielectric in nature [23] so crossed-polarization transmissions exhibit highly correlated fading, unsuited for diversity.

- Multiple repetitions will increase the required bandwidth with the same amount for each retransmission (this is comparable with the routing problem in section 2.2) and introduces considerable delay (0.25 ms for each retransmission, using eq. 3.14 with $\lambda$=5 mm. and v=10 m/s).
- Transmission of the same information on two or more seperated frequencies is intolerable wasteful of spectrum.

If we assume that the number of channels M is two or more, we can make use of a proposed diversity technique called <u>frequency hopping</u> [24]. A good way to describe the proposed diversity technique is to think of a multichannel TDMA system with frequency-switching diversity, where each user would jump to a independent fading channel (carrier frequency) each time a deep fade is encountered. To avoid collisions among the various users of the system, a base station would need to coordinate their frequency jumping and their transmission time slots.Figure 3.4 shows that the same performance as an l-fold diversity system could be attained using a code correction capability of c.

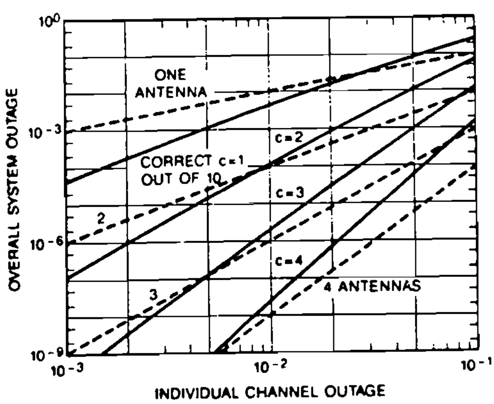


*Fig. 3.4: Comparison of the outage performances of a single-frequency system with multiple antenna diversity (dashed lines), and a ten-frequency system using hopping and coding that is capable of correcting one to four errors.*

Since all users are expected to continue to switch their frequencies, an approach is to force each user to hop cyclically through a preset sequence of frequencies. In this case, the diversity technique is called cyclical frequency hopping. The key idea beyond the forced hopping is that, without forced hopping, long random runs of errors could occur (at any given frequency) which, unless randomized, would require excessive redundancy and delay to correct. This delay would set strong delay contraints on the acknowlegment method.

**Supplementary base stations.**

To fill in large holes caused by shadowing, the conventional answer in mobile communications has been to deploy more base stations. The principle of what is in effect base-station diversity can be built into cellular architecture through the use of corner-sited antennas with directional transmission to provide alternative paths for reading terminals in shadowed areas. In fact this is the same idea as beyond the distributed antenna [16]. The advantage of this architecture is that the number of cell-cites is not increased, which holds down costs and avoids problems like hand off or interference problems at fuzzy cell boundaries.

### 3.3.2 Digital countermeasures.

In a digital system the inherently robust format and the oppertunity to apply intelligent signal processing open up new avenues for counteracting the effects of the mobile environment. The most important of these are:

1: robust voice coding,
2: robust modulation,
3: error correction,
4: adaptive equalization.

The use of robust techniques is based on the principle of making the transmission more error resistent by limiting the damage caused by errors. As all techniques are rather complex, it is difficult here to present a survey of all different approaches. We will only discuss adaptive equalization because this technique is absolute necessary when we want to use high bitrates (> 2 Mbit/s) in the indoor radio environment.

The fundamental principle of equalization is to obtain, prior to the beginning of the transmission, certain information about the distortion and attenuation characteristics of the channel, and then to employ this information to correct the received signal. A typical way of accomplishing this is to begin the transmission with a brief 'training sequence': a known data sequence is tranmitted, and the demodulator compares the actual received signal with what it knows to be the original input sequence. The observed error is therefore channel-induced. From these data, the system derives sufficient information to characterize the channel and uses this information to set its processing algorithms to substract the expected error from all subsequent samples.

Among the many structures used for equalization [25], the simplest is the transversal (tapped delay-line or nonrecursive) equalizer shown in figure 3.5.

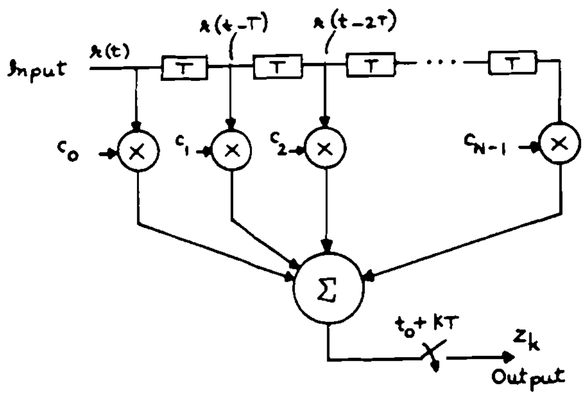


*Fig. 3.5: Linear transversal equalizer.*

In such an equalizer the current and past values of the received signal r(t) are linearly weighted by equalizer coefficients (tap gains) $c_n$ and summed to produce the output given by:

$$z_k = \sum_{n=0}^{N-1} c_n \, r(t_o + kT_s - nt) \tag{3.23}$$

Where N is the number of equalizer coefficients en $T_s$ is the symbol time.

The equalizer coefficients $c_n$ may be chosen to force the samples of the combined channel and equalizer inputs response to zero all but one of the N instants in the span of the equalizer. Such an equalizer is called a zero-forcing (ZF) equalizer. The ZF criterion neglects the effect of noise and is guaranteed to minimize the worst case ISI only if the peak distortion before equalization is less than 100%, i.e. if the eye pattern is initially open. However, at high speeds on bad channels this condition is often not met.

The least-mean-square (LMS) equalizer (fig. 3.6) is more robust. There the equalizer coefficients are chosen to minimize the mean-square error - the sum of squares of all the ISI terms plus the noise power at the output of the receiver. Given a synchronized version of the known training signal, a sequence of error signals $e_k = z_k - x_k$ can be computed at the equalizer output and used to adjust the equalizer coefficients to reduce the sum of squared errors. The algorithm to compute the coefficients may be iterative or be another stochastic update methode, like the Kalman algorithm. The last appears to be the fastest known equalizing adaption algorithm [26].

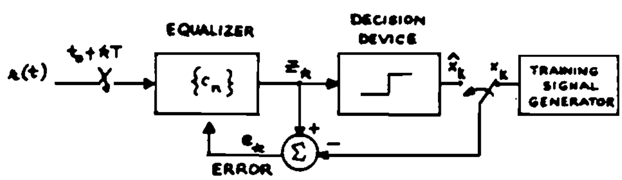


*Fig: 3.6: adaptive equalizer based on mean-square error estimation.*

A disadvantage of the Kalman algorithm is its complexity. Therefore a fast Kalman algorithm has been developed to overcome this problem. If the channel characteristics are expected to be fairly stable over the duration of the transmission, the initial training may suffice. If, however, the channel characteristics are expected to continue to change during the course of the transmission - as in indoor radio - then the initial training must be supplemented by an ongoing adaptive equalization algorithm that is capable of measuring the channel characteristics and continuously tuning the equalizer parameters to maintain a good pulse shaping and low error rates. This can be achieved by using <u>decision feedback</u> equalization (DFE, see figure 3.7). The equalizer is assumed to start out with the right setting immediately after training is completed. The first sample will be analyzed and extracted from the incoming signal. Once the first pulse has been decided and its value is known, the error between that value and what was actually received can be recomputed (past decisions are assumed to be correct). This actual error is now compared with what had been the expected error. This difference - between expected error and observed error - can now be used to update the setting of the equalizer coefficients.

Analyses and field evaluation data are not yet available to permit a full comparative quantitative evaluation of linear equalizers and DFE's in a multipath environment with dynamic fading, but theoritically the DFE's are superior to linear equalizers.

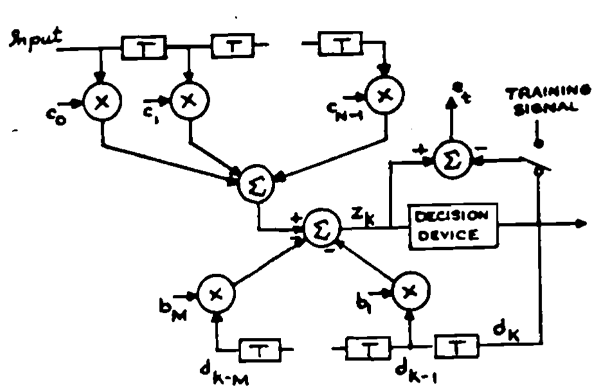


*Fig 3.7: Decision-feedback equalizer.*

### 3.3.2.1 Equalizer convergence.

The convergence behaviour of stochastic updata methods is hard to analyze. In the paper of Godard [27] an algorithm based on Kalman filtering is proposed for obtaining fast convergence of the tap gains of transversal equalizers to their optimal settings. It is shown that the convergence of distortion is obtained, under noisy conditions, within a number of iterations determined only by the number of taps, without prior information about the channel. The expression for the speed of convergence is given by:

$$\varepsilon_k^2 \approx \varepsilon_{opt}^2 (1 + N/k) \tag{3.24}$$

Where $\varepsilon_k^2$ is the expected mean square distortion after k iterations, $\varepsilon_{opt}^2$ is the optimal expected mean square distortion, N is the number of taps and k is the number of iterations. $\varepsilon_{opt}^2$ cannot be known a priori, but may be considered as the variance of a white noice process. Usually after equalization the SNR at the output of the equalizer is between 20-30 dB, so that to compute the Kalman gain one can use an estimated value $\hat{\varepsilon}_{opt}^2$ of the optimal mean-square distortion included between 0.01 and 0.001. The value of $\varepsilon_{opt}^2$ has no influence on the speed of convergence, provided that the value chosen is reasonably small but not zero.

The number of taps is determined by the amout of intersymbol interference. If the interference affects N symbols, then at least N samples should be weighted, independend of the equalizer or algorithm used. Therefore the minimal number of equalizer coefficients N is given by:

$$N = ISI \tag{3.25}$$

More information about can be found in [14,24 t/m 31].

## 3.4 Conclusions.

* The radio environment can be characterized by free space

loss, shadowing and multipath effects, like delay spread, Rayleigh fading and Doppler shift.

* The delay spread and the Doppler shift determine the frequency selectivity and the coherence time of a fading channel. For the indoor environment, all channels of interest are slow-fading channels.

* The bitrate to be used on a slow, non-selective channel is 1 a 2 Mbit/s maximum. Assuming Rayleigh fading and a NCPSK modulation scheme, the SNR at the receiver should be 47 dB to achieve stringent BISDN requirements (BER<$10^{-6}$).

* Slow selective radio channels are very hard to analyse. The fading effects will be less then the fading effects in non-selective channels, because of the inherent diversity. An equalizer has to be used to remove intersymbol interference. A possible candidate is a DFE equalizer using the fast Kalman algorithm.

* The SNR at the receiver can be increased by using diversity. The most suitable diversity methods are space diversity and cyclical frequency hopping. Another option is to use a distributed antenna system, which also decreases the effect of shadowing.

* The most appropiate modulation scheme will be a member of the PSK family.

## 4. Multiaccess protocols.

Packet communication is based on the idea that part or all of the available resources are allocated to one user at a time but for just a short period of time. Here each component of the system is itself a resource which is multiaccessed and shared by the many contending users. To achieve sharing at the component level, customers are required to divide their messages into small units, called packets or cells.

The need for multiaccess protocols arises whenever a resource is shared by many independent contending users. Two factors contribute to such a situation: the need to share expensive resources in order to achieve their efficient utilization and the need to provide a high degree of connectivity for communication among independent subscribers. In the past two decades, various multiaccess protocols have been developed and applied to different types of multiaccess networks. The kind of network we studie is a local packet radio network (number of terminals upto 200), based on ATM. As mentioned in chapter 2, access to the different timeslots in an ATM message channel is obtained through signalling. It is this signalling channel, the multiaccess protocol is applied to. The main issue of concern here is how to control access to the common signalling channel(s), to efficiently allocate the available communication bandwidth (ATM cells) to the many contending users.

In indoor radio networks, the propagation delay is relatively short compared to the transmission time of a packet. As we shall see in the sequel, this can be of great advantage in controlling access to a common channel. It is important however to distinguish that the indoor radio propagation characteristics are subject to important variations in received signal strenght so that system connectivity is at all times difficult to predict. This limited connectivity is important to device access schemes and system control mechanisms that allow the system to adapt itself to these changes. The local area environment is further more characterized by large and often variable number of devices requiring interconnection, whose data traffic is usually bursty.

Multiaccess schemes can be evaluated according to various criteria. The performance characteristics that are desirable are, first of all, high bandwidth utilization and low message delays. But a number of other attributes are just as important. The ability for an access protocol to simultaneously support traffic of different types, different priorities, with variable message length, and differing delay constraints is essential in ISDN networks where higher bandwidth utilization is achieved by the multiplexing of all traffic types. It is therefore that we have choosen ATM as the multiplexing method on the message channel. Also, to guarantee proper operation of the scheme and robustness, defined here as the insensitivity to errors resulting in misinformation, is also most desirable.

For so far we discussed briefly the basic characteristics and system requirements underlying the indoor environment. We now preceed with a discussion of multiaccess protocols.

The performance of a particular protocol depends on such factors as the channel propagation delay, the transmission delay, the traffic arrival process, the number of terminals, etc. Existing multiaccess protocols may be categorized according to how much coordination is required between the potentially conflicting terminals. Multiaccess protocols differ by the static or dynamic nature of the bandwidth allocation algorithm, the centralized or distributed nature of the decission making process, and the degree of adaptivity of the algorithm to changing needs. Accordingly, the protocols can be examined in three ways:

- The degree of control exercised over the user's access, i.e. fixed assignment or random access.

- The (centralized or distributed) nature of the decision making process

- The degree of adaptivity of the algorithm to the changing need.

Not all techniques will be discussed (in detail). An excellent survey is given in [32].

### 4.1 Fixed assignment techniques.

Fixed assignment techniques consist of those techniques which allocate the channel bandwidth to the user in a static fashion, indepently of their activities. These techniques take two common forms: orthogonal, such as frequency division multiple access (FDMA) or synchronous time division multiple access (TDMA), and 'quasi orthogonal' such as code division multiple access (CDMA).

<u>FDMA</u> c.q <u>TDMA</u> consists of assigning to each user a fraction of the bandwidth c.q. timeslots. Orthogonality is respectively achieved in the frequency and time domain. <u>CDMA</u> allows overlap in transmission both in the frequency and time coordinates. It achieves orthogonality by the use of different signalling codes in conjunction with matched filters at the intended receivers. The multiple orthogonal codes are obtained at the expense of increased bandwidth requirements.

In <u>polling</u> systems a central controller sends polling messages to the terminals, one-by-one, asking the polled terminal to transmit. If the polled terminal has something to transmit, it goes ahead; if not, a negative reply is received by the controller, which then polls the next terminal in sequence. Polling is efficient only if 1) the round-trip propagation delay is small, 2) de overhead due to polling messages is low, and 3) the user population is not a large bursty one.

The throughput - defined as the number of messages which is successfully transmitted per unit time - can reach 100% in fixed assignment techniques if all users want to send, and there will be no collisions between the messages. The delay - defined as the time between a message is ready for transmission and the time it is actually received by the receiver - is very small in the case of FDMA and CSMA and is fixed, depending on the number of users and the size of the timeslots, in the case of TDMA and polling.

We note that, even though the allocation can be tailored to the relative need if each user, fixed allocation is wasteful if the users' demand is highly bursty, as we assumed. It is therefore that we use asynchronous time division multiple acces as the technique for operating the message channels.

**4.2 Random access techniques.**

In this class the users is provided a single channel to be accessed randomly; since collisions may result which degrade the performance of the channel, improved performance can be achieved by either synchronizing users so that their transmissions coincide with the boundaries of time slots, by sensing carrier prior to transmission, or both.

The simplest of its kind is pure ALOHA, which permits a user to transmit on the channel at any time it desires. If they do so, and within some appropiate time-out period the user receives an acknowledgement from the destination, then it knows that no conflict occurred. Otherwise it assumes that a collision occured an it must retransmit. To avoid continuously repeated conflicts, the retransmission delay is randomized across the transmitting devices, thus spreading the retry packets over time.

A slotted version, referred to as slotted ALOHA, is obtained by dividing time into slots of duration equal to the transmission time of a single packet (assuming constant length packets). Each user is required to synchronize the start of transmission of its packets to coincide with the slot boundary. When two packets conflict, they will overlap completely rather than partially, providing an increase in channel efficiency over pure ALOHA.

The efficiency of ALOHA and slotted-ALOHA are respectively

0.184 and 0.364 (fig 4.1), and the average delay (fig. 4.2) becomes rather heigh when the channel load increases. Various approaches have been developed to improve the performance, like 'announced retransmission ALOHA [33]', 'reservation ALOHA [34]' and 'ALOHA with capture effect [35]', but all of them are exceeded by CSMA.

In CSMA (carrier sense multiple access), users attempt to avoid collisions by listening to the carrier due to another user's transmission before transmitting, and inhibiting transmission if the channel is sensed busy. When a terminal learns that its transmission had incurred a collision, it reschedules the transmission of the packet according to the randomly distributed delay. At this new point in time, the transmitter senses the channel again and repeats the alogorithm dictated by the protocol.

There are two main CSMA protocols known as non persistent and p-persistent CSMA depending on whether the transmission by a station which finds the channel busy is to occur later or immediately following the current one with probability p. The parameter p is chosen to reduce the level of interference while keeping the idle periods between two consecutive nonoverlapped transmissions as small as possible. A slotted version of these CSMA protocols can also be considered in which the time axis is slotted and the slot size is $\tau$ s, where $\tau$ is the maximum propagation delay among all pairs.

The performance of the different protocols is summarized in the figures 4.1 and 4.2. [36]. S denotes the aggrerate rate of packet generation from the entire population of users, G is the rate of packet transmissions (new and repeated, hence $G \geq S$), and D is the packet delay normalized to the (fixed) packet transmission time. We note that the behaviour of these schemes is typical for contention systems, namely that the throughput increases as the offered channel traffic increases from zero, but reaches a maximum value for some optimum value of G, and then constantly decreases as G increases beyond that optimal value. From figure 4.2 we note that D increases as the throughput increases, and reaches infinite values as the throughput approaches the channel capacity.

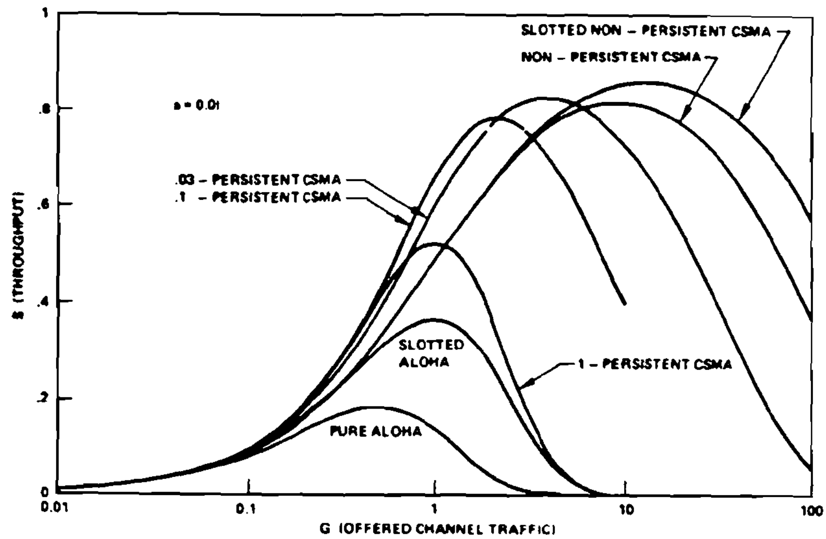


*Fig. 4.1: Throughput for the various random access modes (propagation delay a = 0.01).*

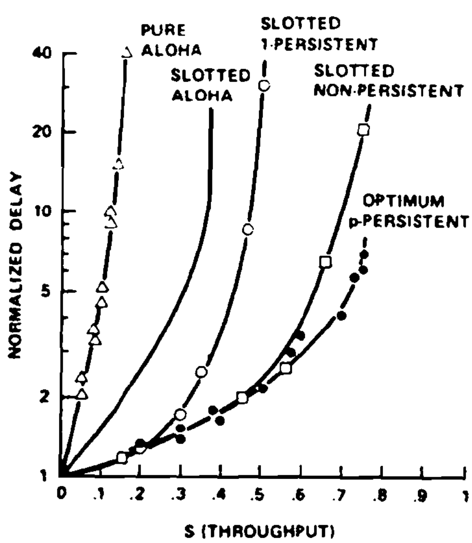


*Fig. 4.2: CSMA and ALOHA; throughput-delay trade off (propagation delay a = 0.01).*

The results show the evident superiority of CSMA over the ALOHA scheme. The CSMA channel capacity in some cases may be as high as 90% of the available bandwidth. It is clear, that, as expected, the channel capacity and the throughput-delay trade off for the CSMA schemes degrade as the normalized propagation delay $a = \tau/T$, where $\tau$ is the propagation delay and T is the packet transmission time, increases. Figure 4.3 illustrates the sensitivity of the channel capacity to a.

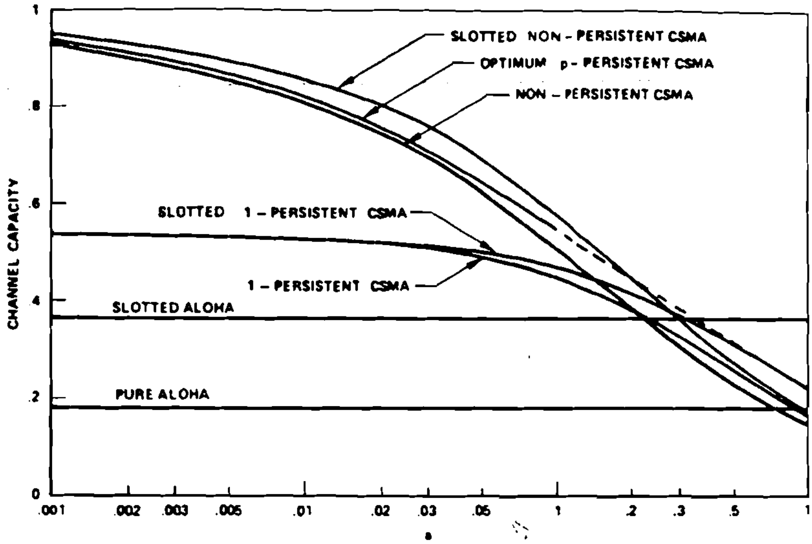


*Fig. 4.3: CSMA and ALOHA; effect of propagation on channel capacity.*

In indoor radio systems it is likely that not all users are in line-of-sight and within range of each other. Typically, two terminals can be within range of the intended receiver, but out-of-range of each other or seperated by some physical obstacle opaque to radio signals. The existence of hidden terminals in a radio environment significantly degrade the performance of CSMA. Fortunately, in environments where all users communicate with a single central station, the hidden terminal problem can be eliminated by frequency dividing the available bandwidth into two separate channels: a busy-tone channel and a message channel, thus giving rise to so called busy-tone multiple access (BTMA).

The operation of BTMA [37] rests on the assumption that a base station is, by definition, within range and in line-of-sight of all terminals. The total available bandwidth is to be divided into two channels: a message channel and a busy-tone (BT) channel. As long as the base station senses (terminal) carrier on the incoming message channel it transmits a (sinewave) busy-tone signal on the BT channel. It is by sensing carrier on the BT channel that terminals determine the state of the message channel. The action pertaining to the transmission of the packet that a terminal takes (again) is prescribed by the particular protocol being used. As in [32], we restrict ourselves to the nonpersistent protocol because of its (relative) simplicity in analyses and in implementation, as well as its relatively high efficiency.

In CSMA the difficulty of detecting the presence of a signal on the message channel when this message used the entire bandwidth was minor and therefore neglected. It is not so when we are concerned with the detection of the (sinewave) busy-tone signal on a narrow-band channel. The detection time, denoted by $t_d$ is no longer neglegtible and must be accounted for.

The nonpersistent BTMA protocol is similar to the nonpersistent CSMA protocol and corresponds to the following: Whenever a terminal has a packet ready for transmission, it senses the busy-tone channel for $t_d$ seconds at the end of which it decides whether the BT signal is absent (in which case it transmits the packet); otherwise it reschedules the packet for transmission at some later time incurring a random rescheduling delay. At this new point in time, it senses the BT channel and repeats the algorithm. In the event of a conflict, which the terminal learns about by failing to receive an acknowlegdement from the base station, the terminal again reschedules the transmission of a packet for some later time, and repeats the above process. Detailed analyses in [32] indicates that the delay performance of BTMA is comparable with that of CSMA but the throughput performance is 70% maximum.

**4.3 Distributed or centralized control.**

The basic element underying distributed algorithms is the need to exchange control information among the users, either explicitly or implicitly. Using this information, all users then execute independently the same algorithm resulting in some coordination in their actions. With distributed control the system is not dependent on the operation of a central scheduler. The performance of distributed control is superior to that of centralized control, especially when dealing with long propagation delays such as those using satellite channels.

Clearly it is essential that all users receive the same information regarding the demand placed on the channel and its usage in order to achieve a global optimum, and thus distributed algorithms are most attractive in fully connected systems. This attribute is not (always) present in indoor radio environments. It is therefore that we choose centralized control and do not examine distributed control algorithms. To assure connectivity with the central station, all stations have a line-of-sight link with the central station. Such a line-of-sight link will improve the channels quality because its fading characteristics are Ricean.

### 4.4 adaptive strategies and mixed modes.

It is possible to improve a specific access scheme to a certain extent by dynamic control. Such an adaptive control scheme does not change the nature of the access scheme nor the nature of its limitation. Dynamically controlled random access schemes provide improved packet delay over uncontrolled versions but still exhibit channel capacity less than 1. Actually what one really needs is a strategy for choosing an access mode which is itself adaptive to the varying need so that optimality is maintained at all times. Clearly, in order to accomplish adaptivity, a certain amount of information is needed by the decision maker(s). In the case of BTMA we could use the busy tone for this purpose: If for some reason the system is overloaded, the the busy tone can be switched on to reschedule packet transmission at the user's side.

### 4.5 Which multiaccess protocol ?

We have so far examined a number of multiaccess schemes and compared their performance. One thing is clear: each of these schemes has its advantages and limitations. No one scheme performs better than all others over the entire range of system throughput. If a scheme performs nearly as well as perfect scheduling at low input rates, then it is plagued by a limited achievable channel capacity. Conversely, if a scheme is efficient when the system utilization is high, the overhead accompanying the access control mechanism becomes prohibitibly large at low utilization. This controversy is illustrated in figure 4.4 for a polling system and two multiaccess schemes.

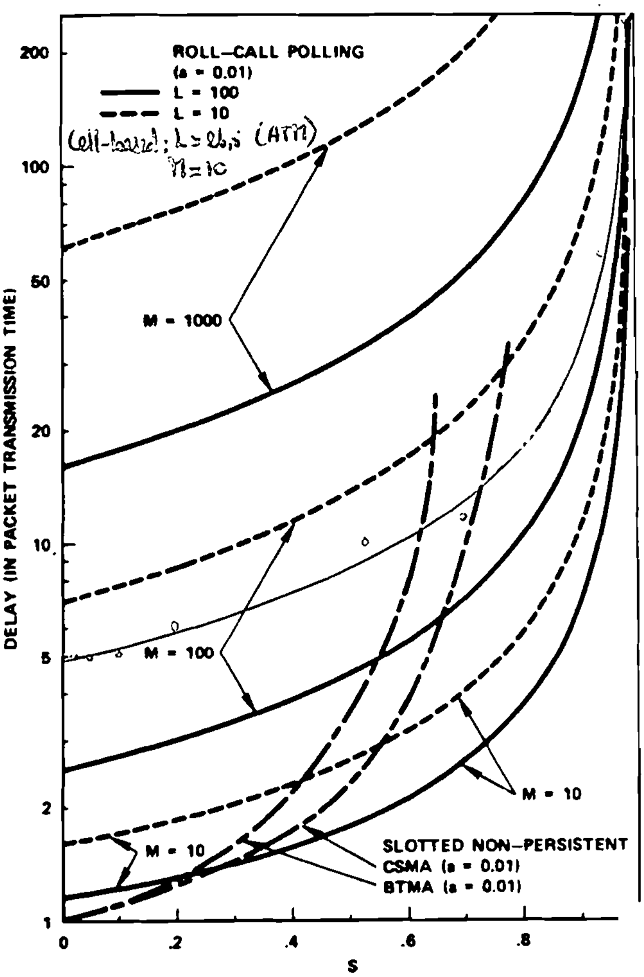


*Fig. 4.4: Packet delay in roll-call polling. L = ratio of data message length to polling message length. a = normalized propagation delay, M = number of stations [32].*

In the future BISDN, much data traffic is characterized as bursty, e.g. interactive terminal traffic. Burstiness is a result of the high degree of randomness seen in the message generation time and size, and of the relatively low-delay constraints required by the user. If one were to observe the users' behaviour over a period of time, one would see that the user requires the communications resources rather infrequently;

but when he does, he requires a rapid response. That is, there is an inherently large peak-to-average ratio in the required data transmission rate.

If fixed subchannel allocation schemes are used, then one must assign enough capacity to each subscriber to meet his peak transmission rates with the consequence that the resulting channel utilization is low. A more advantageous approach is to provide a random access scheme. The strong law of large numbers guarantees that, with a very high probability, the demand at any instant will be approximately be equal to the sum of the average demands of that population. The existence of some positive ackwowledge scheme permits the transmitter to determine if its transmission has been successful or not.

As mentioned, the performance of a random access scheme depends on the propagation delay, or $a = \tau/T$ (see figure 4.3). We can express a as:

$$a = \frac{\tau}{T} = \frac{l}{c\ T} = \frac{l r_s}{c b_m} \tag{4.1}$$

where l: is the maximum distance between a terminal and the base station,

$r_s$: is the symbolrate,

c : is the speed of light,

$b_m$: is the packetlength in bits,

T : is the packetlength in seconds.

It should be noted that while in CSMA sensing the presence of a transmission involves a one-way propagation delay a, BTMA requires a round trip delay 2a to perform the same operation. We now try to estimate the (maximum) value of a.

We assume $l = 30m$ and $r_s = r_b \le 2$ Mbit/s (without equalization). The random access protocol is applied to the signalling or request channel. Such a packet will at least contain a starting delimeter (8 bits), the source adress of the transmitting terminal (8 bits) and a control field indicating which service has to be performed (8 bits). The miminum value for $b_m$ is then 24 bits. With $c = 3.10^8$ m/s, the worst case propagation delay becomes 2a = 0.017. In fig. 4.3 it is shown that for this value of a, CSMA performs better than (slotted) ALOHA. As said, BTMA'S performance is comparable with the performance of CSMA and can be applied to radio networks. It is therefore we choose BTMA as the multiaccess scheme for operating the signalling channel.

## 4.6 Conclusions.

* A multiaccess scheme has to be used to allocate ATM cells to different users. The best option for this multiaccess scheme is 'Busy Tone Muliple Access (BTMA)', applied to a common signalling channel. The delay performance of BTMA is comparable with CSMA but the throughput will be 70% maximum.

* The radio network will use centralized control because of the varying network connectivity. Each user will have a line-of-sight link with a base-station. The fading characteristics of these links will be Ricean. The busy-tone channel can be used for adaptive strategies to improve the network performance during heavy load.

## 5. A radio-based ATM protocol.

As yet the protocol to be used has the following properties:

* There are two types of channels to be used:
  - a: The message channel over which the ATM cells are transported,
  - b: The signalling channel. This channel is split in two subchannels: A request channel on which BTMA is performed and an acknowledge channel which performs all acknowledgement functions.
* The control of the signalling procedure is centralized. To assure the connectivity with the base station we assume all terminals have a line-of-sight link with the base station.

According to the ATM principle explained in chapter 2, a terminal uses the signalling channel to make a reservation for a certain amount of capacity in the message channel. A user begins the signalling procedure by listening to the busy tone signal, and if the signalling channel is free, by sending a request packet to the base station. The base station performs a scheduling function; it compares the capacity request with the actual available capacity of the message channel. if there is sufficient capacity in the message channel, the requesting user is granted a number of time slots in which it can transmit its ATM cells, or in ISDN terms: The user is granted a virtual channel. At the end of the communication session one of the communicating users sends another request to abort the virtual channel.

At the point of transmitting ATM cells, an essential difference between fiber-based ATM and radio-based ATM arises. In fiber-based ATM (chapter 2) the offered cells of many users are buffered, and, according to which cell is first to leave the buffer, the cells are transmitted via the message channel. The message channel will be idle if the buffer is empty. In radio-based ATM it will be different: A user reserves certain time slots in which the user (and only that user) may send ATM cells. if a user has no cell to send within a particular time slot it will stay idle, independent of the offered cells of other users.

A comparable problem arises when a user (temporary) sends more cells than he is allowed to. In fiber-based ATM these extra cells are also put in the buffer. The resulting effect upon the system is nihil when the user makes use of the idle periods of other users, or increasing (temporary) the message channel load when the other users deliver cells at the normal rate. In radio-based ATM a user can never send more cells than he is allowed to, because he has reserved a limited number of time slots. Therefore the cells accumulate at the user's side, creating a waiting queue and increasing the delay per cell.

To avoid unacceptable delay of cells or idle time slots in the message channel, one can use the following strategy: Whenever the queue at the user's side becomes too long (depending upon the computable delay time), the user makes a request for additional capacity by sending a new request to the base-station. If the base station in turn detects that too many time slots are idle it decreases the capacity assigned to the user.

The next problem is to synchronize the different users with the base station. By synchronizing we don't mean the bit timing control, but rather the following question: How does a user know that a particular time slot is assigned to him ? One option is to make use of internal counters on behalf of which a users decides when to transmit. However, this means that time slots have to be assigned to the users in a predetermined fashion according to a limited set of time slot patterns. But, when the supported services use different fractions of the message channel capacity, this may not always be possible. Another option is to use the signalling channel and to acknowledge a user every time one or more time slots are assigned. This method will make the schedulers task less stringent and synchronizes the network easily; after receiving an acknowledge, a user may begin transmitting a cell.

In figure 5.1 a real time diagram is given of the radio-based protocol. We assume two users (termals T1 and T2) and a base station (B). The propagation delays between the users and the base station are $\tau_1$ and $\tau_2$ respectivily. We assume further that:

- There is one common signalling and message channel,
- a request is made for every ATM cell to be transmitted on the message channel,
- the busy-tone detection time and the packetgeneration times are zero,
- the propagation delay between T1 and T2 is $\tau_1+\tau_2$.

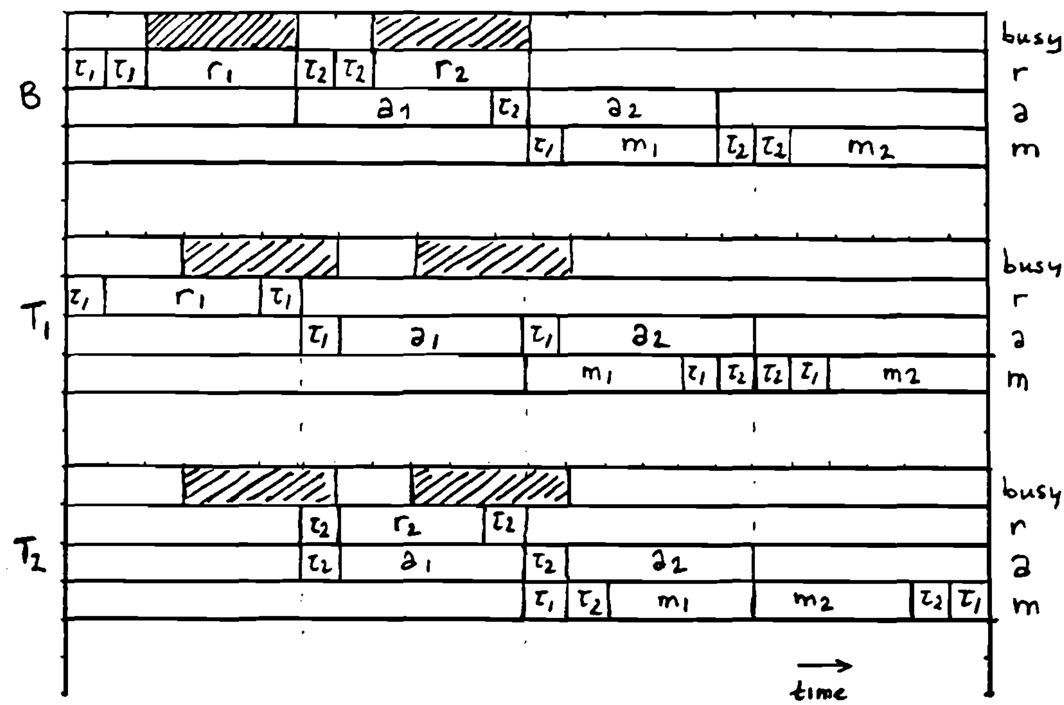


*Fig. 5.1: A real time diagram of the radio-based ATM protocol.*

Figure 5.1 can be understood as follows: At the beginning the request channel is idle, so the base station sends no 'busy-signal' on the busy-tone channel; T1 detects this after $\tau_1$(s) and makes a request which arrives $\tau_1$(s) later at the base station.; the base station senses the request channel busy and turns on the busy-tone; after having received the request the base station turns off the busy-tone and sends an acknowledge which is received by both T1 and T2; after having received the acknowledge, T1 starts sending the ATM cell. The same procedure is executed for user T2.

From figure 5.1 we learn several things:

- The propagation delay plays an important role in synchronizing the network. The time between succeeding transmissions of the same station is at most $2\tau$ ($\tau$ is the maximum propagation delay between any terminal and the base station).
- The transmission time of the acknowledge, $T_a$, has to obey

$$\tau + T_a = 2\tau + T_m \tag{5.1}$$

  Where $T_m$ is the transmission time of an ATM cell.
- The efficiency of the message channel is at most:

$$e_m = \frac{T_m}{T_m+2\tau} \tag{5.2}$$

  and the efficiency of the acknowledge channel is at most:

$$e_a = \frac{T_a}{T_a+\tau} \tag{5.3}$$

  The efficiency of the request channel depends on the access scheme used (BTMA, which is comparable with CSMA), the propagation delay and the offered load. The efficiency can be approximated by combining the figures 4.2 and 4.3.

### 5.1 System efficiency.

To determine system efficiency we have to estimate some basic parameters. We will begin with the length of request ($b_r$), acknowledge ($b_a$) and ATM packets ($b_m$). We assume we will not use an equalizer for the signalling channel i.e. the bitrate of both the request and the acknowledge channel is limited by 2 Mbit/s.(using a 1 bit/s/Hz modulation scheme). We further assume that the line-of-sight links between terminals and base station provide a signalling channel with very small BER and that the amount of packet retransmission is neglectible ($P_e<10^{-3}$).

A 'request' makes a reservation for a certain capacity (time slots) for a certain user. Therefore a 'request' should contain an address field and a control field. The addressfield should be able to discriminate about 200 users which requires 8 bits. The control field may have a multifunctional use, so we reserve again 8 bits. To indicate the beginning of a request message we use a flag (8 bits). To assure the information of the 'request' is received correctly, we include an error check (8 bits). the format of a request message will therefore be $b_r$=32 bits.

The 'acknowledge' is used to assign a particular time slots to a user. It therefore requires an address field and a control field to, for instance, indicate the number of time slots assigned or to indicate the retransmission of an incorrectly received ATM cell. Including a flag and an error check, the 'acknowledge' will contain $b_a$=32 bits. For the ATM cell we assume the format is alike the specifications of fig. 2.2 i.e. $b_m$=53x8=424 bits (bruto), though the header will perform different (address) functions.

The efficiency of the system will further be influenced by:
- The propagation delay,
- the overhead created by the signalling channel,
- the throughput and delay performance of the signalling channel,
- traffic statistics and cell generation statistics,
- acceptable delay.

We will discuss the influence of the first three points for the worst case situation. The influence of statistics and delay will be discussed in the following two sections.

The propagation delay $\tau$ influences both the signalling and de message channel. The influence on the request channel is described in section 4.2. Assuming a 2 Mbit/s bit rate, $b_r$=32 bits and l=30 (m), then a=6.25.10$^{-3}$. Using BTMA gives 2a=0.0125 and an effeciency of 70% for the request channel. The efficiency of the acknowledge is given by eq. 5.3 and gives approximately 99% when using $r_a$=2 Mbit/s and $b_a$=32 bits. If we assume that each ATM cell is accompagnied by an acknowledge which obeys $T_a \approx T_m$ (eq.5.1) then the message channel bitrate is found as:

$$r_m = \frac{b_m}{b_a} r_a \tag{5.4}$$

where $b_m$ = 424 bits,
$b_a$ = 32 bits,
$r_a$ = 2 Mbit/s.

Therefore $r_m$ will approximately be $r_m$=26.5 Mbit/s. The efficiency of the message channel will also be approximately 99% (eq. 5.2).

Thus, to operate a 26,5 Mbit/s ATM channel (with these parameters) we require a 2 Mbit/s acknowledge channel, and, depending upon how many ATM cells can be reserved by one request, a request channel bitrate of 2 Mbit/s maximum. We consider two cases:

1. A request is made for each ATM cell. Using a 2 Mbit/s request channel with an efficiency of 70%, we only assign 70% of the ATM cells: The system efficiency therefore becomes 0.7x(26.5/26.5+2+2)≈ 0.61.

2. A request is made solely for beginning and ending a session which involves many ATM cells. The request channel capacity used to operate the whole system will then be a fraction (few percent) of the total capacity and therefore neglected. By overdimensioning the request channel the usage of the available ATM cells can reach almost 100%. The system efficiency then becomes (26.5)/(26.5+2)≈ 0.93.

The nett capacity i.e. the bits to be used by the users, are the bits in the information field of fig. 2.2. Therefore the nett capacity of a 30.5 Mbit/s message channel will vary between 16.8 and 22.3 MBit/s. Actually it will be even a few percent less when we use an equalizer because this introduces extra overhead.

The two cases presented can be related to the kind of traffic using the network. A group of highly bursty users will often make either requests for starting a (short) session or requests for additional capacity because of their inherent high peak-to-average traffic ratio. Therefore the system efficiency reaches about 60% for highly bursty users, which is comparable with ,for instance, the 70% of BTMA. for non-bursty users (for example: telephone) the system efficiency can reach 90% because of their low utilization of the request channel. As the system is used by both bursty and non-bursty users the system efficiency will be somewhere in between. The exact dimensions of the request channel have to be determined using traffic data and the radio link quality; the radio links are characterized by packet error probability which in turn determine the number of requests for packet retransmission.

Finally we present some general methods to improve the system efficiency:

- By increasing the ATM packet length ($b_m$) the relative overhead, which is proportional to $(b_a+b_r)/(b_m)$, decreases. The message channel efficiency denoted by eq. 5.2 increases too. However, the ATM cell length is limited by the acceptable information delay and system flexibility.
- Using lower bitrates will at a given point remove the necessity for equalization. Equalization causes some additional overhead. The transmission time of a packet increases, thereby increasing the channel efficiencies of the request channel (parameter a decreases) and both the acknowledge and message channel (eq. 5.2 and 5.3).
- If we use one acknowledge to assign several successive timeslots to one user, we reduce the relative overhead. The drawbacks are the same as in the first method.

## 5.2 Message delay.

In the ATM-based indoor radio network we can distinguish two kinds of delay:

1. Session-delay: the time between the moment a user wants to start a session and the moment he is granted enough bandwidth by the network to start one.

2. Cell-delay: the time between an ATM cell is ready for transmission and the moment it is actually received by the addressed station.

We will successively discuss both kinds of delay.

**5.2.1 Session-delay.**

As stated in the explanation of the ATM protocol a user starts a session by sending a request to the base station, asking for a certain amount of capacity. If this capacity is not available at the moment of request, the request packet joins a queue of other contending request packets. If extra capacity comes available in the message channel (because other users terminate their session) this capacity is allocated to one of the requesting users, depending upon the discipline used. Therefore the session-delay can be seperated in:

$D_1$: The time for a request packet to be successfully received at the base station.

$D_2$: The time a request packet waits in the queue.

In the analysis we assume that all the users collectivily form an independent Poisson source with an aggregate mean session generation time of $\lambda$ sessions/s.

ad $D_1$: Consider the request channel operated in a BTMA mode. Let $S_r$ denote the packet input rate of the request channel normalized with respect to the request transmission time $T_r$. This input rate is composed of three kinds of requests:

- requests for starting a new session, i.e. $\lambda * T_r$,

- requests for ending a session. In the long run, all sessions that enter the system will finally end, so we approximate the contribution for ending a session by $\lambda * T_r$.

- requests for retransmission of incorrect received cells. The mean number of incorrect received cells per second is $(P_e\ e)/T_m$, where $P_e$ is the cell error probability mentioned in section 3.2, e is the total message channel efficiency and $T_m$ the cell transmission time. Therefore the contribution of these requests to the mean input rate is $(P_e T_r e)/T_m$.

If we let $D_{BTMA}(S_r, a_r)$ denote the normalized delay of the BTMA operated request channel as a function of the normalized input rate $S_r$ and the normalized propagation delay $a_r$, then:

$$D_1 = D_{BTMA}(S_r, a_r) . T_r \tag{5.5}$$

$$S_r = (2\lambda + \frac{P_e\ e}{T_m}) . T_r \tag{5.6}$$

$$a_r = 2\tau/T_r \tag{5.7}$$

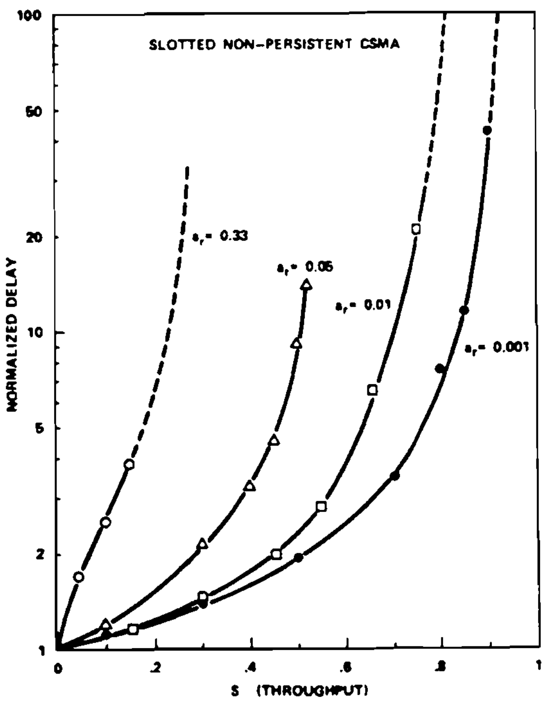


*Fig. 5.2: Slotted non-persistent CSMA: packet delay for various* $a_r$.

The delay performance of BTMA($S_r$,$a_r$) may be approximated by that of CSMA($S_r$,$a_r$) for $S_r$ < 0.5 so we may use the results of fig.5.2 [34].

ad $D_2$: To estimate the delay $D_2$ we make the following assumption: The output of the request channel (defined as the process corresponding to the arrival of succesful requests at the base station) is Poisson with a mean of $\lambda$ requests per second. In [34] it is verified that for interdeparture times (i.e. time between successive succesfull received requests) longer than one or two cell transmission times, this assumption is acceptable. This assumption becomes more realistic if $S_r$ is decreased. Under this assumption, the message channel can be modelled as an M/G/m queuing system, depending upon session traffic characteristics.

As an example of such a M/G/m system we have simulated the following system: The request arrival process is an independent Poisson process with an aggregate mean session generation rate of $\lambda$ sessions/s. The service time of each session is exponentially distributed. All sessions can be divided into several service-classes i, which are each characterized by $\lambda_i$ = the mean generation rate of service-class i, $1/\mu_i$ = the mean service time of service-class i and $c_i$ = the required capacity for service-class i. These service-classes reflect the different users of a future ISDN network. As all service-classes are independent, the aggregate mean generation rate becomes:

$$\lambda = \sum_i \lambda_i \tag{5.8}$$

The message channel has a capacity CMAX. Each time a session i enters the message channel, the resulting message channel capacity C ($C = CMAX - \sum c$) is diminished with $c_i$. If at one time the resulting message channel capacity C is less than an incoming capacity request $c_i$, the request packet joins the queue. The moment a session j in progress ends, the resulting message channel capacity C is increased with $c_j$ and is compared with the first capacity request in queue. If this capacity request is less than C, the session can enter the message channel. If the capacity request of the first session in queue is bigger than C, the second session in queue is compared with C. This goes on until one or more sessions in queue may enter the message channel, or, all waiting sessions stay in queue and the free capacity in the message channel stays idle. This algorithm can be repeated every time a session enters or leaves the system for a certain amount of (entering) sessions NMAX.

The algorithm described above has been simulated by means of a pascal program, called SESSION on a personal computer. The program text, the principles it is based on and the way of operating it, are explained in appendix A. The input parameters are:

- The number of service-classes,
- the total message channel capacity CMAX,
- the parameters $c_i$, $\lambda_i$ and $1/\mu_i$ of each service class and
- the number of sessions NMAX to enter the system.

During the execution of the simulation the following parameters are calculated:

- the utilization factor $\rho$, which gives the fraction of the message channel in use, i.e. is assigned to incoming sessions.
- the mean waiting time $\bar{w}$ for each service class.

If just one service class is chosen, the system behaves like the well known M/M/m system, with m=CMAX/c. If m=1, the utilization factor $\rho$ and the mean session delay $\bar{w}$ of the M/M/1 system are given by [38]:

$$\rho = \lambda/\mu \tag{5.9}$$

$$\bar{w} = 1/(\mu-\lambda) - 1/\mu \tag{5.10}$$

As a comparison between simulation and theory, we have tabulated $\rho$ and $\bar{w}$ calculated by eq. 5.9 and 5.10 and values for $\rho$ and $\bar{w}$ obtained by simulation, for different values of NMAX for a single simulation run.

*Table 5.1: Comparison between $\rho$ and $w$ obtained by theory and simulation (M/M/1).*

| | | theory | simulation | | |
|---|---|---|---|---|---|
| NMAX: | | | 100 | 1000 | 10000 |
| $\lambda$ | $\mu$ | $\rho$ | $\rho_{sim}$ | | |
| 1 | 100 | 0.01 | 1.03E-2 | 9.66E-3 | 1.03E-2 |
| 1 | 10 | 0.10 | 8.26E-2 | 9.88E-2 | 9.92E-2 |
| 1 | 2 | 0.50 | 4.93E-1 | 5.02E-1 | 5.09E-2 |
| 1 | 1.11 | 0.90 | 9.67E-1 | 9.41E-1 | 9.21E-1 |
| 1 | 1.01 | 0.99 | 8.17E-1 | 9.57E-1 | 9.91E-1 |
| $\lambda$ | $\mu$ | $\overline{w}$ | $\overline{w}_{sim}$ | | |
| 1 | 100 | 1.01E-4 | 1.19E-4 | 9.31E-5 | 1.06E-4 |
| 1 | 10 | 1.11E-2 | 7.77E-3 | 1.10E-2 | 1.09E-2 |
| 1 | 2 | 5.00E-1 | 2.74E-1 | 5.30E-1 | 5.17E-1 |
| 1 | 1.11 | 8.19 | 9.39 | 9.59 | 8.37 |
| 1 | 1.01 | 99.0 | 7.42E-1 | 9.19 | 63.53 |

Looking at table 5.1 we see that, in general, the approximations obtained by simulation are more accurate for growing NMAX. This is not surprising because 5.9 and 5.10 have actually been derived for steady state conditions. In the initial period of the simulation this steady state will not have been reached, introducing lower values for utilization and waiting time. When we compute the averages, the effect of these deviations will decrease for increasing NMAX. This effect is clearly shown for the waiting time of $\rho$ =0.99. In this case the initial period is rather long because of the high ratio $\lambda/\mu$. This causes long waiting times for the cells in queue. A second effect is, that a lot of cells will still be in queue at the end of the simulation, contributing only a fraction of their supposed waiting time to the average. In conlusion, table 5.1 shows that the relative deviation between simulation and theory is approximately 5% or less, for both $\rho$ and w, taking $\lambda/\mu$ between 0.1 and 0.9 and NMAX=10000 (single run). The effect can be reduced by averaging the output of several simulation runs.

In the real system, variations of $\lambda$ and $\mu$ will occur. For the M/M/1 case, the sensitivity for small variations can easily be expressed by the derivations of $\rho$ and $\bar{w}$:

$$d\rho/d\lambda = 1/\mu \tag{5.11}$$

$$d\rho/d\mu = -\lambda/\mu^2 = -\rho/\mu \tag{5.12}$$

$$d\bar{w}/d\lambda = 1/(\mu-\lambda)^2 = 1/\mu\ [1/(1-\rho)^2] \tag{5.13}$$

$$d\bar{w}/d\mu = 1/\mu^2 - 1/(\mu-\lambda)^2 = 1/\mu^2\ [1-1/(1-\rho)^2] \tag{5.14}$$

In general we see from equations 5.11 t/m 5.14 that the sensitivity to both $\rho$ and $\bar{w}$ increase with growing mean service time $1/\mu$ and growing $\rho$. When the system is joined by several session classes, the sensitivy of the whole system will be less.

With the help of the program SESSION it is possible to estimate system dimensions. By setting limits to the mean session delay $\bar{w}$ of each session class, one can estimate the necessary message channel capacity CMAX. As an example we will use the program in section 5.3 to design an imaginary system.

**5.2.2 Cell-delay.**

The moment a user is granted some amount of capacity, the base station has to assign time slots for ATM cells to that user. The various service-classes, which are assumed to be bursty, have very different traffic statistics such as average bit rate, peak bit rate and traffic periodicity. At this point two problems can be distinguished:

1. How should time slots be allocated to a session with fluctuating bit rate ?

2. What are the time slot reservation criteria for bursty service-classes ?

To solve these problems, one should have knowledge of two processes:

1. The traffic statistics of the cell arrival process at the transmitting user.

2. The algorithm the base station uses to assign timeslots to a virtual channel.

To achieve insight in these problems we start a (relatively simple) model. We assume the cell arrival process of a session to be a Poisson process with a mean cell arrival rate of $\lambda_c$ cell/s. $\lambda_c$ can be expressed as:

$$\lambda_c = c/b_m \tag{5.10}$$

where c: is the mean number of bits per second generated at the user's side,

$b_m$: is the number of bits in an ATM cell.

To transport these cells through the network, the user requests a capacity of $c_r$ bits/s, or $c_r/b_m$ timeslots/s. The requested time slots can be assigned to a user in various ways. We assume that the time slots are allocated to a user in such a way that the interarrival time between assigned time slots is a constant. Therefore the requested time slots are uniformly spread over the total amount of time slots in the message channel. The constant service time of each cell (transmission time plus interarrival time), denoted by $1/\mu_c$, can therefore be written as:

$$1/\mu_c = \frac{CMAX/b_m}{c_r/b_m} T_m = \frac{CMAX}{c_r} \frac{b_m}{CMAX} = \frac{b_m}{c_r} \tag{5.11}$$

where: CMAX: is the total message channel capacity in bits/s,

$T_m$: is the transmission time for an ATM cell.

So far, we have modelled the cell arrival process as a Poisson process (M) and the cell service time being constant or deterministic (D). For simplicity, we are going to make use of results of the standard M/D/1 queuing system theory [39]. At the point of time slot allocation there is only a slight difference between the two models (fig. 5.3).

In figure 5.3 $t_1$, $t_2$ and $t_3$ represent arrivals of cells at the user's side, which have to be allocated to time slots of seize $T_m$. The cell transmission time and the interarrival time between time slots is $1/\mu_c$ and denotes the constant service time of each incoming ATM cell.

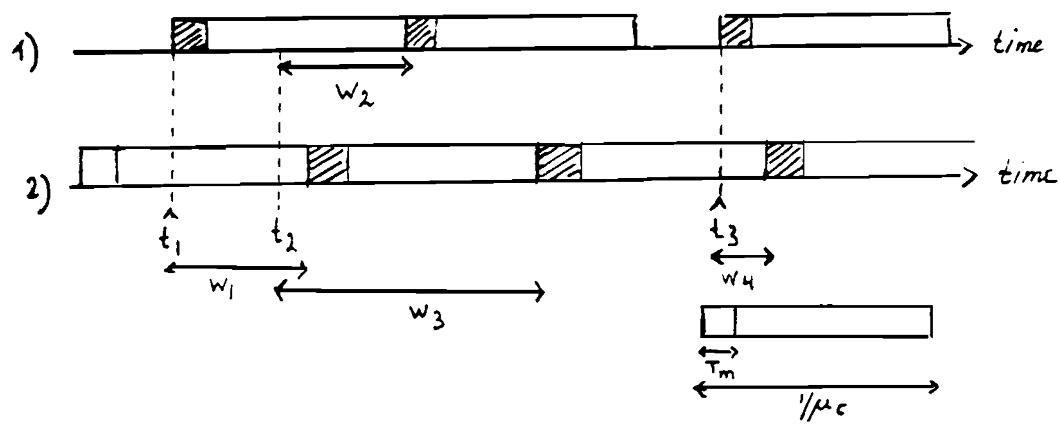


*Fig. 5.3: Time slot allocation in a virtual channel according to 1) the M/D/1 queuing system 2) a possible ATM allocation scheme.*

In the M/D/1 system, the assignment of time slots to cells is instantaneous ($t_1$, $t_3$) if the cell does not arrive within the service time of a previous arrived cell. Therefore, the waiting time in this case will be zero. In the uniform ATM allocation scheme, an incoming cell has to wait a time $w_1$ before transmission, where w ranges from 0 to $1/\mu_c$. If the arrival time takes place in the service time of the previous cell, the difference in waiting time between the two systems again ranges from 0 to $1/\mu_c$. Therefore the cell delay $D_c$ of the ATM allocation scheme is less or equal to the cell delay of the M/D/1 system raised with $1/\mu_c$:

$$D_{M/D/1} \le D_c \le D_{M/D/1} + 1/\mu_c \tag{5.12}$$

where $D_{M/D/1}$ is the delay caused by the M/D/1 queue. In both systems, the idle time of the virtual channel is the same, which means that both systems will have the same utilization factor $\rho_c$, defined as the fraction of assigned time slots which are actually used by ATM cells. This utilization factor can be expressed as:

$$\rho_c = \lambda_c/\mu_c = c/c_r \tag{5.13}$$

In [39] a detailed analysis is given of M/G/1 systems. The results can be applied to the M/D/1 case. It is stated that the Laplace transform W(s) of the queue waiting time distribution w(y) is given by:

$$W(s) = \frac{s(1-\rho)}{s-\lambda+\lambda B(s)} \tag{5.14}$$

where B(s) is the Laplace transform of the service time distribution b(x). Applying this theory to the M/D/1 system gives:

$$b(x) = \delta(x-1/\mu_c) \tag{5.15}$$

$$B(s) = e^{-s/\mu_c} \tag{5.16}$$

$$W(s) = (1-\rho_c)\frac{s}{s-\lambda_c+\lambda_c e^{-s/\mu_c}} \tag{5.17}$$

The problem here, is to transform W(s) into the desired waiting time distribution w(y). So far, it has not been possible by the author, to transform eq. 5.17 by the standard methods. As advanced numerical analysis is very time consuming, we use another result of [39]: The z-transform Q(z) for the distribution of the number of cells in an M/G/1 queue is given by the Pollaczek-Khinchin (P-K) transform equation:

$$Q(z) = B(\lambda-\lambda z)\frac{(1-\rho)(1-z)}{B(\lambda-\lambda z)-z} \qquad (M/G/1) \tag{5.18}$$

where $B(\lambda-\lambda z)$ is the Laplace transform $B(s)$ of the service time distribution where s is substituted by $\lambda-\lambda z$. Applying equations 5.15 and 5.16 for the M/D/1 case, we obtain:

$$Q(z) = (1-\rho_c) \frac{1-z}{1-ze^{\rho_c} e^{-\rho_c z}} \qquad (M/D/1) \qquad (5.19)$$

If we want to transform eq. 5.19 into the desired distribution, we encounter the same problems as mentioned earlier in the case of the Laplace transform. However, in the case of a z-transform, we can make use of the intermediate value theorem. This theorem states that, if the z-transform is given by:

$$F(z) = \sum_{n=0}^{\infty} f_n z^n \qquad (5.20)$$

then $f_n$ can be expressed as:

$$f_n = \frac{1}{n!} \frac{d^n F(z)}{dz^n} \Big|_{z=0} \qquad (5.21)$$

According to this theorem, we can obtain the probability P(q=k) of having q=k cells in the M/D/1 queue as:

$$P(q=k) = \frac{1}{k!} \frac{d^k Q(z)}{dz^k} \Big|_{z=0} \qquad (5.22)$$

where Q(z) is given by equation 5.19.

Analytical expressions for P(q=k) have been derived for k = 0 - 9 with the help of a program called REDUCE. The probability of having Q or more cells waiting in queue is given by:

$$P(q \geq Q) = 1 - \sum_{k=0}^{Q-1} P(q=k) \qquad (5.23)$$

and is plotted in figure 5.4 as a function of $\rho_c$ for Q=1,..,10. Fig. 5.4 shows clearly that if we desire a high utilization of the assigned capacity $c_r$, the probability of having a high number of cells waiting, increases rather fast.

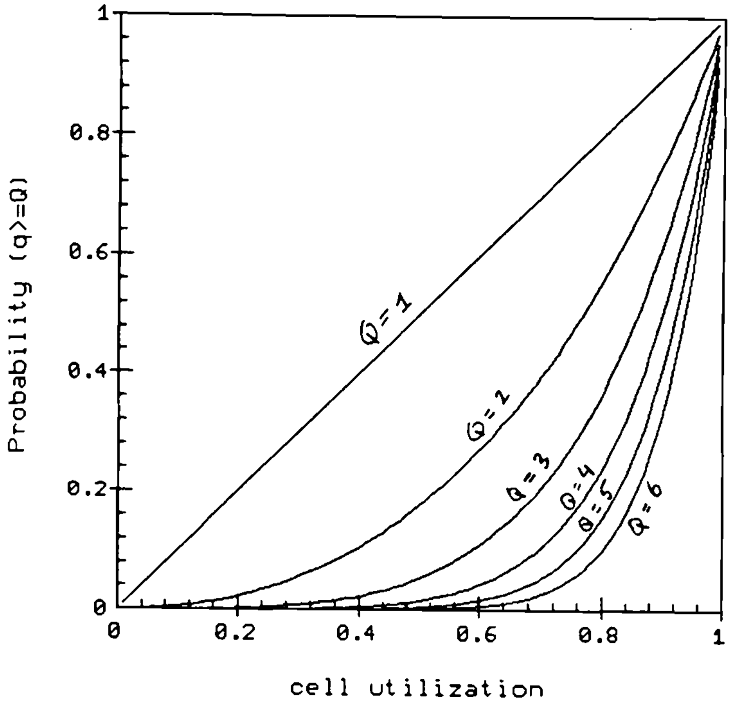


*Fig. 5.4: P(q≥Q) as a function of $\rho_c$ for the M/D/1 system.*

Having obtained expressions for the probability of having Q or more cells in the M/D/1 queue, we now try to relate P(q≥Q) to cell-delay. The P-K formula given in eq.5.18 has actually been derived at the instants a cell has just been served, so the state variable q is the number of cells left behind in queue by such a served cell (fortunately, the solution at these points happen also to provide the solution for all points in time, and therefore $\rho_c$ =1-P(q=0)).

Now consider the case of the M/D/1 queuing system and assume, that there are Q cells in queue, the moment a cell has been served. The time lapse between that moment and the moment the Q-th cell in queue will have been served, is $Q/\mu_c$ maximum. Furthermore, the arrival of this Q-th cell had to be in the serving time of the previous cell, because else it could not be the Q-th in queue. As a result of this, the maximum delay of a Q-th cell in queue will be $(Q+1)/\mu_c$, and hence:

$$P(D_{M/D/1} \geq (Q+1))/\mu_c) \leq P(q \geq Q) \tag{5.24}$$

and when we make use of equation 5.12 we get:

$$P(D_c \geq (Q+2)/\mu_c) \leq P(q \geq Q) \tag{5.25}$$

Equation 5.25 gives the upperlimit to the probability that a cell has a certain delay when he is the Q-th in queue (Q>0). Note that the accuracy of eq. 5.25 is in the order of $1/\mu_c$, so it is not valid if the cell delay constraints are a fraction of the service time $1/\mu_c$ of a cell.

What we finally want is a figure in which the probability is plotted versus increasing delay, for several utilization factors $\rho_c$. Such a plot enables us to see the direct relationship between acceptable delay and the efficiency of the granted capacity. For this purpose we need a reference on the delay axis, because $1/\mu_c$ is dependent on $c_r$ and therefore on $\rho_c$. We define:

$$1/\mu_{ref} = b_m/c \tag{5.26}$$

The idea behind $1/\mu_{ref}$ is that all delays are normalized to the service time of a cell in the case we would (theoretically) chooze $\rho_c=1$. Hence, we vary $\rho_c$ by keeping $\mu_c$ constant and changing $\lambda_c$. In the real system $\lambda_c$ will be the constant and $1/\mu_c$ varies, but the effect on $\rho_c$ will be the same. We can write:

$$\rho_c = \lambda_c/\mu_c = \lambda_{ref}/\mu_{ref} = c/c_r \tag{5.27}$$

$$P(D_c \geq (Q+2)/\mu_c) = P(D_c\mu_{ref} \geq \rho_c(Q+2)) \leq P(q \geq Q) \tag{5.28}$$

In figure 5.5 we have plotted $\rho_c(Q+2)$ versus $(P(q \geq Q))$ for severa values of $\rho_c$ ranging from 0.1 to 0.9 and connected the values fo the same $\rho_c$ value with straight lines. The accuracy of 5.28 is i the order of $\rho_c/\mu_{ref}$ which is always smaller as $1/\mu_{ref}$. Figure 5. is therefore, at any case, valid for normalized delays longer tha 1, which is denoted by the dashed line in figure 5.5.

Fig. 5.5 can be used as follows: Suppose a user wants to transmit 512 kbit/s over the ATM network, such that the maximum

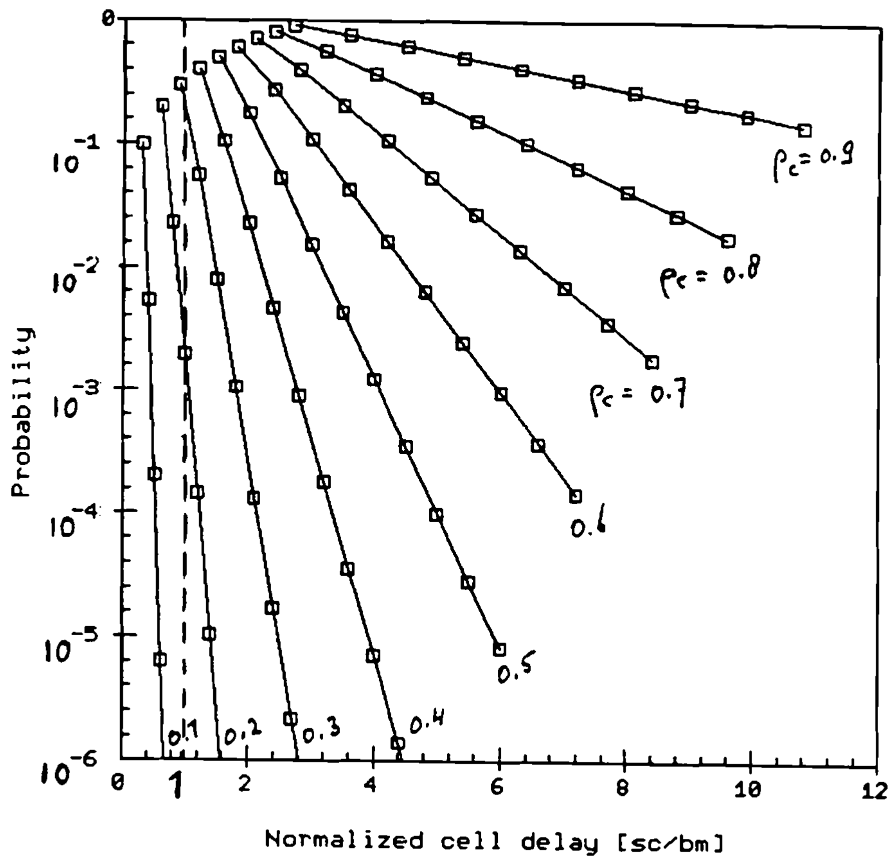


*Fig. 5.5: maximum normalized cell delay probability for different utilization factors.*

cell delay does not exceed 5 ms with a probability of $10^{-4}$. The question is: How much capacity should the session ask for to satisfy this condition ? To answer this question we calculate $1/\mu_{ref}$ as $b_m/c$=424/512000=0.828 ms. So the normalized delay becomes 5/0.828=6.04. The probability becomes $10^{-4}$. Now two straight lines can be drawn in figure 5.5, dividing the figure in four quadrants. All acceptable solutions will be in the left quandrant below the line P=$10^{-4}$. To achieve maximum utilization we choose $\rho_c \approx$ 0.54, so $c_r$ should be 948 kbit/s.

In conclusion we look at some general properties of cell delay, as a result of this time slot allocation method.In fig. 5.5 it is shown that increasing utilization of assigned time slots causes longer cell delay. The increase of cell delay itself becomes higher with increasing utilization. Methods for keeping $D_c$ small are first of all, choosing low utilization by requiring more capacity. Another option is to make $b_m$ smaller (choosing small cells), which lowers $1/\mu_c$. However, the relative overhead of the signalling channel would then increase. From eq. 5.11 it is clear that session classes which use low bitrates $c_r$, have great difficulty satisfying stringent cell delay requirements, because $1/\mu_c$ is relatively large. We will illustrate this problem in section 5.3.

If we assign cells to a user, based on its peak bit rate, cell delay will be minimal but system efficiency will be poor. Though it looks more appropiate to base the cell assignment on the average bit rate. However, as traffic fluctuates, temporary congestion of cells in the internal buffers of the stations leads to unacceptable cell delay. The best option will be to assign cells to a station according to its actual demand; if a bursty user has one cell to send it is quite likely he has more cells to send (burst). In fact this strategy has already been proposed in the explanation of the radio based ATM protocol, by asking additional capacity every time the queue at the user's side becomes too long. A result of this approach is that a user has a certain chance that the additional capacity can not be granted by the system and the session in question suffers from cell loss.

In fact, the option to ask for extra capacity whenever a burst of cells arises, makes the cell congestion problem a session congestion problem. Each burst can be seen as a short additional session for which certain limits on delay or blocking can be set which influence the dimensions of the total system. In fact, the problem of congestion at session, burst and cell level seem to interact and, because of their compexity, can be a subject for further study.

## 5.3 System design; an example.

To illustrate the use of different results of the previous sections we are going to design an ATM system. Therefore we define some input parameters which represent the different users of the system. We assume the system will be used by 200 users which can request the following session classes:

1: Telephone

2: Interactive dialog

3: File transfer

4: Video (smallband).

For each session class both session delay and cell delay constraints can be specified. We will assume that for each session class, the mean session waiting time is approximately 1 s. The cell delay constraints are different for each session class. Session classes which support speech like telephone and video have stringent delay constraints in the order of 6 ms, with a probability of $10^{-4}$. Interactive dialog and file transfer have less stringent cell delay constraints, which we assume to be approximately 100 ms with a probability of $10^{-4}$. The input parameters of the service classes are given in tabel 5.2.

*Tabel 5.2: Input parameters of assumed session classes.*

| session class i | bitrate $c_i$ (kBit/s) | $\lambda_i[s^{-1}]$ | $1/\mu_i[s]$ |
|---|---|---|---|
| 1 | 64 | 1.67E-3 | 120 |
| 2 | 9.6 | 7.50E-3 | 40 |
| 3 | 1000 | 5.55E-4 | 1 |
| 4 | 1500 | 8.33E-4 | 0.5 |

We start the design by estimating the capacity request of each service class satisfy the cell delay constraint. For telephone the cell arrival process is not Poisson, but deterministic. The maximum delay is therefore $1/\mu_c$ = 6ms maximum, and the reservated capacity for 64 kBit/s speech should be 70.7 kBit/s. To estimate the capacity requests of the other service classes we maken use of figure 5.5. We find:

*tabel 5.3: Request capacity and efficiency of assumed session classes.*

| session class i | normalized delay | $c_r$[kBit/s] | $\rho_{ci}$ |
|---|---|---|---|
| 1 | - | 70.7 | 0.90 |
| 2 | 2.26 | 32 | 0.30 |
| 3 | 235.8 | 1053 | >0.95 |
| 4 | 21.2 | 1875 | 0.80 |

In tabel 5.3 we see that the utilization factor decreases considerably when we have to meet stringent cell delay constraints (session class 2). As a result, the mean cell efficiency, which can be expressed as:

$$\bar{\rho}_c = \sum_i \lambda_i \rho_{ci}/\lambda \qquad (5.29)$$

will be $\bar{\rho}_c$ = 47%.

In tabel 5.4 the input parameters are given for the program SESSION. Note that the session generation rate $\lambda_i$ of each session class has be multiplied by the number of users (200).

*Tabel 5.4: The input parameters of the program SESSION.*

| session class i | $c_r$[kBit/s] | $200x\lambda_i$[1/s] | $1/\mu_i$[s] |
|---|---|---|---|
| 1 | 70.7 | 0.334 | 120 |
| 2 | 32 | 1.5 | 40 |
| 3 | 1053 | 0.111 | 1 |
| 4 | 1875 | 0.166 | 0.5 |

By increasing the total channel capacity we lower the mean waiting time per session. When CMAX is approximately 7.5 MBit/s, all session classes have a mean waiting time less then 1 s. The utilization factor is then 65 %.

As a result of the delay constraints the message channel efficiency will become 0.47 x 0.65 = 0.305, or 30.5 %.

To obtain the total system efficiency we have to know the bandwidth of the signalling channel. We assume that the mean delay of the signalling channel has to be less then 1% of the session delay, so the mean delay has to be less then 0.01 s. Therefore we use equation 5.6 and figure 5.2. From tabel 5.4 we compute $\lambda$ = 2.11 [1/s]. We choose BER to be $10^{-6}$ so $P_e$ will satisfy 4.24E-4 (equation 3.19) and $P_e e/T_m = 2.75$. In figure 5.2 we have to find the maximum value for $S_r = 6.97\ T_r$ such that the mean normalized delay expressed in $T_r = b_r/r_r$ is below 0.01 s. If we choose $T_r \approx 7.14$ ms then $S_r \approx 5\%$. The request channel bitrate becomes approximately 4480 bit/s. The bitrate of the acknowledge channel has to obey 5.4 so $r_a$ = 943 kBit/s. The system efficiency without delay constraints then becomes 7.5/(7.5+0.943+4.48E-3) ≃ 88.7%. The total system efficiency will be 0.305 x 0.887 = 0.27 or 27%.

## 5.4 Conclusions.

* An essential difference between radio-based ATM and fiber-based ATM is that in radio-based ATM a user has to reservate time slots in which it can transmit its ATM cells. A time slot will stay idle if its particular user has (temporary) no cells to transmit. If the cell generation rate is higher then the number of requested timeslots, accumulation of cells occur at the user's side.

* System efficiency will mainly be influenced by:
  - the propagation delay,
  - the overhead created by the signalling channel,
  - the performance of the signalling channel,
  - traffic statistics and cell generation statistics,
  - delay constraints.

* If there are no constraints for either session nor cell delay, the system efficiency will be somewhere between 60% and 90%, depending upon traffic statistics.

* Stringent delay constraints cause drastic reduction of system

efficiency.

* Session delay is determined by traffic statistics. A high number of users cause better resource sharing and therefore increases system efficiency.

* Cell delay is determined by the cell arrival statistics and the method of time slot assignment. Increasing utilization of assigned time slots causes longer cell delay. A time slot allocation method with small delay is crucial in a ATM radio based radio network that handles services with high delay constraints like video.

* The problems of delay and congestion at session, burst and cell level seem to interact and, because of their complexity, can be subject for further study.

## 6. Conclusions.

ATM is a communication technique which can support many services using different bitrates. ATM takes advantage of multiplexing a high number of users and low cell error probability.

The main problems of an indoor radio network are the broadcast effect, the time varying connectivity and bad radio channel performance.

An essential difference with fiber-based ATM is, that in radio-based ATM each user has to reserve certain capacity because the radio channel must be shared and therefore allocated among the various users. This causes less sharing of capacity on cell level and decreases system efficiency. To make reservations, a multiacces scheme has to be used for the signalling channel. The best option for such a scheme is 'Busy Tone Multiple Access'.

The ATM-based network will be centrally controlled. To limit the effects of time varying connectivity a line-of-sight connection between each user and the base station is recommended. The busy tone takes care of hidden terminals.

Radio channel performance is mainly limited by multipath effects. Assuming a slow-nonselective Rayleigh fading channel, the bitrate is limited by 1 à 2 Mbit/s and the SNR at the receiver should exceed 47 dB, using PSK, to meet stringent BISDN requirements. Slow selective channels are very hard to analyze. Fading effects will be less then in nonselective channels but an equalizer has to be used to remove intersymbol interference. A possible candidate is a DFE equalizer using the fast Kalman algorithm. The SNR at the receiver can be improved by space diversity, cyclical frequency hopping or a distributed antenna system.

Assuming low cell error probability and no constraints for neither session nor cell delay, the system efficiency is likely to be somewhere in between 60% and 90%, depending upon traffic statistics. Stringent delay constraints cause drastic reduction of system efficiency. Increasing system efficiency causes longer delays.

Session delay is determined by traffic statistics. A high number of users cause better resource sharing and therefore increases system efficiency.

Cell delay is determined by the cell arrival statistics and the method of timeslot assignment. A time slot allocation method with small delay is crucial in an ATM radio-based network that handles services with high delay constraints.

The problems of delay and congestion at session, burst and cell level seem to interact and, because of their complexity, can be subject for further study.

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APPENDICES.

A: The program SESSION.

Operation: The program SESSION simulates the allocation of message channel capacity (CMAX) to different sessions, taken from a limited set of (v) service classes. Sessions of service class i are assumed to:

- have interarrival times modelled as an independent Poisson proces with mean session generation rate $\lambda i$,
- have service times which are assumed to be exponentially distributed with mean service time $1/\mu i$,
- require a capacity ci during the time they are served.

The input of the program is by keyboard, after the program has asked for input parameters. After simulation both the utilization factor and the mean waiting time of each class are printed on the screen.

Input parameters:

| | |
|---|---|
| v: | number of service classes ($v \le 25$) |
| CMAX: | total message channel capacity |
| ci: | capacity request of service class i |
| $\lambda i$: | mean session aggregation rate of service class i ($i \le v$) |
| $1/\mu i$ | mean service time of a session of service class i. |

output parameters

$\rho$: utilization factor: defined as the mean fraction of the message channel capacity assigned to incoming sessions ($0 \le \rho \le 1$).

$\overline{w}i$: mean waiting time in queue for each service class i.

Queuing discipline: All incoming sessions join the queue. Th queuing discipline is showed in the following pseudo-pascal text.

```
  WHILE (sessions enter or leave the system) DO
  BEGIN
    take first in queue;
10  IF (capacity request ≤ idle message channel capacity)
```

```
    THEN BEGIN
           assign requested capacity to the session and
           decrease idle message channel capacity with assigned
           session capacity
         END
    IF (there is a next in queue) THEN GOTO 10
  END.
```

procedures:

PARC

inputparameters: tijd, capaciteit.

outputparameters: global (somcapac).

PARC adds the idle capacity of the message channel during the simulation and puts it in the global variable 'somcapac'. At the end of the simulation the ratio of 'somcapac' and the total simulation time gives the utilization factor.

PARW

inputparameters: wklasseno, wachttijd.

outputparameters: global (teller,noemer).

PARW adds the queuing time of each service class i in the global variable 'teller[i]' and the number of generated sessions of service class i in 'noemer[i]'. At the end of the simulation the ration of 'teller[i]' and 'noemer[i]' gives the mean waiting time of service class i.

GENERATE

inputparameters: global (lambda,klasse,duur,v,kans).

outputparameters: klasseno,Ta,Ts,capac.

GENERATE generates sessions according to the inverse distribution methode [40,blz 460] with 'klasseno' indicating the service class i, 'Ta' the interarrival time (exponentially distributed with mean $1/\lambda i$),'Ts' the service time (exponentially distributed with mean $\mu i$) and 'capac' the capacity request of the generated session.

INSERTSESSION

inputparameters: Te,ci,inT,no.

outputparameters: none.

INSERTSESSION puts sessionparameters in a register which keeps record of the sessions in the message channel. The recorded parameters are 'Te': the time a session will end and leaves the system and, 'ci': the capacity request of the session. The sessions are sorted on the values of 'Te'. The session with the smallest value for 'Te' is being pointed at by the global pointervariable 'Teminpointer' and is the first session to leave the sessionregister.

REMOVESESSION

inputparameters: none.

outputparameters: Te,ci.

REMOVESESSION removes the session, being pointed at by the global pointervariable 'Teminpointer', from the sessionregister(see also INSERTSESSION).

INSERTWACHTRIJ

inputparameters: wklasseno,wci,wts,inT.

outputparameters: none.

INSERTWACHTRIJ puts incoming sessions in a register which keeps record of the sessions in queue. The session class i, the capacity request and the generated session service time are denoted by 'wklasseno','wci'and 'wts' respectively. 'inT' denotes the time the session was generated and enters the queue. The first and last session in queue are pointed at by the global pointer variables 'wachtrijbegin' and 'wachtrij-eind.'

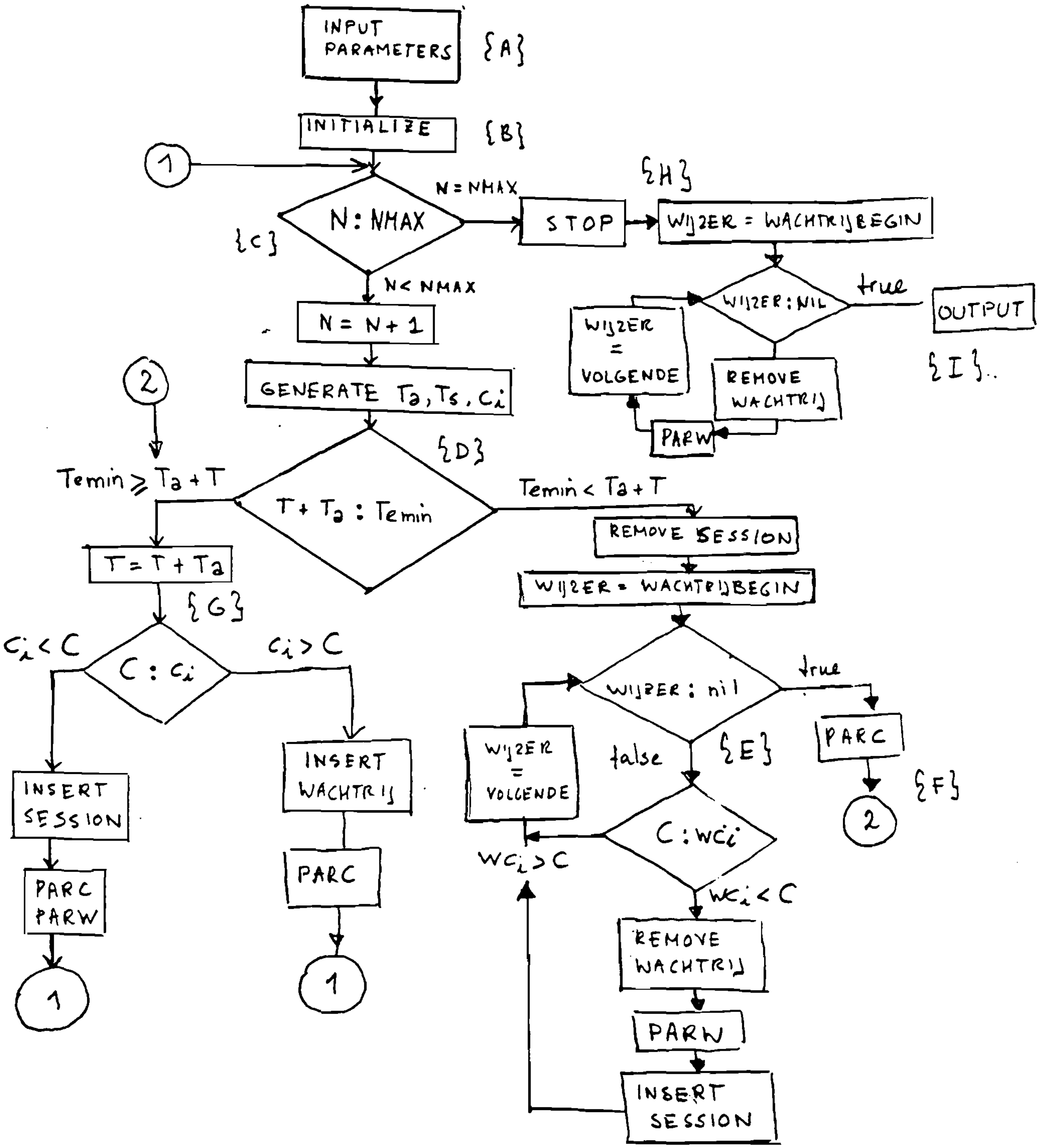
REMOVEWACHTRIJ

inputparameters: none.

outputparameters: wklasseno,wci,wts,inT

REMOVEWACHTRIJ removes the session, being pointed at by the global pointervariable 'wijzer', from the queue register. The session to be removed depends upon the queuing discipline.

<u>flow diagram</u>: Each blok in the flow diagram is assigned a letter which corresponds with parts of the program text indicated by the same letter in brackets {}.




**Program text:**

```
program SESSION(input,output);
{$M 2000,0,655360}
```

```
type vector = array[1..25] of real;

     sessiepointer = ^sessie;
     sessie = record
                capaciteit:real;
                eindtijd:real;
                volgende:sessiepointer
              end;

     wachtrijpointer = ^wachtrij;
     wachtrij = record
                  klassenummer:integer;
                  capaciteit:real;
                  inkomsttijd:real;
                  servicetijd:real;
                  volgende:wachtrijpointer
                end;

var teller,som,klasse,vraag,intens,duur,kans: vector;
    NMAX,C,CMAX,T,lambda,Ta,Ts,capac,Te,ci,wTs,wci,
    hulptijd,inT,wachttijd,somcapac,bezet,T1,T2: real;
    V,i,N,no,wno,wklasseno,klasseno: integer;
    Teminpointer: sessiepointer;
    control,nietleeg: boolean;
    wachtrijbegin,wachtrijeind,wijzer,
    hulpwijzer: wachtrijpointer;

 procedure PARC(tijd,capaciteit:real);

 begin
   T2:=tijd;
   somcapac:=somcapac+(bezet*(T2-T1));
   T1:=T2;
   bezet:=(CMAX-capaciteit)/CMAX
 end;

 procedure PARW(wklasseno:integer;wachttijd:real);

 var  j :integer;

 begin
   for j:=1 to v do
   begin
     if (wklasseno=j) then
     begin
       som[j]:=som[j]+wachttijd;
       teller[j]:=teller[j]+1
     end
   end
 end;


 procedure GENERATE(var klasseno              :integer;
                        var Ta,Ts,capac       :real);

 var k: boolean;
     Pr: array[0..25] of real;
     r: real;
     i: integer;
```

```
begin
  r:=random;
  Ta:=-(1/lambda)*ln(1-r);

  Pr[0]:=0;
  for i:=1 to v do Pr[i]:=Pr[i-1]+kans[i];
  r:=random;
  k:=true;
  i:=v+1;
  while ((k)and(i>0)) do
  begin
    i:=i-1;
    if ((Pr[i-1])<r)and(r<=(Pr[i])) then
    begin
      k:=false;
      capac:=klasse[i]
    end
  end;

  klasseno:=i;

  r:=random;
  Ts:=-duur[i]*ln(1-r)
end;


procedure INSERTSESSION(Te,ci:real);

var wijzer1,wijzer2,newsession : sessiepointer;
    notinserted : boolean;

begin
  wijzer1:=Teminpointer;
  notinserted:=true;
  if (wijzer1=nil) then
  begin
    new(newsession);
    with newsession^ do begin
                          capaciteit:=ci;
                          eindtijd:=Te;
                          volgende:=wijzer1
                        end;
    Teminpointer:=newsession
  end
  else
  begin
    while (notinserted=true) do
    begin
      if (wijzer1^.eindtijd<Te) then
      begin
        wijzer2:=wijzer1;
        wijzer1:=wijzer1^.volgende
      end
      else
      begin
        if (wijzer1=Teminpointer) then
        begin
          new(newsession);
```

```
          with newsession^ do begin
                                    capaciteit:=ci;
                                    eindtijd:=Te;
                                    volgende:=wijzer1
                                  end;
          Teminpointer:=newsession;
          notinserted:=false
        end
        else
        begin
          new(newsession);
          with newsession^ do begin
                                    capaciteit:=ci;
                                    eindtijd:=Te;
                                    volgende:=wijzer1
                                  end;
          wijzer2^.volgende:=newsession;
          notinserted:=false
        end
      end;
      if (wijzer1=nil) then
      begin
        new(newsession);
        with newsession^ do begin
                                  capaciteit:=ci;
                                  eindtijd:=Te;
                                  volgende:=wijzer1
                                end;
        wijzer2^.volgende:=newsession;
        notinserted:=false
      end
    end
  end
end;

procedure REMOVESESSION(var Te,ci:real);

var remove: sessiepointer;

begin
  remove:=Teminpointer;
  ci:=remove^.capaciteit;
  Te:=remove^.eindtijd;
  Teminpointer:=remove^.volgende;
  dispose(remove)
 end;

procedure INSERTWACHTRIJ(wklasseno:integer;
                         wci,wts,inT:real);

var newcomponent: wachtrijpointer;

begin
  new(newcomponent);
  with newcomponent^ do begin
                          klassenummer:=wklasseno;
                          capaciteit:=wci;
                          inkomsttijd:=inT;
                          servicetijd:=wts;
```

```
                              volgende:=nil
                            end;
  if (wachtrijbegin=nil) then wachtrijbegin:=newcomponent
  else wachtrijeind^.volgende:=newcomponent;
  wachtrijeind:=newcomponent
end;

procedure REMOVEWACHTRIJ(var wklasseno:integer;
                         var wci,wts,inT:real);

var remove: wachtrijpointer;

begin
  new(remove);
  remove:=wijzer;
  wklasseno:=remove^.klassenummer;
  wci:=remove^.capaciteit;
  inT:=remove^.inkomsttijd;
  wts:=remove^.servicetijd;
  if (wijzer=wachtrijbegin) then
  begin
    wachtrijbegin:=wijzer^.volgende;
    wijzer:=wachtrijbegin
  end
  else
  begin
    if (wijzer=wachtrijeind) then
    begin
      wachtrijeind:=hulpwijzer;
      wijzer:=nil;
      hulpwijzer^.volgende:=nil
    end
    else
    begin
      wijzer:=wijzer^.volgende;
      hulpwijzer^.volgende:=wijzer
    end
  end;
  dispose(remove)
end;


begin

{A}    {INVOEREN VAN DE PARAMETERS}

 writeln('Hoeveel verkeersklassen wenst u in te voeren
         (<25) ?');
 readln(v);

 writeln('Hoe groot is de kanaalcapaciteit ?');
 readln(CMAX);

 writeln('Geef van de verkeersklassen de volgende
         parameters:');
 writeln('a: de capaciteitsaanvraag,');
 writeln('b: de intensiteit (lambda) van deze klasse
         (per seconde),');
 writeln('c: en de gemiddelde duur (1/mu) van een sessie
```

```
                (seconden).');
        writeln('Per klasse dienen de parameters a t/m c op een
                regel');
        writeln('gescheiden te worden door een spatie');
        for i:=1 to V do
        begin
          writeln('klasseno.:',i,' ?');
          readln(klasse[i],intens[i],duur[i]);
        end;

        writeln('hoeveel sessies wilt u uitvoeren ?');
        readln(NMAX);

{B}     {INITIALIZEREN VAN DE PARAMETERS}

        randomize;
        N:=0;
        T:=0;
        lambda:=0;
        for i:=1 to v do lambda:=lambda+intens[i];
        for i:=1 to v do
        begin
          kans[i]:=intens[i]/lambda;
          teller[i]:=0;
          som[i]:=0
        end;
        bezet:=0;somcapac:=0;T1:=0;
        C:=CMAX;
        Teminpointer:=nil;
        wachtrijbegin:=nil;
        wachtrijeind:=nil;

        {HOOFDPROGRAMMA}

{C}     while (N<NMAX) do
        begin
          N:=N+1;
          control:=true;
          GENERATE(klasseno,Ta,Ts,capac);
          if (Teminpointer=nil) then control:=false;
          while (control=true) do
          begin
{D}         if (Teminpointer^.eindtijd)<(Ta+T) then
            begin
              hulptijd:=Teminpointer^.eindtijd;
              REMOVESESSION(Te,ci);
              C:=C+ci;

              wijzer:=wachtrijbegin;
              if (wijzer=nil) then nietleeg:=false
              else nietleeg:=true;
{E}           while (nietleeg=true) do
              begin
                if (wijzer^.capaciteit<=C) then
                begin
                  REMOVEWACHTRIJ(wklasseno,wci,wts,inT);
                  wachttijd:=hulptijd-inT;
                  PARW(wklasseno,wachttijd);
                  Te:=hulptijd+wTs;
```

```
              INSERTSESSION(Te,wci);
              C:=C-wci
            end
            else
            begin
              hulpwijzer:=wijzer;
              wijzer:=wijzer^.volgende
            end;
            if (wijzer=nil) then nietleeg:=false
          end;
{F}       PARC(hulptijd,C);
        end
        else
        control:=false;
        if (teminpointer=nil) then control:=false
      end;
{G}   T:=T+Ta;
      if (capac<=C) then
      begin
        Te:=T+Ts;
        INSERTSESSION(Te,capac);
        C:=C-capac;
        PARW(klasseno,0.0)
      end
      else
        INSERTWACHTRIJ(klasseno,capac,Ts,T);
        PARC(T,C)
    end;

{H} wijzer:=wachtrijbegin;
    while (wijzer<>nil) do
    begin
      REMOVEWACHTRIJ(wklasseno,wci,wts,inT);
      wachttijd:=T-inT;
      PARW(wklasseno,wachttijd)
    end;

{I} somcapac:=somcapac/T;
    writeln(output,'gemiddelde bezettingsgraad = ',somcapac);

    for i:=1 to v do som[i]:=som[i]/teller[i];
    writeln(output,'gemiddelde wachttijden:');
    writeln(output,'klasse','':12,'gemiddelde wachttijd');
    for i:=1 to v do writeln(output,i,'':16,som[i])
  end.
```