FLEXIBILITY AND OPTIMISATION IN BROADBAND ISDNs

Carsten Rasmussen and Kenn Kvols
Copenhagen Telephone Company, UB
Nørregade 21, DK-1199 Copenhagen K

The role of ATM in INTEGRATED COMMUNICATION SYSTEMS is discussed. Some QOS MEASURES are compared. The choice of TRAFFIC PARAMETERS and QUEUEING MODELS is discussed. The possibility of BLOCKING FREE ATM networks based on BUSY HOUR is considered.

1. INTRODUCTION

1.1 Multi-bit-rate switching

At a first glance multi-bit-rate switching looks like an easy task. If $2^n$ times 64 Kbit/s is needed one just gets $n$ narrowband ISDN connections, of course they could be multiplexed into the same access fiber: e.g. for 30 times 64 Kbit/s one gets a primary access.

Unfortunately two 64 Kbit/s connections passing just one switch are likely to suffer different delays. This means that it is necessary to compensate for the delay difference at the receiving end.

This technique is reasonable for smaller values of $n$. Copenhagen Telephone Company (KTAS) has developed an equipment doing this for any bandwidth between 128 Kbit/s and 2 Mbit/s. The equipment can also be used in connection with a digitally connected non ISDN PABX. It is being used by KTAS in the RACE application pilot RESAM.

The advantage of such an approach is obvious. On the other hand, for large bandwidths a lot of signalling is needed.

To avoid too much signalling one could switch a whole frame at a time. But to do so, it is necessary in every switch to arrange all 64 Kbit/s connections arriving in the same ingoing frame into the same outgoing frame.

Especially when mixing several bit rates in the same switched network, this needs a lot of administration and is not very efficient, because, an outgoing frame can not be used as a whole as long as a single 64 Kbit/s connection is still present. To avoid this effect connections using less than the highest possible bandwidth should be constantly rearranged during the call, causing loss of information.

It should be noted, that this is the case not only in PCM systems but also in SDH systems with 64 Kbit/s replaced by some higher bitrate.

1.2 Asynchronous transfer mode as "flexibility protocols"

In order to multiplex and switch several bit rates in a flexible and "future-proof" way without these complications it is natural to use one of the properties of packet switching, namely:

- the possibility for a switch to delay the incoming information until an outgoing line is free

To simplify as much as possible, other features such as error correction and "store and forward" are neglected. They are either not necessary or can be performed by higher layer protocols.

This makes it possible in a flexible way to switch channels with different bit rates in the same switch. But channels can no longer be separated by their different position in time. They must now be separated on a given link by different labels in the header of the packets belonging to different channels.

Such channels are known from data networks as virtual channels.

1.3 ATM

According to CCITT, ATM is the transfer mode for the B-ISDN.

In ATM the information is packed into fixed length data packets called cells, with a header of 5 octets and an information field of 48 octets.

The main information in the header is a label to separate virtual channels on a given link. For practical reasons this information has been separated into 2 parts allowing grouping of a bunch of virtual channels (VC) into a so called virtual path (VP).
An ATM switch is in principle a black box with a number of ingoing links and a number of outgoing buffers each served by an outgoing link at a constant rate. If the buffer is full the cell is discarded. The labels must if necessary be changed in each switch, to avoid 2 channels getting the same label.

An advantage of creating ATM as a network primarily fitted for connection-oriented services is, that it is relatively simple (see [3]) to determine whether a new connection will cause the delay and or cell loss to exceed a tolerated level. This could be done by defining a connection by the maximum allowed average of cells over a short interval (virtual peak bit rate) and the maximally allowed average over a longer interval (virtual mean bit rate).

If the network "polices" these parameters for each connection at the "entrance" and for example discards a cell as soon as one of the parameters are violated, it is easy to calculate a relatively tight upper bound on the delay and cell loss in a given buffer based on the peak bit rates and mean bit rates of all connection passing the buffer.

The condition is that all connections passing the buffer are independent. This condition is seriously violated if many terminals with a peak bit rate much higher than their mean bit rate are likely to transmit at peak bit rate simultaneously and their output passes the same buffer. A solution is to consider such connections as having a mean bit rate equal to the peak bit rate.

If a major part of the connections come from terminals with constant bit rate and a peak bit rate not too high compared to the total bit rate of the link, ATM can save a considerable part (in the order of peak bit rate divided by mean bit rate) of transmission resources as well as switching resources by statistical multiplexing.

If the peak bit rates are high compared to the total bit rate of a link, the gain could be increased significantly by letting the network split each cell stream into more than one link. With a significant amount of variable bit rate traffic in the network, a realistic increase of statistical multiplexing gain would be a factor 2 on as well switching- as transmission resources. For the moment this service is not defined.

2. QOS MEASURES

The following measures have been used in various models:

- **congestion probability** (the fraction of time, where a finite queue of length b is unaccessible)
- **overflow probability** (the fraction of time, where an infinite queue is at state b or higher)
- **cell loss ratio** (the proportion of lost cells)

To illustrate the relation between these parameters we will prove two inequalities the second being defined as a "virtual" inequality.

Inequality 1:

\[
\text{overflow probability} \geq \text{congestion probability} + \text{load} \ast \text{cell loss ratio}
\]

Proof: Consider at some point of time t an infinite buffer, and a finite buffer of length b, both feeded by identical traffic streams. As only the finite buffer may have suffered cell losses, it contains no more cells than the infinite buffer. Any cell arriving at the infinite buffer at state b-1 or higher will cause 1 extra service time of overflow, whereas the same cell arriving at the finite buffer will either be lost, cause 1 extra service time of congestion or arrive at a state lower than b-1.

The definition and the conjecture stated below are both needed for the formulation of inequality 2.

**Definition:**

In measure theory one uses the notation \( p(x) \) a.e. (almost everywhere), meaning that the set of \( x \) where \( p(x) \) is true has measure 1. A similar notation relevant to ATM can be defined as:

\[ p(x) \text{ is virtually valid, meaning that the set of } x \text{ where } p(x) \text{ is true has measure almost } 1. \]

**Conjecture:**

Consider a n\(D/D/1.b\) queue and a M\(D/1.b\) queue both with the same load greater than 1.

Then the former will have at most the same cell loss ratio and at least the same congestion probability as the latter.

For on/off sources with long on and off durations the following inequality is virtually valid if burst scale fluctuations dominate.
Inequality 2:
congestion probability ≥ cell loss ratio

Proof: Due to the long-burst assumption the queue can be interpreted as an iD/D/1.b queue when the momentaneous load is i/D, see [3].

The M/D/1.b queue has the property that the congestion probability is equal to the cell loss ratio. This property combined with the conjecture shows that inequality 2 is valid when the load is larger than 1.

When burst scale fluctuations dominate, it is very unlikely that cell loss and/or congestion occur if the momentaneous load is less than 1. According to the above definition we therefore say that inequality 2 is virtually valid.

Comments on the two inequalities:

Inequality 1 shows that as the number of lost cells during a service time (load * cell loss ratio) can be arbitrarily large, causing only 1 service time of congestion, so can the overflow be arbitrarily large compared to congestion probability.

Inequality 2 shows, that for on/off traffic with long bursts, this is no longer the case.

3. ATM TRAFFIC
3.1 Characterization

In many traditional traffic systems the state distribution or the interarrival time distribution characterizes the system. In ATM this is not the case as the following examples shows.

Example 1:

Example of two traffic streams with different effects on a queue but with the same state distribution and the same interarrival time distribution:

<table>
<thead>
<tr>
<th>Unknown start</th>
<th>Unknown start</th>
</tr>
</thead>
<tbody>
<tr>
<td>State distribution</td>
<td>Interarrival time distribution</td>
</tr>
<tr>
<td>Support Probability</td>
<td>1</td>
</tr>
<tr>
<td>Stream 1</td>
<td>1/2</td>
</tr>
<tr>
<td>Stream 2</td>
<td>1/2</td>
</tr>
</tbody>
</table>

It is easy to modify the example to regular arrival processes. This illustrates that no set of the usual traffic parameters like mean, variance, form factor etc. are sufficient to precisely characterize ATM traffic.

Example 2:

It is well known that for poissonian streams the system behaviour is determined by the state distribution or the interarrival time distribution.

It is often claimed that mixing sufficiently many independent sources, one gets a poissonian stream.

To see that this is in fact not the case (even for identical sources) consider $X_i = \text{the number of arrivals during a window of some given length } w \text{ from source } i$.

Consider a periodic on/off source with a peak cell rate $R_p$, on period of $2/R_p$ and on+off period of $100/R_p$.

Then always 2 cells will arrive in a window of length $100/R_p$. $X_i$ has mean 2 and variance 0.

We then consider a mix of $n$ similar independent sources each with on period of $2/R_p$ and with on+off period of $100n/R_p$. The sum still has mean 2 but variance greater than 2 as $n$ tends to infinity.

Such a mix is not poissonian, as mean and variance over a window of length $100/R_p$ are different.

But it is locally poissonian with locality interval length $1/R_p$.

Locally poissonian is a useful property. For instance a mix of sufficiently many independent sources of any kind each with a peak rate equal to say 1% of the link rate gives a locally poissonian stream with a locality interval length 100 time units, in which the number of arrivals follows a poissonian distribution and the number of arrivals in sub sets of a locality interval are independent, see [1] and [2].

3.2 Realistic parameters and queuing models in ATM

As the characteristics of future ATM traffic is unknown, in principle the total joint distribution should be needed in order to characterize the system behaviour.

As this is of course impossible one should find parameters that are either predictable or can be easily controlled and which lead to fast models.

Unfortunately this is not an easy task.
It seems that a reasonably good compromise is to use only peak and mean rate (in spite of example 1 above) but combine this with the remarks in example 2 above of a mix of many independent sources being locally poissonian. By this approach some convolution of binomial tails becomes an upper bound on the congestion probability for any kind of source mixes, see [2].

This model gives typical system utilizations from 40 to 80% with cell congestion probability guaranteed less than 10^{-9}.

Note that this approach is not equivalent to leaving out the buffer but to modeling the queue by a window of the same length as the buffer, i.e. modeling the queue by a loss system with the number of servers equal to the length of the buffer and with lost calls held (Molina).

Other methods give as a tight upper bound for the time congestion for on/off sources with long burst durations, a sum of a M/D/1 congestion probability describing cell scale phenomena and a binomial tail describing burst scale phenomena, see [3] and [4], and for a comparison [5]

It is conjectured that the M/D/1 also applies to inhomogenous sources, see [3].

The binomial tail either describes the probability, that more than a certain number of sources deliver a cell during a given interval of time or the probability that more than a certain number of independent on/off sources are active at a given point of time.

We conjecture that for all practical purposes (i.e. weights of the order of magnitude 10^{-9}) a poissonian tail is always heavier than the same binomial tail for distributions having the same mean.

This implies that, using the poissonian tail instead of the binomial tail, forgetting about the number of sources and only considering the total load in a given peak rate class, one still gets an upper bound. For sources from different peak rate classes a convolution should be carried out.

This upper bound is tight, if the peak rates are small compared to the link rate.

4. INDEPENDENCY AND BUSY HOUR IN ATM

Other queueing models consider other parameters such as burst durations. But all papers on ATM queueing known to the authors build on the assumption of stationarity per source and independence between sources.

These assumptions are closely related to the notion of "busy hour". To illustrate this we give two examples:

1. Consider an ergodic not stationary process with constant mean but periodic variance. To estimate a tight upper bound for the mean of the random variable "number of arrivals in some given window length" for a single source it would be sufficient to take the average over a long period.

2. Consider for the above example 2 sources and some window length. If an extreme number of cells from source 1 is present in some window, we have probably observed the processes during a period with high variance, and then the probability that a cell from source 2 is present in the same window, has increased. This means that the sources are not independent.

This leads to consider a maximum period of stationarity with maximal load, the well known busy hour. Not busy milisecond, because as the queueing models above describe a queueing system by a loss system and a M/D/1 queue, only the cell traffic and not the cell duration determines the cell loss.

Unfortunately a strong correlation between sources even with a lag of very few cell service times (the so called football game effect) may occur in ATM because of interference between machines or synchronous broadcast effects. Such events will spoil any queueing model based on any other parameter than peak rate. How often such effects will occur in real life is a very interesting problem.

5. FORECASTING OR CONTROL (POLICING)?

It is commonly accepted, that the peak rate should be controlled. Lack of control would allow a single user to spoil the entire network.

5.1 Some arguments in favour of replacing mean rate control by mean rate forecasting: a blocking free ATM network

As the load in busy hour is an average, it is much more robust to predict than peak, which is a maximum value. During almost a century it has proven possible to base network dimensioning on forecasted average traffic over the worst busy hour.

Because of a limited number of resources (trunks or timeslots) it was necessary to include a call blocking of say 1%.

By some reason this idea has survived so strongly in ATM that no one has challenged call acceptance, which means that the necessity of call blocking has become commonly accepted.

After skipping 1% of the cell traffic one then dimension the network to 99% of the offered cell traffic with a cell congestion probability of say 10^{-9}, instead of dimensioning to 1% more cell traffic getting a blocking free network well fit also for on line data services.
Note that with dimensioning based on peak and mean as described in this paper, it does not matter that the 1% lost calls belong to overflow traffic.

Even the effect of independent malicious users can like today easily be predicted and accounted for in the busy hour average. Normal users are accounted e.g. by traditional Carlson charging. Therefore they have no interest in using more mean bit rate than they need.

How to handle some special situations:

A huge amount of users calling one subscriber may cause problems today, but with end-to-end signalling this will no longer be the case.

An emergency situation may generate much more traffic than forecasted. In such a situation preferent connections could be used, in first step only telephony could be made preferent, in next step only selected telephones etc.

Naturally it will still be possible for organised malicious users (so called terrorists) to send synchronised traffic from several sources. But probably it is easier to blow up a central office.

As a conclusion: a busy hour based blocking-free ATM network seems possible. Special situations seem to be better handled in traditional ways than by several billions of mean rate policing chips. Because of difficulties in prediction, sources with high peak rate compared to the link rate should be kept in a logically separated network, unless they are spread onto a number of links as proposed in the introduction. If a new service develops faster than predicted it must be controlled by charging.

5.2 Some arguments in favour of mean rate control (policing)

It is too dangerous without.

In traditional telephony systems a too low forecast only results in a too high call blocking, whereas a too low prediction in ATM systems may result in a temporary degradation of the overall QoS.

It is possible and simple.

A robust mean rate policing, based on the leaky bucket principle, which never exceeds the mean:

It has been pointed out that mean rate policing using a large bucket size will allow a lot of extra cells to enter the network during its initial transient state, with the consequence that the mean can be exceeded.

One way of avoiding this would be to start with a full bucket. But this method is inconvenient to the costumer. A very simple way of avoiding both drawbacks is the following:

During the time $t_e$, it takes to empty a full bucket at leak rate, which for mean policing equals mean rate, register the call to 2 times mean rate. For the rest of the call duration register the call to mean rate. If the call is shorter than $t_e$, register to 2 times mean until the call stops, and register to mean for the rest of the period $t_e$.

The averaging period (bucket size) should be as close to busy hour as possible, as on one hand a longer period gives a better estimate, and on the other hand busy hour is the longest averaging period, where independency between sources is not lost.

Other methods for call acceptance "between" forecasting and control have been proposed, e.g. Fast reservation scheme [6] or Ticketing [7], and Continuous short term traffic measurements proposed by Villy Bak Iversen.

6. CONCLUSION

ATM is possible and will provide flexibility in future multi-bit-rate networks. Peak rate policing is necessary. Mean rate policing is possible. If only one logical ATM network exists mean rate policing is necessary.

It might be advantageous to separate ATM into more than one logical network, where services with high peak rates are in one logically separate network with call acceptance, and other services are in another logically separate blocking free network. The extra cost of dimensioning to a blocking free ATM network will be compensated by avoiding call acceptance procedures and -delays.

The size of the first network can be reduced and a considerable amount of switching and transmission resources saved by spreading cells from services with high peak rates into several links.

7. FUTURE ROLE OF ATM

For technical and market reasons it seems likely that the broadband network will be an inhomogenous network with different protocols interworking in different parts and/or on top of each other. At least for a period future broadband networks are likely to involve some kind of asynchronous transfer mode, ATM being one candidate.

In such a network one can foresee a need for traffic "arbitrage" choosing the best subnetwork for the demanded service, and for end to end models and traffic management centers getting information safely through complicated inhomogenous networks at the negotiated quality.
Traditional control methods based on busy hour are still relevant for controlling fast variations relative to the propagation delay, whereas online management is appropriate to control slower variations relative to the propagation delay.

REFERENCES


