DISCUSSION RECORD

Session No. 63 – TDM-SWITCHING SYSTEMS

PAPER No. 631
Authors: H INOSE, T SAITO and Y YANAGISAWA

Question by A MYSKJA
When calculating blocking, you have assumed the binomial distribution in all stages. Is this for simplicity, or because it is considered the best approximation? Is your optimization sensitive to the distribution chosen?

Answer
The only reason is for the simplicity in performing the optimization procedures on a large variety of network configurations under consideration. For better approximation, we should take into account the dependency of calls between stages or at least the distribution of calls in each of the stages. In our previous papers, we checked the distribution of calls by means of computer simulation on some restricted types of network configurations with the result that the binomial distribution was a rather good approximation. The sensitivity of the optimization to the distribution of calls however, is subject to future studies.

Question by I MOLNAR
In your Fig. 2 a you show an arrangement requiring (n-1) memory elements. By introducing the subframe in Fig. 2b this is reduced to (2m-1). Could you, by analogy with 2 a reduce this further to (m-1)?

Answer
In principle we can. The increase in the amount of control memory as well as the increase in control complexity result from that, however, hardly justify the realization of such principle.

PAPER No. 632
Authors: W T DUERDOTH and C A SEYMOUR

Question by I MOLNAR
The title of this Contribution has greatly puzzled me, which did not lessen after reading its corpus. To me the import of "quasi-non-blocking" recalls the consistency recurring furly raised when in numismatic trade publications coins are being advertised as 'nearly uncirculated', "about uncirculated", "almost uncirculated" or "quasi uncirculated".

According to IEEE nomenclature "Non-blocking switching network: a switching network in which a given output can always be reached from any given input under all traffic conditions". This is unequivocal, not derived from the behavior of networks in the limit. Conversely, internal blocking is defined as "the unavailability of paths in a switching network between a given input and any suitable free output". The degree of blocking in other than non-blocking networks is dependent on its configuration and traffic conditions. They represent a perfect dichotomy with no gray area in between.

Charlie Clos, the inventor of non-blocking networks, recognized (and so stated in 1952) that such networks compare unfavourably costwise with conventional networks. However, by omitting certain paths, blocking can deliberately introduced and still may obtain economies over certain approaches.

Neither is the classification contingent on the numerical value of blocking. "That which we call a rose by any other name would smell as sweet or sour". Because any network yields a very low amount of blocking under design-specified traffic conditions, is not "almost" or "quasi" non-blocking: its overall behavior has no similarity with non-blocking networks.

Not so many years ago (and occasionally even today) conventional switching systems were routinely specified and operated by numerous administrations at 0.001 blocking (for some components even at 0.0001) and unabashedly called so. Recently one may hear about "nearly non-blocking" networks aiming at a blocking probability 0.001.

Nor does the practice of underprovisioning the number of input sources to achieve a desired degree of blocking makes the system resemble non-blocking; this has always been done with any of the conventional switching networks.

If I interpret the description and specifically figure 2 correctly, this network seems to be particularly blocking sensitive. A small increase of 10% in mean external channel occupancy (whether due to growth, or administrative limitations, temporary overload, or other reason) increases the blocking probability about ten thousand-fold. I cannot recall to my knowledge any practical switching system with such sensitivity to blocking.

It would also be helpful if the authors would show the formula for the blocking probability $E$ on page 2 were derived.

Answer
The very rapid change in the $G$ of $S$ which occurs with change of loading is a characteristic of switching systems with a high availability. The switch we described offers 512 internal channels to the incoming channels, a situation which is only likely to occur with TDM systems. This rapid change of $G$ of $S$ is no disadvantage, the switch must be designed to give the required $G$ of $S$ on overload and a very much better performance is obtained with normal loading.

We must accept Mr. Molnar's comments regarding the use of the term quasi-non-blocking. However, there is a feature of the switch which we did not adequately describe in the paper. Where the number of input channels $N$ is less than the internal channels $m$ a useful $G$ of $S$ can be maintained even when the input channels are saturated, i.e., when loading is 1.0. For $N = 448$ the blocking probability is 1 in $10^{-5}$ and for 435 channels it is 1 in $10^{-3}$. Thus the switch is of the form that it can maintain any desired $G$ of $S$ even when the input channel loading is 1.0. This feature is practical only with high availabilities. The connection of 13 PCM systems (416 channels) to a 512 channel internal highway results in a blocking probability of better than 1 in $10^{-4}$ which is maintained under any loading conditions.

We need a term, better than quasi-non-blocking, which describes such a switch.

Question by A MYSKJA
As I understand your Fig. 2 the parameter $m$ is the number of internal time slots available, while $N\cdot \alpha$ is the offered traffic, where $N=512$ and $\alpha$ is the abscissa value. To me it seems that for a given offered traffic the blocking increases with available time slots ($m$). Could you please explain this?
Answer
This question, no doubt, arises from errors in the printed paper, Figures 2 shows curves for various incoming circuit traffic loadings and several values of N for the case where m = 512. In the figure m and N have been interchanged, m = 512 and N = 480, ..., 1000.
The expression for \( E_m(A) \) should read
\[
\frac{A^m}{m!} \sum_{i=0}^{m} \frac{A^i}{i!}
\]

Apologies.

Question by W MILOERT
Have you ever considered replacing the one square array space switch (max dimensions 64 x 64) by a 2-stage switching array \( b \times b \times b \times b \times b \times b \), thereby reducing the number of crosspoints? If so, have you estimated the influence of such an arrangement on the blocking probability?

Answer
Consideration has been given to both 2 stage and 3 stage switching arrays for use with the larger switches. Some economy in crosspoints is obtained with two stage arrays when compared with single or three stage arrays, for a given blocking probability, provided that the channel loading is not high. The proportion of cost which is attributable to the crosspoints is not high but work in this area is in hand. To obtain reliability the switch block is divided into four zones and a possible organization is to provide single stage intrazonal switching with a limited amount of two stage interzone switching. This necessitates distributing the circuits of each junction or trunk route over all zones but this can have considerable influence on the size of the space switch which must be provided on the initial installation, a factor which can have significant influence on the initial cost.

PAPER No. 633
Author: R SCHEHRER

Question by F SCHEIBER
If the block stream has passed already several nodes, the assumption of randomly distributed blocks within a frame does no longer hold. Instead some kind of clustering of blocks can be assumed. This situation seems to be important for practical applications of a TDM-node-switching-network. In remark 2 of the paper it is stated that dropping the assumption of randomly spaced blocks yields only a slightly increased loss probability. Can the author please indicate the basis for this conclusion?

Answer
The problem of regarding the clustering effect in the calculation of the loss probability was not the aim of this paper and may be a topic for future investigations. From preliminary considerations it follows that the clustering effect will result in a slight increase of the loss probability, provided that the system is operated at a reasonable load. It can be shown that the main influence of the clustering effect turns up in central parts of the network. Therefore in these central nodes there must be more waiting places than in peripheral nodes.

Thus, the statement of a slightly increased loss probability is mainly based on the following items:
1. Network operation at a reasonable load
2. Provision of larger stores in central nodes
3. The absence of a correlation between the clusters on different incoming trunks.

Question by A MYSKUJA
Your Concept looks very interesting and non-conventional.

1. Do you have any estimates as to average time slot efficiency of your system?
2. What will be the effect of cumulated delays of samples in the case of speech transmission?

Answer
Thank you for your kind comment, Mr. Myskja.
To your first question:
In the considered system, the admissible time slot efficiency depends mainly on the prescribed loss probability, the number of waiting places, the structure of the network and the geographical situation of the subscribers. A typical value for the time slot efficiency would e.g. be 0.6, but it can also rise to higher values.

As to your second question:
In the case of speech transmission, the cumulated delay of samples has the effect of a phase jitter of the originally equidistant samples, e.g. for the usual sampling frequency of 8 kHz and 10,000 time slot per frame, a delay of 80 time slots corresponds to 1/4s which is 1% of the sampling period. This is practically of no influence to the quality of speech transmission. Actually, the phase jitter is much smaller, because it is caused by delay changes in succeeding frames and not by the absolute delay values. If, however, for special purposes strictly equidistant samples are desired, then this can be easily achieved with the aid of a buffer store which is in any case provided in the decoder of the subscriber stations.

PAPER No. 634
Authors: G LAJTHA and A MAZGON

Question by E JENSEN
It is well known that in the case of Engset sources, as actually treated in your paper, one might distinguish between call congestion and time congestion.

1. Have you considered whether the discrepancy between the two concepts of blocking is of any importance in the practical cases?
2. And if this is the case, how should the blocking B used in your paper be specified?

Answer
As we have given in the written paper, the used formula is from a paper find in the NTZ. We are sorry that we have given a wrong reference under number 8. The improved version is: (6) D BAZLEN, "The Dimensioning of Trunk Groups for Standard Gradings of the German GPO in Case of Finite Number of Traffic Sources" - NTZ 25, 1972, No. 1.

In the original paper - as we know - Mr. D Bazlen is using Call Congestion. So we have NOT considered the practical importance of the difference of the two blocking concepts, only used the original Bazlen formula.

Remark by D BAZLEN
You mention in your paper the MPG-formula. This formula is valid for PCT 1 (infinite number of sources), and can be adapted for the various grading types as it was shown yesterday in the lecture of A Lotze (paper No. 521). For a handy use this MPG-formula has been tabulated. For the case of PCT 2 (finite number of sources) a Bächle and U Herzog have developed the EQ-Formula (Bermouilli-Quotient-Formula), which bases (instead on an Erlang-distribution) on an Erlang-Bernouilli-distribution. This EQ-Formula has been published already at the 5th ITC in "History and Development of Gradings" by A Lotze. This formula is up to now not tabulated. To use the existing MPG-Tables being adapted for various grading types also for PCT2, we have developed, as also already mentioned in the lecture of A Lotze yesterday, the "Finite Source Transformation-Method (FST-method)". This FST-Method allows to calculate the admissible offered traffic for given probability of loss in case of finite number of sources starting from given tables for the adapted MPG-formula. This transformation is done by the empirical formula which is...
given as equation (1) in your paper, I am very glad that you are using this formula, but unfortunately citing this method you have taken a wrong reference. This FST-method was invented by myself and therefore your reference No. 6 has to be replaced by the following reference:

D BAZLEN: The Dimensioning of Trunk Groups for Standard Gradings of the German GPO In Case of Finite Number of Traffic Sources

NTZ 25, 1972, No. 1.

(Copies of this publication are available from the author.)

In addition to this I have checked in the last months that this FST-Method is valid for all sequentially hunted grading types, (e.g. "Perfect", Standard, O'Dell and AT & T gradings resp.), when starting the calculation from that adapted MFU-value which stands for the considered grading type.

Answer

Thanks for the remarks, and the failure, we made, at the references, will be corrected at the lecture. We are very sorry for it.

PAPER No. 635
Authors: O ENOMOTO and H MIYAMOTO

Question by A MYSKUJA

An important variable in your diagrams is the proportion of wide band calls (or middle band calls). Can you explain how this relates to traffic? Is the call proportion a sufficient specification when you have different bandwidths?

Answer

In our diagrams, we have many parameters. These are not only the proportion of wide band calls (K) but also total traffic offered (A), the number of time slots required for a wide band call (m), the number of time slots in a frame (M), etc. Let Az be amount of wide band traffic offered. Az relates to total traffic offered A through the proportion of wide band calls K, that is Az = KA/m.

Mr. Gimpelson suggested this expression of the diagram in his paper titled "Analysis of Mixtures of Wide- and Narrow-Band Traffic" published September 1965.

PAPER No. 636
Authors: L MANERA and R PELLEGRINI

Question by T NORDAHL

In your paper you have simulated a TDM/PAM system and from the computer output derived a mathematical model relating system parameters. I find it interesting to note that you have adopted time-true simulation as a fundamental part of your work.

My questions are:

1. What is your opinion about using time-true simulation compared to other methods?
2. In what respect, do you check your simulated results to gain confidence in them?
3. In your paper you state that you use Pulse-Amplitude Modulation. This mode of transmission is in my opinion inferior to other TDM modulation methods as regards crosstalk and signal to noise ratio. Could you say something about why pulse amplitude modulation has been adopted?

Answer

1. Time-true simulation was used in our case because we intended to investigate the waiting behaviour of subscribers and this is very difficult to take into account with ordinary simulation methods (e.g. the so-called Koster method or roulette method). In addition, in the system to be simulated we had a mixture of subscriber's holding times (which may be taken as negative exponentially distributed) and of constant delay times due to the operating mode of the exchange. Furthermore we had concurrency between calls for the occupation of time slots. The simulation language used (namely SIMULA) was particularly well suited to deal with these problems. I think that the classic method of simulation is superior when the pure poissonian traffic assumption is realistic and we have not concurrency between events. In this case we may achieve a great saving in computer memory and computer time for the simulation program to be run. In other cases the scheduling of events and the discretization of the time axis require auxiliary programs and a theoretical analysis which may lead to programs very efficient with respect to CPU time and memory occupancy but not optimized with respect to model implementation time, debugging ease and self-documentation.
2. The simulation has in our opinion, two objectives:
   1) With respect to system analysis it has the same function of the experiment in natural sciences.
   2) With respect to system synthesis it is a powerful designing tool. In both cases it has a great usefulness in the conception and validation of analytical models. Without simulation it is impossible to collect the fundamental facts which must be taken into account in an analytical model, and it is impossible to demonstrate that the model correctly takes account of these facts. Without a starting hypothesis it is impossible to know what to look for with simulation and it is impossible to extend the studies at a reasonable cost. We think that the central problem will be solved when it will be possible to use a method of structural analysis which permits a mutual validation of simulation results and analytical results.
3. I am not an expert of transmission techniques and I am not able to answer to this question.

Question by A MYSKUJA

To formula (2): Your K (capital K) is not defined. Is it the same as k (lower case k)? Is it correct that you get negative terms in your binomial coefficients?

To formula (9): Is the traffic per subscriber, A_g, in the parenthesis compatible with the relative numbers 1 and R?

Answer

1) Thank you for your question. There are errors in the formula you mentioned (formula 2). The appearance of capital K is due to a typing error; it should in fact be lower case k. There is also an error in the specification of the summation limits. The correct formula reads:

\[ S = \sum_{j=1}^{\infty} \sum_{k=1}^{\infty} \frac{j}{k} \cdot \binom{K}{j+K} \cdot \binom{R}{k} \cdot \binom{R}{j+k} \cdot \binom{R}{k-j} \cdot \sum_{p=1}^{\infty} \frac{p^j}{j!} \cdot \frac{p^k}{k!} \cdot \frac{p^R}{R!} \]

2) In the formula 9 the variable A_g is the traffic per subscriber and its value is always compatible with the other numbers in the expression. This formula gives a first rough approximation of the value of the traffic carried by the exchange.

Question by A MYSKUJA

The group subdivisions of N_g into N_sg and of N_s into N_sg seem to be important in your formula. However, in your discussion they are very little touched. According to Fig. 6 it seems that the smaller N_sg the better. Are these subdivisions rather unimportant, and how do they influence the dimensioning?

Answer

In fact the values of N_sg (number of junctions per group) and of N_sg (number of subscribers per group) are important not only in the formulas but also for the system design.

In Fig. 6 it is clearly seen that for little values of N_sg (e.g. N_sg=32) the loss for incoming calls is less than the loss for the outgoing calls. The loss of outgoing calls is nearly constant throughout the range of values of N_sg and is equal to the standard design loss value. This fact permits to take the value of N_sg in the range 32/250 paying attention to technological considerations.