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TABLE OF CONTENTS

A G E N D A

Chairman: Mr. Louis Pouzin

Monday, 21 October

Public Data Networks, the Next Decade	Kirstein
The Future of Computer Networks	Davies
Austrian Computer Networks in the Scientific Academic Fields	Firneis
Data Communications in Sweden	Haug

Tuesday, 22 October

A Proposal for Interconnecting Packet Switching Networks	Pouzin
Some Conceptual and Technical Aspects of an Integrated System of Scientific and Technical Information Storage and Retrieval	Kulikowski
A Study of Line Overhead in the ARPANET	Kleinrock
Some Aspects of Networking Economics in Relation to the Dissemination of Scientific and Technical Information	Page
Some Problems of Multimachine Systems	Zhelezov

Wednesday, 23 October

Private Computer Networks	Hughes
A Practical Example of Distributed Computer Systems	Molnar
Variables and Constraints in Data Communication Systems Design	Porizek
Decentralized Control in Communication and Data Processing Systems	Aven/Stetsura
An Adaptive Routing Technique for Channel Switching Networks (Abstract)	Bell/Butrimenko

Thursday, 24 October

The Computer Conferencing of the Institute for the Future	Vallee
KUIPNET: In-house Computer Network for Information Processing and Some Applications	Sakai
Evaluation of Terminal Nets	Blatny
Overview of an Experimental Loop System for Data Transmission	Kaye
Views on the IIASA Programme in Computer Communications and a Review of Some Canadian Projects	Kaye/Cowan/Hanna

Friday, 25 October

General Discussion

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Introductory Remarks by the Computer Science Project Leader

A. Butrimenko

This publication contains the papers presented at the Conference on Computer Networks organized by the Computer Science Project of IIASA in October 1974.

This area is one of the main concerns of the Computer Science Project, but also has a broader significance for the Institute. IIASA deals with various problems of industrialization and its consequences, and inevitably depends on permanent working contacts with a number of national institutions. The interdisciplinary and international character of the Institute is essential to success in finding solutions to these problems. Applied research today depends heavily on the use of large amounts of data and data processing. We believe that connecting computers installed in various national institutions will contribute significantly to the achievements of the main goals, allowing for the exchange of data and programs, and in this way facilitating the understanding of problems, resulting in faster solutions.

This Conference was the first of a series of conferences and workshops to be held on this topic. In addition to the exchange of ideas and the discussion of problems arising in networking, it was intended also to identify people and institutions that were interested in establishing links contributing to the achievement of the goals of the Institute.

In addition to the presentation of papers and formal discussions, discussions on the periphery of the conference were probably of equal importance. The papers presented do not reflect the spirit of cooperation which was very characteristic of the conference. However, we feel that the publication of these papers will be useful to the scientific community and give a picture of recent developments in this area.

Mr. J. Sexton and Miss U. Sichra are staff members of the Computer Science Project who devoted a great deal of their time and efforts to the editing of the papers. Mrs. H. MacKinnon was especially helpful in the technical editing and polishing of a number of the papers presented.

Public Data Networks, The Next Decade

Peter T. Kirstein*

I. INTRODUCTION

This paper presents a brief overview of planned public data networks. First, (see Section II) the facilities commonly available over the present analogue telephone network are discussed. Then two special line-switched networks which are already operational are presented (Section III). The present generation of data networks is being developed slowly on an experimental scale only, because it is universally realized that synchronous networks with digital transmission are the way of the future. Because of the need to interact internationally, careful attention is being paid to common standards of such networks. So far most of the telecommunication authorites have been convinced to provide line-switched services only, and some of these are described in Section IV.

While many countries have schemes in this area, only the Scandinavians, Swiss, Americans, Canadians, British, and Japanese have given any indications about their plans. The US activities in data communications are well in advance of other countries. Their developments differ from others because the unique position of ATT versus other carriers in the US; elsewhere the PTTs usually have a monopoly of all communications.

The US activities in packet-switched services are discussed in Section V. Elsewhere the open data networks are the responsibility of the PTTs. Some of the experimental PTT activities of France, UK, Spain and others are mentioned in Section VI.

International activities are very complex, because of the different constraints imposed by the National PTTs and the International Regulatory bodies. Some future international activities are discussed in Section VII. Some of the impact of data network offerings on such networks is considered briefly in Section VIII. The PTTs usually attempt to distinguish between "data" and "message" traffic. In the context of computer networks, these distinctions become blurred. This subject is considered briefly in Section IX. Finally, some conclusions are presented in Section X.

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II. PRESENT DATA FACILITIES ON THE ANALOGUE TELEPHONE NETWORK

Many countries now offer data facilities, dialled and leased lines, on the public switched telephone network (PSTN). A brief summary of these and signalling modes recommended by the international body CCITT (Comite Consulatif International Telegraphique et Telephonique) is shown in Table 1.

TABLE 1.

Commonly Available Facilities on the Analogue Telephone Network According to CCITT Recommendations

Service	CCITT recommen dation	Rate K bps	Rate PSTN	On Leased Lines	Async/ sync Operation	Duplex or Half Duplex on 2 wire lines
1	V21	0.2	*	*	A	D
2	V23	0.6 1.2	*	*	A	H
3	V26	2.4		*	S	H
4	V26	2.4	*		S	H
5	V27	4.8		*	S	H
6	V35	4.8		*	S or A	H
7	V30	up to 40 ch/sec parallel	*		A	H

*Denotes Service Available

Recommendations for automatic answering and calling on the PSTN are given in Rec. V25, and for the Telex Network in Rec. V11.

The PTT recommendations are somewhat conservative. Any service offered is to be available over most lines in the country. In fact on most circuits of the PSTN of many countries, speeds of 4.8K bps are obtainable with certain high quality proprietary modems, and on leased lines 9.6K bps or even 19.2K bps. All transmission between exchanges on the PSTN are four-wire, and only on local lines are the PSTN circuits two-wire. Even over these, full duplex speeds at 0.3K bps is usually obtainable. Over leased lines, four-wire lines can be rented to the local exchanges, making duplex facilities at the relevant speed available. For the 48K bps special lines are required, and a full primary group of 12 telephone channels is usually used for long-distance transmission. In fact when digital transmission is used for the inter-exchange transmission, 64K bps is used for ordinary voice traffic (see Section IV).

Most of the PSTN connections of Table 1 imply electrical connections of the data terminal equipment to modems, attached eletrically to the telephone line, by bypassing the telephone handset (except in the call set-up phase). At the lower speeds, acoustic coupled modems can be used; they are connected electrically to the terminal equipment, and acoustically, through the telephone handset to the line. In most countries acoustic coupled modems can reach 300 bps. Higher speeds, up to 600 or 1200 bps in one direction, can be attained by exchanging the microphone in the telephone for one of better quality. Unfortunately this substitution is illegal in most countries.

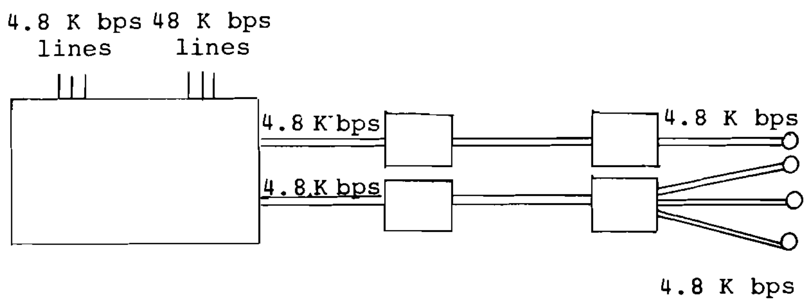
On the PSTN the call set-up time is usually 20 secs., due to mechanical switching in the exchanges. For data traffic, particularly with short transactions, a much shorter call set-up time would be advantageous.

III. PRESENT LINE-SWITCHED DATA NETWORKS

Two countries, France and West Germany, have started introducing special line-switched data networks without digital transmission.

The French introduced the CADUCEE system early in 1972 (Figure 1). It is based on an Ericsson CP 410 crossbar switch. It is a circuit switched system for the speed range 2400-72 K bps using analogue transmission. It is aimed at an interim solution to their data problems, and will be superceded by an all-digital system in the early 80's. It is being put in now mainly because of the poor quality of the normal telephone circuits. The system is based on 2000 line exchanges.

Figure 1. Schematic of French CADUCEE



There are four classes of subscribers:

A. Those within 30 km of an exchange using ordinary modems;

B. Those within 30 km of an exchange using a base-band modem;

C. Those at greater distances from an exchange with heavy traffic using ordinary modems;

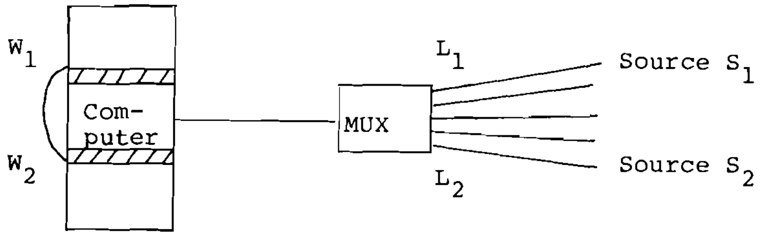
D. Those at greater distances from an exchange with less traffic, connected to it through a concentrator, again via ordinary modems.

Although any modems can be connected, the French PTT will supply modems at 2.4K bps and later 4.8K bps for Classes A, C and D, and a modem at 19.2K bps for B. Later they hope to provide higher speed modems. The same quality of four-wire lines will be used as for leased telephone service. Call set-up time is six seconds, and there is out-of-band signalling via a touch-tone telephone.

Each CADUCEE exchange can handle about 2K duplex circuits; 1600 are ordinary lines, 240 lines with remote concentrators for calling in, 240 with remote concentrators for calling out. The concentrators map forty channels onto twenty lines, and are envisaged for circuits with 0.2 erlang traffic or less. Initially there is one exchange, but as needs grow additional ones will be installed initially in remote towns. These will increase the area over which high speed traffic is possible.

A far more ambitious system, the EDS system is being developed by Siemens for the German Post Office. A fairly complete description in English is now available [1,2]. The system was developed initially for handling German Telex and Datex traffic at speeds up to 200 bps, but was designed to take also higher speed traffic. It will not be described in detail here. The important point is that it is essentially asynchronous. Schematically it is shown in Figure 2.

Figure 2. <u>Schematic of EDS Exchange</u>



A computer is attached to up to 13K lines by a special multiplexor. To each line there is allocated a work core W. Any data input from L will affect W. Making a connection from a terminal S_1 to S_2 would be accomplished by setting a pointer in W_1 to W_2 and vice versa. Once a connection has been made, the MUX will detect any change in state of the bistable line L_1. If a number of lines change state simultaneously, since the computer can service only one line at a time, there will be a delay in some of the changes being passed on; this delay will show up as a distortion in the transmitted signal. Provided this distortion is not too serious there will be almost complete transparency in data speed or format with this network. It is almost complete transparency, because signalling to make a connection will be at a specific speed with ISO Alphabet No. 5. While the connect time is designed to be 100 ms, the time to terminate a call may take 250 ms, or 300 ms if remote concentrators are used.

Typical numbers of lines active on 13K line exchange envisioned by Siemens in early studies are shown in Table 2.

TABLE 2. Speeds and Signalling Speeds of Different Classes

Class	Speed K bps	Signallling Rate K bps	Number of Active Connections	Percent of Machine Cycles
1	$S < .05$	.05	4300	13
2	$.05 < S < .2$	.2	375	5
3	$.2 < S < 2.4$	2.4	275	39
4	$2.4 < S < 9.6$	2.4	75	43

Clearly a small proportion of higher speed traffic would saturate this system. The German PTT has stated that for the time being it will only use this system asynchronously for classes 1 and 2 up to 200 bps. Above that speed Siemens is now developing synchronous units. It seems clear that the growth of the EDS system [3] will be in accordance with the other networks of Section IV.

Siemens claim that when the EDS exchange is working synchronously, its throughput is much higher than indicated in Table 2. When it is equipped with a 200 ns memory and a single set of buffers it can handle the following traffic:

TABLE 3.

Capacity of EDS Exchange for Synchronous Traffic

Speed (K bps)	0.6	2.4	9.6	48
No. of full duplex connections	25K	6.24K	1.56K	312

However, the time to set up calls is comparatively long--20 ms. Thus at most 50 calls/sec could be set up. This implies that the EDS system is not ideal for short transactions. For comparison, in the packet switching ARPANET IMP [4] it would be possible to generate shorter messages to different sites by order of magnitude.

In the EDS system there will also be a remote controlled concentrator, with 10 lines to the exchange and 100 to subscribers. It will have facilities for abbreviated dialling, multi-address messages, calling station identification and closed user groups. Facilities for packet switching, hot-line and delayed delivery have been announced to the EDS exchange by Siemens, but the German PTT has not said that they will provide these facilities. A prototype exchange is being operated in Munich, and the first real exchange is scheduled for late 1974. Some ten exchanges are scheduled by 1976.

IV. SYNCHRONOUS DATA NETWORKS

Many of the wide-spread computer networks have implemented a communication subsystem, which operates over the analogue telephone network. Some of these subsystems are described briefly in this paper. For some years the international body "Committee Consultative Internationale de Telegraphe et Telephone" (CCITT) has studied the requirements for specialized data networks in their working party on New Data Networks (NRD). The CCITT is a consultative committee to the International Telecommunications Union, and its recommendations are usually followed by the National Telecommunications Administrations (PTTs).

The NRD has concluded that it is the intention of a number of PTTs to provide data services over synchronous networks [5]. Currently only facilities provided by most countries will be circuit-switched connections--a single channel between two terminals. On the present analogue public switched telephone network, full duplex facilities are usually provided only up to 200 or 300 bps, with half-duplex facilities up to 1200 or 2400 bps. The new networks will use pcm digital transmission, and plan to provide a number of user classes with full duplex capability as indicated in Table 4. These will be introduced between 1975 (US) and 1985, with many starting about 1980. Thus it is felt that some information about these networks is of some importance. Further details are given elsewhere [6, 7].

TABLE 4.

Classes of User Services Recommended*

Class	Data	Address Selection and Service Signals (Alphabet No. 5)
1	200 bps, 11 units/char start/stop	200 bps
2	50-200 bps, 7.5-12 units/char start/stop	200 bps
3	600 bps synchronous	600 bps
4	2400 bps synchronous	2400 bps
5	9600 bps synchronous	9600 bps
6	48000 bps synchronous	48000 bps

It is proposed that these new data networks have a public switching capability of making a call "reasonably fast;" however, the present recommendations can not guarantee a call set up in less than 10 sec; or shut down in less than 1 sec. It is supposed to be symmetrically duplex, bit sequence independent, with automatic calling and answering. It is recommended that direct call, abbreviated address and closed user groups be provided. Remote terminal identification, multi-address, and delayed delivery for Class 1 service may be provided.

The actual transmission between exchanges will be a multiple of 64K bps (usually 2.048 M bps in Europe which uses 32 64K bps channels). It will usually share the same long distance transmission as the telephone system, with different exchanges, and different terminating units to subscribers' premises from the present modems. In many countries the new network is intended to carry the telex traffic, and possibly also facsimile.

*Certain specific recommendations have been made on Class 2, so that the combinations of speed and unit/character match present terminals.

There are a number of consequences of the need for inter-working between countries, together with facilities for complete bit transparency [5]. Several countries (particularly US and Canada) have opted for a frame consisting of some synchronising or control bytes followed by data bytes.

SYN data data data ... SYN data data data SYN.

Others have preferred to have each byte carry information on whether it is control or data e.g.:

TABLE 5.

Frame Format for Synchronous Data

Bit	1	2-7	8
Content	Frame	Information	Status

Here for one value of status the information is data, for another control. Denmark, Finland, W. Germany, Norway, Sweden, and the UK have opted for this system; France and Italy will use both methods. The two systems will be kept capable of inter-working, because it has been agreed to use a 32 bit frame consisting of four 8 bit bytes. In the first scheme there will be one control byte followed by three data bytes/frame; in the second each frame will have four bytes of which up to 24 bits are data.

Since the control and framing information is put in by the switching exchanges, the relative transmission rates in the classes are shown below:

TABLE 6.

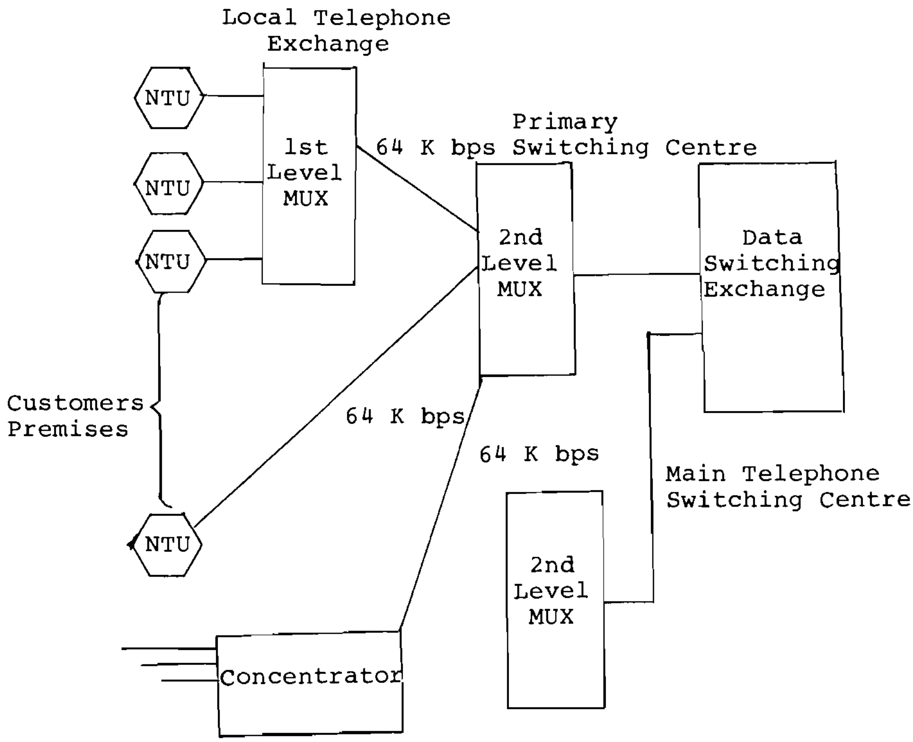
Signalling Speeds and Number of Bytes in 64K bps Channel for Data for a Signal User Class

Class	1	2	3	4	5	6
Rate bps	266	266	800	3,200	12,800	64,000
Repetition in 64K bps stream	240	240	80	20	5	1

The typical schematic of a system of this sort is indicated below:

Figure 3.

Schematic of Data Network Multiplexing

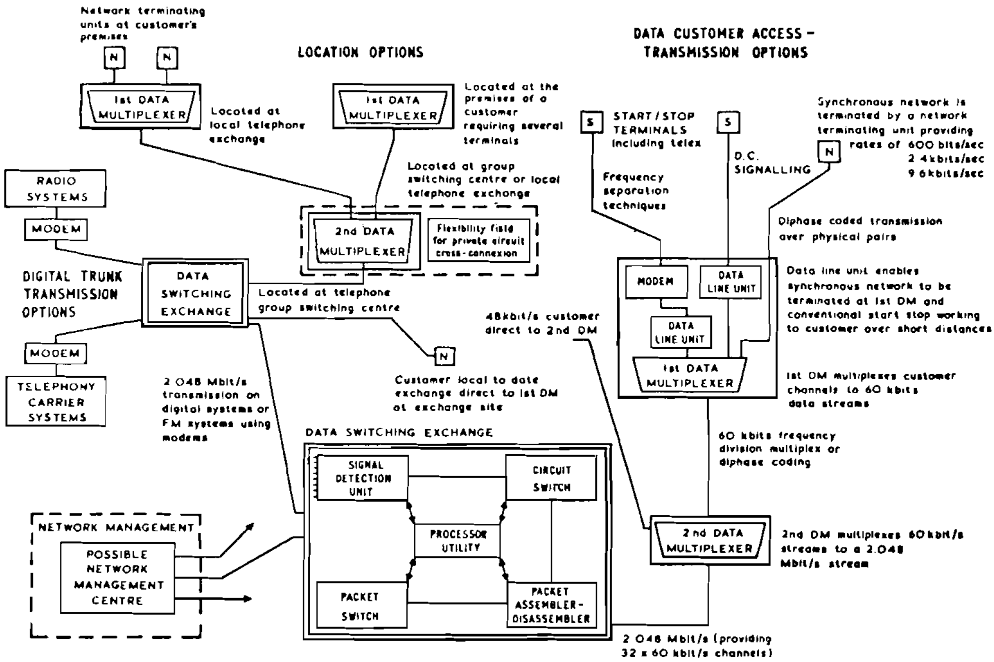


It would be possible to add packet switching to such a system, but the NRD has not made any recommendations on this point.

The UK BPO claimed that while most of the traffic in a network can be best handled by circuit switched techniques, others can be handled better by breaking up the data into packets; these are then sent by fairly standard store and forward techniques. They claimed that a DSE built to handle packets would be only 15% more effective than one without, and the DSE costs are only 25% of the total network costs. At that time (1971), they aimed at an installed capital cost of switch of $600/line. A schematic of the system for mixed packet and circuit switching is shown in Figure 4.

Figure 4.

Schematic of Packet and Circuit Switched Operation in the Data Switching Exchange (DSE)*



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*This is an example of how packet switching could be added to synchronous line-switched networks. No European PTT has committed themselves to provide such a service, however.

The PTTs of a number of countries have announced definite plans to introduce trial or production networks of this kind (without packet working). The Nordic countries, Denmark, Finland, Norway and Sweden are planning their systems together [8], although they will be trying out different technical solutions. An explicit intention, even in their trial networks, is a high level network, connecting Oslo, Stockholm, Copenhagen and Helsinki. Finland, Norway [9] and Sweden [10] will each have 3 nodes on their original trial systems. The Norwegian one will have initially 72K bps transmission between nodes at Oslo, Trondheim and Bergen. At each city there will be a 512 port multiplexor to subscribers' equipment, and between the subscriber and the multiplexor analogue transmission via modems will be used. There will be initially only leased line service in 1974, and switching may be added later. The Swedish system (also commencing in 1974) is somewhat ambitious. It will have a switching exchange in Stockholm, attached by 4K bps lines to concentrators in Stockholm, Malmo, and Gothenburg. It will have limited capacity, initially for only 100 terminals of different types. Transmission speeds of 2.4K bps will be offered, with 9.6 K bps and 600 bps following later; the customer interface will be as for synchronous modems. This system will be similar to the CCITT envelope scheme, and have a call set-up of 100-200 ms.

Elsewhere in Europe, the Swiss have started experiments [11] to produce an integrated synchronous network including data, telegraph and telephone services. The UK has put forward projects on special synchronous data networks [6], and the French have a similar development. The West Germans have considered how they could run the EDS network of Section III synchronously. Many European countries, including the UK, France and Scandinavia, expect to offer digital transmission on a leased line basis to customers' premises within the next four years.

Outside Europe, the activities nearest fruition are in the US and Canada. In the US, a leased line digital service is under construction [12]. The speeds offered for digital synchronous transmission are at 2.4, 4.8, 9.6 and 56K bps. The proposed installation schedule is 5 cities by mid-1974, 24 by the end of 1974, 60 by the end of 1975, and 96 by the end of 1976. This system uses an unused portion of the frequency spectrum of the standard long-distance telephone transmission systems. This is the reason why ATT is able to propose such a large-scale installation in such a short time scale. The question of adding digital switching to the system is also under study. DATRAN [14] has a rival switched data network under construction. It is finding it very expensive to compete with the ATT Dataphone Digital Service, because it has to build its own special long-haul microwave transmission system. Originally it proposed to put in its own local distribution into customers' premises, but that idea seems to be in abeyance; it now proposes to use ATT's local lines.

The Canadians have put in a cross-country digital data transmission system, Data Route [15] and are proposing to add switching units to give a service similar to DATRAN. Japan has made tentative plans for a similar service to start in 1967/7 [16]. It seems clear that many countries will follow this route in the future.

V. US PUBLIC PACKET-SWITCHED SERVICES

New data networks where circuits are provided between the two ends have been discussed. An added, or alternate, facility is to store the data in an exchange near the subscriber, and send it at higher speeds in packets to the recipient. This form of service, called "Packet-Switched Service," permits a number of features to be added. These include speed changing between a sender and receiver, and many sending stations mutliplexing their data to one receiver (and vice versa). Moreover for bursty traffic, this system makes better use of the long distance transmission medium--at extra cost in the switches. In the past these systems have been used for message switching; in which whole long messages are stored and then forwarded. More recently, the same principle has been applied to short packets, typically 1-2K bits. Many large private computer networks adopt packet transmission in their data communication subsystems, (e.g. the General Electric [17] and TYMNET [18] systems). One large data transmission system for the airlines SITA [19] has used it for its high level network. The best-known of the Packet-Switching Networks, ARPANET [20] now connects 50 centres and has nodes not only in the US but in Norway and the UK.

The success of the data communications aspects of ARPANET encouraged several US private firms [21, 22] to propose large scale switched services, with transmission facilities leased from the telephone carriers. These services are being developed by groups with substantial experience of ARPANET, and are to offer similar systems with the appropriate modifications for commercial security and reliability. The extent and time scale for their services depends on the availability of capital. Both plan to offer their services by late 1975, and both have received FCC permission. TELENET is unique, since it relies heavily on long-distance domestic satellite data transmission.

TYMESHARE [18] is offering a differently designed packet-switched service. It already has national coverage, with additional switches in Paris and London. It operates also as a Value Added Network, since it gives terminal access through its TYMNET not only to its own computers, but also to others such as those at the National Library of Medicine, Batelle, Lockheed, and Systems Development Corporation.

Any of the special networks between universities or banks, or other closed systems, which are not available to the public at large are not included as they are beyond the scope of this survey.

VI. NON-US NATIONAL PACKET SWITCHED SERVICES

For a time the PTTs were unconvinced of the need for public packet-switched services, but many seem to have been persuaded recently. The requirements for Packet-Switched Services is now the official concern of Study Group VII of CCITT, and several PTTs have announced experimental or prototype services. There seems a fair measure of agreement on the types of service which will be offered. One will be a "datagram," in which an individual packet will be sent to a destination, and the sender may or may not be informed of its delivery. The second is a "virtual circuit," in which the same sort of service is offered as in a circuit-switched connection, but the sending and receiving speeds may be different once the critical circuit has been established, and messages may arrive over several interleaved virtual circuits.

On the whole the PTT's are more used to the "virtual circuit" concept, and two PTT initiated experimental services in France [23] and the UK [24] use mainly this technique (though the UK system can be used to send datagrams also). The French experimental service, RCP, is nearly operational [23]. It will have nodes in Paris, Rennes and Lyon, with remote concentrators in Bordeaux, Marseilles and Lille. In all, it will have scope for carrying 120 connections. The internode line speeds will be 9.6K bps, and the speeds to the concentrators 4.8K bps. Customers will access the nodes over the telex or switched telephone networks (at speeds up to 300 bps), though some leased lines at 1.2K bps and 4.8K bps will also be available. Subscribers will come into the network with arbitrary data streams of 8 bit bytes. RCP will start offering service in early 1975. It will be replaced by a more sizeable system, TRANSPAC; this is scheduled to have its nodes operational in late 1976, and be open for service in 1977. TRANSPAC is designed to have 6K terminals at all speeds, and have its procedures downward compatible with RCP. It will also offer the datagram service.

The UK EPSS is intermediate between RCP and TRANSPAC in scale [24]. It is again based on three nodes, in London, Manchester and Glasgow, and is scheduled to be operational in late 1975. It has been designed to have very reliable packet switches, with at least duplexed operation (unlike RCP, which has only single computers). In the first version, there will be 166 terminations with character interfaces, and 84 with packet interfaces. The character interfaces will be similar to those at RCP, and will all come over the PSTN (or a related telegraph service). The packet interfaces will be accessed via leased telephone lines at 2.4, 4.8, 9.6 and 48K bps. Unlike RCP, both datagrams and virtual circuits are offered. The EPSS nodes will be connected by 48K bps lines, and the switch design is expandible. It is possible to expand the number of nodes and their capacities. The procedures used are compatible with later replacement of the analogue transmission facilities by digital ones. One of the first places where leased digital service facilities will be installed is between the EPSS nodes.

One packet switched service is already in service in Europe, being operated by the Spanish PTT [25]. At present it has two nodes, in Madrid and Barcelona, with 500 terminals. Because it uses only leased lines, software in the nodes can deal with the present twelve specific terminal types without too much difficulty. The system is somewhat intermediate between a packet-switched and message-switched service, with both sets of facilities provided. The initial growth envisaged is rapid, to 5K terminals by 1978. At that time there will be 9 remote concentrators to the present switching centres. Eventually the aim is to have switching centres in each Spanish province.

Australia is already operating a Common User Data Network for a number of Government establishments, and Japan is planning to introduce a packet-switched service [16] in the late 70's. Canada is committed, via the Trans Canada Telephone system, to provide a packet-switched service from 1975. It has the banking community in mind as its first target, but few details have been released yet. Even the Federal German Republic is investigating how packet-switching can be added to the EDS system.

Another network, the largest packet-switched system now operational in Europe, is CYCLADES [26] in France. This currently has five nodes, in Paris, Rennes, Toulouse, IRIA and Grenoble. It is planned currently to attach some sixteen computers to these nodes. At present CYCLADES would fall outside this survey, because it has no access to switched telephone network, and is not yet the basis of an open national system.

VII. INTERNATIONAL DATA NETWORKS

The present National Data Networks already have their political difficulties; the international scene is yet more hazardous. For example, Western Union International has used the potential saving from satellite digital transmission to offer private line services (IDDS) allegedly at substantially lower rates [27]. It is not yet established that the PTTs will permit the savings of this type of service to be passed on to customers who require also access to the switched telephone network. Early examples show that the PTTs will put on a rate surcharge as a condition to permitting such interconnection. The IDDS service still has both cable and satellite hops; the difference in real cost can be seen by the fact that a single telephone channel over the satellite portion can carry 56K bps, while an analogue telephone cable channel can carry only 7.2 or 9.6K bps. A problem is that the really substantial savings are realised only when satellite channels are used. It is still PTT policy not to differentiate in tariffs (and ideally not to give the customer a choice) between cable and satellites.

TYMESHARE is trying to offer an open Value Added service internationally. In France it has had little trouble. In the UK the BPO has proposed to allow their service only if they charge TYMESHARE a substantial surcharge for each hour of connect time at a terminal; moreover, the BPO would be licensing TYMESHARE to act as an agent of the BPO.

Two shared use networks have been permitted, SITA [19] and SWIFT. The first is a message-switching system for the airlines which has many of the characteristics of National Packet-Switched Networks. It has some eight packet-switched nodes, at New York, London, Paris, Madrid, Rome, Frankfurt, Brussels and Amsterdam in its high level network; terminals and computers of the airlines are attached into the system which has been in existence for some years. There are other shared use networks of this type in other industries. SWIFT is a system for banking, using two switches in Brussels and Amsterdam, due to go live in 1975. This has needed approval from all the various PPTs in Europe and North America. These systems are again essentially beyond the scope of this paper because of their closed nature.

Two other activities should be mentioned here. Already the SPADE system [28] is providing 56K bps of data traffic over a voice channel. Unfortunately this data rate is not available to customers yet at the corresponding low tariff. The long distance data transmission costs can be expected to be reduced drastically due to satellite techniques [29]. By using some of the new broadcast satellite techniques [30], yet more substantial savings should be possible, particularly for low activity channels.

Several international experiments are planned to improve compatability and to examine these techniques. An international link between the French EPSS and the UK EPSS is planned for late 1975. An experiment with the broadcast satellite techniques of Reference 30 is being discussed with, currently, the UK, the US, and possibly other countries. On international compatability, several protocols have been defined [31, 32] and experiments to investigate their relative advantages are under way between research groups in the US, UK and France. It is too early to say whether these experiments will lead to international standards, but the PTTs are well represented on the IFIF Working Group 6.1.

This section would be incomplete without mentioning one major European development, the European Informatics Network (EIN) [33]. At present, while international, this network is to be a closed one, with nodes in London, Paris, Zurich, Milan and ISPRA. The nodes are designed to have interfaces with the national data and PST networks and the EIN may in the future play a role as an international high level network. So far, only the specifications for the communications subnet have been drawn up, and by the time of this conference, the contracts for the communication computers will have been placed. However who will be able to access this network, how it will be accessed, and what will be its role in relation to the National PTTs and their plans are still a moot point.

VIII. COMPUTER NETWORKS AND THEIR AIMS

In Sections II-VII we surveyed the status of the various Public Data Network activities. At first sight, these seem a motley collection of incompatible projects. However, there is excellent collaboration between the PTTs, and great emphasis is being placed on the ability to interconnect these networks. For the interim networks typified by Sections III and V, interworking will be hard and special gateways must be employed. However, such gatway experiments are already being planned, and will be constructed within a year or two of the national networks being operational. The synchronous networks of Section IV are being designed with international compatibility in mind. It seems clear that with these networks, interworking at much higher speeds should be possible and even economic. In the US, national working at speeds up to 64K bps are being promised for 1974, and a full T1 carrier, at 1.34 M bps, should be available fairly shortly. The use of domestic satellites should make these speed available at rates relatively independent of distance. In Europe, one may expect the 48K bps rate to be standard by 1980, as the digital services come into being in the various countries, but the dependence of rates on distance will remain longer--though it should not be as steep as now.

The impact of these changes will be striking. Already now a US carrier (MCI) is offering a magnetic tape transfer service; at 50K bps, a full length tape can be transferred in half-an-hour, at 1.5 M bps, it takes only one minute! At 50K bps, with data compression, an A4 page of facsimile typescript of text takes 4 secs; at 1.5M bps, one can send 8 pages/second. With machine readable text, the rates are at least a factor of ten faster! Thus the advent of these data services will make facsimile and file transfer technically achievable, and probably even economical. Having such communications facilities, the possible use of computer networks increases dramatically.

At the lower speeds, terminal use of remote computers is easily achieved, but lengthy output is a nuisance. There is a premium on interactive browsing through output, and a need to use information retrieval techniques. At the higher speeds, file transfer becomes a reality, and it is no longer as necessary to have the data bases and processing systems co-located--though this may still be convenient and more economical, of course.

In the context of research, there are three basic reasons for computer networks:

1. Data acquisition;

2. Data bank collection;

3. Data processing and data bank access.

In many applications there is a requirement to collect data. With the rapidly reducing costs of mini-computers this can usually be done most economically digitally, and is often convenient to transfer it on-line to a large data bank--though the transfer can be done off-line. The construction of the data banks can be very expensive, and for many purposes it is desirable to have the banks widely accessible. There is a large saving in cost/bit when one is able to use stores of the order of 10^{12} to 10^{13} bits. However, the use of such large stores may mean that their use must be shared amongst widely dispersed groups--perhaps even on a world-wide basis.

There is not necessarily the same gain in central processing of the data--though this can still be a major factor. The local or national availability of large processors may make it more desirable to do some of the processing nationally, but to build up data banks internationally. This type of use would require the fairly rapid transmission facilities mentioned above.

A further reason for computer networks is the need for extreme reliability in the on-line environment. It is this need which led to the large complexes of processors in the commercial systems of [17] and [18]. While their reliability makes such configurations desirable, their added power forces the use of data networks to increase the number of users able to access them. It is a combination of these technologies together with those of ARPANET [20] and its derivatives, which will lead to the optimum networks.

IX. MESSAGE FACILITIES

In many applications of networks, it is not only that data must be moved between sites, but also that people in different locations must interact. Often there may be several different people at different sites who must take part in conferences. The same requirements exist for scientists trying to analyse results, who may need to consult with their colleagues at other centres. This type of case is greatly aided by good telex-like facilities, often with multi-destinations, the possibility of teleconferencing, the ability to add facsimiles to data streams, the ability to display graphic data, and the ability to introduce files already archived into the conference. Even packet-switched voice may be a useful adjunct.

The present telex facilities are much too slow for this; there are often substantial amounts of data to be sent, so that, at least 30-100 ch/s terminals are useful--much faster is too fast for people to absorb the information. Moreover, the present PTT facilities do not have interactive editing facilities with archived material, the capability to support facsimile, or the ability to teleconference. The parties using the network belong to different countries and organizations. The use of computers and shared lines for the above purposes is not permitted by the PTTs in this regard. The special facilities mentioned above may only be required by too few people to justify a national service from the PTT, yet they may be vital for the proper interaction of certain users. There will probably be increasing pressure from users to persuade the PTTs to provide the basic transmission facilities, but to permit specific sites--maybe as a commercial service in some different country--to supply the services. These types of service have started to be seen on the ARPA Network [34] e.g. a Network Message System, a Network Editing Service, and a Network Teleconferencing System. The form of service is a little like the Value Added Networks (VAN) of [22]. The licensing of such VANs may be repugnant to the PTTs, but it may become necessary.

X. CONCLUSION

Present computer networks have been established and made to work; in eight years' time they will be easy to put together and give far better facilities; however, the PTTs may limit the facilities which may be available.

The new digital data networks being developed by the PTTs will give the basic data communication facilities they require. The next generation of operating systems will make it easier to attach computers, and will make unnecessary the contortions undertaken to attach some of the ARPANET Hosts. The next generation of applications will start requiring computer networks to achieve economies due to functional specialization when it makes sense, and high reliability. These applications and the close interaction required between people will, one hopes, either persuade the PTTs to add much more sophisticated man-man communication facilities, or to permit such services to be added by entrepreneurs.

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DISCUSSION

Question: What kind of problems could be solved if an international organization were to consider public data networks.

Answer: I do not think that an international organization can help very much. The need for cooperation is well known, but developed only in some aspects. This conference therefore perhaps starts with a deeper dialogue.

Comment: Most of the international conferences have been attended by national representatives. Now the IIASA people are really international, and could define the differences between systems, standards, etc. There is a major problem with the CCITT papers, which are normally not concerned with concrete problems. IIASA may well be in a position to address such problems in specific discussions, and the Institute may also be well placed to bring different organizations together.

Comment: The last TYMNET statistics are about 40 computers on that net which do not belong to time-shared corporations, and 36 computers which do. There is a feeling that TYMNET could cut off the European connections if things go badly.

Answer: According to TYMNET, they are planning an approach center. Another problem is that PTTs in different countries have different levels of sophistication, and that their development plans are not the same.

Question: The price curve given in the paper is based on old equipment which will fail with age and have to be replaced. Are the cost predictions realistic?

Answer: This is not the main problem. To fill the big channels, and give new facilities to the users remains unsolved. Until now, users have had no control over the type of transmission, although they should have.

Question: Explain why a TV cable network cannot support the distributional function of a local computer network?

Answer: Cable TV is not developed enough to reach all parts of a population. It is assumed that cable TV can reach all houses in a large area, but in reality it is not able to reach even half. Investigations have tried to see if a clutch of cables reaching nearly every house in a large area can be converted into nodes of an interactive system. It is unrealistic to think that everybody could have a terminal to receive and store information. The single channel crossovers are also a problem. In the USA, cable systems are larger than in Europe.

Comment: In the USA TV cables are a useful way of distributing computer power. In Japan TV and tape equipment costs $400. This shows that technology is developed sufficiently to offer necessary equipment at a reasonable price. The problem, therefore, is not technical but one of distribution.

Answer: The cost to extend this is high, and there are also problems with the CCITT members to decide which technique should be established. There are severe constraints if one takes a certain system design, and then applies it to other areas, because it is only suitable to that specific aspect.

Comment: There still exist other possibilities. The current system uses air waves. Instead of cables one could use more band width, or an optical cable; the air waves should be saved for when they are absolutely necessary.

Comment: The cost curves mentioned in the paper not only depend on the proximity but also on the tax structure of the corresponding country. They had been calculated with a 40 year amortization period.

The Future of Computer Networks

D.W. Davies

The recent evolution of computer networks has largely been due to new concepts in systems design. Networks extend the architecture of computers to a new level at which the computer itself becomes one component of the system.

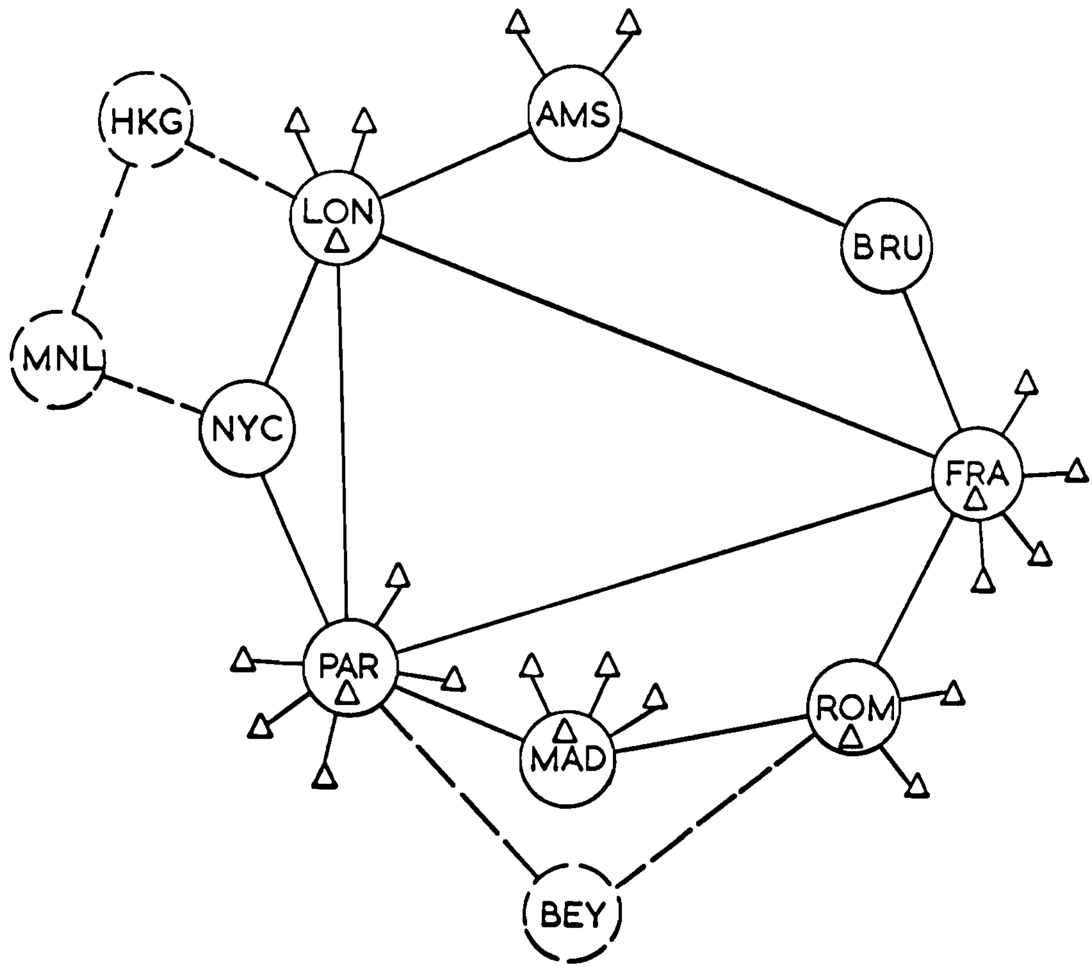
These developments in systems design have been based on new techniques. First there were modems for data transmission using the telephone network, then later there were digital transmission techniques and new methods for switching and multiplexing.

In this paper we first review the techniques which have contributed to the evolution of computer networks. Then we consider the architecture of networks and problems of interconnection and standards.

The earliest examples were private networks. In the United Kingdom, each of the major banks required a central accounting system, thus radially-connected distribution networks were built around their central computers, designed for this particular application. Later new functions were added to their distributed computing systems such as on-line cash dispensers. Many large organizations require, in effect, several overlaid, distributed computer systems. The design approach which will be used in the future is to provide, from the outset, a single communication subsystem for the whole range of anticipated teleprocessing needs. The correct separation of the communication function is the first step in improving the architecture of computer networks.

If the scheme of combining the communication functions of several users is carried one stage further we arrive at the concept of a public data network, designed for computer communications. The aim is to gain the economy of using higher transmission speeds and averaging the traffic over a larger group. This change to public networks requires that a set of computer communication needs should be defined which will be valid for a large part of the user community. This, we believe, has now emerged and so the next decade should be a period of growth for public data networks.

One of the first modern networks was the SITA high-level network [1], shown in Figure 1. SITA is owned by a large group of airlines and provides a world-wide communication system. The high-level network was built to obtain greater economy in the region with the densest traffic, namely Europe with a connection to New York. The network is now being extended to the Far East. Figure 1 shows the computer and



Topology of the SITA network

Figure 1

terminal concentrators attached to the SITA network by the European Division of British Airways. Many computers, each with its own set of terminal concentrators, can be connected in the same way. Each user provides his own terminal concentrators because there are no standards for the terminal interfaces and procedure, therefore the network handles messages between intelligent devices. In fact, the design of the SITA high-level network provided two kinds of service: one (packet switching) serves intelligent devices, while the other is a form of message switching between character terminals. A similar combination of message and packet switching is also found in the Spanish CTNE network.

DIGITAL TRANSMISSION

Economical transmission, whether of voice or data, is only possible in a high capacity link. Communication costs per telephone channel decrease as the size of the group increases. For example, coaxial cables began around 1950 with 600 circuits per pair of tubes and have currently reached 10,800 circuits with the same size of tubes at about one-fifth of the cost per circuit. For transatlantic cables the published figures show that TAT 1 with 36 channels costs £ 294K per channel, CANAT 1 with 80 channels costs £ 100K per channel and the recently opened CANAT 2 with 1840 channels costs £ 16.5K per channel.

Figure 2 shows how transmission economy has steadily improved as a result of employing larger groups of telephone circuits and therefore higher bandwidths. The continuation of this trend will require semi-conductor or similar higher bandwidth devices and there is every prospect that these will become available.

The capacity of the main trunk transmission systems is very large. The bundle of 18 tubes used in the "supercable" provides 97,000 channels and the waveguide system which is now at an advanced stage of development provides 105,000 channels although only one third of its potential bandwidth is being exploited. The need for this capacity results from the magnitude of telephone traffic.

Whatever is available for telephony can be exploited for data. Computer communication has been given a bonus by the trend in new telephone transmission systems towards digital transmission.

The reason for adopting digital transmission is the same as the reason for the much earlier shift to digital computing. In digital logic signal distortion is corrected as often as necessary to avoid errors. In digital transmission regenerators are fitted at necessary points to restore the digital signal to a near perfect shape. Thus the siting of

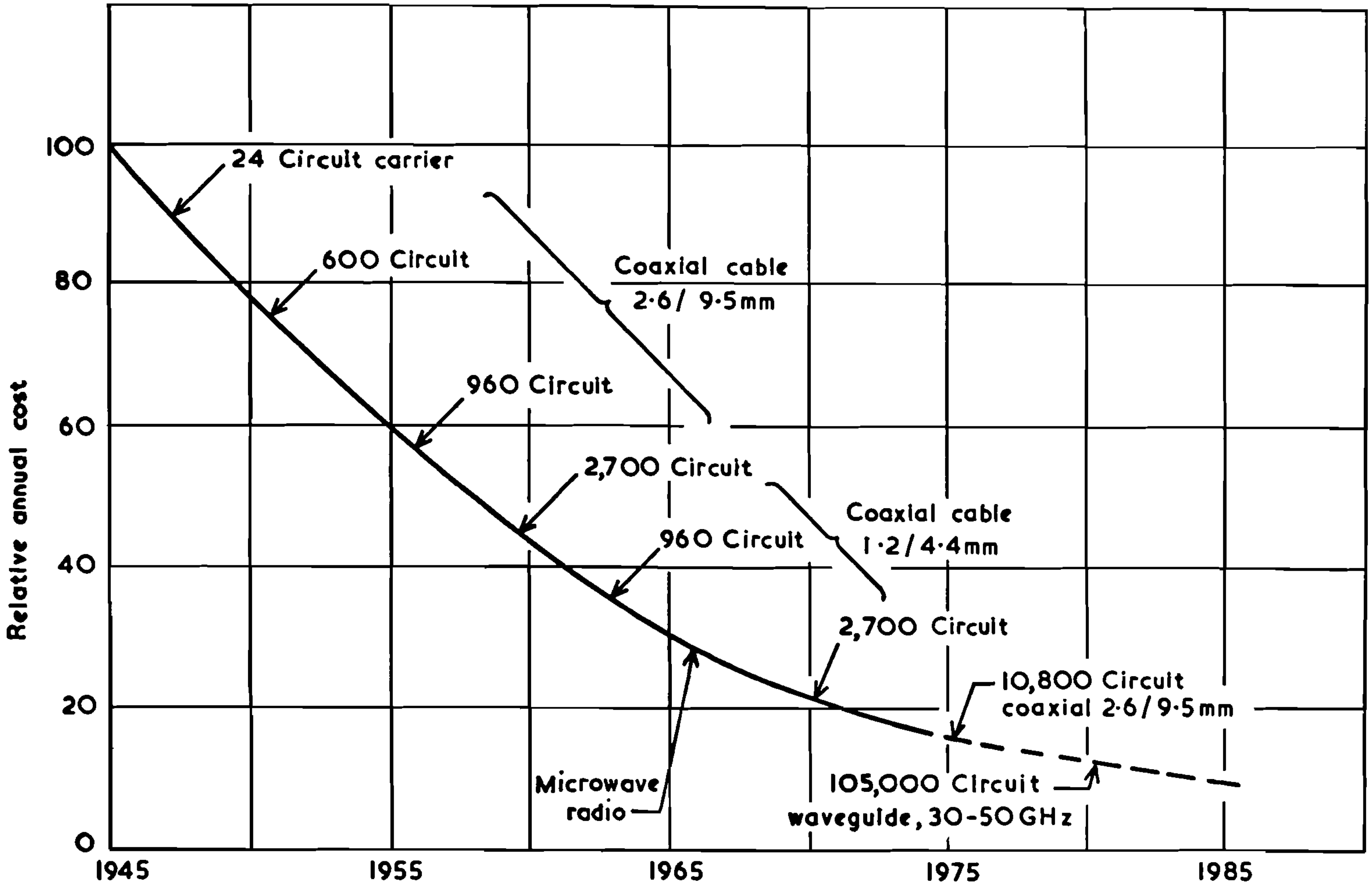


FIGURE 2

COST OF LINE PLANT FOR TRUNK CIRCUIT

regenerators adds another dimension to the design of the system. The older, analogue, method required that line noise and distortion be controlled from end to end, this became increasingly difficult as the lines became longer and passed through more countries. With digital transmission, only very small bit-error rates accumulate with distance, and in the more modern data communication systems this is kept well under control by error correction procedures.

Every new kind of transmission technology uses digital transmission--coaxial cable, waveguide, microwave, satellite relay and optical fibres. Since the digital transmission (by pulse code modulation) of one voice channel occupies 64,000 bit/s this provides a big advance over the bit carrying capacity of lines with modems.

At present, there is a large amount of PCM transmission in the short trunks which interconnect between telephone exchanges in a local area. But with very few exceptions, users do not get access to these digital streams at 2.048 Mbit/s.

In the UK, the earliest development in digital transmission which is intended for the main trunk network and could therefore form the basis for computer communications is a 120 Mbit/s system which, used for telephony, carries about 1600 channels. When a part of this is diverted to data transmission, the backbone of an effective computer communication network in which trunk transmission costs are very low will be established.

Therefore, the general picture in transmission capacity is one of future plenty based on a demand for large bandwidths which comes from telephone traffic.

SWITCHING AND MULTIPLEXING

The telephone network has employed circuit switching because this is appropriate for long calls compared with switching time and transit time. In the past, with analogue transmission, frequency division multiplexing was used but this will be replaced by time division multiplexing and digital transmission. Because all telephone connections have about the same bandwidth which they occupy throughout the call, it makes sense to allocate each one a fixed band in FDM or a fixed time-slot in the TDM cycle.

However, for computer communication none of these techniques is appropriate. Common data rates range between 100 bit/s and 4800 bit/s or more, and the individual messages are often very short because many are concerned with control functions, red tape or acknowledgements, both at the communications and the applications level. All these factors point to a more flexible method of multiplexing and switching.

If we regard circuit switching with FDM or fixed-cycle TDM multiplexing as one extreme of a spectrum of switching and multiplexing techniques, we find that there is a general drift towards a greater degree of asynchronism and towards dynamic assignment of channel capacity. The latter goes by many names, such as "demand assignment," "statistical multiplexing" or "dynamic multiplexing"--all varieties of the same sort of technique. Packet switching is the natural switching method to go with demand assignment multiplexing because it also handles small units of information together with their addresses, and these units compete for the use of the switch, in the same way that they compete for the use of the communication channel.

A good example of demand assignment in an extreme form is the ALOHA system, first used with radio links in the University of Hawaii [2] and now proposed for use in satellite connections as part of a packet switched network [3, 4]. It is a broadcast technique in which all the stations can hear all the others, either by direct radio links or via a satellite transponder. Particularly in the satellite case, there is an insoluble logical problem which is how to allocate time on the transponder when random demands arise in different places. The audacious technique used in ALOHA allows any station to speak when it wishes, and relies on statistics to get most of the messages through uncorrupted by mutual interference. An elementary theory shows that, with no synchronisation at all this system reaches its limit when the channel is occupied to an extent of 1/2e or 18% of its potential capacity. This may seem a low figure, but there is a compensating advantage due to the dynamic allocation. As the traffic reaches the limit, an ALOHA system collapses spectacularly [5] with every message being misheard and repeated indefinitely. Some interesting theoretical work is going on to determine the best strategies for repetition, making use of the fact that each station hears all the traffic and can adjust its parameters according to the traffic level.

The multi-drop line is such a broadcast system which could easily have used the ALOHA technique instead of traditional polling, and might have worked more satisfactorily.

Wherever computer communication employs short messages, packet switching will be the preferred method, however, it will show a clear economy only over long distances. The wider use of packet switching will come about because of a desire for a unified system and because it fits in with the general philosophy of computer system design.

This does not mean that circuit switching is inappropriate for digital data. There are many kinds of digital traffic, such as facsimile or viewphone, which have long holding times and have no need of elaborate error control. Moreover, they employ constant bandwidth and depend on preservation of time, unlike true data communication. The future therefore seems to include a requirement for both kinds of switching, the circuits carrying more bits and packet systems serving computer networks.

Circuit switches are increasingly controlled by computer, and packet switching is entirely carried out by computers. There is an apparent trend towards greater efficiency and economy in small computers even though the speed of stores is no longer evolving rapidly. Telecommunication switching may come to use groups of small processors, but the techniques for efficient collaboration of many processors have yet to be worked out. One promising approach in packet switching is to employ a group of autonomous switches making up a small local network. At the present time, a small processor can switch approximately 1 megabit per second or 500 packets per second. The "high speed modular IMP" being developed for the ARPA network is an example of a similar approach, and the UK Post Office EPSS switch is also administered by a group of small processors. If a small economic processor would be designed and manufactured, packet switches would certainly assume this modular form in the future.

ROUTING AND FLOW CONTROL

From the earliest studies by Baran [6], packet switching has been associated with adaptive routing. This association may have arisen because a packet transport system provides a convenient way to implement a complex routing algorithm. A more limited form of adaptive routing of circuits has been present in the telephone network. It can be used to recover from the effect of line failures or to accommodate changes in traffic patterns.

Recent studies both of circuit switched networks [7] and packet switched networks [8] have thrown some doubt on the value of the extreme forms of adaptive routing. If, in response to overload on one route, packets are sent by routes involving more hops, the network as a whole is less efficiently used. This type of adaptation works only over a very small regime close to overload and it contributes eventually to a more sudden congestion.

The method employed in the SITA network [1] calculates in advance optimal routing strategies for the undamaged network and a number of damage conditions such as any single node or two link failures. Alternative tables are held at each node and brought into operation when a report of failure or recovery is received.

The investigation of routing techniques continues. Adequate methods are available for small or experimental networks but more knowledge is needed before large or hierarchically organized networks are built.

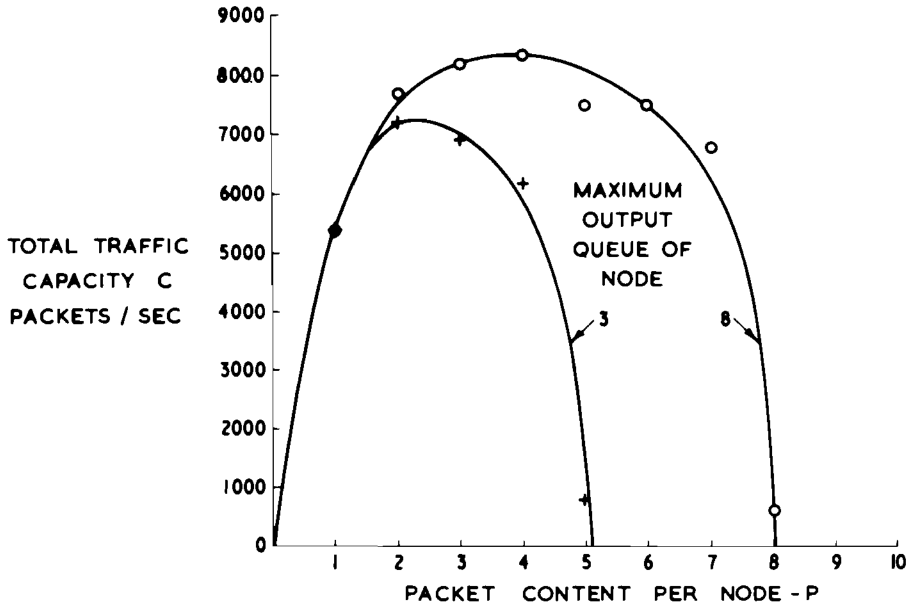
Control of network flow is closely related to routing. It is a problem that arises only in packet switched networks where the "virtual circuit" between two communicating computers can carry a varying amount of traffic. There are analogies between these flow control problems and the control of traffic on road systems (see Figure 3). In Figure 3 the throughput of a network is plotted against the number of packets currently in transit. Roughly speaking, this number of packets is also a measure of the delay in transit through the network. The interesting feature is that above a certain packet content the flow capacity falls. Clearly there is no possible advantage in allowing a network to go beyond this point because it carries less traffic at a higher delay. A special technique [9] was developed to control the total packet content of the network. At the same time, it has been found [10] that careful attention to details of the functioning of the network's switching nodes can produce the same effective kind of flow control and the need for a special mechanism is doubtful.

DESIGN OPTIMIZATION

When all the components of a communication system have been designed they provide only a "kit of parts" for the construction of an actual network. The location of nodes, the choice of links between these nodes and link capacities, and the choice of sizes of switching equipment or location of multiplexers can all make a considerable difference to the economy of the network. This aspect of network design can be called "design optimization." There is already a considerable literature on optimization problems in computer communications, beginning with Kleinrock's classic book [11]. For all but the most elementary problems no analytic solutions exist and the methods employed are essentially heuristic. Nevertheless, it has been shown that these methods approach closely to the performance of the theoretically best designs [12] and, rather paradoxically, that optimal network topology is not very sensitive to the traffic matrix [13].

Until recently, all this work for data communication was based on the concept of a mesh with arbitrary topology. On the other hand, the development of telephone networks had already evolved a form of hierarchical organization in which the longer distance lines of larger capacity formed a "supernet" and, developing this concept further, typical telephone systems had four levels of network. The top level in a country contains a few centers fully connected for the utmost reliability. The bottom level contains inter-exchange links in highly populous areas.

The basic reasons for hierarchical organization of network layout is to concentrate traffic on high density routes, and to make use of the economy which is obtained with transmission systems of extremely high capacity.



CHARACTERISTIC OF AN 18-NODE ISARITHMIC NETWORK

FIGURE 3

Hierarchical organization in the telephone network has been developed largely on empirical lines but a recent theoretical treatment in a thesis by Schlumberger [14] shows similar results. For a network with d links per node and n nodes there are connection patterns and associated routing algorithms which produce an average number of hops proportional to $\log_d n$.

Undoubtedly, as computer communication networks expand they will take on a hierarchical form. On the basis of simple trials it seems that hierarchical organization can be an advantage when the number of nodes in a network reaches a figure between 50 and 100. Hierarchical structure has a profound effect on the choice of routing algorithm and there is probably much to be learnt in this connection from telephone networks.

THE FUTURE OF NETWORK ARCHITECTURE

Ultimately the architecture of networks is determined by economics and by the customers for whom the network is designed --their geographical distribution and traffic pattern. The cost of a network in a highly developed or compact country is determined largely by the local distribution system and not by the trunk network. This is because of the economy of high density trunk routes and the high cost of the wires joining the local switches to the users' terminals. Even in countries of the size of the USA, because of the non-redundant nature of data traffic and the efficient use of lines by demand assignment, the cost of long lines for a public data network will be low.

Trunk communication is more costly for a large country which is either less developed in telecommunications or sparsely populated. Examples would be Northern Canada, much of the USSR and Australia. For these countries, mircrowave transmission with satellite relay becomes attractive. This can also be an alternative to cables under oceans.

Because of the delay of approximately .25s for a satellite hop, it would generally be unacceptable to have two satellite hops in a link between a terminal and a computer. It is likely that even one satellite hop might be troublesome in some computer to computer interactions. The schemes [3, 4] which are being discussed for the incorporation of satellite relay to replace the mesh connections of a packet switched network would probably cover a continent or ocean. In planning for a world coverage of data communication it would be wise to retain cable connections as an equal alternative for satellites wherever these exist. The development of digital transmission on under-sea cable has been started by KDD in Japan and may become important in the future.

The availability of cheaper logic will reduce the cost of switching, and encourage the use of local intelligence, including the more complex forms of multiplexing such as demand assignment. These trends point to the present approach represented by packet switching.

The most costly aspect of a communication system is usually the local distribution network. It has been suggested that the further development of satellites with higher power and multiple access will assist in local distribution by allowing individual users to have their own radio transmitting and receiving equipment. This is doubtful, not only because it requires a considerable extrapolation of present technology, but also because broadcast systems depend critically on the reliability of each of the communicating devices. In a widespread time-division multiple access system, one faulty station can damage the whole network. Thus it seems unlikely that satellite relay will be used for local distribution except where it is economically imperative.

The design of the local network depends on the distribution and traffic generated by the users. At the present time when large industrial companies or governments are likely to be the major market for computer networks, the local distribution problems are eased by the clustering of terminals in large groups. However, they become increasingly complex with shops and offices, and reach their highest level of cost and difficulty when the computer networks spread to homes.

The timing of the development of these new markets is highly speculative. In the author's opinion, the domestic use of teleprocessing is likely to develop very slowly and may not become important before the end of the century. Even if the technology to make acceptable, compact terminals were available, the human factors problems for interaction of untrained people with computers are very little understood. Nevertheless, the future possibility that data communication may have to reach the majority of homes could influence our thinking about local networks.

The suggestion that cable television networks could take over this distribution function seems technically naive even though it may have a strong commercial interest for the companies concerned. The cable system does illustrate the low cost of connecting houses to a network by a high bandwidth single connection. As the cost of copper increases, it will become economical to replace the multi-core cables which connect our telephones to their switching centers by high bandwidth connections with suitable logic in the telephone set, and the junction points of the network. The most likely arrangement seems to be digital conversion in the telephone and something like the Newhall loop [15] for the first level of multiplexing. The time scales of such developments are uncertain, but in recent years the telephone authorities seem to have been overtaken by events, having grown used to a rather slow rate of technical change. Therefore it would not be surprising to find major changes in the local distribution system in the next decade. Since these changes will be brought about by the economics of the telephone service, it is very important to provide at the same time for the future of data transmission.

NETWORK INTERCONNECTION AND STANDARDS

Interworking between networks requires agreement on a large number of standards such as character codes, packet format, message protocol, terminal protocol, flow control, file transfer, data structure, job control and so forth. Only a small beginning has been made in establishing these standards. Since there are now many designs of packet switched networks under development throughout the world it is necessary to push the level of agreement at least to the point where these networks can inter-communicate as communication subsystems.

It would be quite impossible for the user of a set of interconnected networks, whether he is the terminal user or the system programmer building a computer network on the subsystem, to know the individual details of each network through which his messages pass. In an adaptive routing system he may not know which networks his message goes through. Therefore he must be able to regard the complete collection as one, well-defined single communication system, which Pouzin [16] has called the "Catenet." To achieve this with the interconnection of packet switching systems we have to decide at what level they will interwork. The levels chosen could be the character stream, packet transport or the virtual circuit. After some discussion, a group including ARPA, NPL and CYLCADES is trying out a scheme of interconnection based on a packet transport network with an agreed protocol [17, 18] for message transport in the "Catenet."

The appropriate body at which international agreements are reached is the CCITT, which is part of the ITU, an organ of the United Nations. This now has set up, in Study Group VII and its Working Groups, a powerful machine for arriving at common standards. CCITT works mainly by recommendations, but in Europe at least these have almost the force of standards for the PTTs.

A number of recommendations already exist in the X series, defining for example data rates, customer interfaces and multiplexing and signalling schemes. In the work prior to 1973 which led to these recommendations packet switching was not sufficiently established to feature in them. Now there is a Rapporteur's Group which is beginning to identify the main questions to be solved. Perhaps the first of these will be to establish a maximum size of data field and, for this, the figure that seems likely to emerge is 255 bytes (of the common or 8 bit variety) with the length of the data field, in bytes, given in a separate one byte field. It seems likely that two kinds of service will be defined, one of these a simple packet transport system in which any end-to-end control is handled by the user and the other a "virtual circuit" or "call" type of system which preserves the sequence of packets.

Simple packet transport is most appropriate to computer-computer communications or to communications involving intelligent terminals. Thus it is probably the method of the future. A virtual circuit facility is needed for a simple terminal that is merely a source or destination of bits or bytes.

At present these are mere conjectures because the real discussion has not begun. It is an unfortunate fact that definitive CCITT recommendations for leased lines and circuit switching may not be with us until 1975 and packet switching recommendations will be realized several years later. National networks need not wait on CCITT recommendations, but in Europe they usually do wait in order to avoid incompatibilities, or the need to change standards later. It really seems that a full-scale switched data network, with international inter-connections, will not be with us until 1980, although elements of the system (such as leased digital circuits) will come sooner. Therefore the present tendency towards private networks serving particular communities of interest will not be reversed for a long time.

IFIP has established a Technical Committee 6.1 which is coordinating the experiments in interworking between packet switched networks by means of an internetwork protocol. Thus the present time is a period of rapid development in this critical matter of international standards for data communication. However, the establishment of methods for interworking at the communications level will only be a beginning, it must be followed by work, in which ISO and IFIP together with the manufacturers associations are likely to be involved, leading to the definition of several higher levels of protocols for computers in networks.

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DISCUSSION

Question: What kind of new facilities can be provided by networks?

Answer: Computer networks could eventually replace the mail service between organizations. New facilities are being made operational. For example, we collaborate with other laboratories using communication networks. We do not send them lengthy reports, but simply tell them where to find these in the system. In this way information is extended between organizations, and computers will take on a large part of total traffic. They will also incorporate part of the telex network, which does not work as successfully in the UK as in Germany. The British Post Office, for example, will bring professionals to speak to the CCITT, and specify its demands that the European carriers face an issue together, and come up with policy solutions. On the other hand, one could not stop people using value added networks (NW in the USA), and PTTs in Germany, Holland and other countries are planning to go on with time-sharing. Time sharing NWs do have access control by which you can stop a node in a country, for example, but the question is whether PTTs will continue to trust the provided service.

Austrian Computer Networks in the Scientific Academic Fields*

H. Jeram**
F. Firneis**

In Austria local computer centers offer various services. This study will show the ways in which these facilities can be made accessible to the greatest number of users in the scientific-academic world. The following considerations are applicable also to other fields. The computer centers up to now have offered limited time sharing services, restricted by geographical limitations. We will show how to increase computer efficiency by simultaneous use of the capacity offered by various computer centers [22]. To achieve this a computer network is established, linking centers with various types of computers, giving the user access to all the network resources. This is possible, either through a computer linked into the network, or directly via a terminal. To optimize use, data exchange between any point of the network has to be possible with a high degree of reliability, limited delays, and at an economically feasible level. Different emphases placed on the individual qualities of the network are decisive for the methods to be chosen [1, 2, 6].

Not only those experiences gained from existing or developing networks have to be taken into consideration for present and future studies but also those network requirements necessary to fulfill conditions in the scientific and academic field in Austria. ARPA network [3, 7], NPL network [9], network Cyclades [5, 10], and the COST-11-Project [11], require special mention from among approximately three dozen networks currently either in operation, or in the developmental stage [4].

In Austria heterogenous networks are particularly relevant, i.e. networks encompassing various types of hardware. To cope with any difficulties network control computers were introduced (e.g. the Interface-Message-Processors of the ARPA network [8, 12]), which operate homogeneously within the network. The various resources can be distributed between the network control processor, and the host computer. Communication between the network and control

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computers on the one hand, as well as the host--and network control computers on the other hand, is determined by trace programs, which contain the links between both the operating system of the host computer (process identification, storage allocation, etc.), and the network (linking and routing information).

Present operations, and future computer potential in the USA, have led to the development of large-scale networks whose transmission lines are rented from telephone networks. Currently they are being used commercially [13].

An important network element is the transmission channel provided by existing post and telegraph lines. According to Shannon's Law, channel capacity is determined by the frequency margin, and the given interference zone. Capacity can be increased either by relay switching, or by improving the quality of the transmission lines. The latter allows for parallel transmission of several individual channels in a single line by means of frequency or time multiplexers (see [14, 15, 16, 17]).

For the frequency multiplex procedure individual low speed channels are grouped by hierarchically ordered frequency converters. AT&T's L5 System, for instance, consists of 10,800 single channels in 18 master groups. One advantage is that individual groups can be cascaded easily. However, the frequency is not fully used, due to the necessary blocked zones between individual channels and groups. Also, this procedure requires a high quality transmission channel, such as a coaxial cable with amplifiers and microwave links built in at regular intervals.

Alternatively, for multiplex systems time intervals are allocated to individual channels. By further subdividing time periods a digital hierarchy can be established which reaches into the Gigabeaud range. For instance, in Pulse-Code-Modulation, the voice channel is sampled 8,000 times per second, and the amplitude is digitally coded. This gives a data rate of 64 kilobauds for 24 equal coupled channels, the transmission rate is 1.5 mega bauds, which can be cascaded. Again, the transmission quality of the channel is the essential criterion for a capacity. Thus regenerators have to be built into the lines at regular intervals.

Synchronous or asynchronous time multiplexers allocate a given or variable number of time periods to individual channels. Synchronous time multiplexers, channels linked to the system, are sampled cyclically. However, with inactive channels, potential transmission capacity is wasted. One way to use them is to apply demand-adaptive multiplexers [18]. A channel is declared inactive by repeated blanks. This is used to limit the frequency of error, such as duplicate transmission, on other channels.

The asynchronous multiplexer utilizes transmission capacity more efficiently. Time periods are allocated according to demand. Depending on the amount of data to be transmitted it increases capacity two to fourfold. However, identification of each data block, and thus the transfer of address information is required. Address decoding, and the inclusion of control circuits for checking end marks and buffer zones complicate the multiplexer.

Functional extensions of an asynchronous multiplexer allow for entire messages to be multiplexed instead of individual characters or bits. It is called a data concentrator since it buffers incoming messages. It collects information blocks of various lengths in its buffers until fully assembled and a high frequency line is available for transmission of these blocks including all the data address and control information.

With line or circuit switching the route has to be completely cleared before data can be transmitted. But with store and forward message switching, address information is sent with the message, and these can be transmitted on sections of the route [24]. In this way the capacity of the entire network is increased. However, the drawbacks are longer transmission intervals for individual messages the to intermediate buffering and the formation of queues. Depending on the capability of the message switching concentrator it is able to perform other tasks especially line control procedures, error detection and correction, code conversion, routing, etc. [19].

Store and forward message switching networks have effected the creation of new line check procedures. These are rules relating both to hardware and software which secure proper data interchange between either two computers, or between a computer and a terminal.

Line check procedures, such as the ANSI subcategory 2.4, existed in a variety of dialects because of differences in hardware, the basic goals of the new line check procedures are full transparency code independence, and full duplex operation [20, 21]. This is possible if each message is bordered by flag bytes. The initial flag byte is followed by a master block containing the address and control byte. The latter defines the instruction and sequence number of a message. The master block is followed by data of unlimited length. As flag bytes form the sequence 01111110 the message is clarified by inserting a zero bit after a sequence of five single bits thus distinguishing data from flag bytes. The transmitting unit does this, and the receiving unit removes the inserted bits again. Error control is achieved by a two byte field, directly following the data bits check for cyclical redundancy. This control includes both the master block and the data.

Full duplex transmission is achieved by numbering the messages. Both single and double numbering systems may be used. The given examples show the possibility of using mini-computers, or existing front end processors as concentrators.

Message routing and selecting an optimum site for concentrators depends on network topology. The following investigates communication network structure and optimization [23]. The communication network is given as a graph G(N,M). N-nodes are the user's geographical location at first and links can be chosen freely. From among the great number of possible networks only those which satisfy specific performance criteria should be chosen. The following should be mentioned in particular: average and maximum tolerable delays; average and maximum required throughput; required reliability, and its definition; as well as types of communication and line topologies. Among all networks which would satisfy these criteria the one ensuring maximum throughput, and greatest reliability at minimum cost, should be chosen.

Delays are so complex to analyze that adequate approximation algorithms were developed only recently. Single-server queues can be solved analytically, however, network queueing causes additional difficulties, because flows within the network influence one another. One way to solve it is to postulate the length of a discrete message as an independent random variable with Poisson distribution (dispersion). This assumption is perhaps restrictive, but the results can be verified in actual situations. Thus network performance can be traced back to single-server queues. This model also helps to study the dispersion characteristics of messages and additional traffic due to control data. One difficulty arises from the impact of routing problems. For tree structured networks this problem is less relevant, as there is only one specific route possible between any two nodes.

An analysis of network reliability depends upon the reliability of the individual elements. The ability to link any two nodes in a network depends upon the number of independent routes between these nodes. Accordingly, redundancy can be assumed for either the entire network, or for individual parts. Network reliability models take into account specific failure probabilities of individual components [28, 32, 33]. Thus the impact of the failure of individual components on the entire network can be studied. These models can be implemented on the computer in the conventional way, with random number generators. Basic structural differences result depending whether either one central computer, or several computers spread out over the network are provided.

In the case of a single computer (central network), a tree-like structure is adequate. If the span for all trees is determined by the total length of branches, the network with the lowest possible span is the optimum choice. When designing minimum spanning trees, individual flow requirements cannot be taken into account. Also, the spans between nodes on various branches are extremely long. To overcome these difficulties, STEINER-nodes are being introduced. It has been established that the optimum topology deviates from the minimum spanning tree in such a way that fully utilized bounds can be inserted into the network. In order to further optimize the topology, iterative exchange methods are applied, which change small parts of the network in order to lower costs and/or delays, while satisfying all given pre-requisites [25, 26, 27, 30].

In a decentralized network, several computers are available. The essential problems of this network design are the specification of message routing and topological structures. The routing algorithm not only has to utilize fully the available transmission capacity, but also has to have an efficient computing technique to allow for repeated application. This can be achieved in two ways: the message is transmitted either via routes with low loading and small number of nodes, or via that shortest route already used by other messages until cut-saturation is achieved.

Topological considerations greatly depend on the required degree of reliability. For larger networks, reliability studies are one of the most important criterion. As the nodes increase the chances of good communication between them decrease drastically [29, 31].

Network design should try to achieve a desirable degree of reliability. This can be met by inserting redundancy of network elements. It can be defined as the number of independent routes between any two nodes of the network, or as the minimum number of network elements whose failure would cause a communication breakdown between two nodes. However, since insertion of additional network elements increases costs, the main criterion should be to obtain this reliability at the lowest possible price. Similar to the "travelling salesman" problem, no practical solution had been found.

Finally, the following method has proved feasible: first, arbitrary networks are established, which satisfy reliability requirements, and then using a step-by-step optimization the most economical network is determined.

To generate a satisfactory network, the degree of a node may not be smaller than the maximum number of independent routes radiating from it. All adjoining nodes are tested on the order of their redundancy requirements, and from all possible combinations the cheapest is chosen. The reliability requirement for this node is then decreased by one, and the procedure is continued until all reliability requirements are satisfied. Thus a reliable network is established.

Optimization procedures are trying to find local optima, because global optimization for large numbers of nodes requires too much computer time. The usefulness of the optimization algorithms depends upon whether the reliability of proven combinations does not have to be checked again. In large networks, the procedure is modified so that only individual subsections are checked at a time. The criterion determining the size of the various subsections is the computer time required for their optimization.

For future development, which is guaranteed by the postal and telegraph authorities, the establishment of long distance, high capacity communication links will not require the highest financial outlay because of the constant spread of modern technologies. The local network elements will be most cost intensive. Thus special care has to be taken in designing these. This depends basically on the users' geographical distribution and on their respective demands.

Because of the complexity of future networks, neither the individual terminal user, nor the systems' programmer will know through which parts of the network a specific message will run. Therefore, it will be necessary to view the entire heterogenous system as one well-defined communication system. This is achieved by standardizing many individual elements, such as character codes, data structure, packet format, message and terminal format, job control, etc. Since this standardization is based generally on CCITT recommendations, and since they usually are adhered to in Europe, it will be advantageous to take these recommendations into consideration during the detailed drafting of computer networks.

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DISCUSSION

Question: In Japan, the PTT is assisting with a project to connect several universities. Is it possible to achieve load balancing via a network?

Answer: This is difficult because each user wants different access to various possibilities which can result in an unstable network.

Comment: From scientific and technological points of view, the connections are in order, but commercial considerations are great. The major problem is to choose the most beneficial connection. Experimentally, the network is very interesting, but from a cost point of view it is not very profitable.

Answer: Universities in Austria are financed differently. In any case, a network in an academic domain would never be commercially infeasible.

Data Communications in Sweden

Thomas Haug

I. BACKGROUND

The Swedish Telecommunications Administration has been offering data transmission services since 1962, and in accordance with established policy, it always provides modems. The rapid development of data processing systems led to the introduction of a new generation of computer systems, offering multiple access and time sharing. Because of this and of the increased processing capacity, interest in remote processing via telephone lines grew rapidly.

However, public telephone networks cannot be used for all types of data communications, and in many cases leased circuits are used to build up private networks. Unfortunately, these networks vary in design, in procedures, and in operation and monitoring facilities. Intercommunication therefore becomes almost impossible in many cases. As a result of this proliferation of systems, the Administration started in 1969 a study of suitable forms for a public data network in which all types of data communication could be included in the future.

The first phase of this study was concluded in 1971 with the following results:

A switched data communication service was needed to provide access to and from other networks. It was decided to include those data transmission rates proposed by CCITT but only up to 9600 bits/sec, as the need for higher rates would remain small for a long time. The new network would have to be code transparent, it must be possible to transmit any bit sequence between two points in the network, and thus there must be no limitations on the number of consecutive ones or zeros.

The choice between buffered and non-buffered systems was considered very carefully. A packet-switching system would offer the advantage of a higher degree of circuit utilization. Because of the moderate size of the Swedish network, and because circuit-switching is a technique familiar to any telecommunications administration it was decided that the network should be built on this principle, but the possibility of introducing packet-switching later should be retained. As for the important question, whether the network should be synchronous or not, it was decided after extensive study that a synchronous network would in the long run be preferable, particularly because it was considered desirable to use future digital transmission systems as carriers for the data channels.

The structure of the network was to be hierarchical, with a high-level network linking a small number of exchanges. Each exchange would control a local network, consisting of subscriber circuits connected via concentrators or directly to the exchange (see Figure 1).

II. TRIAL NETWORK

As the proposal involved a partly new technique, it was deemed necessary to perform limited trials before a large scale data network could be built. It was therefore decided to build a trial network accommodating up to 100 subscribers, connected via digital concentrators in the three largest cities in Sweden: Stockholm, Gothenburg and Malmö. The concentrators are connected to a computer controlled exchange with a very limited switching capability, located in Stockhom.

The main principles used in the design of the trial network are the following:

1. The network is synchronous, implying that the data transmission from the subscribers is controlled by a clock in each concentrator. Synchronization of the entire network is performed by a centrally located clock controlling the other clocks.

2. Envelope transmission of the 8+2 type is used, the 2 bits being added to each octet by adaptation units on the subscribers' premises. This principle ensures a flexible signalling system and full code transparency.

3. Data rates of 600, 2400 and 9600 bits/sec are accommodated.

4. Time-multiplexed circuits are used between concentrator and exchange.

5. Set-up times of 100-200 msec are attained.

6. A fault detecting and locating system has been built into the network.

7. The network is dimensioned for a grade of service of 0.1 per cent per link.

The network operation was started during the spring of 1974 on a very limited scale, with only Administration-owned terminals. Today, work is in progress in order to connect a number of subscribers outside the Administration.

Simultaneously with the design of the trial network, two other important activities have been going on. Phase II is

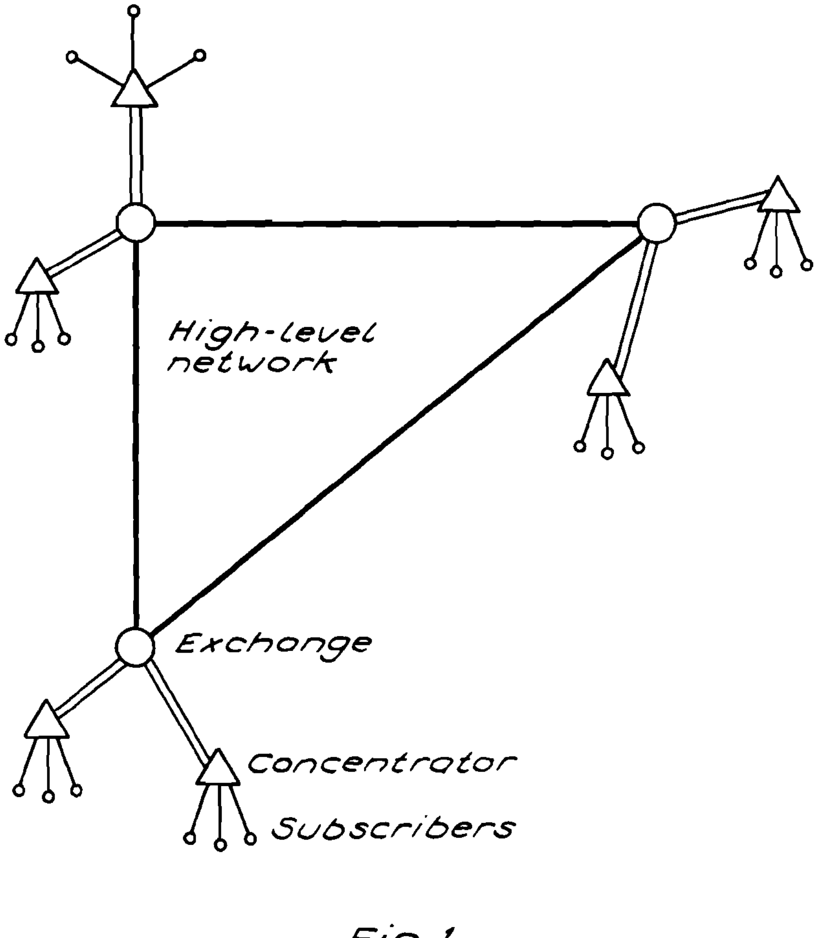

Fig. 1
Structure of data network

a detailed study of the structure of the future data network, which closely examines the technical and economic aspects of a number of alternatives. The other activity is a study of the interworking between trial networks in the Nordic countries: Denmark, Finland, Norway and Sweden.

III. DETAILED STUDY OF THE NETWORK STRUCTURE

The starting point in the structure study was to list a number of feasible alternatives. These solutions were then analyzed taking into consideration the expected subscriber and traffic model.

The structure of the high-level network is determined mainly by the number of exchanges. Four main alternatives were investigated:

1. One exchange (Figure 2);
2. A small number (2-3) of exchanges (Figure 3);
3. A larger number (4-10) of exchanges in a one-level system (Figure 4);
4. One transit exchange and a large number of local exchanges (Figure 5).

The outcome of the study was as follows:

1) In a network with only one exchange, there are no alternatives regarding the high-level network. However, the local networks can be arranged in several different ways, with or without concentrators, multiplexors etc. Common to all of these alternatives is the fact that all traffic passes through a central exchange.

2) In this alternative all exchanges have direct connections with each other and thus there is normally no transit traffic through the exchanges. It is necessary, however, to route traffic between any two exchanges through the third in case their direct connection goes out of operation.

3) Several variations are possible for this alternative. One extreme is to connect every exchange only to two other exchanges in order to simplify the network and the routing procedure while still maintaining a fair degree of reliability. This method will in many cases lead to long set-up times. The other extreme would be to connect every exchange directly to every other. In that case, there will be no transit traffic.

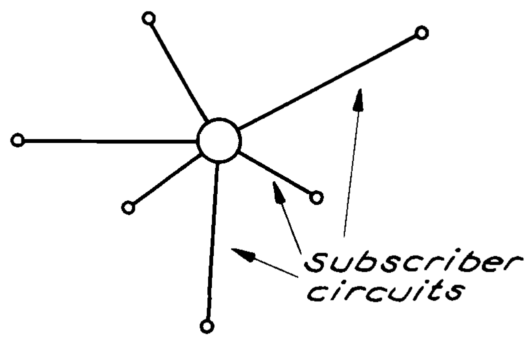

Fig. 2
Network with one exchange

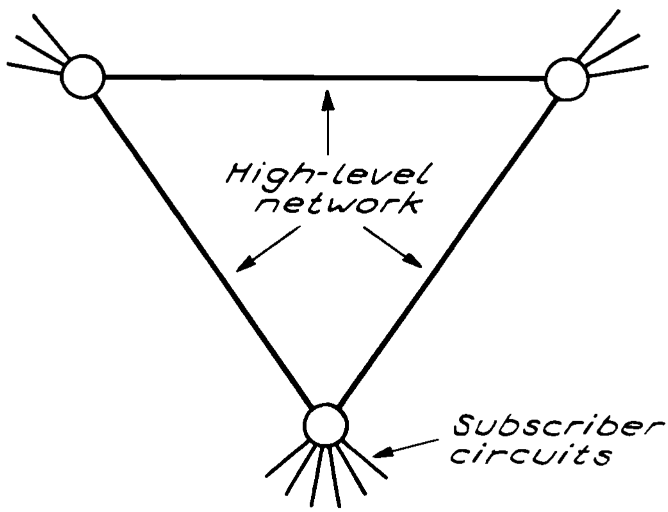

Fig. 3

High-level network with a small number of exchanges

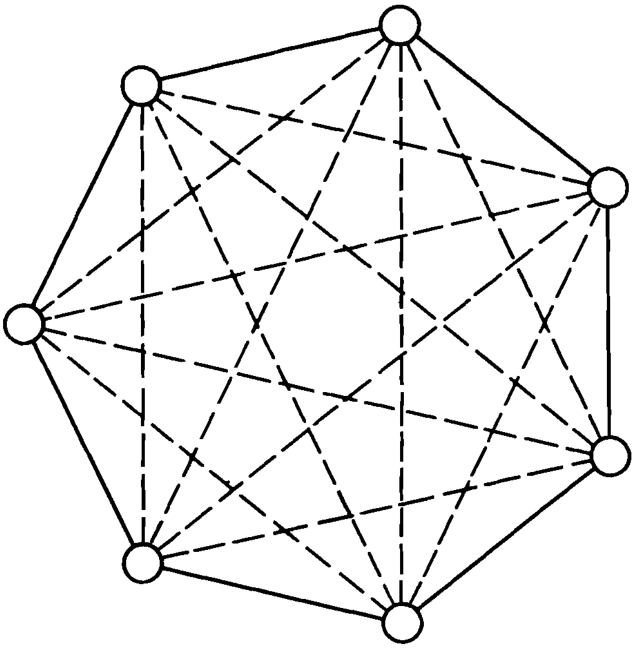

Fig.4

High-level network with a fairly large (4-10) number of exchanges

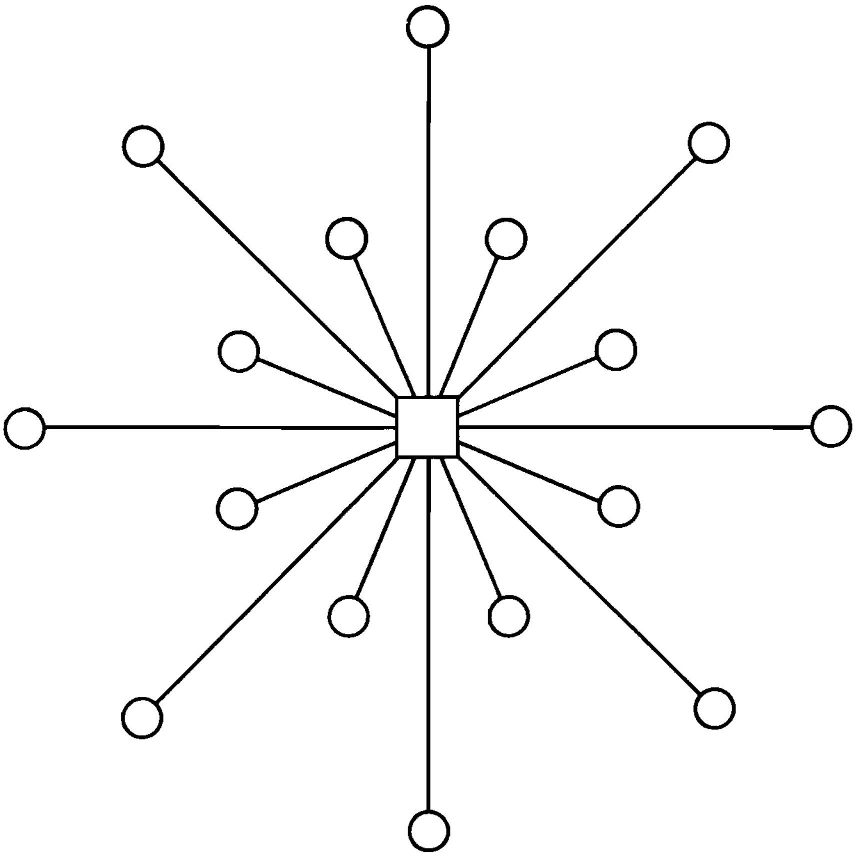

Fig.5

One transit exchange with a large number of local exchanges (2-level system)

None of these extremes was found to be attractive economically. A much better solution would be to install direct connections between those exchanges which have a large volume of traffic with each other and route other traffic through one or more exchanges.

4) When the number of exchanges grows beyond a certain limit, it becomes necessary to introduce a transit level in the switching system structure. Since the data network is expected to remain small for a long time we limited our efforts to the study of the alternative with only one transit exchange. It was assumed that direct connections between exchanges would exist in a few cases where the traffic would be sufficiently large.

From a subscriber point of view a solution involving as few exchanges as possible in a call set-up would be preferable, as this would result in the shortest possible set-up times. This would indicate that alternative 1) would best serve the subscribers' interests. However, with the expected number of subscribers in 1985, that alternative would be difficult to implement because of the very heavy traffic load on the exchange. It would also be unfavorable with regard to reliability, although it would probably be very easy to monitor because of its simple structure. These drawbacks would to some extent be present even in a 2-level network with one transit exchange. In a network with 2 or 3 exchanges the traffic load on any one would be much less although the uneven distribution of subscribers throughout the country would still leave about 12000 subscribers on the Stockholm exchange. The conclusion was therefore that a network with a limited number of exchanges, circa 3-10, would be preferable.

The next step was to study the economic aspects of the various solutions. First the total cost of the network for a varying number and location of exchanges was studied. 15 alternatives were examined, ranging from 1 to 6 exchanges. Because of the expected subscriber distribution, in each case the three first exchanges were located in Stockholm, Gothenburg and Malmö. Figure 6 shows that the largest savings are gained when the number of exchanges is increased from one to three. After that, a further increase will result in rather marginal gains, due to increasingly poor utilization of the equipment in a large part of the network at a fixed blocking percentage.

In the next phase the most suitable geographical locations for the concentrators and multiplexers was examined. EURODATA was used as a basis, which enabled us to estimate the traffic for 272 communities into which Sweden is divided for administrative purposes. Based on the known costs for lines and equipment, and the geographical distances between the community centers, the computer program calculated the most economically favourable locations for the concentrators and multiplexers.

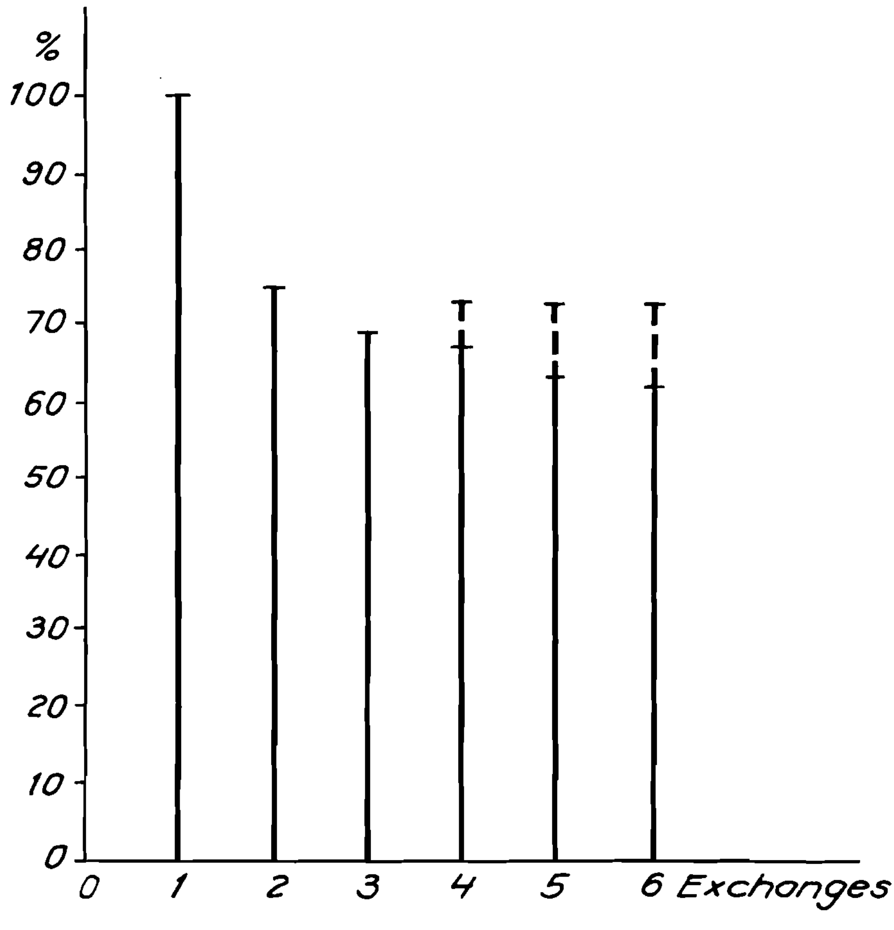

Fig.6
Relative cost of network vs. number of exchanges

The final phase consisted of a more thorough calculation of the costs together with a sensitivity analysis, which investigated how the network cost changed when some important parameters were varied.

The following conclusions were drawn from our study:

1. The most favorable number of exchanges would be in the range of 3 to 6;

2. The savings obtained by introducing more than one level of concentration in the local network are quite insignificant;

3. Reasonable errors in the estimates regarding figures for the data rates and traffic do not appreciably alter the total network cost;

4. The gains in transmission costs that can be made by introducing dynamic allocation are fairly small (about 5 percent). They are probably non-existent when the increased complexity in the switching equipment is taken into consideration;

5. Increasing the blocking percentage from 0.1 percent to 0.5 percent results in a decrease of the total costs of only 1.6 percent.

IV. NORDIC STUDY

The four Nordic Administrations, with a tradition of close cooperation, have carried out a joint study of data networks with the aim of making a specification for a public data communication service between the planned networks in the Nordic countries, in order to facilitate the movement of the large amount of expected intra-Nordic traffic.

The networks planned in the four countries differ somewhat both in service functions offered and in technical solutions. Thus, the joint Nordic specifications are based upon a wide range of ideas and practical experience with data networks and, for the traffic, upon the EURODATA study which gave important traffic forecasts up to 1985.

As for the technical considerations, it was decided that the data network must rely upon using special transmission facilities within the telephone network, i.e. selected pairs in local telephone cables and dedicated channels in the long distance transmission system. Even in 1985 subscribers of the data network will have less than 1 percent of the number of lines which the telephone network will have. This explains why separate transmission facilities are not feasible.

Pulse code modulation systems for speech signals are now being introduced, offering the data network a basic two-way channel capacity of 64K bit/s at the price of less than one telephone channel currently. As the data network is basically a digital service, the technical solutions must be adapted towards the PCM systems.

As for TDM-switching for telephone PCM systems it is expected to be introduced in the 1980's but it will take a long time before such switching systems are distributed country-wide. Because of this, and because the pattern and the functional requirements of data traffic will differ considerably from telephone traffic, it is believed that integration of medium speed data and digital telephone services in the same switching equipment will not be practical.

Three main levels of services may be defined for data transmission networks: (1) transmission, (2) switching and (3) data communications processing (Figure 7).

Transmission is fundamental to all data communications services. Binary data signals will be transferred between two locations with a fixed, small delay in full duplex high quality channels, with bit sequence transparency, and at a given rate (600, 2400, 9600 and possibly 4800 bit/s). Bit timing and (8-bit) byte-timing will be given from network to the data terminals; and timing will be maintained end-to-end throughout the network. The subscribers will have free choice of communications code, procedure and protocol, when utilizing this service level only.

Circuit switching will enable customers at any time to establish or release transmission connections between alternate selected locations. Connecting and releasing delays will be very short, about 100 ms for rates of 2400 bit/s and higher. The subscribers will have to adhere to standardized signalling codes and procedures when requesting the setting up and clearing of a connection, but will have free choice of codes, procedures and protocols after the call has been set up.

Data communications processing includes temporary, random duration storing, sorting, and possibly also simple processing of data transmitted from subscriber into the network. Some possible facilities of this type are temporary storing of messages, code conversion and various packet switched facilities. The subscribers using this service level will have to observe extensive regulations of codes, formats and transmission procedures to be used.

Although a new, simplified interface between data terminals and new synchronous data networks are currently being defined internationally by CCITT in recommendation X.21, the current modem-type interfaces will also be offered.

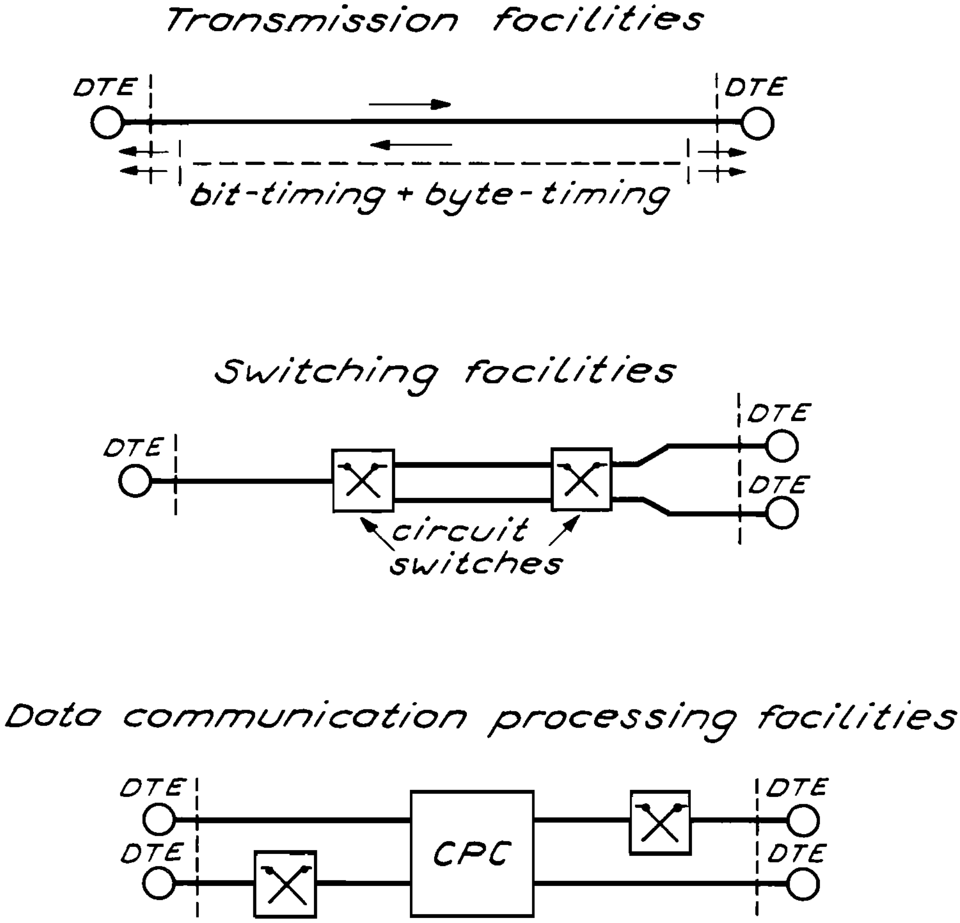
Transmission facilities
DTE
DTE
bit-timing + byte-timing
Switching facilities
DTE
DTE
DTE
circuit
switches
Data communication processing facilities
DTE
DTE
DTE
DTE
CPC

Fig.7

However, the modem interface specifications are not well suited for addressed call establishing procedures. Addressed calling will therefore require manual operation of a special key-set provided as an extra part of the standard network equipment situated on subscribers' premises.

Many users of data transmission today are leasing lines between two or more locations. This facility will be offered also in the Nordic data network. However, the "lines" will still be routed through the switching equipment, and special provision may therefore be made to re-establish automatically a "line" which has been disconnected, e.g. due to a faulty circuit. Furthermore, all such leased lines will then be included in the general network fault control and maintenance system. Thus, specialized private data networks utilizing for example multipoint polling techniques, can be accommodated by the transmission facilities of the new data network without any significant changes to the user's method of operation.

Subscribers currently using the public switched telephone network would experience very few problems in changing over to the data network, the manual calling and automatic answering procedures being basically unaltered. However, the very short call connection and release time, and the possibility of full duplex transmission up to 9600 bits/s will be new to these subscribers.

Figure 8 shows a possible configuration of the high-level network in 1985. Data switching exchanges (DSE) are expected to be situated in Copenhagen (Denmark), Helsinki, Turku and Tampere (Finland), Oslo, Bergen and Trondheim (Norway), and Stockholm, Gothenburg, Malmö and possibly one or two additional cities in Sweden.

In the high-level transmission network between the DSE's, groups of multiplexed (TDM) data channels will be established, each having an aggregated bit rate of 64K bit/s. Several parallel multiplex systems will be installed on each route. Each system will have from 5 to 80 data channels, with data signalling rates of 9600 to 600 bit/s respectively. In principle, different mixes of data channel rates may be found within each system.

Even though PCM transmission systems will be introduced rapidly on short distances, offering the data network cheap 64K bit/s transmission capacity, it is expected that particularly on longer distances, the analogue transmission systems will in most cases be the only alternative for the next 10-15 years. For these distances the data network will use groups of 48 kHz bandwidth in the existing carrier frequency systems, equipped with 64K bit/s digital modems. For the subscribers' lines, double pairs in local telephone cables will normally be used between subscriber and nearest concentrator or multiplexer.

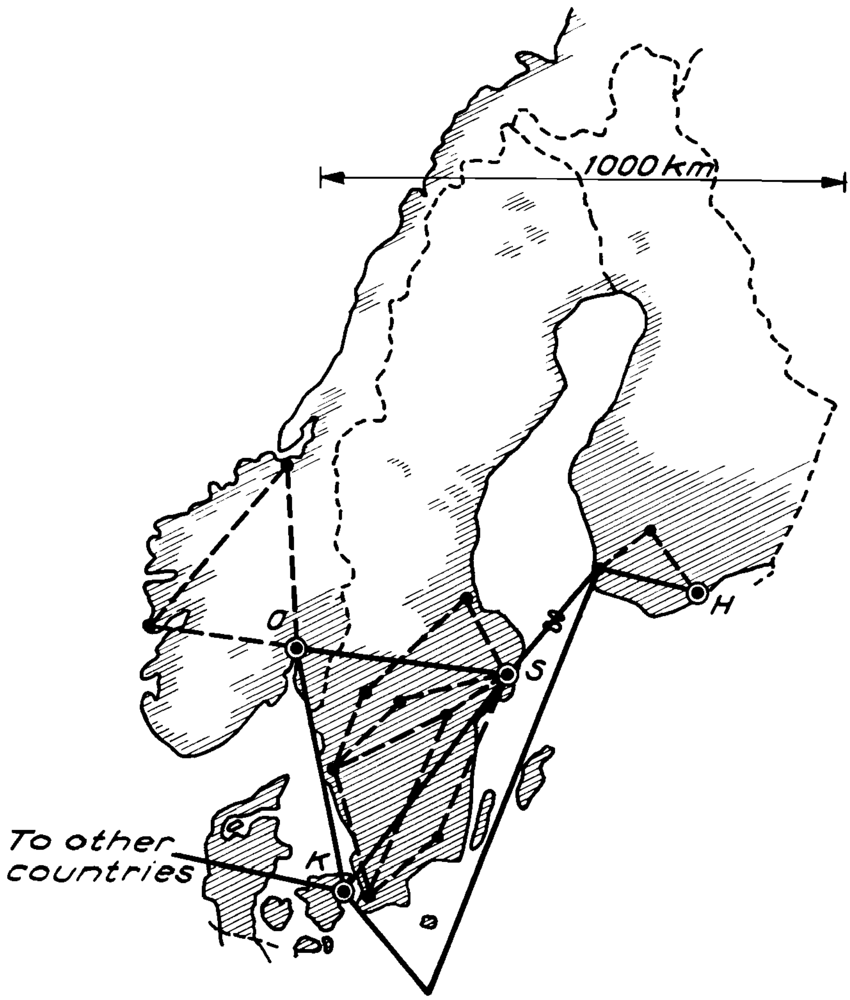

Fig. 8

Possible interchange network configuration 1985

Transmission will be with a digital base-band signalling technique. Several types of codes are currently being field tested in the trial data networks.

Subscribers' data processing centers having multiplexed connections to the network will have 64K bit/s base-band transmission on special cable pairs. Such transmission will be used also when only a short distance (i.e. within a large city) separates concentrator and DSE. The bit-stream from the subscriber will be rearranged into sequences of 8+2 "envelopes." One envelope will consist of 8 databits from the DTE, one synchronization bit (envelope alignment bit) alternating between 1 and 0 and one status-bit to indicate whether the envelope is conveying data or signalling information. The synchronization bit pattern may also be monitored as a transmission pilot signal within the network. The envelope format technique facilitates a uniform logic signal transmission code throughout the entire network, regardless of what transmission system (e.g. multiplexer, local cable lines, telephone channel) is conveying the signal.

The network described is influenced by the conditions in the Nordic countries, i.e. a moderate number of subscribers, a widespread network, and extensive use of analogue transmission facilities for many years to come. The design has been influenced by great requirements of equipment modularity and uniformity to enable easy network growth.

V. CONCLUSION

From a PTT point of view, it is interesting to note that the development of the data network is very different from the development of the telephone network. While the latter grew very gradually with little real long-term planning, the data network has been planned in great detail even before there is any traffic on it. This has made it possible, among other things, to integrate domestic and international traffic from the start, as opposed to the telephone network, where automatic telephone traffic could only be dealt with by the addition of special equipment to the existing system. Hence, use of careful planning and analysis has enabled the system designers to make important progress towards an international network.

References

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DISCUSSION

Question: How was the problem of interfacing the trial network with a packet switching network solved? Was general storage, or packet switching implemented?

Answer: This is solved by an intermediate storage possibility. Circuit switching is well known. If a data switching service is to be implemented, it will have to work without too many initial problems. Experimentation with data switching has only just started, and will continue to be introduced a long time after circuit switching.

Question: Where do the control bits join the message, and how is the modularity problem avoided?

Answer: A stream of bits is buffered by adding two extra bits in the network. The data rate is then reduced to cope with the two extra bits.

Question: There is some confusion as to the time scale of the implementation, and the way in which Sweden proceeded from the current trial to a full system. How and when will Sweden connect with the Nordic System?

Answer: The trial system is in operation for PTT terminals, but we now are working also with some outside connections. Sweden will make use of an interim solution to connect to the remainder. It should be implemented by 1977-78.

Question: Is the trial network free to connect with international networks such as TYMNET, MARK III?

Answer: This is a policy matter. There are no direct answers, and no specific directives to work on it have been given.

Question: Are satellites used?

Answer: Our study on a satellite link showed the following disadvantages: higher delay time, and to repair a satellite or to send a new one would take 2-3 months.

Question: Was Cost 11 associated with the project, and what are the benefits?

Answer: The members are free to participate, and to get information from Cost 11, especially about packet switching, and available software. No decision has been taken whether to join Phase 2, or to establish a node.

A Proposal for Interconnecting Packet Switching Networks*

L. Pouzin

Abstract

Interconnecting packet switching networks may be a puzzle, when they are built as stand alone objects, and mix various other functions with the transport of packets. In working out international standards, agreement will likely be reached over a minimum set of basic characteristics, such as: addressing plan, packet size, header format, and accounting. Packet transport will be the basic service. This allows each network to keep complete autonomy in internal design, technology, and operation. With a hierarchical address space, packets can be shipped across networks at a minimum overhead. Options in header format cater for shorter or longer subscriber addresses, depending on network size. The maximum length of information carried within a packet would be 255 octets. With a limited set of constraints, an aggregate of networks behaves as a single logical network, called CATENET. Interfacing with existing networks usually requires a front end adaptor, or gateway, to wrap CATENET packets into local formats, unless the standard format is accepted internally. Other specific services, such as end-to-end control, are to be stopped at the gateways so that they do not interfere with CATENET workings.

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DISCUSSION

Question: Is network (NW) communication best on a packet switching level?

Answer: Packet switching is easier at present, whereas circuit switching is more difficult. We do not have high speed circuit switching systems. It is necessary to have a virtual circuit at the user level. NW circuits would heed user management to detect errors, and to keep the path open which is now performed by the NW. If the NW crashes constant synchronization is necessary, thus circuits must be managed twice. The only way to use them is to have a connecting interface or map users and ports, because at the application level users do not create circuits. Users' long addresses could create an allocation problem on the NW.

Question: Are mapping tables needed if there are virtual circuits?

Answer: Yes, but they should only work at the source and port, and not need to be made twice. If we had the option, to interconnect ports at the NW level, we need three mapping tables: NW systems, and host level.

Question: If one had a datagram with an address must the destination be selected?

Answer: The NW is concerned only with its own addresses; any special address remains in the text.

Comment: It is rather difficult to argue that packet switching is simpler to implement since many of the actual systems do assign to the ports virtual circuits as part of themselves. It is simpler to connect one port to another port and keep track of the connections, than to try to connect different things together, such as the datagram mechanism to a port. It is arguable both ways and not evident at all, that to have to do it more times is more difficult. The same problem occurs when you want to connect an operating system. You have all the problems associated with connecting two automatic NWs.

Comment: It would be desirable if the CCITT came up with a world-wide definition of virtual circuits in order to create a compatible system.

Question: What progress has been made to attach terminals having their own virtual circuits to a processor? Is an interim stage needed during which such terminals would be handled by the NW?

Answer: The CCITT had compatible schemes, and perhaps a solution exists, but decisions were taken only every four years. They would continue with new developments and NW designs. It might be valuable to implement virtual circuits within the NW, if economically feasible.

Question: The terminals should be handled at NW level. Were the PTTs able to put as many character-handlers on the market as there are terminals?

Answer: Probably not.

Question: Whom would they support?

Answer: If they only supported some, it would be very unfair towards the others, as the large manufacturers would be preferred before the smaller ones.

Question: Are large manufacturers interested in designing for all NWs?

Answer: IBM, General Electric, etc. tend to produce items which require other machines, equipment and terminals which can be supported only by them.

Question: Do users want circuits or terminal handling?

Answer: This is a problem of highly distributed or concentrated intelligence. In virtual circuits the user might or might not want them. In telex machines a virtual circuit is desirable. If you did not have it and a message was broken into packets, you would not know whether you received two mixed messages. Virtual circuits, except periphery ones handle different terminals through different programs, therefore the NW should not be concerned with them. At the periphery you could worry about virtual circuits, and terminals. To solve the problem, standard hardware interface units should be provided for each terminal, operated entirely with microprogram logic. You could then handle different types of terminals. Then, various programs could be used for different terminals with a different kind of program.

Some Conceptual and Technical Aspects of an Integrated System of Scientific and Technical Information Storage and Retrieval

Julius L. Kulikowski*

I. INTRODUCTION

This paper will present a problem of practical importance, which can be solved by modern computer networks.

In 1972 the Ministry of Science, Higher Education and Technology ordered the elaboration of the organization principles of a modern system for storage, retrieval and distribution of scientific, technical and economic information in Poland. This task was given to a commission created for this purpose and this paper is based on the Report presented early in 1974. The system under consideration, called SINTO, is one of four government computer systems which are going to be implemented in the next few years in Poland.

II. PROBLEM FORMULATION

The optimal system is subject to the following restrictions:

- It should be based on the existing traditional system (using almost no computing techniques) of storage, retrieval and distribution of scientific, technical and economic information (INSTE);
- It should be suited to the increasing number of publications for future inclusion in the system;
- It should satisfy the increasing demands of customers of different types and educational levels;
- It should be consistent with international standards and similar (collaborating) systems;
- It should be realistic, from the economic and technical point of view, of the situation which can be forecaste up to approximately 1990.

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Several attempts to solve this were made earlier but were rejected mainly because they were unrealistic. However, this does not mean that the proposals now discussed are optimal.

The situation is complicated additionally by the fact that the basic source documents of the system under consideration, are disseminated to about:

- 55,000 libraries (including scientific, school and public);

- 500 governmental archives;

- 2,000 centers of scientific, technical and economical information (STE). These usually belonging to the institutions.

The libraries, archives and centers of STE information are administratively responsible to different institutions. They are also not autonomous from the point of view of the technical facilities available to them.

The total volume of information stored by the system can be characterized by the following: 40 million volumes stored by the libraries (to be doubled in about 15 years), about 11 million licence descriptions (to be doubled in about 20 years) and about 100,000 standards (to be doubled in about 10 years). The number of legal, and government documents stored in archives is not yet defined.

It becomes clear that in the very near future no solution, which is based on the principle of storing full texts of the source documents in computer memories, will be available. Nevertheless, in particular cases the source documents can be reproduced and distributed on microfiche. The system automation could thus be based on the derivative forms of information and documents such as:

- Bibliographic notes;

- Indexes and catalogues;

- Signal notes, etc.

accompanying the source documents. The derivative forms of information could be extended to:

- Compiling elaborations;

- Factographic data, etc.

However, the two systems managing the source, and the derivative documents should be closely coupled; it would be better to consider them as two sub-systems of an integrated system.

III. ORGANIZATION OF THE SYSTEM

Any document stored by the system can be considered in the following ways:

- Its form (a book paper, standard, legal act, computing program, etc.);
- Its general bibliographical character (the author, year of publication, volume, etc.);
- Its content;
- Place of storage (address in the system);
- Recommendations concerning its distribution in the system (wide distribution, certain class of users, conditionally attainable, secret, etc.).

The general structure of the system should be chosen in such a way that any document, before being delivered to the user, passes through several stages of information processing performed by centers representing different points of view. This is possible
is established. It is hierarchical, consisting of the following functional elements:

- The document-form oriented information centers (DFOIC);
- The document-content-oriented information centers (DCOIC);
- The user-oriented information centers.

The last-mentioned information centers can be sub-divided into two classes:

- The territorial centers: - regional
  - local
- The professional centers: - general (branch centers)
  - institutional.

The general scheme is given in Figure 1. The information flow in the system can be considered for two different situations.

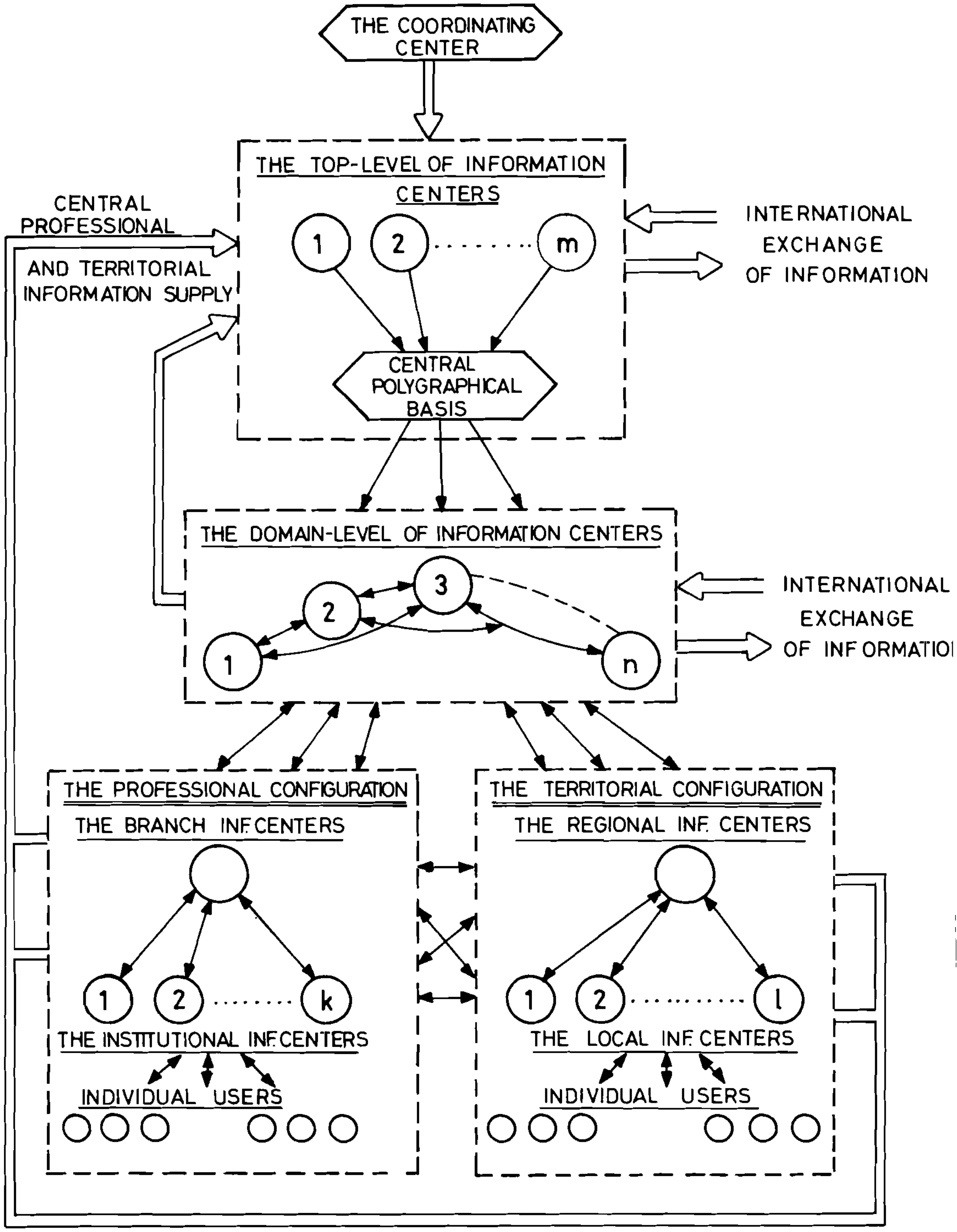

FIG. 1: THE GENERAL ORGANIZATION OF THE PROPOSED SYSTEM S I N T O.

	DFOIC	DCOIC	reg.(gen)	loc.(inst)	user
DFOIC	x	l p g	l r s	l r s	x
DCOIC	x	l p g	l p g	l r s	l r s
reg.(gen)	m r s	l r s	l r s	h p g	m r s
loc.(inst)	l r s	l r s	m r s	l r s	h r s
user	x	l r s	m r s	h r s	x

It is clear that at the higher levels of system organization large volumes of messages of low urgency are dealt with. On the other hand, at the lower levels, the urgent, randomly-called messages of small volume will predominate. A transmission of large records of bibliographic notes will be typical of the exchange between the DFOICs and regional (general) information centers, while short demands and short or medium-size responses (usually urgent for the users) will be typical between users and local or institutional centers.

IV. CONSEQUENCES FOR COMMUNICATION NETWORKS

At first, it may seem that there will be no need for data transmission between the DFOICs. However, this is not so. The DFOICs, according to the general concept of the system, will be responsible for editing the bibliographic records of the documents of fixed forms. The records will be available in print and on magnetic tapes, which can be distributed in the traditional manner (spedition). In both cases, the production process should be automated. As all DFOICs will probably be located in Warsaw it would be reasonable to have a common computer-controlled printing office for all DFOICs. This means that all DFOICs should be connected to a star-like computer network as illustrated in Figure 2. The central computer with printing equipment could be located, for example in the National Library (which will also constitute a DFOIC for historical documents).

The second type of computer network is for direct user servicing. It would be reasonable to concentrate most of the bibliographic records at regional (general) information centers. Therefore, a star configuration is desirable for the sub-systems of high traffic. However, it is hardly possible to install local computer networks of this type everywhere. The traditional communication methods (data spedition), telephone and telex will coexist with computer facilities for many years to come. The reason for this is the great number of territorial and professional information

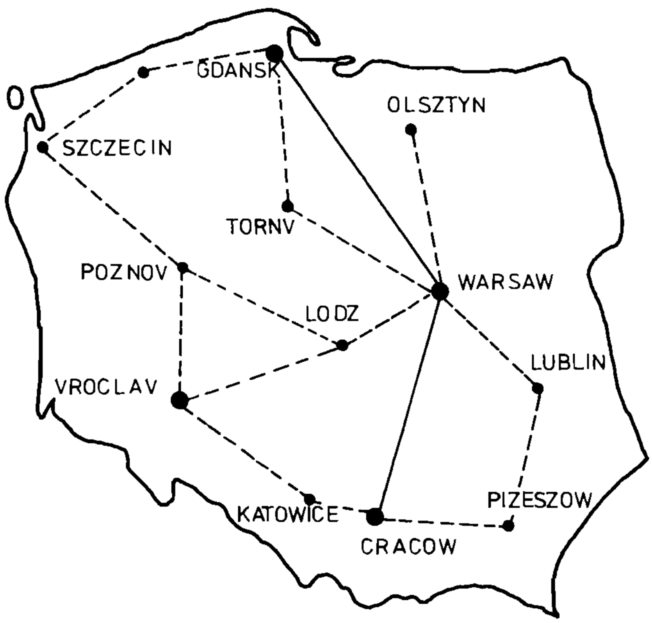

FIG. 2. THE EXPECTED EVOLUTION OF THE SYSTEM SINTO IN POLAND ● - UP TO 1980, • - UP TO 1990.

A. Information Storage

Any source document included in the system is registered by the corresponding DFOIC in the form of its derivative bibliographic note or (for some types of documents such as reports, standards, etc.) is directly stored in the DFOIC and its bibliographic note is produced there. The DFOICs are responsible for periodical editing of these records. They are distributed among the DCOICs.

The DCOICs are scientific information centers located at institutions, universities and industrial centers supplied by specialists in the given fields. The DCOICs will be responsible for content analysis, full indexing and classification for adaptation of foreign (imported) records, for collecting and ordering the records of different documents, and for selective distribution of sub-records collected according to a subject key.

The sub-records, are distributed among the various centers to include them in the user-oriented banks of derivative information (of bibliographic data). Any source document included in the system is thus registered in the form of its derivative note in all professional or territorial centers who decide that it is of interest to their users. It is also registered in one DFOIC and one DCOIC according to its form and content. The redundancy of document registration makes it possible to guarantee delivery of the information concerning any source document, independent of its form, content or system address.

B. Information Retrieval

It is evident that user demands will be concentrated mainly at the lower levels of the system. This should imply the separation of information flows inside the system according to their statistical parameters. In fact, if we take into account the following characteristics of messages transmitted between the different kinds of information centers:

- Urgency (h-high, m-middle, l-low);

- Calls (p-periodical, r-random);

- Volume (g-great, s-small)

then the following matrix of traffic characteristics for different ordered pairs of centers is obtained:

centers, as shown below:

Type of Information Center	Total Number (approximately)
document-form-oriented	10
document-content-oriented	30
territorial:	
- regional	17
- local	35,000
professional:	
- general	180
- institutional	8,000

The total cost of computer installations for all these centers is measured in astronomical numbers. Nevertheless, it was decided to start some experiments concerning computer networks, including their application to information retrieval systems.

An experimental multiaccess conversational system was built up in the Technical University in Wroclaw. The system, called WASC, is a smaller version of a Polish-produced computer ODRA-1325 started this year. A larger version, based on the computer ODRA-1305, will begin next year. Both systems will use the GEORGE-3 operational system (the computers ODRA 1325 and 1305 are compatible with ICL-1900 system). The WASC system is used for management, education (teaching of computer programming and numerical solving of simple mathematical tasks), computer aided design, experimental data collecting and information retrieval. Probably it will be recommended for medium-sized universities. Next year it will be extended to the Medical Academy in Wroclaw. Therefore, in several years time the number of local computer networks will multiply and the problem of their stepwise integration will arise (see Figure 2). The experiments concerning the connection of the ODRA computers through a public switched telephone network have been continuously carried out in Poland for about two years with satisfactory results, and the problem lies in insufficient number of lines rather than in the technical difficulities. However, only simple computer configurations have been proved until now, such as starwise terminals--computer or double computer team configuration. Some experiments have been performed in cooperation with the WASC system of the University of Toronto (Canada) computing center.

Recently a direct satellite connection with the Soviet Union was also established. This provides a new possibility of direct cooperation with the International Information Center (MCI) in Moscow; and the exchange of bibliographical records is possible via the satellite. Some DFOICs are also interested in direct cooperation with other foreign information centers. The records of medical information based on the MEDLARS system could be obtained through Stockholm, and corresponding experiments are being initiated.

V. THE PROBLEM OF DATA SETS ALLOCATION

This presentation would be incomplete if no remarks were made on information allocation. It is clear that the records of bibliographic data can be multiplied and stored in many system centers, and that redundancy reduces data transmission in certain directions. The problem thus arises of what is the best policy for storage of derivative information. Under certain assumptions the problem can be formalized leading to a discrete variable optimization problem with a non-linear cost function and linear constraints. It can be solved exactly for a moderate dimensionality (several dozens of variables) or approximately for a higher dimensionality. The dimensionality is, obviously, connected with the level of aggregation content. However, the real problem is not a particular optimization but an investigation of the relationship between the data base and the traffic parameters in a computer network. We have only a very general idea of this kind of relationship.

The other problem is a formal language for information retrieval in the future integrated system. As there exists a correlation between a record content and its possible address, any demand for bibliographic data will also contain some indications concerning the possible address of the record in the system. In other words, the address is suggested while the derived content of the documents is pointed out. This leads to an information retrieval language which could, at the same time, automatically direct the information demands inside the computer network to a center where a response is possible. The address, thus, should be defined semantically. Until now, if the user's demands could not be satisfied by local records, this will be directed to another center where it can be met. This is a type of disseminated intelligence of a computer network for a special use.

Therefore, it can be concluded that the technical problems of data transmission and those of computing systems application and philosophy are strongly dependent on each other and should not be considered separately.

Poland is very much interested in international cooperation in this field. The Institute for Organization and Control Sciences of the Polish Academy of Sciences, and of the Ministry of Science, Higher Education and Technology, coordinating the investigations concerning computer applications for management, economy and science will strongly collaborate with IIASA in this field.

DISCUSSION

Comment: If there is a large data base with diagrams and charts, most equipment would have no possibility to give information or store it.

Question: Does the computer terminal help to find material, or is the system only an on-line replacement of books?

Answer: The first stage was to establish a few centers with a local network for local users. By 1980 all the networks should be connected, and only then will all the user's demands be satisfied. The integration of a common information language is a problem. Until now it has been effective only in a small area; the addressing is specialized for a particular region. If one has separate semantic codes deriving information through the system then the effectiveness is increased.

Question: Is there one center in Warsaw with several terminals to other centers, provided with concentrators, or are there different computers at different sites connected together?

Answer: There is one Cyber 74 in Warsaw, one machine in Wroclaw and one will be installed at Krakow. Integration of the sub-systems is based on the government communication network. There is a connection between Warsaw, Wroclaw and Krakow, which will later be extended to serve several systems. VASK is one of four systems; the other three are administration, central planning and document statistics.

Question: How many types of terminals can be serviced?

Answer: VASK can serve up to about 30 terminals.

Question: What different kinds of information are delivered and stored?

Answer: We are interested only in books, papers, legal acts, licence descriptions - no managing information will be provided.

A Study of Line Overhead in the ARPANET*

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ABSTRACT

In this paper we consider the form, extent and effect of the line overhead in the ARPANET. We classify this overhead into levels of protocol hierarchy and summarize the characteristics of each level. We then study the line efficiency for various models of system use.

Lastly, we present some measurements of line efficiency for the ARPANET and extrapolate these measurements to anticipate overhead in a heavily loaded network.

I. INTRODUCTION

The choice of control characters and control messages is one of the most important decisions in the design of computer communication networks. The more elaborate the control procedure or protocols, the smoother and safer (supposedly) is the exchange of data in a well-designed system. However, as with sophisticated operating systems, there is a limit to the complexity of the control procedures which is determined by the amount of overhead these procedures introduce. Beyond a certain point, the increase in overhead is too large to justify additional complexity for an intended improvement of service. Therefore it is important to carefully analyze the overhead characteristics of a given system.

It is a well-known fact that the CPU overhead of time-sharing systems depends critically on the behavior of the running programs. Similarly, the line overhead in communication networks depends critically on the characteristics of the exchanged messages. A large number of small messages clearly involves more overhead than a small number of large messages to transfer the same number of data bits. The line capacity and protocol must therefore be selected with respect to the expected characteristics of the message exchange.

In this study, we will focus on the line overhead in packet-switched computer communications networks, using the ARPANET [ROBE 70, CARR 70, HEAR 70, KLEI 70, FRAN 70] as our

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basic model. We define line overhead as all those characters transmitted on lines that are not exchanged between user processes in the attached computer systems (or HOSTs). A user process is here defined to be any process that makes use of the system calls provided by the Network Control Program (or NCP) in each HOST. (The NCP controls the exchange of messages between HOSTs.) With this definition of a user process we exclude from this study some higher level processes that involve types of overhead that are different from those discussed below. For example, a process that controls the transfer of files generates control messages that contribute to the total overhead. However, these higher level control messages cannot be recognized as such by only observing the traffic in the communications subnet. Therefore they have been excluded since our overhead study is based on subnet measurements at the UCLA Network Measurement Center only [KLEI 74].

The discussion will focus on the overhead induced by the subnet protocols and the HOST-to-HOST protocol. The overhead due to the control messages of TELNET (terminal access), FTP (file transfer), and other higher level protocols will be excluded.

A discussion of the effect of flow control on the delay experienced by user data packets is beyond the scope of this paper. This after all is overhead as far as the user is concerned. We will, however, show some effects that the HOST-to-HOST flow control mechanism places on line efficiency (and therefore throughput). We will derive a simple formula for the line efficiency as a function of the traffic characteristics. This will allow us to do a best and worst case overhead analysis. Then a breakdown of the overhead for the current traffic characteristics is presented. The same data is calculated for a saturated net under the assumption that the traffic characteristics do not change.

II. LEVELS OF OVERHEAD

In the following we give a detailed description of the line overhead in the ARPANET. This overhead will be classified according to the following four categories:

1. Level-0 overhead. Control of packet transmission between adjacent IMPs (the communications processors in the ARPANET).

2. Level-1 overhead. Message control in the subnet, i.e. transmission control between source IMP and destination IMP.

3. Level-2 overhead. Message control between HOSTs.

4. Background traffic overhead. Routing messages, line status messages, status reports.

In the following discussion we will only be concerned with line overhead that is either constant or an increasing function of the network load. Thus we will ignore the following two kinds of line overhead that actually decrease with increasing user traffic: (1) "null packets" in which only the acknowledgement bits contain relevant information (these packets are sent only in the absence of other traffic) and (2) excess routing messages which are sent only if the line utilization is less than 80%.

Table 1 gives a detailed explanation of the line overhead on all three levels of communication and for the background traffic. The format of a single-packet user message is further illustrated in Figure 1. As Table 1 and Figure 1 show, there are nine hardware-generated characters for each transmitted packet. In addition, there are 16 bits for the IMP-to-IMP acknowledgment and 16 bits for the software checksum which contribute to the level-0 line overhead.

The level-1 line overhead consists of two parts: 64 bits for the packet header of each user message and 48 bits for each subnet control message. The 64 bits of the packet header are not exclusively used for message control in the subnet. There is, for instance, a field of message identification that is passed unmodified from the source HOST to the destination HOST and might therefore be considered level-2 line overhead. For simplicity, however, all 64 bits of the packet header are counted here as level-1 overhead.

The level-2 line overhead also consists of two parts: 40 bits for the extended leader of HOST-to-HOST protocol and an average number of 93.5 bits for each HOST-to-HOST control message. We will assume that all messages adhere to the official HOST-to-HOST protocol. Although it is known that other private protocols are in use, they represent only a very small fraction of the total traffic. Therefore this assumption appears to be justified.

The background traffic consists mainly of routing messages that are exchanges between any pair of adjacent IMPs at least every 640 msec (per 50 KBPS lines). The I-heard-you messages which are sent to test the status of the phone lines and the status reports which are sent by each IMP every 52.4 sec to the Network Control Center represent a much smaller fraction of the background traffic. Since the background traffic is assumed to be independent of the network load, the level-0 and level-1 overhead of the background messages is included in the line overhead for the background traffic. This distribution of line overhead will later facilitate our calculations.

Level-1 overhead and most of the background traffic overhead is characteristic for a packet-switched communication

CATEGORY	NAME	# BITS	DESCRIPTION
Level-0	SYN	16	2 hardware generated SYN characters for clock synchronization
	DLE/STX DLE/ETX	32	4 hardware generated control characters for message delimiting
	H-checksum	24	Hardware generated checksum
	ACK-header	16	carries acknowledgment bits
	S-checksum	16	Software generated checksum
Level-1	Packet Header	64	Four 16-bit words of packet header in each non-control message
	Subnet Control	48	Three words for each subnet control message (see Table 2)
Level-2	HOST/HOST Protocol	40	Per message overhead specified by the HOST/HOST protocol
	HOST/HOST Control	Average 93.5	Messages of different lengths for control of HOST/HOST traffic (see Table 3)
Background	Routing	1160	Routing message sent every 640 msec (includes level-0 and level-1 overhead)
	IHY	152	I-heard-you message sent every 640 msec to determine up/down status of line (includes level-0 and level-1 overhead)
	Status Reports	1696 (+ 304 for 2 RFNMs)	Status reports (2 packets) sent every 52.4 sec to the Network Control Center (NCC) (includes level-0 and level-1 overhead)

TABLE 1: LINE OVERHEAD CLASSIFICATION

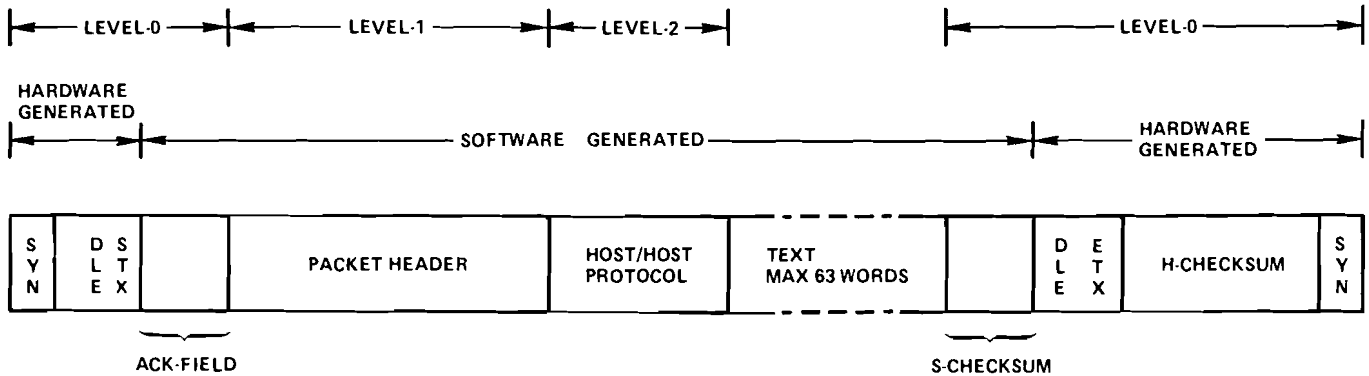

Figure 1. Format of Single-Packet User Message.

network. This type of overhead corresponds to the signalling overhead incurred in a circuit-switched communication network.

III. SUBNET CONTROL MESSAGES

A week-long measurement experiment in May 1974 showed that 49.15% of all packets transmitted in the ARPANET were subnet control messages. For a detailed description of the subnet control procedures the reader is referred to [BBN 69, HEAR 70, MCQU 72]. A list of all the subnet control messages, their frequency of occurrence and their function is shown in Table 2. The relative frequency of these subnet control messages was determined by means of a new measurement feature in the IMPs called "packet tracing" which was suggested by the UCLA-NMC and has recently been implemented by BBN. Using this packet tracing mechanism, about 75,000 subnet control messages were sampled from 35 different IMPs at different times of day and on different days of the week. The data of Table 2 represents the average over all these samples. Though not shown in the table, the deviation of the individual samples from the mean was remarkably small.

As one would expect, the RFNMs for single-packet messages represent, by far, the largest fraction of subnet control messages. It is interesting to note that we never observed a RFNM for a multi-packet message that did not carry a "piggbacked ALLOCATE." This means that there is so much reassembly buffer space available that in almost all cases 8 packets can be allocated for the next transmission within 1 second after the first packet of a multi-packet message has been accepted by the destination HOST.

Table 2 shows that only about 2-3% of all messages are multi-packet messages. This number is about 1% smaller than the percent of multi-packet messages reported in [KLEI 74]. This discrepancy can be explained by the fact that the "incest traffic" [KLEI 74] consists of proportionally more multi-packet messages. Since the RFNMs generated by this incest traffic do not travel over any line, they cannot be observed by means of our packet tracing method.

If our sampling technique were perfect, then the fraction of REQ-ALL messages, the fraction of ALL-M messages, and the fraction of GIVEBACK messages would all be the same. This is because every request will sooner or later be granted and every allocation will, possibly after repeated use, be returned. The amount by which these fractions differ gives an indication of the accuracy of our sampling method. The general conclusion we can draw from this data is that slightly more than 50% of all multi-packet messages which enter the ARPANET do not have to request a buffer allocation at the destination IMP since such an allocation is already waiting at the source IMP to be used. Phrased differently, we can say that slightly

NAME	% OF TOTAL	FUNCTION
RFNM-S	88.77	Sent from destination IMP to source IMP to signal the correct receipt of a single packet message (RFNM = Request-for-Next-Message)
RFNM-M	0.00	Sent from destination IMP to source IMP to signal the correct receipt of a multi-packet message)
REQ-ALL	1.09	Sent from source IMP to destination IMP to request the allocation of 8 buffers for a multi-packet message
ALL-S	3.98	Sent from destination IMP to source IMP to signal the allocation of one buffer for a single-packet message
ALL-M	1.19	Sent from destination IMP to source IMP to signal the allocation of 8 buffers for a multi-packet message
RFNM-ALL	2.35	Combined effect of RFNM-M and ALL-M; the allocation for the next multi-packet message is piggybacked in the RFNM of the previous one
GIVEBACK	1.04	Sent from source IMP to destination IMP to give back an unused buffer allocation that was received via a RFNM-ALL
INCTRANS	1.13	An incomplete transmission message sent from destination IMP to source IMP for each message that could not be delivered correctly to its destination HOST
DESTDEAD	0.46	Sent from destination IMP to source IMP for each message that was sent to a dead destination HOST

TABLE 2: SUBNET CONTROL MESSAGES

less than 50% of all piggbacked ALLOCATES are not used by the source IMP and returned after a time-out of 125 msec to the destination IMP. This means that a much larger fraction of multi-packet messages must wait for the necessary buffer allocation than one would have hoped in order to achieve a high throughput. There are two possible explanations for this behavior: (1) transfer of files which demand a long sequence of multi-packet messages are relatively infrequent, (2) the time-out interval of 125 msec is too small compared to the HOST reaction time.

The large number of ALLOCATEs for single-packet messages is in agreement with the observations reported in [OPDE 74]. Since the IMPs obviously are not short of reassembly buffers all these ALLOCATEs are due to single-packet messages which arrive out of order at their destination IMP. The fraction of ALLOCATEs for single-packet messages has decreased lately since the IMP program has recently been modified in such a way that continued retransmission of single-packet messages is no longer possible.

A suprisingly large fraction of subnet control messages are INCTRANS which signal the source HOST that a message could not be delivered correctly to its destination HOST. The data of Table 2 indicates that, on the average, every hundredth message which enters the ARPANET will not reach its destination. The reason for this undesirable behavior is that many destination HOSTs are tardy in accepting messages. A HOST is declared down by its IMP if a message waits for more than 30 sec on the HOST output queue to be accepted. When this occurs, an INCTRANS control message is returned for every message that is waiting on the HOST output queue. (Future messages which reach the destination IMP after the HOST has already been declared down will generate a "destination dead" control message.) The frequent occurrence of incomplete transmissions is therefore not due to a failure of the subnet but to the unresponsiveness of some of the attached HOST computers.

Compared with the number of incomplete transmissions, the number of destination dead control messages is rather small. These DESTDEAD messages appear to be generated mainly in cases where one HOST wants to find out which other HOSTs are currently responding to net traffic, and thereby sends a "probe" message to dead HOSTs.

IV. HOST-TO-HOST CONTROL MESSAGES

Some 41% of all HOST-to-HOST packets that traverse the network are NCP control commands. Here, we examine the frequency with which each command type is sent, in order to determine what might be done to reduce this type of overhead.

In Table 3 we list the control commands, their length and a short description together with bounds on their frequency of transmission. (For a more detailed description of NCP control commands, see [MCKE 72]).

Most striking among the bounds in the table is the high frequency of the ALL (allocate) command. This phenomenon was reported, for instance, for the HARVARD-10 HOST in [TAFT 74]. We now see that this is a network-wide characteristic. Let us consider the impact of this phenomenon.

As for the using up of network bandwidth, this currently has little effect, since at the present time there is plenty to spare [KLEI 74]. In Section V we show the effects of allocation size on available capacity. If a user message is required to wait in the sending HOST until an ALL arrives from the receiving HOST, the effect would surely be noticeable. This may in fact be the case, and as such would contribute to some excessive delay as seen by users but not attributable to network delay alone. The fact that there are a large number of ALL commands in proportion to data messages indicates that the allocations contained therein are small and are often exhausted by sending a single data message. Note that the allocation size is a variable which depends on the NCP implementation. Another consideration (possibly more important than the two above) is what portion of the HOST I/O and CPU bandwidth is spent in sending these overhead messages. The fewer messages sent and received by the NCP the smaller is the degradation to overall HOST performance. We noticed that even though the HARVARD-10 [TAFT 74] sent a significant number of ALL type commands its data buffer utilization was in the range of 3 to 4% indicating that there is some excess capacity to store more data per connection. So it would appear that the HARVARD-10 in particular could send ALLs containing much larger allocations and thus send many fewer control messages. Perhaps not all HOSTs have more storage to allocate to the NCP input buffers, but it would be well to examine these considerations in detail for each HOST.

Such an examination is often impossible without the aid of an instrumented NCP. There are several arbitrary decisions to be made by NCP implementers. Among these are buffer size connection, fixed or dynamically allocated buffers, maximum number of allowable connections, etc. In each case these arbitrary decisions must be tested to ascertain their validity or to suggest improvements. The above arguments can be made with regard to any software project; indeed to any engineering project at all.

In conclusion, we notice that most NCP control commands sent are of the ALL type. Therefore if one wants to reduce the overhead due to HOST-to-HOST control messages the most effective first step (within the current protocol) is to reduce the number of ALL type messages. So for those HOSTs which can afford to use larger buffers the answer is: send larger allocations!

NAME	LENGTH IN BITS	% OF TOTAL	FUNCTION
RTS	80	3.17-7.53	Sent from receiving HOST to sending HOST to set up a connection
STR	80		Sent from sending HOST to receiving HOST to set up a connection
CLS	72	3.17-7.53	Exchanged between receiving HOST and sending HOST to close a connection
ALL	64	63.82-78.96	Sent from receiving HOST to sending HOST to signal the allocation of message and bit space
GVB	32	0-9.99	Sent from receiving HOST as a request that the sending HOST give back all or part of its current allocation.
RET	64	0-9.99	Sent from sending HOST to receiving HOST to return all or part of its allocation (response to GVB)
INR	16	1.24-7.31	Interrupt command sent from the receiving HOST to the sending HOST
INS	16		Interrupt command sent from the sending HOST to the receiving HOST
ECO	16		Echo command to determine if some other HOST is ready for a network conversation
ERP	16		Echo reply command returns data from the echo command to its sender
RST	8		Reset command for the reinitialization of NCP tables
RRP	8		Reset reply command (response to Reset command)
ERR	max 96		Error command
NOP	8		No operation

TABLE 3: HOST/HOST CONTROL COMMANDS

V. CALCULATED LINE OVERHEAD

Let us derive a simple formula for the line efficiency as a function of the traffic characteristics. The ARPANET HOST-to-HOST protocol provides for a connection-oriented message exchange. Thus, whenever one process decides to send data to another process, a connection must first be set up between these two processes (by means of the HOST-to-HOST control messages RTS and STR). When all the data has been sent, the connection is closed (by means of two CLS control messages). Furthermore, data can be transmitted only after storage has been allocated by the receiver by means of an ALL HOST-to-HOST control message.

Let N be a random variable representing the total number of bits that are to be transmitted and let A (also a random variable among HOSTs) be the number of bits that is allocated per ALL control message by the receiving HOST. Then the number of ALL control messages which must be sent from the receiver to the sender is*

$$a = \lceil N/A \rceil \quad . \tag{1}$$

Define X to be the random number of data bits in a data message. Note that X must be smaller or equal to:

a. N, the total number of transmitted bits,

b. A, the number of allocated bits, and

c. 8023, the maximum number of data bits per message.

Define Y to be the number of packets per message; we have

$$Y = \lceil (X + 40)/1008 \rceil \quad . \tag{2}$$

Define M to be the total number of messages to be transmitted; we have

$$m = \lceil N/X \rceil \quad . \tag{3}$$

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*$\lceil s \rceil$ is the usual ceiling function and is equal to the smallest integer greater than or equal to s.

We denote the mathematical expectation E[ ] by an overbar, thus defining $\bar{N}$, $\bar{A}$, $\bar{a}$, $\bar{Y}$, and $\bar{m}$.

Note that we have ignored all the overhead bits used for message padding. The number of padding bits depends on the word length of the HOST computer. They have only a small effect on our computations below; as a consequence, our results may be viewed as being slightly optimistic.

Table 4 summarizes the line overhead involved in opening and closing a single connection and sending ALL control messages and data messages:

MESSAGE TYPE	NUMBER OF MESSAGES	OVERHEAD PER MESSAGE				TOTAL OVERHEAD PER MESSAGE
		LEVEL-0 OVERHEAD	LEVEL-1 OVERHEAD	LEVEL-2 OVERHEAD	RFNM OVERHEAD	
RTS	1	104	64	40 + 80	152	440
STR	1	104	64	40 + 80	152	440
ALL	$\bar{a}$	104	64	40 + 64	152	424
Data	$\bar{m}$	104Y	64Y	40	152	168Y + 192
CLS	2	104	64	40 + 72	152	432

TABLE 4: LINE OVERHEAD PER CONNECTION

We assume that no HOST-to-HOST control messages are "piggy-backed" together as for example sending the first ALL together with the RTS which is done by several HOSTs. Our measurement data shows that over 80% of all HOST-to-HOST control messages contain only one control command. If HOSTs maximize their message lengths (an assumption we shall make), we have

$$X = \min (N, A, 8023) \quad . \tag{4}$$

We define the average line efficiency E as the ratio of the total average number of data bits to the total average number of data plus overhead bits. Assuming that all the connections in the

ARPANET can be described by the two variables $\bar{N}$ and $\bar{A}$, we make the following simple definition for the average line efficiency (See Table 4):

$$E = \frac{\bar{N}}{\bar{N} + 424\,\bar{a} + (168\bar{Y} + 192)\,\bar{m} + 1744} \quad .$$

Table 5 shows the line efficiency E for selected values of $\bar{N}$ and $\bar{A}$. For $\bar{A} \geq \bar{N}$ only one ALL control message is necessary. Therefore the line efficiency is independent of $\bar{A}$ in this case. Note the low line efficiency for small values of $\bar{N}$. The line efficiency is only 0.32% if connections are used to transmit single characters. Even for large values of $\bar{N}$ the line efficiency is very low if the allocation size $\bar{A}$ is small. This shows what a drastic effect an NCP controlled parameter (A) can have on the efficiency of the communications subnet. A buffer shortage in the HOST computers can therefore directly lead to a decreased line utilization in the subnet. For the transfer of large quantities of data with a sufficiently large allocation size, the average line efficiency can be almost as high as 84%

Since our definition of average line efficiency does not include the background traffic, we must subtract the average bandwidth for the background traffic from the given physical bandwidth before applying the calculated percentages. In the ARPANET, the background traffic is 2.16 KBPS. The best possible bandwidth for process-to-process communication is therefore 83.74% of 47.84 KBPS or 40.07 KBPS. This corresponds to an 80% utilization of the 50 KBPS lines.

$\bar{N}$ BITS \ $\bar{A}$ BITS	8	10^2	10^3	10^4	10^5	10^6
8	0.32	0.32	0.32	0.32	0.32	0.32
10^2	0.83	3.81	3.81	3.81	3.81	3.81
10^3	0.99	9.45	27.06	27.06	27.06	27.06
10^4	1.01	11.09	47.03	65.62	65.62	65.62
10^5	1.01	11.29	50.78	79.40	81.88	81.88
10^6	1.01	11.31	51.18	80.90	83.47	83.74

TABLE 5 : AVERAGE LINE EFFICIENCY E IN PERCENT

Let us consider the case where connections are used for a long interactive use of a HOST computer. In this case the overhead for opening and closing connections can be neglected. The line efficiency is not determined by N_ℓ the total number of transmitted bits, but by the average size $\bar{X}$ of each interactive message. The formula for the average line efficiency can now be simplified

$$E_I = \frac{1}{1 + 424/\bar{A} + (168\bar{Y} + 192)/\bar{X}} \ .$$

Table 6 shows the average line efficiency, E_I, for interactive use as a function of $\bar{X}$ and $\bar{A}$. Part of this table is empty since the average message size $\bar{X}$ can never be larger than the average allocation size $\bar{A}$. We again notice the decreased line efficiency for small values of $\bar{A}$. However, even for $\bar{A} = \infty$ the line efficiency is only 2.17% if the messages are sent one character at a time, as is frequently done in the ARPANET. It now becomes obvious that it is very important for the designers of higher-level protocols to understand the overhead characteristics of the underlying communications systems. Protocols that are designed without such an understanding may present numerous problems when they are used.

$\bar{A}$ BITS / $\bar{X}$ BITS	8	100	1000	∞
8	1.01	1.99	2.15	2.17
40		7.02	9.59	10.00
100		11.31	19.90	21.74
200			31.02	35.71
500			46.64	58.14
1000			51.23	65.45
2000				74.18
5000				82.89
8023				83.93

TABLE 6: AVERAGE LINE EFFICIENCY E_I IN PERCENT

CATEGORY	NAME	LINE OVERHEAD		% OF LINE CAPACITY	
		BIT/SEC	SUM	%	SUM
Level-0	SYN	68.32	444.08	0.14	0.89
	STX/ETX	136.64		0.27	
	H-Checksum	102.48		0.20	
	ACK-header	68.32		0.14	
	S-Checksum	68.32		0.14	
Level-1	Packet Header	138.88	239.68	0.28	0.48
	Subnet Control	100.80		0.20	
Level-2	HOST/HOST Protocol	75.53	158.72	0.15	0.32
	HOST/HOST Control	83.19		0.17	
Background	Routing Messages	1812.50	2157.52	3.63	4.32
	IHY	237.50		0.48	
	Status Reports	107.52		0.22	
Data	(Non-Overhead Bits)	295.56	295.56	0.59	0.59
		TOTAL SUM:	3295.56		6.59

TABLE 7: LINE OVERHEAD IN THE ARPANET (MAY 1974)

VI. MEASURED AND PROJECTED LINE OVERHEAD

As we have seen, the line efficiency in the ARPANET lies somewhere in the wide range between less than 1% and almost 80%. Let us now turn to measurement results that will allow us to calculate the current line efficiency. These refer to the ARPANET as of May 1974 with 46 IMPs and 51 full duplex channels. To simplify matters, we make the following additional assumptions:

a. All lines have the same speed (50 KBPS);

b. All IMPs and lines are up;

c. The overhead for status reports can be equally allocated to all lines.

Table 7 gives a breakdown of all the bits transmitted per second in the ARPANET according to the line overhead classification of Table 1. These numbers represent an average over all 102 simplex lines. The contributions of the background traffic to the total traffic can be directly derived from Table 1. For the status reports we assumed that they are, on the average, sent over 6.25 hops before they reach the Network Control Center. (This number was computed for the topology of the ARPANET in May 1974.) The average number of packets per second per channel was measured to be 4.27 pkt/sec (excluding status reports). From this we easily derive the level-0 line overhead. The fact that 49.15% of all transmitted packets represent subnet control messages allows us to determine the level-1 line overhead. The average number of bits per second per channel, excluding level-0 and level-1 line overhead and background traffic, was measured to be 454.28 bits/sec. 87.02% of all packets are the first packet of a message and therefore carry the additional 40 bits of HOST-to-HOST protocol overhead. As previously stated, 41 of all packets exhanged between HOSTs are HOST-to-HOST control messages with an average length of 93.5 bits (excluding the 40 bits of HOST-to-HOST overhead). These numbers allow us to determine the level-2 line overhead and from this we can determine the level-2 line overhead and from this we can determine the number of data bits exchanged between processes.

As can be seen from Table 7, most of the traffic currently being carried by the ARPANET is background traffic. A large percentage of the background traffic is due to routing messages. Most of the line overhead is incurred on level-0. The number of data bits per second is only about one half of one percent of the line capacity. The line utilization including all types of overhead is 6.59%. (As mentioned before, this does not include the extra routing messages that are sent when the line utilization is low).

Because of the low line utilization, some of these numbers might be misleading. Therefore let us try to assess the effect of increasing the load on the subnet. While the background traffic is held constant, we will assume that the level-0, level-1, and level-2 line overhead as well as the data bits are increased proportionally until the line utilization is 100%. This way we obtain an estimate for the overhead characteristics in a saturated net, the traffic characteristics being unchanged. The result of this traffic projection is displayed in Table 8.

Table 8 shows that the background traffic is only a very small part of the total traffic in a saturated net. It is interesting to note that about 37% of all transmitted characters are now due to IMP-to-IMP (level-0) transmission control. Because the delay increases indefinitely as the net saturates, the best line efficiency one can hope to achieve is about 20% (a conservative estimate of the 24.85% shown). This, of course, is an average number. In particular cases one may get a far better line utilization. However, if the overall traffic characteristics remain constant, not more than 10 of the 50 KBPS will, on the average, be available for process/process communication. Note also that the background traffic which currently represents more than 65% of all the traffic becomes almost negligible as the net saturates.

VII. CONCLUSIONS

It has been argued that the method of interprocess communication should be independent of whether or not the two communicating processes are running on the same HOST computer system [MCKE 72]. From a logical point of view this may be the right approach. As far as the efficient use of resources is concerned, such an approach may have disastrous results. In this sense the network is not transparent to interprocess communication. The HOSTs must rather be aware of the fact that the allocation of network resources requires the same care as the allocation of any other resource. It appears that in some cases the freedom which the ARPANET protocols provide its implementers has been misused. In order to reduce the overhead, much more thought must be spent on the efficient implementation and use of network protocols, rather than only on their feasibility.

CATEGORY	NAME	LINE OVERHEAD		% OF LINE CAPACITY	
		BIT/SEC	SUM	%	SUM
Level-0	SYN	2872.13	18668.84	5.74	37.34
	STX/ETX	5744.26		11.49	
	H-Checksum	4308.19		8.62	
	ACK-Header	2872.13		5.74	
	S-Checksum	2872.13		5.74	
Level-1	Packet Header	5838.43	10076.00	11.68	20.15
	Subnet Control	4237.57		8.48	
Level-2	HOST/HOST Protocol	3175.23	6672.49	6.35	13.34
	HOST/HOST Control	3497.25		6.99	
Background	Routing Messages	1812.50	2157.52	3.63	4.32
	IHY	237.50		0.48	
	Status Reports	107.52		0.22	
Data	(Non-Overhead Bits)	12425.15	12425.15	24.85	24.85
		TOTAL SUM:	50000.00		100.00

TABLE 8: PROJECTED LINE OVERHEAD

Although the overhead can be decreased, the designers of computer networks must realize that a significant percentage of the line utilization will always be needed for control information. The exact amount of the overhead depends critically on the type of traffic (or traffic mix) the network is intended to carry. Only a careful study will reveal what part of the physical bandwidth is actually available for user-process to user-process communication. For the ARPANET we have shown that the average line efficiency can be as low as 1% for single character traffic and as high as 80% for efficient file transfers. Assuming that the traffic characteristics remain unchanged, we were also able to show that the ARPANET is able to support little more than 10 KBPS user-to-user traffic on its 50 KBPS lines. In view of these results we hope that the designed protocols will in the future take more account of the effect of overhead on the user-to-user throughput and thereby improve the network performance.

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DISCUSSION

Question: How do the costs presented apply to the slower lines on ARPANET?

Answer: At low speed, overhead information costs are high. 38% of the through-put is used for overhead when there is little traffic. Referring to Mr. Haug's remarks earlier, a satellite link would not increase the costs. It has recently been observed that deadlocks do occur, although it is not designed to let this happen. Consequently, the ARPANET does not have the expected efficiency.

Comment: Systems are often badly designed because the major concern is the local logic part. This did not fail but the whole system did.

Comment: We considered local optimization, but had worried about special things. The scene was, therefore, disasterous.

Question: What kind of interactive connections are being considered, and what are the conditions for speech transmission now?

Answer: There are long delays. However, referring to the simplest kind of interactive traffic: keyboard-computer-traffic, very long delays for characters printed on a screen or a paper cannot be sustained. Also, interactive response to small requests should be fast in a time-sharing system. The designers could not sustain more than 0.5 sec. average delay time for interactive messages. NW real time is difficult to control for the delivery of speech traffic.

Question: Why were the high measurement costs not analyzed earlier?

Answer: There was no way to analyze the deadlocks,or to find them. The simulation would have contained many compromises, and would have caused too much work. At first, one should have written the whole simulation. Now the costs of measurement are not too high. Simulations are useful for calculations on delay, throughput and hardware failures but not for overload situation and deadlocks. Good measurements are now being entered into the NW, and they are paying off.

Question: How important is the multi-packet facility; and does it need a lot of control and reassembling?

Answer: Multi-packet transmission is not commonly used. Control and reassembling costs are enormous. One could eliminate the multi-packets and introduce only two sizes. One size alone causes buffer allocation problems. Reassembling and sequencing are also problems. The use of multi-packets depends on the application.

Comment: It is obvious that resequencing and reassembling must be done, but why in a communications NW where there are not only few buffers, but also other traffic? If a user wants to carry many simultaneous streams, let him handle it in his own code, and provide as much buffer as needed. A small user, with few connections can predict the amount of traffic and buffers needed; he is in a much better position than the NW to allocate the buffers in a sensible way. Multi-packet messages can be solved with a fountain processor, it is just a matter of having a system better organized than the current one.

Some Aspects of Networking Economics in Relation to the Dissemination of Scientific and Technical Information

J. Page

I. BACKGROUND AND INTRODUCTION

The dissemination of scientific and technical information (STI) is now a vital part of research, development and planning activities, not only in research but also in areas in which scientific discovery and applications of technology become intertwined with vital public policy. G. Anderla [1] analyzed the present situation and projected it, to produce scenarios for the next decades. He concluded that the so-called information explosion would be likely to continue during this period. In order to meet the access needs, and to transfer this increasing body of STI to integrate its traditional forms with actual data, and to produce the basic inter-disciplinary attack on planning problems, interactive networking has been suggested as one possible means.

Forecasts of traffic over computer to computer networks indicate very large increases, particularly in the commercial data processing area. Although STI will always form a rather small proportion of this total traffic, it is suggested that its needs cannot be ignored in network design. Search costs play a large part in STI networking: most clients of such services are libraries, research departments, etc. with limited budgets, and, the acceptability of STI services is limited by costs. Although some government agencies do not pass on charges to users, the great majority of such services operate with a cost recovery policy. Hence, at the policy level, the question of growth of these services has become less a technical problem than a cost problem. Thus economic parameters in network development have caused more concern than in other areas of computer networking. Many commercial services for banking, airline reservations, investment, etc. have been privately financed, and no doubt cost/benefit considerations have been taken into account. However, for scientific computation, networks have been experimental and form major research projects which up to now have been required to be funded by actual users.

II. NATURE OF STI SYSTEMS

The design of interactive STI systems will not be discussed in detail, but some of the features by which they differ from the more general form of computer time-sharing services will be noted. Interactive STI programs require more core storage; data files are large and expensive to maintain on the central system since many must be stored on fast-access devices. A typical on-line storage requirement (fast and slow access) is 2.4×10^9 bytes. D.B. McCarn and J. Leiter [2] estimated that the average per hour cost of connect time for retrieval services might be a factor of 3 or 4 above normal time-sharing costs.

STI systems vary considerably in flexibility and degreee of interaction. A system designed for use on video display terminals operating at 2.4 to 4.8 Kb/s will involve circa fifty interactions, each based on short commands from the terminals, perhaps interspersed with data displays from the computer occupying a full screen. For a 30 minutes search, circa 25,000 characters will be transmitted by the network.

III. THE DATA BASES

In 1973 the Science Information Association (SIA) established that there were about 100 commercial data bases in machine-readable form in the USA [5] Fourteen of the data bases are from the following organizations. (Growth rates are given in the parenthesis as thousands of items per year):

- Chemical Abstracts Service (Chemical Abstracts Condensates) (360)

- Biosciences Information Service of Biological Abstracts (BA Previews) (240)

- National Technical Information Service (56)

- Science Information Exchange (30)

- Engineering Index (COMPENDEX) (85)

- American Institute of Physics (Searchable Physics Information Notes, SPIN) (30)

- National Agricultural Library (Cataloging and Indexing Systems) (120)

- National Library of Medicine (MEDLARS) (150)

- National Library of Medicine (TOXICON) (50)

- The Institution of Electrical Engineers (INSPEC, Information Service in Physics, Electrotechnology and Control) (150)

- Pandex (250)

- Educational Resources Information Center (ERIC) (27)
- Excerpta Medica (250)
- Institute for Scientific Information (ISI) (374).

Thirteen bases show an annual increase of approximately 1.8 million records. The fourteenth, ISI, increases annually by about 374,000 more records (4 million citations). The remaining bases increase by about one million records annually. According to SIA, the annual rate of increase for all commercially available data bases is well over 3 million records. In numbers of characters, this could amount to about 10^9, but with 30 to 50 percent duplication. This estimate does not include limited clientele data bases of US government agencies, such as the Defence Documentation Center, the Environmental Protection Agency, the National Aeronautics and Space Administration, etc.

In 1972 in the UK, there were 35 information services offering English language information retrieval services. Many data bases were identical with those listed above, but the UK list also included the NASA file from ESRO/SDS, INIS, etc.

Previously these data bases have been interrogated by batch processing methods, however, this has become increasingly ineffective especially retrospective searches, i.e. searches of an entire data base by subject. However carefully the search logic is constructed, the amount of "noise" generated by a Boolean search increase linearly with the size of the file. Moreover, computer time, and therefore cost, increases almost linearly with the file size, and it is not uncommon to find a retrospective search of a large file costing several hundreds of dollars. Thus, at present, batch searching is carried out on file updates, to produce so-called current awarenesss services. Since the introduction of interactive file searching techniques, and the simultaneous growth of networking possibilities, the economies stemming from simultaneous searching by many terminals of a single large data base have permitted per search cost decrease by at least an order of magnitude.

The interim results presented here are part of a study which will treat the dissemination of STI as a system in itself, operated mainly by networks. The full study will take into account not only the relative economics of STI in a networking environment, compared with other possibilities, but will also examine user-system interfaces, particularly the demography in distribution of the user population, and the most cost effective networking methods used to reach a large proportion of it. An important interface is the optimum interaction betwen the user and the STI system. Obviously, from both the networking and STI systems aspects, there are numerous trade-offs between technical sophistication and cost which are important for future STI policy implementation. This paper describes the initial work on some of these trade-offs.

IV. A SAMPLING OF STI NETWORK OPERATIONS

The cost of a retrospective search of a large STI file can vary widely--$100-$150 is not uncommon. For interactive networking, data is available which relates per search cost to the total volume of traffic. Three different STI network operations are compared to derive indications of the economies of scale currently being achieved.

V. SWEDISH MEDLINE

This is a simple star network using mainly dial-up slow-speed terminals, but is being integrated with a more extensive semi-distributed network for interlibrary operations (LIBRIS) still under development. Details of LIBRIS and Swedish MEDLINE are given in Annex I. Central system costs are given below. They do not take into account terminal or dial-up costs:

Volume	Cost per Search ($)
300 monthly searches	22
700 monthly searches	11

At the 700 search level, the network is operating at about 36% of its capacity.

VI. EUROPEAN SPACE RESEARCH ORGANIZATION, SPACE DOCUMENTATION SERVICE (ESRO/SDS)

This is a fairly extensive leased line network of developed star configuration operating in Western Europe. Twenty high-speed terminals are connected to a single host computer. The network is described in Annex II. At an early developmental stage, with about half the present terminals in a simple star configuration, cost per search was $112, for about 4,000 annual searches. In the first half of 1974, the network had doubled in size (both geographically and in number of terminals), and employs concentrator and multiplexing devices. Per search cost had been reduced to $17 for a volume of 20,000 annual questions. It is difficult to estimate the unused capacity in the present system. Since the network has operated in its present configuration at a uniform speed of 2.4 Kb/s without a concentrator, multiplexing two channels over the final link from communication to computer center as this was achieved with an average response time no greater than 4 seconds, it would not be unreasonable to infer that there is probably 50% spare capacity in the present configuration.

VII. US MEDLINE OVER TYMNET

TYMNET is a commercial computer network extending from Hawaii and the continental US to Paris. It is essentially a high volume operation with low access speed (300 b/s): daily traffic estimates amount to 110 million characters. It is however, circuit switched, the circuit from the terminal to its host is established by the network controller on log-in, and then remains unchanged, unless line failure occurs.* It is one of the few commercial networks to carry a large variety of interactive STI services, including a number of data bases offered by information brokers, such as Systems Development Corporation (SDC), and Lockheed Information Systems. Since the end of 1971 TYMNET has been the main distributor of the MEDLINE medical information system stored on host computers at the National Library of Medicine, Bethesda Md. Costs for the MEDLARS batch processing system which preceded MEDLINE are difficult to estimate since the service was carried by as many as 20 different computers in the USA and other countries. However $50 to $100 was probably the mean cost (see [2]). After one year, the service was supporting an average of 25 simultaneous users, 43 hours per week, and by October 1972 the number of searches had risen to an annual rate of 140,000. Although no direct statistics are available in [2] for later periods, it now appears that the annual volume is at least 400,000. 1973 figures have been computed based on the cost to NLM for providing a search. These include central system costs, payments to the network, etc. In 1973 these costs amounted to $4: the users' costs will vary according to the method of connection used, but in the USA this could be about $5 per search.

VIII. ECONOMIES OF SCALE

An obvious consequence of networking is that central system costs may be spread over a large number of users, and therefore economies of scale are expected to the extent the network costs per search are not large enough to absorb these benefits. However, it is interesting that in all three cases, with very different networking arrangements, these economies are realised, although at different levels. Generally, the extent to which search costs may be reduced

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*A packet-switched version of TYMNET is under development will depend upon the breakdown of the cost elements. These will normally include the following costs: file preparation or acquisition, inversion, maintenance and updating, computer (program residence and CPU time), general system overheads, and network. While the major study is concerned with the variation of all these costs, in this paper only those costs attributable to the network itself are discussed.

Search modes, and therefore search times, differ widely: for example, for ESRO/SDS the average time is 30 minutes, whereas for US MEDLINE, it is 12 minutes. TYMNET charges amount to $10 per hour in the USA, according to the schedule issued by SDC for the STI services offered: therefore network charges per notional half hour search should amount to $5. However, this is not entirely consistent with the NLM computation, since their estimate of half hour cost connect time is about $3.50. For ESRO/SDS, line cost plus hardware costs in the network may be assessed at approximately $300,000 per year; most of this is represented by line costs. On a 20 terminal network basis the average annual utilization rate per terminal is 1,000 hours: thus, the cost of network services per hour, per terminal is about $15, or $7.50 per notional half hour. Obviously, these relative costs are affected by a number of factors, including the relationship of network sophistication (roughly measured by the cost and number of mini-computer nodes) to the overall volume of traffic.

A difference in per search costs could also be attributed to variations of spare capacity in the two networks. This is difficult to quantify with present knowledge. However, one important element in these cost differentials is the difference in line costs in Europe and the USA. Network planners must take this into account in their economic projections. The available data is examined in the next section.

IX. LINE COSTS

At present, when there is so much international commercial business relying on internationally leased line networks one could expect that it would be simple to compute line charges for planned or hypothetical networks. This however is not so, and it is very difficult to disentangle the charging policies of the various European PTTs, especially when planning an international network. The following summary is based on non-PTT sources.

X. BASIS FOR EUROPEAN CHARGING SYSTEMS

In 1972, J.G. Thompson reported that the current charging system for international leased circuits was composed of three elements:

1. A fixed figure per kilometer to cover the distance between the international terminal cities in the countries concerned;

2. A fixed figure for the international terminal equipment involved;

3. A variable figure to cover the cost from an international terminal in the country to the actual line termination.

These three figures are added together to form a notional per minute telephone cost. Various multipliers are then applied to this figure depending on the type of circuit: e.g. for an ordinary quality circuit the multiplier was then 7,500 minutes. The monthly circuit charges thus calculated are further multiplied by a factor which may vary between 0.5 and 1.8. Thompson stated that new charging arrangements were to be introduced but the basic method and the end results would remain substantially the same.

Apparently this is presently under study by the CCITT. From submissions made to them by users such as the International Press Telecommunication Council (IPTC), it has become apparent from the user's point of view, that charging basis is defective and that monthly rental costs are excessive. The reason for these charges is if the PTTs want to be compensated for the loss of income obtained from a public telephone circuit taken out of service, then the multiplier factor (minutes per month) should be related to the average number of minutes of normal telephone conversations on international circuits. The IPTC stated in March 1974 that this would be circa 4,600 minutes per month. They also pointed out that one method of computing rental charges recommended in D 2 of the CCITT resulted in a cost nearly six times higher than that for a public telephone circuit in the automatic or semi-automatic service. It is difficult to determine how international leased circuits charges are actually established, but it is hoped that this can be illuminated in the future phases of the study.

XI. INTERNATIONAL PRESS TELECOMMUNICATION COUNCIL (IPTC) ESTIMATES

In a document submitted to the Study Group III of the CCITT in January 1974, the Council gave a comparative table showing international leased circuit costs compared with the national circuits of about the same length. These leased line circuits were taken at random in September 1973 (see Table 1).

TABLE 1. European Leased-Line Circuit Costs

NATIONAL CIRCUITS

Route	Distance in KMs	Approx. Monthly Rental in Gold Francs per KM
London-Manchester	368	2.9
Paris-Arles	601	3.5
Brussels-Arlon	160	4.6
Milan-Rome	498	6.0
Cologne-Munich	446	6.2
Copenhagen-Esjberg	262	3.6
Amsterdam-Maastricht	168	2.5

INTERNATIONAL CIRCUITS

Route	Distance in KMs	Approx. Monthly Rental in Gold Francs per KM
London-Amsterdam	368	10.9
Paris-Milan	618	13.1
Brussels-Cologne	185	29.2
Milan-Frankfurt	498	19.1
Cologne-Amsterdam	216	20.3
Copenhagen-Hamburg	278	19.5
Amsterdam-Düsseldorf	176	24.8

In interpreting Table 1 the Council summarizes their position as follows:

> "This table reveals that a very considerable premium is incurred whenever a leased circuit crosses a frontier and this is particularly difficult to understand when each international circuit only involves an Administration in the expense of maintaining one terminal instead of two.
>
> Recommendation D 2 stipulates that rentals should be related to a prescribed number of telephone call minutes per month--as if the

cost of a leased circuit and of providing a telephone service were in any way comparable! It also permits each Administration to multiply its terminal charges by a coefficient not in excess of 1.8 'to adapt the rentals to the national levels of tariffs where necessary.' Furthermore, it allows opportunities for local extensions to be priced at tariffs which may exceed those for domestic circuits of the same length.

Besides, the researches of the Teurem Group also show that the cost of providing an international leased circuit will be proportionally less the longer its distance may be, yet the foregoing table shows how international circuit rentals are sometimes assessed without regard to this basic principle.

It is clear from the table--and our examples have been taken at random--that the relationship between the per kilometer rental for an international as opposed to a national circuit is seldom, if ever, less than 3 and sometimes as high as 10."

XII. COMPARISON WITH US RATES

It is important to compare the European situation with the US situation where for the past few years, commercial networks have been in operation. The American charges are based on two simple factors: the inter-city mileage on a sliding scale, the cost per mile decreasing with distance, and a fixed terminal charge. Circuit costs are based on the actual service cost plus a return on invested capital, the latter being subject to control by the FCC. Thompson gives two US rates, that charged by AT&T (the Bell System) the common carrier responsible for long distance telephone facilities, and the rates proposed by Microwave Communications Inc. (MCI). This company, at the time the paper was written, had been authorized by the FCC as a specialized common carrier.

The results of the comparison are set out in Table 2. In order to more easily associate these results with those discussed above, miles have been converted into kilometers and their cost has been computed. This is given both in Gold Francs (GF) and in Dollars, based on a rate of 2.8143 GF per Dollar.

TABLE 2. Comparison of Leased Line Charges (July 1972)

European and US Rates

(ordinary quality circuits)

From	To	Distance KM	European Rates			Bell System			M C I		
			Total GF	Per KM GF	Per KM $	Total GF	Per KM GF	Per KM $	Total GF	Per KM GF	Per KM $
London	Paris	365	5700	15.78	5.61	1402	3.84	1.36	690	1.89	0.67
Amsterdam	Frankfurt	365	6377	17.47	6.21	1402	3.84	1.36	690	1.89	0.67
London	Madrid	1246	16603	13.33	4.74	2941	2.36	0.84	2229	1.59	0.56
Paris	Brussels	274	4912	17.93	6.37	1134	4.14	1.47	529	1.93	0.69

When comparing the USA and European tariffs, one should mention that in a purely commercial environment the AT&T rates are almost twice as high as necessary.* The proponents of satellite communications for voice and data traffic have argued that for busy long-distance routes the cost should be considerably cheaper than undersea cable or land line/ microwave connections. Unfortunately, these predictions, while apparently well founded, are not substantiated in the charging policy of organizations operating communications satellites. Their rates seem to be geared to the current rates of the common carriers/PTTs for non-satellite communications between the same points. However, following the "open skies" policy announced by the US government in which private organizations are allowed to submit proposals for domestic satellites the rates should be lowered. Telecommunications Reports of 20 June 1974 announced that Western Union has filed an application with the FCC for a domestic communications satellite system. The tariff given in this application quoted a monthly rental for a private voice grade, two-way line between New York and Los Angeles at $1,000 as against $2,100 normal AT&T rates, i.e. $0.21 as against $0.45 per kilometer.

It should be emphasized that all previous discussion has been about the cost of leasing voice grade lines. Such lines, if conditioned to M102 standards, can in principle operate up to about 9.6 Kb/s. If however, a higher speed network is constructed (e.g. 48 Kb/s) then it is necessary to group some 12 voice grade channels together and transmit them via frequency division multiplexing (FDM). This gives a so-called groupband circuit and in Europe costs about eight times as much as a normal conditioned voice circuit.

XIII. ACCESS MODE--LEASED VERSUS DIALLED CONNECTIONS

This is important for the future development of STI networks. There are a number of trade-offs involved: adding a dial-in capacity at nodes of a dedicated STI network gives access possibilities to a wider user population, some of whom may be relatively low volume users not able to amortise the cost of high speed video terminal over a large number of searches. As is indicated by the NLM experience, dramatic increases in the total volume of use may be obtained by providing relatively cheap terminal dial-in facilities. On the other hand, with large files and the possibility of switching searches between them to obtain the greatest subject coverage, the maximum degree of interaction is necessary

*It should however be pointed out that the AT&T tariffs are universal, i.e. they apply over both low and high density routes. MCI on the other hand may concentrate on high-density traffic routes, and this may account for the lower unit prices quoted.

and this is more easily obtained over a high speed video terminal than a slow speed TTY printer.

Probably STI networks must provide both facilities based on a sample demography of the user population. Table 3 illustrates the break even point for several countries in Western Europe and the USA between dialled and leased connections, in terms of month/hours at which the cost of a leased line is equal to that of a dialled connection over a distance of 300 km.

TABLE 3.

Approximate Break-Even Point-PSN/Leased Voice Grade
For 300 KM [3]

Country	PSN-3 Min.	Leased	Equivalent Hrs. of PSN
France	$0.98	$630/mo	32.2 hours/mo
Germany	$0.95	$1690/mo	89 hours/mo
Italy	$0.97	$1108/mo	60.9 hours/mo
Sweden	$0.38	$ 670/mo	88 hours/mo
UK	$0.42	$ 580/mo	69 hours/mo
USA (ATT)	$0.70	$ 360/mo	25.7 hours/mo
USA (WUBB)	$0.60	$ 360/mo	33.3 hours/mo

The difference in terminal equipment costs must also be taken into account and for this purpose two services available in Europe are compared--ESRO/SDS as a typical example of high speed terminal rental and CEGOS/TYMSHARE (the French subsidiary of TYMSHARE INC.), offering dial-in services to a variety of data bases from the TYMNET node in Paris (see Table 4).

TABLE 4.

Comparison of Terminal Equipment Costs

ESRO/SDS	CEGOS/TYMSHARE
high speed video, 2.4 Kb/s	low speed video, max. 300 b/s
local printer	local printer
rental plus maintenance $13,200 per year	rental plus maintenance $4,325 per year

Notes: (i) The ESRO figure generally may include leased line charges to the nearest network node;

(ii) The CEGOS/TYMSHARE cost quoted is for their most expensive terminal operating at up to 30 characters per second. Simple teletype systems for 10 characters per second rent at about half this price.

These terminal costs now will be associated with other search costs to base a comparison of total costs for dial-up and dedicated line modes of operation for the same system. The ESRO/SDS system will be used since it is the only European system currently offering both modes. The network data base and system costs (averaged over the files available on this network) are \$18 and \$19 per notional half-hour search for leased line and dial-up modes respectively. Assuming in the leased line case that a terminal will work at an average rate of 1000 hours per year, and taking the annual cost of terminal plus maintenance as \$13,200, the per search terminal cost is about \$6.60. For the dial-up case, (see Table 4) the break-even points for several countries in Europe and the USA for dialled versus leased connections at 300 kms. Averaging the costs for Europe for a 30 minute telephone connection, the charge is \$7.40 per half-hour search. For dial-up per question costs, we may assume that use is at the break-even point. Taking the average break-even point for Europe this corresponds to 68 hours per month, i.e. 816 hours per year or 1,632 questions. On a yearly rental of \$4,325, the terminal cost thus amounts to \$2.60 per question. Table 5 summarizes these calculations.

TABLE 5.

Breakdown of Total Cost per Search--ESRO/SDS Network

	Dedicated lines	Dial-up
System costs including network	\$18.00	\$19.00
Line charges	-	\$ 7.40
Terminal costs	\$ 6.60	\$ 2.60
Total	\$24.60	\$29.00

In the dial-up case, we have assumed a rather large distance between terminal and node (300 kms). Reducing it could perhaps lower the total cost by no more than \$5. If the terminal were utilized at less than the break-even rate, terminal costs per search would increase, e.g. at 34 hours per month charges would be circa $5 per question. In general for monthly utilization rates of 30 to 60 hours the total per question saving of using a cheaper dial-up terminal does not have much effect: it appears that the losses in interactivity are not compensated by substantial savings (at the most $1 - $2 per search).

The above analysis may be limited since the example was chosen from only one system and no extra dedicated line charges are assumed, however, this simple model could be applied to actual cases where a user at a given location with an expected volume of traffic could calculate the most cost-effective access to a particular network. One general conclusion can be drawn however: if system and network costs stay about the same for dedicated and dial-up modes, then the per question cost differential will probably not be significant, unless these costs can be reduced well below current European levels. For example, if the system costs were $4 per question as stated by NLM for a total annual volume of 140,000 questions on a network carrying a greater volume of traffic, then $3 - $5 difference of line and terminal costs per question would be important.

This discussion of dial-up cost-effectiveness presents the point of view of a user confronted with a network with a defined tariff structure. The real advantage of the dial-up mode lies in the ease with which a wider user population can gain access to the network. From the system operator's point of view if this potential user population can be tapped by dial-up facilities, this could increase the total volume of use, thereby reducing research costs significantly. The economy of scale argument is used to justify the introductory investment cost; this need not be large, at the most part of a nodal mini-computer and about two man-months of programming effort.

XIV. FUTURE PROJECTIONS

Following the creation of several data processing network experiments in Western Europe (such as NPL, CYCLADES, EPSS, and EIN) and the plans of the European PTTs for specialized high capacity data processing networks employing packet switching and adaptive routing technologies, it should be possible for STI services to operate over publically owned networks of this type by the mid 1980's. The CYCLADES network, for example is planning this year some STI experiments. From a technical point of view the differences in STI traffic are not significant in terms of the basic design elements of the envisaged sophisticated networks.

H. Frank and W. Chou [4] suggested that for a particular configuration of the ARPANET, the incremental charge per kilopacket is in the range of $0.20 - 0.30. It is not clear however to what extent ARPANET can be used as a basis for discussing future economics of commercial or public data processing networks of this type. A more reliable indication of future cost patterns may be given by TELENET, which will shortly begin commercial operations in the USA, and whose tariffs have been agreed on by the FCC. The cost per kilopacket will be $1.25 with a packet size of circa 1,000 bits. To this should be added leased-line port charges of $50 per month for terminals working up to 9.6 Kb/s: thus, for a terminal working approximately 1,000 hours per year and generating two questions per hour, the per question port charge will be about $0.30 per search. If we assume a standard search of some 50 interactions each requiring one packet plus displays of 25,000 characters, circa 300-400 packets will be transmitted during a search. Thus, the total network cost per search will be about $0.75, independent of distance. Networking economics could cease potentially to become a factor in STI transfer if this cost level can be applied. There is no doubt that this operational technology can be transferred to the European situation. As shown, the present problem in Europe is one of tariff structures. No national telecommunication authority in Western Europe has yet indicated the basis for its possible future international charges when sophisticated public data processing networks become available for use. If the tariff structures are not reformed, the major economies of scale expected from the newer networking technologies will not be passed on to the users. This is important, not only for the European STI group but also for the entire European data processing community.

In the meantime, the STI community, particularly in Europe, has a series of problems which could be solved by systems analysis, combining networking economics and the cost effectiveness of various levels of technology. There are obvious attractions in applying distributed networks principles to dedicated STI networking operations. This would permit an increase in subject coverage by associating "centres of excellence" in particular subject or mission oriented areas within the same network. One such distributed STI network is already in the planning stage, the Nordic I and D network covering four Scandinavian countries. The EEC is also interested in promoting such networks, as part of its regional STI policy.

ANNEX I
LIBRIS AND SWEDISH MEDLINE

LIBRIS

The LIBRIS inter-library network, under development by the Swedish Agency for Administration Development (SAFAD), is well known. The following is a brief summary of some of its most interesting characteristics for future networking of STI. The overall concept is the interconnection of some 20 to 25 university libraries in Sweden (to be joined later by government agency and large company libraries). Its main function will be the computerized exchange of cataloguing information, acquisition data, control of periodicals and inter-library loans. It will also enable the automated exchange of bibliographic data on library holdings. This, together with the possible interconnection with other systems, creates considerable interest for future STI networking. Operationally, it is designed so that several data bases, held on different computers, may be interrogated from different types of terminals. Each library in the network will become a node in a large scale information network. A further important feature is the facility for incorporating other STI data bases (e.g. MARC of the US Library of Congress). LIBRIS opens prospects for international networking. The system as a whole is still being developed, and the features of interest for STI networking can best be summarized by reference to the development phases.

PHASE 1--TRIAL PERIOD

Started in 1972 in Linköping, the experimental system now embraces five libraries connected by dedicated lines of 4.8 Kb/s, using as the main computer a SAAB D22 on which the LIBRIS data base resides consisting of 50,000 documents; at the end of 1974 this is estimated to increase to 250,000. At present it is restricted to bibliographic data for literature searches and cataloguing. A prototype node, formed by a SAAB D5/30 mini-computer is installed at SAFAD. This has all the basic functions of future nodes, and acts essentially as a data interchange, but it can also act as a stand-alone computer, with capabilities for time-sharing and programming in BASIC, and file creation. An important point is the development of interchange software known as DISPATCHER; among the other resident software are terminal modules, in which the polling function takes place, and emulator modules and code translation for communication with other devices, such as a host computer or a terminal in another network, (e.g. Swedish MEDLINE or ESRO/SDS). By the end of 1974 three host computers should be operating within the network. The LIBRIS computer at Linköping, the Stockholm Computer Center, and an IBM 360/75 at the Karolinska

Institute on which the Swedish MEDLINE system is operated. The network will give access within Sweden to the LIBRIS file itself, MEDLINE, the data bases operated by the RITL Stockholm and those of ESRO/SDS. Some 15 terminal connections are planned by the end of 1974. At present, LIBRIS terminals are CD-92-414, but INCOTERM SPD 10120 are being introduced.

PHASE 2--LIMITED OPERATION PHASE

Four further libraries in the LIBRIS network will be connected after the completion of Phase 1. LIBRIS terminals will at that stage be standardized as INCOTERM SPD 10/20. These terminals are "intelligent" and, with 4 K bytes main memory and external disc units capable of storing ½ million characters, will perform as mini-computers. This illustrates an important feature of the LIBRIS development, the ability to create, store, and interrogate local information files at the nodes. During this phase, work towards the full integration of LIBRIS services, e.g. computerization of inter-library loans, periodical holdings, etc will continue.

PHASE 3--FULL-SCALE OPERATION

This phase is expected to begin 1976/77 when some 20 to 25 major university and other libraries will be connected. The central computer probably will be upgraded before this phase, and mini-computers will be installed in the network in each of the six university regions. Some of the line communications will be upgraded to 9.6 Kb/s. It has been stated, in order to gain maximum economy, all libraries contributing to the Swedish national accession catalogue should be connected, and therefore dialled connections to the basic LIBRIS network are also envisaged. The possibility of connecting university libraries in other Scandinavian countries is under consideration for Phase Three. A network diagram corresponding to the initial phases of LIBRIS development is in Figure 1 in which are included the locations of the present Swedish MEDLINE terminals.

With regard to LIBRIS economics, cost/benefit studies have shown that in terms of the proposed functions (search, acquisition, cataloguing, periodical control) there may be an annual net saving of about 1.1 million Swedish Kr., and the benefits (measured only in terms of reduced manual operation salaries) are likely to amount to 5.3 million Sw. Kr. Additional non-quantifiable benefits have not been taken into account.

THE SWEDISH MEDLINE

This service, operated by the Biomedical Documentation Center, Karolinska Institute, was started experimentally in 1972 and reached full operational status later that year.

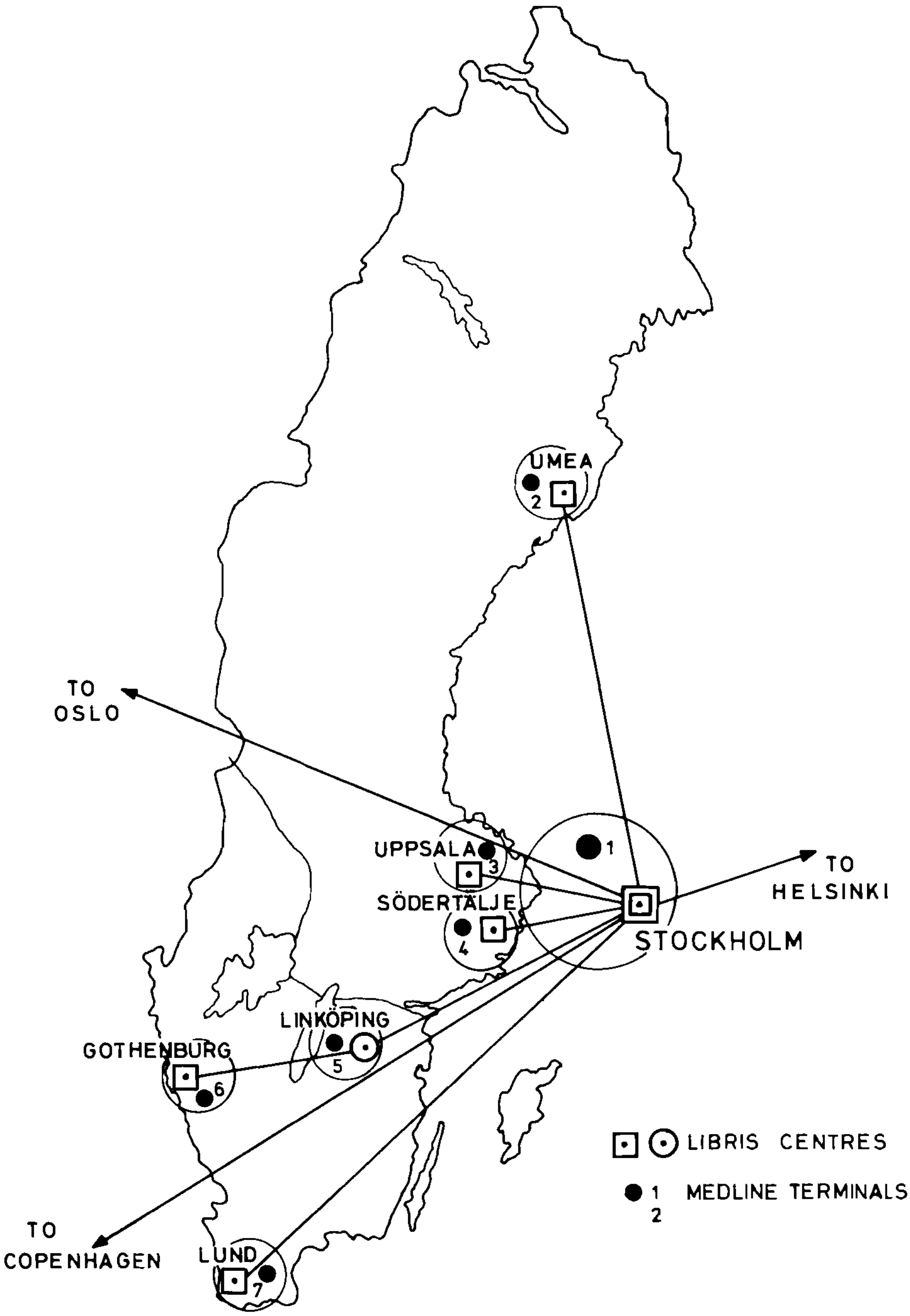

FIGURE 1. THE LIBRIS NETWORK WITH SWEDISH MEDLINE TERMINALS SUPERIMPOSED.

During 1971-73 the MEDLARS file was limited to 370,000 items in Stockholm, owing to storage capacity. 12,000 new references are added each month and the center is responsible for Scandinavian input to the system at the National Library of Medicine. As with the main NLM file, file structure and retrieval techniques permit interactive searches. Files are held on an IBM 360/75 computer at the Stockholm University Center (shared with other users). Twelve terminals are at present linked to the system, including three located in Copenhagen, Oslo and Helsinki. The terminals are asynchronous teletype compatible, operating at 200 bits/s over dialled lines. An additional 4-5 terminals will be installed during 1974. In the future a combination of dial-up lines and a network of leased lines will be used, using the multi-drop technique. The net will be controlled by a SAAB D5-37 mini-computer which will make it possible to use other types of terminals. Germany has also joined Swedish MEDLINE under arrangements made by the Institut für Dokumentationswesen.

ANNEX II

EUROPEAN SPACE RESEARCH ORGANIZATION

SPACE DOCUMENTATION SERVICE

(ESRO--SDS)

GENERAL

This network which first went on line in 1969 with terminals located in Paris (subsequently extended to Noordwijk in Holland) linked to a computer in Darmstadt, has now developed to include 7500 km of dedicated lines extending from Madrid and Barcelona to Stockholm, and from Frascati in Italy to London. The network currently is undergoing changes but the information given should be accurate for March 1974.

INFORMATION CONTENT--THE FILES

The network started operations based on the NASA/IAAA file containing references in a wide area of science and technology and currently six files are operated; they are METADEX, COMPENDEX, NTIS (GRA), NSA, NASA/IAAA and INSPEC. In addition, the Chemical Abstracts file is available to network users under special terms, having been added to the system as a result of a specific requirement of the Dutch organization Stichtung Nederlandse Informatie Combinatie. Over two million references are stored, and experiments with other files are in progress to meet particular requirements. In addition to bibliographic files, a specialized databank giving physical properties of high-quality, high-reliability electronic components is being established. This will enable interactive searches to be run on such questions as--is there a particular component having the following electrical properties and tested to a specific standard, and if not, what is the component most nearly meeting these specifications? It is expected that the input to it will be sufficient to enable full scale retrospective searching by the end of 1974.

HARDWARE

The computer used is a dedicated IBM 360/50 located at Frascati, Italy, with mass storage in the form of data cells totalling about 2400 million bytes. There are 21 terminals in the entire network, all CCI, with video display and slow speed printers. These terminals are buffered and operate in a binary synchronous mode. Dial-up facilities for speeds of up to 300 bits/s could be provided directly into the computer at Frascati, and similar facilities can be established at other nodal points. Dial-up facilities elsewhere will be via an interfacing PDP-11 mini-computer.

NETWORK CHARACTERISTICS

The location of the main processing computer at Frascati is not ideal from the network configuration point of view. The actual central communications node for the whole system is at Darmstadt, Germany, and this means that the Darmstadt-Frascati line must carry the network traffic. The line speed from the central node to external terminals is standardized at 2.4 Kb/s, therefore it is necessary that the central spine operates at higher speeds, although in the past it has proved possible to operate a 17 terminal network with a uniform overall speed of 2.4 Kb/s, with an average response time of about 4 seconds. Nevertheless, because of network expansion and general optimization, the central spine now operates at 9.6 Kb/s. In this configuration the Frascati-Darmstadt line is divided into four separate multiplexed channels, one of which is used for computer-to-computer working outside the normal operations of the service. A PDP-11 mini-computer is installed at the central communications node at Darmstadt to concentrate the traffic from terminals in the rest of the network, but other configurations are possible using multiplexing high speed modems and modem sharing devices. The attached diagram shows the layout of terminal installations in Western Europe (see Figure 2).

NETWORK DEVELOPMENT

An important next step in network development is to provide redundancy by creating alternative paths in case one part of the network suffers line disturbance. For example there are now six terminals in the Netherlands and three in the UK, each group operating on single lines, London-Darmstadt and Rotterdam-Darmstadt. It would be desirable to "close the loop" by an additional line from London to Rotterdam, thus providing an alternative path if any sector of the loop is inoperative. Other possibilities of providing redundancy will be investigated as network traffic builds up. For example a direct line from Barcelona to Frascati could provide a southern loop. A future possibility is a second back-up line from Frascati to Darmstadt, since it is on this link that the network is must vulnerable to line failure. Mention has been made of the possibility of direct dial-up facilities for a network carrying a variety of data bases, since at the present this is the only way to bring the network directly to the ultimate user. Apart from the expense of long distance calls, it is difficult to establish reliable international telephone circuits which will operate a teletype compatible terminal at 300 baud. Experience shows that dial-up speeds of this order are only reliable within a single telephone system: international circuits working at 110 baud are possible but at this speed one is probably below the threshold of usefulness of a sophisticated interactive system. Mini-computers located at the main nodes would facilitate local dial-in and also control

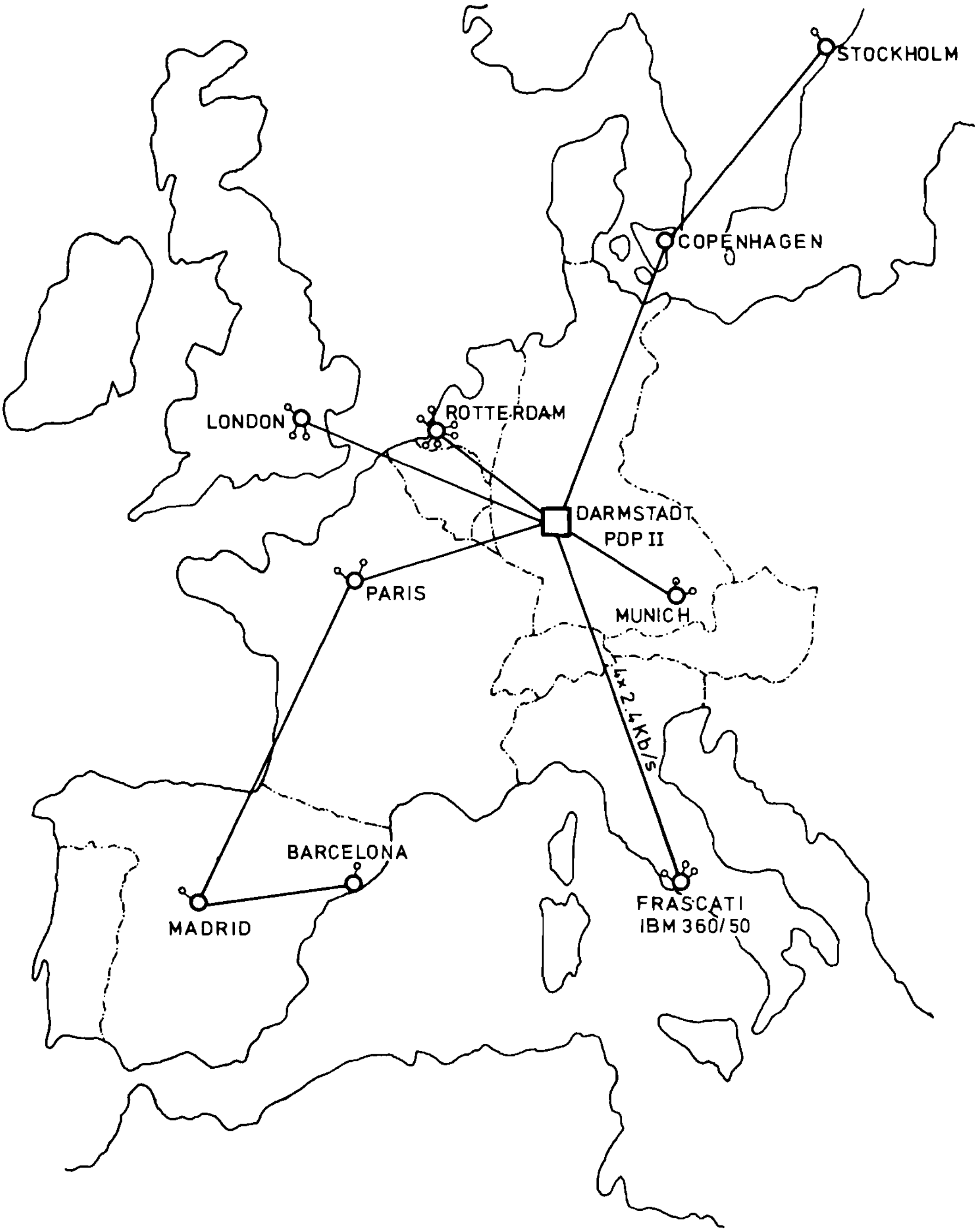

FIGURE 2. ESRO/SDS NETWORK.

such additonal facilities as fast printers, decentralize the print-out operation from Frascati, and enable message switching between terminals when the main computer is out of action. Further, the polling function could also be decentralized. These latter developments could be undertaken within a reasonably short time, i.e. 1974/75, once international agreement has been obtained for expansion of the offered network service: the software for different mini-computer roles has either already been developed or is in an advanced stage.

There is no immediate need to change the basic configuration to that of a fully distributed network, since such a step would not be required until a second main computer enters the ESRO network. On technical and economic grounds this would be a natural development since it would enable other STI organizations, with their own computer and files, to provide additional services for the area served by the network. However, this type of expansion would require international decisions which seem unlikely in the immediate future.

RETRIEVAL SERVICES

The basic software has been developed from the Lockheed Dialog System and is known as ESRO RECON. It provides full interaction and incorporates a file-switch capability, enabling a user to continue his search on different files. Free text searching of titles is also possible. For personalized current awareness services a user can establish an SDI profile at his terminal which is then automatically stored and run in batch mode against subsequent updates. It also provides centrally produced SDI and RB services for users outside the range of network terminals.

COSTS AND PRICES

The ESRO STI network has been committed to a partial cost recovery policy since 1972, and therefore all users pay charges according to their network use, including elements for file acquisition and maintenance, data processing facilities and network costs. Terminal equipment and its maintenance, and dedicated lines are provided by ESRO for a flat-rate fee of 11,000 annual accounting units for each terminal, and all other charges are aggregated into a cost per hour of connect time for each file in the system.

The following prices were quoted in 1974 for SDS on-line services:

File	Cost per hour connect time including network charges (in accounting units--au)	
	Leased line connection	Dialled connection
NASA/IAAA	28 au	30 au
NTIS/GRA	33 au	35 au
METADEX	33 au	35 au
COMPENDEX	38 au	40 au
NSA	43 au	45 au
Electronic Components Databank	43 au	45 au
INSPEC	35 au	37 au

Notes:

(i) For leased line connection, SDS supplies terminal lines and maintenance for a fixed annual fee of 11,000 au's.

(ii) For dial-up connections the user supplies his own terminal and pays local telephone charges.

(iii) For large-volume users with leased-line terminals special discount rates apply.

ANNEX III

THE TYMSHARE SYSTEM (TYMNET)

This network first came into operation in 1970 and now extends over the continental United States and to Europe (Paris). In the USA the network is accessible from over 70 major cities and the total extent of the network has been quoted as over 30 million terminal channel miles. It is essentially designed to provide a large volume of low speed access from the largest possible number of points.

It consists of some 90 Varian 620 mini-computers which form the nodes of the system, knows as TYMSATS. There are two types of TYMSATS; the first, known as a Base TYMSAT, is essentially a message-switching and an interfacing device to attach host computers to the network. It is programmed to interface directly with certain types of hosts (e.g. XDS 940, IBM 370/158, PDP 10). Base TYMSATS can accept up to two hosts each, and are connected together via 2.4 or 4.8 Kb/s leased lines in multiple rings.

The second type of node is known as a Remote TYMSAT and is essentially a terminal concentrator with store-and-forward capability. Each Remote TYMSAT can support up to 31 circuits operating in full duplex, asynchronous mode at 110 or 300 b/s. Thus, while presently it is impossible for the network to support fast synchronous terminals, Remote TYMSATS are capable of speed identification and code conversion. With each terminal attached to the network operating asynchronously, minimization of interrupts to the host computers is obviously important, and this is achieved by a multiple synchronous adapter which assembles bit stream into 16 bits, thus achieving a reduction in interrupts by a factor of 16. By standardization on low speed terminals, a large volume of traffic can be dealt with by the Remote TYMSATS. Since, on the average, interactive users with 10 to 30 characters per second terminals, average only 5 to 6 characters transmitted per second, traffic from up to 40 users can be concentrated on one 2.4 Kb/s line. Store-and-forward techniques allow for above average rates at some sacrifice of response time. Up to 900 users can use the network simultaneously. The TYMSATS also provide for efficient error detection, better than one bit in $4x10^9$ bits transmitted.

There are four supervisor computers, two are located at Cupertino, California, and one each at Englewood Cliffs, New Jersey, and Paris; only one is active at any one time. A user first connects to the active supervisor which after logging-in procedures establishes a circuit which remains unchanged for the duration of the log-in, unless line failures occur, when the supervisor re-routes the circuit.

There are 40 heterogeneous host computers in the network, of 14 different types, including IBM 360 and 370 series, Burroughs, PDP, Honeywell, XDS, Univac and CDC. The future development of the network will be to increase line speeds and capacity, with the emphasis on multi-channel operation over the main leased-line network to provide maximum redundancy. The number of faster terminals (600 bits/s and 1.2 Kb/s) able to dial into the network is increasing; this will necessitate upgrading the TYMSAT concentrators. High speed printing facilities are also being supplied at the network nodes.

The commercial role of TYMNET is to provide interactive computing facilities, using host computers owned by TYMSHARE Corporation, or in some cases, other organizations. It has however an important role in the provision of STI services. In addition to a number of commercial and statistical data bases in the information field, TYMSHARE is the network responsible for distributing MEDLINE not only in the USA, but also via the TYMNET node in Paris to France. TYMNET in the USA makes available data bases operated by Systems Development Corporation, such as CHEMCON, CAIN, INFORM and ERIC.

COSTS

The basic TYMNET tariff is as follows:

Description	Charges (in $)
Each log-on to host computer	0.50
Accumulative time connected to host, for all terminals:	
0 to 500 hours	3.00/hour
next 1500 hours	2.50/hour
next 3000 hours	2.00/hour
next 5000 hours	1.50/hour
each hour over 10,000	1.00/hour
Character transmission between user and host computer	.125/1000 characters
TYMCON-III rental (30 ports)	2150.00/month
AT&T Joint Use Charge (billed by AT&T)	7.50/leased lines used/month
One time installation charge	1000.00

For particular STI services, actual hourly rates of connect time have been quoted by organizations operating STI files over the network. For example, Systems Development Corporation in the USA quotes $10 per hour as TYMSHARE communication costs. In Europe, CEGOS-TYMSHARE, the French affiliate of the US Corporation quotes a price of F55 per hour connect time, F2 per 1000 characters transmitted from a terminal, plus a log-in fee of F5. Other "information wholesalers" offering services over TYMNET suggest that in addition to TYMCOM hardware costs of about $26,000 per year, user costs amount to about $6.50 per hour in the USA and about $15 per hour overseas. However, for the UK the GPO requires a surcharge for interconnection to US TYMNET nodes which would more than double the communication costs.

A network diagram, showing the approximate position of the main nodes, in mid-1973, is given in Figure 3.

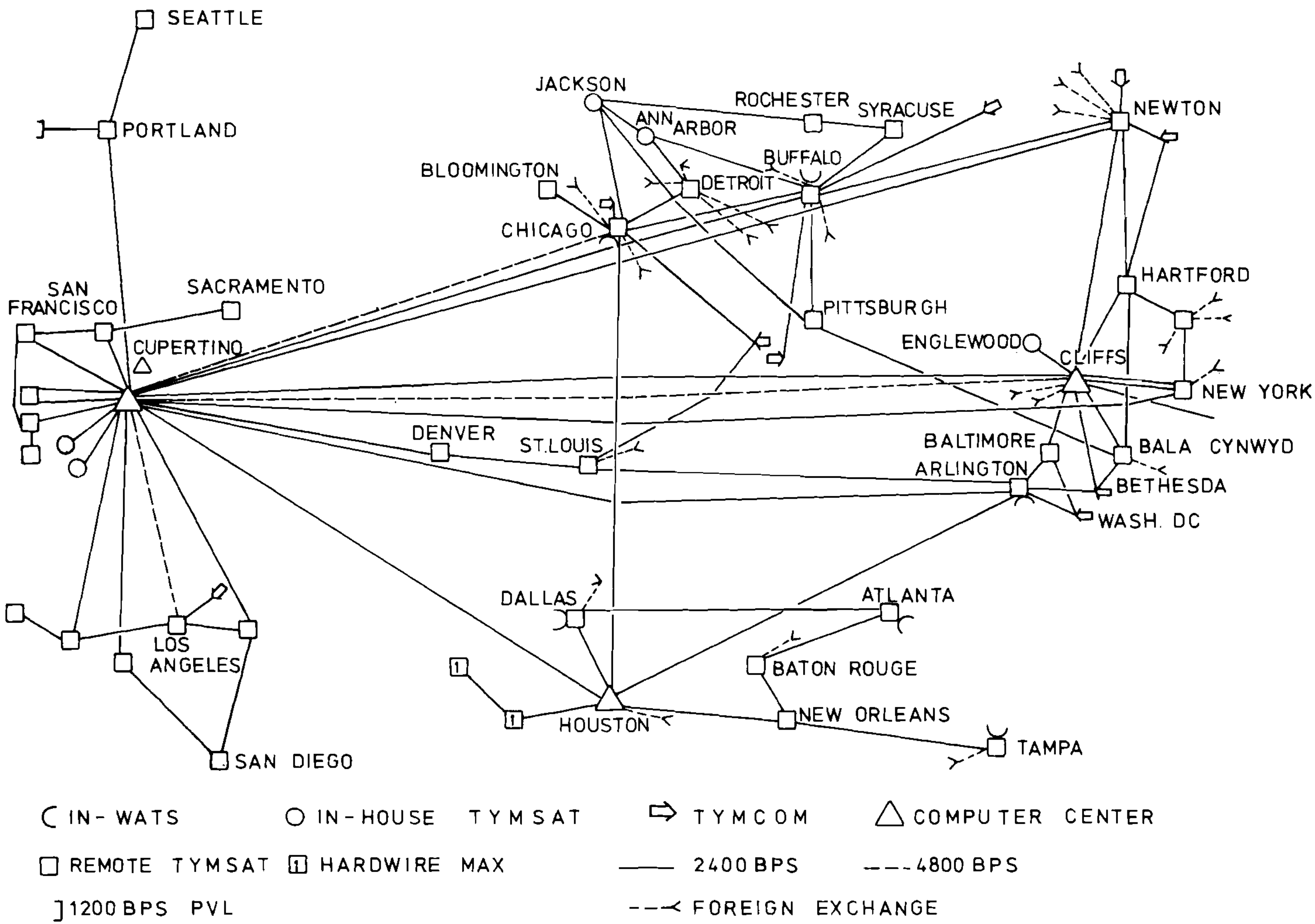

FIGURE 3. TYMNET

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DISCUSSION

Question: Is there a connection between this study and the projects of EURONET and INIS?

Answer: EURONET is a plan not only for a physical NW, but also to organize and rationalize STI activities of the EEC countries. Planning of the physical NW is not yet advanced enough to produce detailed information on operating economics. With regard to the INIS, the file, produced by the IAEA, was not networked by them. However, it seemed very probable that it would form part of the database of STI's European networking operations. The work carried out so far is restricted to output. At a later stage the economics of input and file creation will be examined. Another important and interesting extension of the study would be the STI NW being developed in socialist countries, especially those encountering similar economic and regulatory problems in leasing lines. Their experience would provide an essential third point of reference in these areas.

Comment: The cost of leasing lines in Europe is a problem. However, a major problem not yet discussed is that PTTs did not allow the sharing of lines between several users. The per search cost quotes, e.g. for ESRO, are also affected by other considerations: not only unused capacity, but also the suboptimal configuration of the NW, with the host computer situated on the periphery rather than at the center. This had to be done for political reasons.

Answer: As pointed out in the paper, economics should dictate that the days of single purpose dedicated NWs are numbered. The long range problem is the regulatory and clearing policy of European PTTs, especially if the economies of new technology NWs are to be transferable. In the meantime, the problem is to get from where we now are to some future point where the NWs will be operated by the PTTs. In a dedicated single purpose NW we have to pay for a very high proportion of "silent time," while this can be ameliorated, the economics will always be suboptimal.

Some Problems of Multimachine Systems

Z. Zhelesov*

I. INTRODUCTION

The widespread use of computers in all fields of human activities calls for some changes in their application. The creation of complex information systems requires the use of more than one computer. Walter Finke, one of the technical directors of Honeywell, was correct in claiming that anyone, who wants to be a leader in the computer field has to demonstrate his ability to create multimachine computer systems.

A. Some Definitions

In order to avoid misunderstandings of the terminology, the following definitions are given:

MMS - Multimachine system. A combination of machines connected either locally or remotely to fulfil joint tasks;

MMC - Multimachine complex. A subset of MMS where all connections are local, i.e. a computer center using a few computers;

MMN - Multimachine network. A subset of MMS where all connections between the MMC, or computers are remote, i.e. by means of telecommunications. A network of computer centers.

B. Advantages of MMS

MMS feature some advantages in comparison with the uni-machine structures:

- Effective use of hardware;
- High reliability;
- Possibility for using common control programs;
- Possibility for common use of data files, libraries, archives, etc.
- Decrease of job execution time;

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- Higher computational power; and
- Greater flexibility.

C. Features of MMS

MMS's permit the carrying out of the following tasks:

- Collecting, processing and storage of big data files necessary for control systems such as material supply, bank management, criminology, hotel, plane, and train reservations, different reference systems, etc.;
- Creation of national information storage networks which can be accessed at any point. This is of great importance for social management;
- Optimal solution of strategic problems through combining computer capabilities with human heuristic abilities;
- Education;
- Centralized management of sectors, including administrative, material supply, reservation, and firm management;
- Solution of complex tasks requiring great computing power (e.g. change reactor, physical processes, meteorology). The solution of these problems could exceed the capabilities of a single high performance computer. With a MMS, the task may be split into segments to be executed by different computers. Upon execution of its own segment each machine sends the results to the others and receives new data for processing;
- Creation of a national system for social management.

D. Hardware Requirements

These are:

- Flexible systems featuring real time and multi-programming capabilities;
- Memory protection enabling concurrent execution of different programs without access to other problem files;
- Dynamic organization of the storage capabilities enabling displacement of a program in the course of its execution;

- External high capacity direct access storage, and low access time;

- Special telecommunication units;

- Special units for information exchange between computers within a MMS.

It can be said that the above requirements have been met by current computer installations.

E. Software Requirements

These are:

- Optimal organization of information exchange between user and machine and between individual units;

- Adequate response time for each user;

- High reliability;

- Options for different tasks, i.e. possibilities for uncomplicated reorganization of data files by simple command language, for different algorithmic languages, etc.

- Possibility for concurrent use of a program.

The problems related to the creation of MMC and MMN are specific ones. Although a number of systems have been organized, there are still many problems to be solved. The majority of unsolved problems of MMN have technical and social aspects, therefore IIASA has to consider mainly these.

F. The Optimal Geographical Distribution of Computers Within a MMS

A number of theoretical problems are connected with the design of MMS:

- Geographical displacement of computers enabling effective processing of information requests;

- Capacity of information channels;

- Service sequence of data transfer and processing;

Different solutions can be found for the above problems:

- Connecting computers with available telecommunication links;

- Changing the geographical distribution of computing power;

- Re-designing some of the available communication equipment for a more effective MMS;

- Introducing new computer centers, and communication equipment.

Several approaches have to be considered for an optimal geographical distribution of computing powers. This gives rise to problems connected with the effective operation of the MMS. Answers to this type of question can be given mainly by modelling.

G. <u>MMS Modelling by Discrete Situation Networks (DSN)</u>

A number of reports [1; 2] contain basic definitions of discrete situation networks (DSN). Brief information and some definitions of DSN are presented in this paper.

DSN is a large system characterized by network processes whose nodes take charge of the following functions: input; service; output; and provide DSN services (communications, transport units, etc.). The processes within the network are of a discrete nature and are considered to be changing in time. This occurs as a result of random or control effects. The purpose of these is to optimize the running processes based on a criterion for system effectiveness.

For the definition of DSN, the following sets are to be considered - A, B, C, D, W, - where:

- A elements include only those nodes which take charge of input functions;

- B elements include only those nodes which take charge of output functions;

- C elements include only those nodes which model the service object, and this set is called "decider;"

- D includes all service objects;

- W includes all object characteristics (type, velocity, volume, etc., according to the model's purpose).

Each DSN node (elements A, B, C) is characterized by a specific status. The object status is determined in a parallel way.

The object and node status (elements A, B, C) at a given moment t defines the situation S(t). The DSN operation is considered as a situation change. For this purpose the functions of the set (elements A, B, C), the rules of servicing and others are determined.

The description of a certain class of large systems (in particular MMS) is not only intended to create a model for investigation, but also for control. A more detailed consideration of the so-called situation control can be found in [3].

H. Organization

Some problems pertaining to the creation of MMS are interesting, useful and necessary. Under the auspices of IIASA some of the following questions could be solved:

Preparing a review on the operation of projected MMN's, such as ARPA, CYCLADES, etc. In this the following questions are to be asked:

- How to design it;
- Structure - present and future;
- Network capability;
- Equipment included, and main parameters;
- Basic dialogue principles within the network;
- Pros and cons of the network;
- Packet commutation system (messages or channels);
- Service order in data processing and transmission;
- Data protection of files and transmission;
- Others
- Dispatching the review to IIASA members;
- Discussion of the review;
- Visits sponsored by IIASA to the most appropriate MMN, for acquaintance with their capabilities and peculiarities;
- Preparing requirments based on review information for MMN (drafted by leading MMN organizations).

These must take into consideration the following:

- Ways of data transmission by commutational networks;
- Ways of dialogue between users and MMN.

- Discussion of these requirements together with the International Network Working Group (INWG), Subcommittee of TC-G of IFIP. Requirements needed to expand to the level of international standards of the MMN.

- Periodical experience exchange within IIASA on the MMN creation by conferences, seminars and workshops.

- Providing literature on MMN in the IIASA library, for the information of all members.

No doubt there are many MMN problems to be solved by IIASA. One thing is certain - the drafting of international requirements, and later on, a standard, is of great importance, and IIASA can play a significant role in this.

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(translated from the Russian)

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Private Computer Networks

P. Hughes*
D. Mann*

I. INTRODUCTION

During the remainder of this decade there will be rapid growth in "second generation" private networks, i.e. those which involve the linking of terminals to a single central computer by leased circuits via multiplexers and/or concentrators. However, this paper concentrates on the development of a "third generation" private networks, in which data transmitted into the network may be switched or routed to a variety of different destinations.

A. Requirements for Private Networks

There are two broad classes of user requirement that result in the establishment of third generation private networks:

1. The desire to have an on-line or real time system based on a number of geographically dispersed computers for security or reliability;

2. The desire to share the resources of a number of computers among users.

The demands for these facilities are growing rapidly, albeit at this stage among university/research establishments, government departments, national industries and large private organizations. Moreover, different users' requirements lead to a wide range of needs for data transmission facilities.

We do not see how the Telecommunication Authorities can cope with these demands over the next few years and therefore, many organizations may wish to establish private networks.

B. Background for the Establishment of Private Networks

In planning a private network the user must try to take into account the different relative cost trends of its constituents: communication channels, computer hardware, computer software, and professional services. He also has to cope with Telecommunication Authority constraints, short-term licences and variable tariffs; and, quite likely, with lack of cooperation from mainframe computer manufacturers.

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II. OVERALL DESIGN

A. Choice of Switching Method

Patching of lines (circuit switching) can provide a convenient solution for some simple private networks. However, for most applications packet switching is emerging as the natural means of data communication. (Message switching is more akin to operations performed inside a host computer connected to the network and is rarely an alternative to circuit/packet switching.)

B. Allocation of Tasks to Computers

In a packet-switched private network there are four broad task categories: transmission and switching, monitoring and control, terminal handling, and host interfacing. These can be allocated to network computers in a variety of ways. At one extreme, they can be executed by the host computers; at the other extreme, separate computers can be introduced into the network for each category of tasks. Separating the tasks has a number of advantages and, with the decreasing cost of computer hardware, will become increasingly viable even for small networks.

C. Provision for Fall Back

If there are several computers in the network between a source and destination, then communication may be impossible if any one of the computers fails (e.g. that linking a terminal or host into the network). Hence the availability of the overall service may be unacceptably low unless the accessibility of the individual computer systems is increased (e.g. by duplication) and/or alternative transmission paths are provided.

A judicious combination of these two approaches will probably always be appropriate, although the optimum balance will change with time. At the moment highly reliable computer systems are expensive. With machines like the Pluribus, which is currently being introduced into the ARPA network, this cost should be reduced significantly.

III. STANDARDS

A. Required Communication Standards

The standards required in the communications subsystem can be considered under three headings:

1. Single packet transmission--covering the transmission interface, synchronization at the block level, length determination, transparency, error detection, etc.;

2. Channel control--covering error correction, efficient channel utilization, local flow control, etc.;

3. Network facilities--covering maximum packet length, message assembly/disassembly, addressing, virtual calls, delivery confirmation, global flow control, priority levels, etc.

Interconnection of networks, and use of more than one network would be simplified if common standards were adopted at all levels. However, category (3) has the most fundamental implications for internetworking, and it is at this level that common standards are most desirable. NB: the nature of the facilities is important (e.g. whether sequencing is unavailable, optional or mandatory), while the format of the packet is much less important.

B. Available Communication Standards

Relevant international communication standards have been or are being defined by CCITT, CEPT, ISO, and IFIP. In addition there are standards defined by national bodies (e.g. Post Office for EPSS) and by manufacturers (e.g. IBM). However, progress is slow, possibly because it falls outside the mainstream of interest of both Telecommunication Authorities and the DP industry. Users should be more influential in setting standards and EPSS provides an important vehicle for this.

C. User Strategy on Communication Procedures

At the moment the user must define many of his own procedures, although he can look for guidance to the standards mentioned above and to other sources such as specific research projects (e.g. ARPA, NPL, CYCLADES) or commercial systems (e.g. SITA, SWIFT). Thus the user can make a sensible, if somewhat unsatisfactory, compromise choice of procedures for single packet transmission and channel control. However, the sensible definition of network facilities to ease internetworking is not obvious and is the source of controversy among workers in the field. CEPT, and IFIP TC 6.1 are concerned with this problem and the European Informatics Network should provide a valuable vehicle for experimentation.

D. Other Standards

It should be emphasized that communications are only a small part of the operations of private networks. Standards are also required for other procedures (e.g. terminal handling, file transfer and remote job entry protocols). Unfortunately the computing industry does not have strong bodies such as CCITT to pave the way in these areas.

IV. COMPONENTS

A. Communication Channels

The overall transit time for traffic in a private network is determined largely by the capacity of the individual channels. Hence, in many networks, high capacity channels will be needed to achieve adequate response times, even though the volume of traffic could be handled by smaller capacity links. Thus there will be a rapid rise in demand for 48Kbps services from the current low level.

Because of the high cost, organizations increasingly will seek ways to economize by integrating their private voice and data networks.

B. Host Interfacing

One of the major problems in establishing private networks is, and will continue to be, the interfacing of mainframe computers into the system. This is because the mainframe manufacturers do not currently support private networks. Their standard software requires modifications and these currently are made in such a way that the user is deprived of the minimum of manufacturers' support.

The interface method appropriate to a particular computer depends on the type of mainframe and on the operating software under which it is running. In some cases new access software should be introduced on the mainframe manufacturer's hardware, either on the mainframe itself or on a standard front-end processor. In other cases it is better to introduce a separate communications processor for this purpose. Because of the great variety of mainframe installations, there is a high probability that in the near future, the user will have to undertake development work in this area.

C. Communication Processors

In addition to the front-end processor role, there are at least three other broad classes of application for communication processors in private networks: as nodal switches; as terminal handling computers; and for network monitoring and control. (In any specific network, two or more of these roles may be undertaken on the same computer.)

Unless environmental standard and reliability characteristic requirements are stringent, there are many commercial mini-computers that are suitable for these roles. The characteristics of the machines differ, and hence there is considerable scope for selecting combinations to optimize cost/performance. However, differences in this area are likely to be insignificant compared with the advantages of standardization.

D. Communications Software

Over the next few years there will be increasing availability of software packages for nodes, network monitoring, control centers, and terminal handling computers. The last is more akin to host interfacing and will probably involve most users in development work because of the variety of terminals available.

V. COSTS

Salaries account for a high and increasing component of the total cost for establishing private networks. In contrast hardware costs are a low and decreasing component. Telecommunications rentals are very significant. The total cost for establishing and operating a private network is currently very high. Hence there will be considerable requirements for shared networks like SITA and SWITFT. Controlled sharing should be allowed in private networks and greater flexibility is required from various Telecommunication Authorities.

VI. EXAMPLES

Each of these examples illustrates various factors which are currently applied in the design and implementation of private networks. In each case Logica has been, or is involved in aspects of design or implementation.

A. Swift

This network will connect some 250-300 major banks in 15 countries. It is expected to spread to include more countries. The design is a mixture between message and packet switching.

B. Bins

This is a switched network being implemented by Barclays Bank in the UK. The network, when completed at the end of 1975, will handle well over 3000 terminals and will involve 12 communications processors. It could prove to be the forerunner of large scale banking networks.

C. British Steel Corporation

It produces nearly all the UK's steel. A packet switched network has been designed to link all its computer centers. If the implementation proceeds as planned, this will be one of Europe's largest packet switching networks in 1976.

D. European Informatics Network (EIN)

Logica together with SESA have been awarded the contract to develop the first stage of this network. Some experiences are given in cost breakdown and methodology being used to develop the network switching centers. This development is planned to take about one year. EIN is already having influence on the general development of packet switching networks in some European countries.

NETWORK PROCUREMENT

	£ K
Hardware	
6 nodes at say £ 20K	120
9 THPs at say £ 25K	225
NOC and development system(s)	100
	445
Software	
Node	80
THP	120
NOC	40
NCMs for hosts	150
	390
Staffing	
Supplier Selection	
Design Validation and Progress Control	
Acceptance Testing	
Planning Host and THP Standard S/W	
PTT Liaison	150
User Liaison - training	
- monitoring	
Site Provision	
	£ 985 K

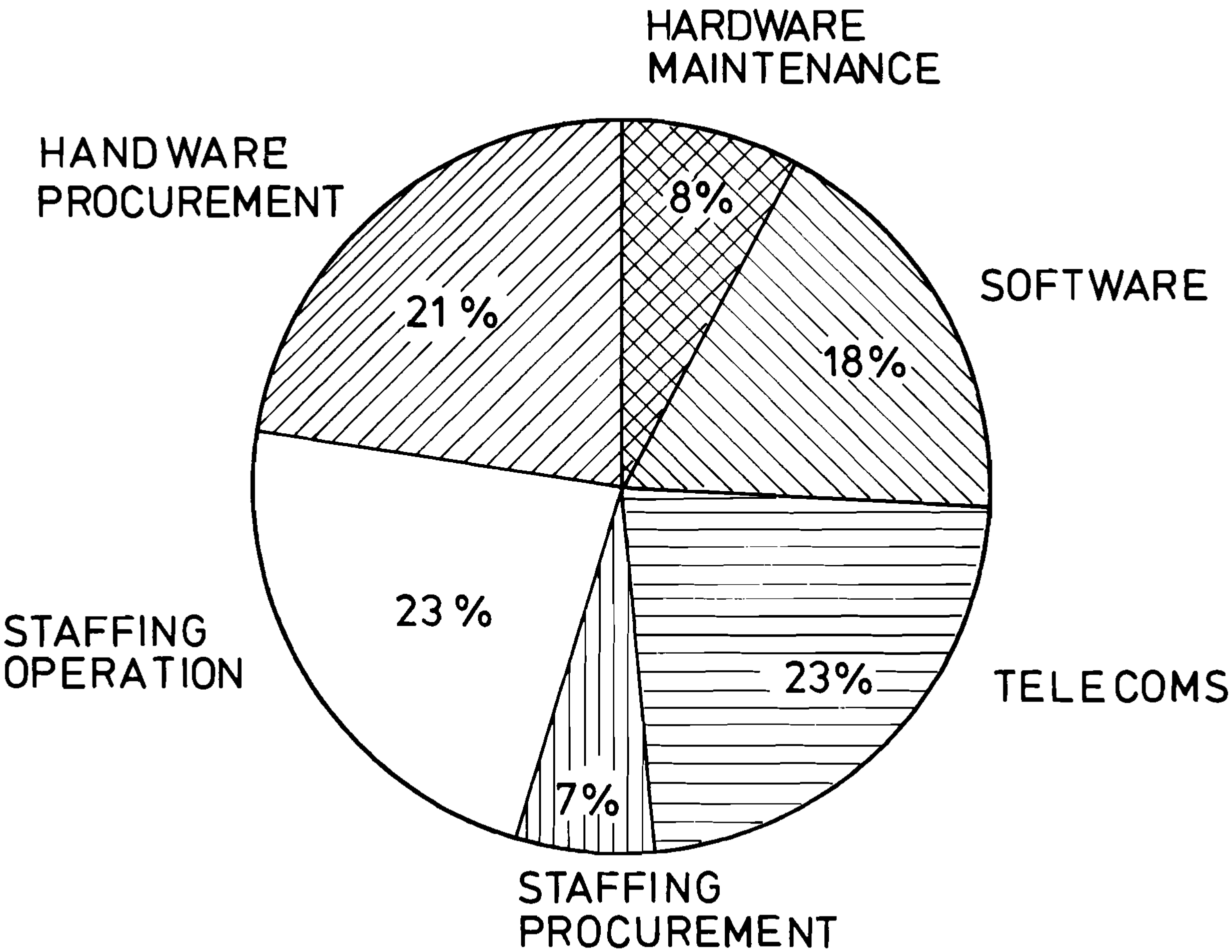
Not included, inter alia: Terminals, Hosts, Buildings, Installation of Communications Channels.

NETWORK OPERATION

	£ K p.a.
Telecommunication Channels	
Trunk	50
Local	50
Maintenance of Hardware	35
Staffing	
Manning NOC	
Organization of Maintenance	
Maintaining/Adjusting NCMs	
Planning for Growth	100
Information, Advice, Training	
Analysis of Use and Performance	
Accounting	
	£ 235 K p.a.

Not included, inter alia: Procurement/Development of New Items, Anciliary Services.

BREAKDOWN OF TOTAL COST (FIVE YEARS)



DISCUSSION

Question: Do the telecommunication costs mentioned depend on the size of the system? Which size do £100,000 refer to?

Answer: They are projected costs for a six nodes NW, with nine terminal processors. Barclay Bank spends one million pounds a year for communication. BINS spend three million pounds to procure its NW. This is only the communication portion, excluding terminals. The main processor costs about 25 million pounds. Communication expenses do not represent a high proportion of the total. This increases only on low traffic Transatlantic NWs.

Question: The cost calculations for ARPA are very difficult to obtain and not precise. It costs approximately 30 million dollars per year. What about privately designed networks, versus public packet switching design?

Answer: The British Steel, as mentioned, have two bids:

a) to make it compatible with EPSS;
b) to design it as they may desire.

What they would do faced with these bids is not known. COST 11 already has influenced the way people think of NWs; besides, private NWs will be established sooner than public ones. The most important thing is not to force everybody to use or build the same type of NW, but to be able to make suitable connections. Private NWs will certainly take the first step.

Question: Who is responsible for the management of SWIFT?

Answer: There is a separate, nonprofit organization in Brussels concerned with it. It is owned partly by that Bank.

Question: Until now there are no tariffs at ARPA, but it soon will change. The financing will be carried out directly, but the tariffs must still be investigated. Who knows about this?

Answer: When ARPA was built, communication costs were the main issue. Packet switching was to reduce the total connection cost. It is not clear, how much ARPA will charge, and how. Attempts have been made to change the way of running the NW by distributing charges, but they are not yet published.

Answer: It does not matter how high the cost of ARPA will be. An important problem is the payment structure. In the SWIFT NW, smaller banks should be associated also for a maximum penetration. If this were not so, the net would be less useful, because greater penetration increases the value. One problem is whether to charge for installation, or only for using it. The same problem existed with telephones in the UK; they have solved it by not charging installation costs, in the hope of maintaining traffic. It will be worthwhile to see the results of ARPA investigations on how people react to changes in tariff structure.

Question: What is the relative security cost in SWIFT?

Answer: In the design state, one third of the effort is used in security planning; the hardware costs for this are not known, as the problem has not yet arisen.

Question: Are there any plans in the UK for future integration of private NWs and EPSS installations?

Answer: It is difficult to connect two private NWs through EPSS, but it is possible to connect one. That is why an EPSS compatible British version is needed. British Steel, as a private NW, wants to go partly through EPSS.

Question: What proportion of British Steel's budget is spent on communication costs?

Answer: Investments for the next 10 years will be 55 million pounds. The procurement for the proposed NW will be two million pounds, it is currently being priced. It is regarded as a very small cost in relation to total costs, and they believe that there will be a more than 4% saving in the effectiveness of their total computing investment project over the next 10 years.

A Practical Example of Distributed Computer Systems

C. Molnar*

Distributed computer systems are now in vogue. They include world-wide computer networks as well as the emulation of standard terminals or large computers running on small ones. Moreover, the last three years have produced a considerable amount of theory which, together with effective implementations, allow us to attempt to come to some of the following conclusions:

1. There is no theoretical difficulty to join different computer systems in a single network; the main problems are administrative, i.e. how to agree on standards, protocols, etc.

2. While computer hardware still continues to decrease in price, communication lines do not show the same trend.

These considerations place more importance on how to distribute information processing capability in an economical way, rather than to find new network principles. In this paper, our modest distributed computer system is described, taking into account some practical considerations originated during its implementation.

In 1969 a small Hewlett-Packard time-sharing system was bought, whose present version is the HP 2000/F. As it could not satisfy our information processing requirements, it was connected to two large computers: the IBM 360/67 close to us in Pisa, and the CDC 6600 in Bologna, about 200 kms away. At the same time, the coexistence of two different operating systems was to be tested. This will be described later. In designing this small network, we worked with the criterion to let the HP process as much as its hardware and software support enabled it to, and to send to the large computers those programs which required a greater core memory or special languages.

The standard HP 2000/F system consists of two HP 2100 computers, one with 32 K core memory and one or more discs, for actual processing, and the other with 8 K core memory dedicated to I/O operations. The system's program language is BASIC with some extensions, with a rather good file-handling capability. Besides the common library, which is

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available for any user, each programmer can build his own, containing programs as well as data files. The only input-output device is the terminal which is either a teletype or an alphanumeric display. The system can handle up to 32 terminals. Optionally, output can be deviated to a line-printer which is available only for one user at a time as there is no spooling facility.

One of our aims was to extend this rather limited information processing tool. First of all, a fast I/O device was needed to speed up the exchange between internal and external information. Without modifying the BASIC language itself, the program has the possibility to call a fast punched-tape I/O equipment by outputting a special non-printing character. As an educational tool, and for system analysts a conversational ASSEMBLER a small HP virtual machine, and a conversational micro-assembler for writing and debugging microprograms were added to the system. For writing and updating text-files, a programmer context editor was implemented. Finally, a SCRIPT program was introduced for formating texts.

However, these facilities still do not allow the processing of large scientific programs, which are currently sent to the CDC 6600. The standard CDC terminal, called USER 200, consists of a card-reader, a line-printer, a display and a keyboard. As it would be of little interest to send a program from the CDC 6600 to be processed by the small HP, these two computers are connected only in one way, which was made possible by simulating a USER 200 inside the HP computer. Through a USER 200, the SCOPE operating system can be accessed either in batch mode or in a conversational manner. The advantage of the latter is the possibility of file-editing, consulting the system, etc., but it is rather expensive. As the possibility of running large programs in the local system was lacking, the possibility of sending programs to the CDC only in batch mode was included. Thus we simulated only one USER 200 in the HP system, giving only one user access to the SCOPE system at a time. (One telephone line could hold up to 16 USER 200.) In other words, the simulated USER 200 belongs to the system, not to the single user. The simulation takes place in the HP I/O processor whose core memory is widened to 16 K. As the central processor of this computer normally works at less than 15-30% of its capacity, availability was no problem. The simulation of the USER 200 occupies about 4 K, mostly used by buffers. The display and keyboard were replaced by a teletype, which is used also for supervising the computer connections through system commands. The line-printer is that of the HP 2000/F. Of course, without a spooling device, when it is used as a CDC terminal, it must be detached from the time-sharing system and vice-versa.

Automatic checking avoids incompatibilities. The card-reader of the USER 200 may be virtually any HP 2000/F user program. Its output can be deviated by special characters to the CDC terminal. If the connection is not activated, or, if it is in use the potential user is informed by messages. When a user leaves the connection, an automatic check detects whether he has sent an end-of-file. If not, the file eventually transmitted is killed to avoid processing bad information. The physical connection between the two computers is a telephone line using tele-selection. The transmission rate is only 1200 bauds, but this limitation is due to the poor quality of public telephone lines; the system should actually be able to handle transmissions of up to 4800 bauds.

For the user this connection enables him to prepare his FORTRAN or SIMULA, etc. programs on HP BASIC files by using a rather powerful context editor, and to send them by a library BASIC program for processing at Bologna. The results come back on the line-printer. A more sophisticated user could write a pre-processing program in BASIC or in HP ASSEMBLER, which could be sent directly to the CDC as data.

The connection with the IBM computer in Pisa has interesting theoretical aspects. Its first pecularity is the very short distance (about 300 m.) between the computers. This is covered by a private line, so that data transmission is practically free of charge. Its other curious feature is that a time-sharing system communicates with another time-sharing system, more precisely, a user-program on the HP with another user-program on the IBM 360/67. Several dialogues can run simultaneously.

The 360/67 uses the CP operating system based on the concept of virtual machines. A user can have access to the system through his terminal, which, normally, is an IBM 2741, covering the function of the console keyboard of a bare 360/65 computer. This virtual machine may have as I/O devices the terminal itself; a spooled card-reader; a line-printer; a card punch; and dedicated devices such as magnetic tapes; telephone lines, etc. A particular program, called CMS, running on the virtual machine, accomplishes the task of an operating system serving one user: it has its own language processors, etc. The physical connection between the two computers is made by 16 parallel data lines. Transmission is asynchronous. As interfaces, a parallel data adapter (PDA) on the IBM computer is used. It is attached to a selector channel, and to two 16 bit duplex registers on the HP. Transfer rate is limited only by the channel speed of the IBM computer. The PDA is attached as a dedicated line to a particular virtual machine, called MVC, which, with some modifications of the CP operating system, dispatches the information to the other virtual machines. The MVC works as the virtual console of the other virtual machines accessed through it: thus it also allows the generation of new virtual machines by the "logon" command. In other words, the MVC works as a concentrator of terminal lines. The same work is done on the HP side of the I/O processor.

When an HP user wants to dial a virtual machine, the I/O has to deviate to the 360. He can access a virtual machine, and, from this moment, two-way communication is established. Therefore, the HP user does not need to specify explicitly his interlocutor each time. The above mentioned fact is an interesting aspect of our connection, i.e. instead of two separate systems, there are two programs in dialogue. Thus, the input command of one program must correspond to the output of the other program. To avoid possible deadlocks between two programs requiring simultaneously either a reading or a writing operation, a priority system has been introduced, giving systematically lower priority to the HP program. More precisely:

1. If both programs request an output operation, the higher priority program waits till the other has finished, but no information is transmitted to the virtual machine;

2. If both programs request an input operation, the lower priority program is immediately warned by means of a special character.

In a more sophisticated computer network, this priority could be made subject to change, and handled by the programs themselves. This type of communication between computer systems is somewhat more expensive regarding data transmission than the one based on rigorously defined protocols. However, its great advantage is its extreme simplicity which allows implementation with relatively little effort. On the other hand, the possibility of enabling a program on one computer to cause another program to run on a second computer, as well as information exchange possibilities between these two programs, is equivalent to calling library programs from, or sending files or specified records to another computer, i.e. it is equivalent to the functions of a computer network. The format of the messages between the computers is practically defined by these library programs, so one or more protocol systems can be introduced without touching the operating system. In the future, we hope to access also other networks.

Another result of this system is that the contents of the message need not be known. The "closed envelope" hold only the addresses of the origin and destination, in order to be able to inform the sender should the message be damaged. It increases security, and--what is more interesting--makes it possible to transmit information following different conventions on the same physical lines.

Our "distributed computer system" is far from being a real network. It is rather a tool for widening "our" information processing capabilities and, at the same time, can work in a satisfactory way.

DISCUSSION

Question: How many connections are there from one process to a virtual machine?

Answer: One process is connected to one machine, if there is one user. There is one virtual machine, implementing a single channel, that is a normal packing channel, a half duplex channel.

Question: Is there any possibility of multiplexing if a process wants to communicate over more than one logical line? Has the process to implement both multiplexing and demultiplexing?

Answer: Multiplexing is implemented at the system level. The meaning of a message for the computer is defined by the user on his level.

Question: Is the connection HP → VM → HP different from the VM → VM communication?

Answer: It is. You gain access to the system through a virtual machine.

Question: Do you also want to study problems of cooperation of the different processes with a large host computer such as IBM or CDC?

Answer: IBM is allowed to send messages and problems to CDC and bring others back, through the HP.

Question: Does one job consist of different steps, and is it possible to have the steps run on different machines?

Answer: Yes, if you know the right sequence of job cards.

Question: Can a user select the terminal in the system according to his process, or is a point connection the only possible solution?

Answer: A user of the HP has access to IBM but it accesses the virtual machine through the system, not directly.

Variables and Constraints in Data Communication Systems Design

R. Porizek
J. Puzman

Present data communication systems are complex, contain many elements and are characterized by many variables and parameters. The design of such a system is a difficult task because of the large number of variables and parameters, and because of the many interrelations not only among these, but also among the parameters themselves. The analytical expressions for these interrelations often are not known, or they are very complicated.

There exist several approaches to the design of an optimal network. First let us formulate the common one:

let Y_j, $j = 1,\ldots,M$ be the variables of a data transmission network and let B_j be a set of values which the variable Y_j can acquire. The sets B_j can be finite or infinite. Such variables can be e.g. a number of nodes, channel capacity, transfer rate, transmission model, a type of data link control procedures, routing algorithms, a type of switching, and others.

The variables are related to, or derived from, the properties of network elements and facilities, both hardware and software. Their values are determined by selecting certain elements from sets of available ones.

Let X_i, $i = 1,\ldots,N$ be the parameters of a data transmission network and let A_i be a set of values of the variable X_i. A_i are sets of real numbers, finite or infinite. Some examples of these parameters are: costs; response time; average time delay; reliability; location of network nodes (including terminal nodes); average throughput; and line utilization factor, etc. With these parameters the overall performance of any network may be described.

In general each parameter X_i is a function of all variables Y_j and of other parameters X_k, $k = 1,\ldots,N$ and $k \neq i$:

$$X_i = F_i \left[Y_1,\ldots,Y_M;\ X_1,\ldots,X_{i-1},\ X_{i+1},\ldots,X_N\right] \quad . \tag{1}$$

In practice the individual parameters X_i do not depend on all variables Y_j and on all remaining parameters X_k, $k \neq i$, but only on some of them respectively. Nevertheless, the functions F_i, when it is possible to express them analytically, are mainly complex.

Let us denote the N · M-tuple of values of all variables and parameters by s, and let us call it a network solution. Generally there is an infinite number of such N · M-tuples, or solutions. Let us denote the set (infinite) of all solutions by S.

The problem of designing an optimal network, as it is commonly defined by present methods, is to find the solution $s \in S$, so that for the chosen parameter X_i the value of X_i is maximal or minimal over S subject to X_k, $k = 1,\ldots,N$ and $k \neq i$, having their values defined by sets $A_k' \subset A_k$; i.e. each A_k' is a proper subset of A_k. Thus parameters X_k cannot acquire all values from their defined sets A_k, but only some parts which are defined by the restricted, or constrained, subsets A_k'.

Some examples of present methods are the following:

1. $N = 2$; X_1 is total cost; X_2 is an average delay; A_2' is the closed interval $< 0, t_{max} >$, where t_{max} is the maximal allowed response time. Variables: channel capacity; network topology. X_1 should be minimal [1].

2. $N = 2$; X_1 is an average delay; X_2 is a fixed cost; $A_2' = const.$ Variables: channel capacity. X_1 should be minimal [4].

3. $N = 2$; X_1 is an average delay; X_2 is the reliability of a network; A_2' is an interval $< 0,p >$ where $p < 1$ and corresponds to the probability that the logical path between any two nodes still exists, even if certain numbers of branches and nodes are failing. Variables: channel capacity; network topology. X_1 should be minimal [2].

4. $N = 2$; X_1 is total cost; X_2 is a processing capacity; A_2' is an interval $< C_{min},\infty >$, where C_{min} is a minimal processing capacity of the network. Variables: geographical location of computers and terminals; types of terminals; size of computers. X_1 should be minimal [3].

In all these examples, X_1 is an objective function (one only), and X_2 is a constraint, so that an analytical solution can be found by mathematical programming. In spite of the simplicity of the given examples the solutions are not easy to obtain, because both X_1 and X_2 depend on mentioned variables, and several input parameters, i.e. constant values of some of the variables (not mentioned in the shortened examples).

The limiting value used with a constraint must be properly chosen, because it can substantially influence the resulting solution. Let us consider the following problem: $N = 2$; X_1 is a total cost; X_2 is an average delay; and transfer rate is the variable. If we minimize X_1, subject to the constraint that the values of X_2 are less than a certain constant (say t_{max}), we generally obtain a local optimum, because the variable (transfer rate) can acquire a finite number of values only. Let us suppose that the transfer rate value corresponding to the optimum value of X_1 is 2400 bit/second. It can happen that after increasing the value of t_{max} a little, the new corresponding transfer rate is only 1200 bit/second; i.e. the total cost will be reduced considerably. Of course, the solution depends on the value of the constraint, i.e.one more variable must be taken into consideration instead of the constant value of a constraint.

If the difference between two local optimum values, let us say costs, is large, then the value of the constraint parameter can be selected easily. If however, this difference is small, and especially if the optimization is performed for more parameters of X_i, then some decision-making tool is needed.

Mastromonaco [5] has discussed the loss function of a time delay, and has pointed out that it seemed to be logistic (a similar phenomenon is estimated with the ageing of data). Suppose that in a specific information system a delay of three minutes is an appropriate time for most customers to wait for an answer and to pay for it. If the delay exceeds this value (e.g. it is four minutes), some customers may not wait, thus a loss in the system is created. This loss can be calculated in the same units, and scale, as the costs. The total costs are then given as the sum of original costs plus the additional costs assigned to the loss.

In a more generalized approach the following loss function can be defined:

$$L = \sum_{i=1}^{N} w_i \cdot g_i \quad , \tag{2}$$

where w_i are weighing coefficients, which take their values from an interval (0;1) and satisfy the condition:

$$\sum_{i=1}^{N} w_i = 1 \tag{3}$$

The coefficients w_i express the influence of corresponding parameters X_i on the overall performance of a network.

The functions or transformations g_i are mappings of individual sets A_i onto the set of real numbers R:

$$A_i \xrightarrow{g_i} R \quad . \tag{4}$$

The various influences of the values of parameters X_i on the overall network performance is expressed here. If we calculate the optimum value of some parameter X_i, the overall network performance will not necessarily be optimum. In fact it can be quite far from it. Normalized functions g_i are used in expression (2), i.e.:

$$g_i = \frac{G_i(X_i)}{\max G_i} \quad . \tag{5}$$

It means that functions g_i are defined in the interval $< 0;1 >$, whereas the definition interval of functions G_i is $< 0;\infty >$.

It is clear that the weighing coefficients w_i and functions G_i can be estimated and selected only from a set of possible values in a way which is most suitable for a given system. Experience is extremely important when doing this. There seem to be no possibilities to bypass this intuitive step in the design procedure. A good system analysis could help. After all, even a bad estimate of these functions is better than none.

The optimization problem is to find $s \in S$ so that L defined by expression (2) will be minimal.

The special case occurs if for several X_i the following conditions are satisfied:

$$G_i(a) = \begin{cases} \text{const. for all } a \in A_k' \subset A_i \quad , \\ \text{otherwise} \end{cases}$$

Such parameters X_i then change into constraints. If for $i = 1,2,\ldots,N - 1$ all X_i are constraints and only X_N is an objective function, then we have a structure of one example.

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DISCUSSION

Question: If one takes into account the relationship of the users of the system, is it necessary to introduce a nonlinear cost function, containing square terms, etc? The dimensionality of the problem is strongly limited by the possibilities of the programs. Can one large program be divided into smalller programs to overcome these difficulties?

Answer: This system was not implemented. There are exact methods which are time consuming. With the results gained by a simpler reasoning one can evaluate some variables and then get an improved similar value.

Question: How can one find the topology using this optimization method?

Answer: The variables y_j, defined by numbers, give the possible topology. But there are more considerations, which have not been taken into account by mathematics, for example, the personal or political aspects. We just use the results derived from known methods to improve the overall NW example.

Comment: If you have 100 nodes and no constraints, the computation will take a long time. Then the problem of capacity assignment arises. The cost function, for example, is discontinuous for the USA; in order to optimize it, it is first made continuous, then optimized, and then restored to discontinuous form.

Answer: A nonlinear approach will give a discontinuous function which causes problems and requires a lot of computing time.

Comment: The discontinuous problem could be solved with a mixed program technique. It can be solved numerically using very many parameters.

Comment: The heuristic approach should give some results, but it was not very successful yet. Kleinrock is working at the same problem, however with little success.

Decentralized Control in Communication and Data Processing Systems

G.G. Stetsura
O.I. Aven

Decentralized control in communication and data processing systems, and new control methods are described in this paper. Many recent publications deal with information exchange between communication technology and data processing. Networks include mini and microcomputers as components. The task of drawing a line between computers and networks is rather difficult, and will not be discussed here.

The following two points are important:

1. In modern real time communication and data processing systems the distribution of resources (memory banks, computing and data exchange units) can result in conflict. This can be due to units (subscribers of the system) which may adversely affect the system operation by their own operation. These units are referred to as active units, or active subscribers. (In a classical von Neumann computer, possibilities for such conflicts do not exist.)

2. The structure of control tools for resource distribution and, particularly, for control of data exchange, defines basic system parameters: complexity, reliability, possibility for structure modifications and continuous power extension, and organization of data processing.

Rather simple and effective systems may be constructed if the system has as many active units as possible, performing control functions. In other words, the system control must either be decentralized or collective.

In the majority of existing systems, control is concentrated in one or several centers. Perhaps one of the reasons for this is past technology. The control center simplifies distributed hardware and improves its reliability. However, modern technology removes this constraint, and in some of the recent network systems control is decentralized, using two methods: messages are switched by relay stations (for example ARPANET), and in the second a Pierce-Newhall loop is used.

A third method, decentralized priority control (DPC) will be described. DPC may also be applied to loop systems to improve their control performance. It may be modified for a wide range of data transmission speeds and length of lines, from long telecommunication lines to intercomputer data exchange.

Consider several units $A_1,\ldots,A_n$ linked through an exchange medium (a line, a radio channel) as shown in Figure 1. All the units are active, any of them may require data exchange with any other at unpredictable times. Simultaneous exit from several units may produce a conflict situation demanding control for its solution. The system considered has no control center. The problem of distributing the transmission capacity of the line among units is solved directly. Units ready to transmit a message survey the line, and if it is busy (occupied), undertake nothing. As soon as it is free, all surveying units start a competition in order to occupy the line. The competitioned is organized in the following manner. Let the units numbers $A_1,\cdots,A_n$ be all distinct and in the binary code be represented by an equal number of bits. The units transmit their codes into the line. Let signals chosen for codes enable the detection a "1", when "1" and "0" are present in the line simultaneously and a "1" is not erased. The units start the competition by transmitting the higher order bits of their number. If the line contains a "1", units transmitting an "0" are cut off to wait until the line is free. This continues until all the number units are transmitted.

Obviously, if all the numbers are distinct the competition is won by a unique unit. This transmits the name of the unit initiating data exchange. Thus the conflict is solved without a control center.

A few notes concerning the codes used for organizing the line entry. It defines the unit's priority in occupying the line. In the example cited above, the higher priority belongs to that unit with the higher number. If signals "1" and "0" are interchanged, the priority shall belong to the unit with the lesser number. Priority control may be significantly expanded in the following way. Assume the priority code has the form

ATN

where the minor code bits N contain the unit's number. Field T contains time variant bits. Field A contains the unit's name (details given below). If all the units have the same values of fields, A and T, the case is reduced to the earlier example. If A is the same, and then T enables the unit's priority to be changed according to the time spent by the associated unit in the queue. The value of T must be changed by timing signals.

Decentralized adaptive control may be realized with the help of field T. Let every active unit have some constraining parameters: mean or maximal time of service expectation, maximal number of nonservice inquiries, etc. If there is a set of correctly selected parameters for active units, then these restrictions must hold for all units. In many practical cases a simple adaptive control may be realized. Let every unit with unsatisfied restrictions increase its own priority parameter

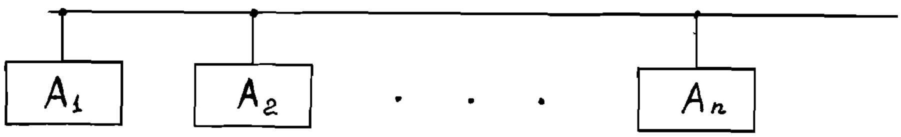


Fig. 1

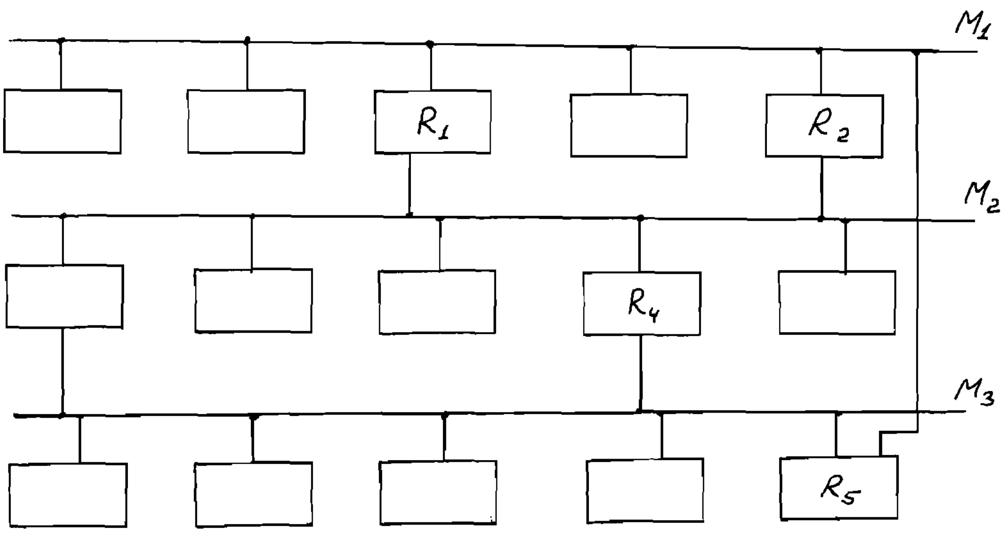


Fig. 2

by one. After taking this step the unit again checks its restrictions. If it is not satisfactory the unit again increases its priority parameter by one, etc. The entire system moves into an admissible region in spite of the units improving only their own situation. Thus some of the system control load may be distributed between active units.

Finally, a few words about field A. If A is distinct, the entry into the line is gained by the unit with the highest priority. Field A may have various structures. For example if has priority, then the unit does also. The A value may represent an even finer structure. It may indicate the priority of the fixed situation of the units, for example, the priority of a fixed message.

By joining fields, connected to distinct lines by means of relay units with simultaneous entry into several lines, complex networks may be obtained (see e.g. Figure 2). Relay units R_1 and R_2 connect lines M_1 and M_2; R_3 and R_4 connect M_2 and M_3; and R_5 connects M_1 and M_3. Some of the units may serve as retranslating units. Data transmission over diverse lines, only differes slightly from transmission with one line only.

It may be useful to introduce the concept of data exchange interrupt, authorising an interruption of data exchange of any unit on request from a source with higher priority entering the system. In a Data Processing Control (DPC) system every new priority code signal may be transmitted only after the preceding one has been received by all units. This restricts the signal transmission speeds (i.e. only a few signals may be present simultaneously in the line).

A DPC modification which deals with high transmitting speeds and very long lines uses the loop system principle. In loop systems the unit (subscriber) is passed by zones of equal length. The beginning of each zone contains a mark, indicating whether the zone is free or busy. If the zone is free, the unit changes its mark to "busy" and places its message there. Thus loop system control is local. In attempting to obtain an entry into the line, every unit investigates only its closed environment. Later advantage will be taken of the fact that there are no time restrictions in loop systems. Three alterations in the organization of a loop system can be suggested:

1. Eliminate the control center;

2. Provide the signal in the line passing the units, and therefore increase the system's vitality and response;

3. Make use of the advantages of priority control, gained by DPC.

In loop systems, the basic function of the center is to delineate the line into zones of equal length. DPC facilitates this. To transfer this function to system units seems to be complicated. The regeneration of zones, the erasure and writing of data may be directly performed by units. Initially, while the line is free, the system units attempt to enter the line according to DPC principles. The higher priority unit marks the line into zones. This system functions in a loop mode.

The main reason for including the unit's hardware in the line is the requirement for it to be able to analyse the information received and transmit the result. For example, the unit must be able to read the busy or free mark, and to change it to "busy" before transmitting the information. Hence the line must contain a memory. This, however, may be eliminated, e.g. consider the function dealing with the mark "zone busy-zone free." Let a free zone carry the mark "0", a busy one, "$1 \wedge 0$" (e.g. both signals "0" and "1" arrive simultaneously). A unit aiming to occupy a zone, by receiving the zone's mark, must send a "1"--the sign of occupation. Only afterwards may it start to analyse whether it has a right to have a zone. If transmitting a "1" the unit receives from the line a "0" (e.g. the zone is free) it may occupy a zone. The mark attached to the zone because of the interference of signals "0" and "1" becomes the mark of a busy zone, "$1 \wedge 0$". If the mark received through the line is "$1 \wedge 0$", the zone is busy and the unit has no right to use it. Obviously, a "1", transmitted by the unit does not alter the mark "$1 \wedge 0$". Thus a memory is not needed.

Now let us introduce occupation priority control. Let the message of every unit written into a zone start with its priority code. It would be desirable to give the higher priority source an opportunity to tap the zone in order to fill it with its message. This may be achieved if a unit's hardware has an entry into the line. The lack of such hardware presents a more interesting case. The way to solve this is described in the example of the occupation marker. Let a free zone be filled by a "0" signals. When the unit writes a "0" into the zone, its "0" signal is preserved; by writing a "1" the signal becomes "$1 \wedge 0$". The unit is permitted to start an analysis if it has a priority code entered into the zone. If the unit wrote a "0", but received through the line a "$1 \wedge 0$", it apparently does not have priority and must be cut off. If the signal received is a "0", it may transmit the next bit. In the case the unit writes a "1" and receives a "$1 \wedge 0$" it may also transmit the next bit. If, when writing a "1", the unit received a "0", it appears to be the most significant one and may occupy the zone. To do this the unit must be able to change the information on its own. Thus by assigning priority codes no memory is required. The examples cited trace a route for constructing various DPC algorithms for loop systems.

In effect, does a loop system necessarily need a loop? Figures 3 and 4 display open structures equivalent to a loop. The signals of Figure 3 extend from left to right in line 1 and in the oppositie direction in line 2, absorbed at both ends. At the onset of work, according to DPC principles, line 1 is occupied by the first left side functioning unit; line 2--by the first right side one. Both then perform a continuous generation of zones as in a loop system. Generation of zones in Figure 4 is performed by a least number functioning unit. Therefore the name "loop systems" does not describe it adequately. "Travelling wave system" (TWS) would be a more appropriate name. Zones pass the units with the regularity of waves and the message moves to its destination on the "peak of wave." The modification of DPC for TWS we shall name DPC/TWS.

In comparison with a loop, an open line provides an automatic clearing of needless information. It also permits non-integer numbers of zones, and part of one zone present in the line. This latter capability may be useful in data exchange between nearby plants. Below some additional examples for DPC applications are given.

The problem of organizing communication between a computer network and mobile subscribers is well established. One of the solutions introduces a random time shear into the repeated attempts of a subscriber to transmit information. The DPC offers a more flexible solution. In Figure 5 a particularly important DPC and DPC/TWS cooperation within the same system is displayed. Part (1) of a TWS line contains two TWS units B - B_i and B_{i+1}. These units can act as radio transmitters and receivers. Their active area is indicated by circles (2). The mobile radio stations C in active area B, and station B form a DPC subsystem. The units C transmit their priority codes to B, and station B retransmits them to all units C in area B. The C with the higher priority may transmit. Station B retransmits C's messages, by radio if destination C is in B's active area. Otherwise B retransmits messages by TWS. The unit C, in Figure 5 is located in the active area of stations B_i and B_{i+1} simultaneously. The neighbour stations B_j must work in different radio channels to be able to connect C_1 to one of them. DPC control units of C may be very small, less complex than an electric wrist watch. The system given in Figure 5 may be useful for governing the transporation flow on highways.

Some examples of computers and computer networks will be given. Consider a computer structure with a broad system of terminals, organized through DPC (Figure 6). The system consists of the computer and units $A_1,\dots,A_n$ of equal priority

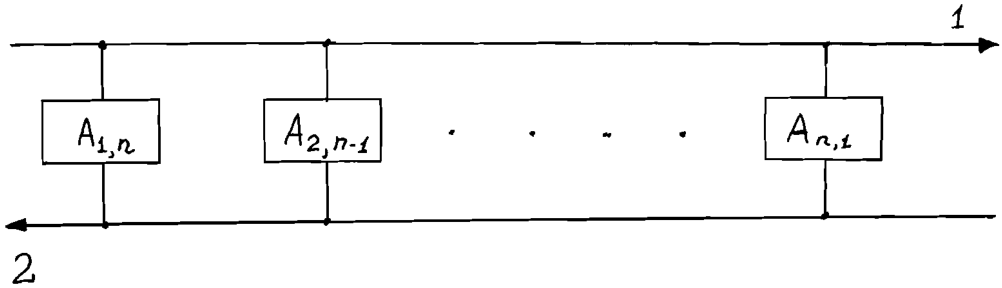


Fig. 3

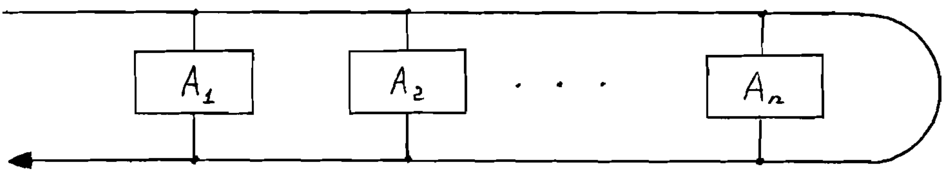


Fig. 4

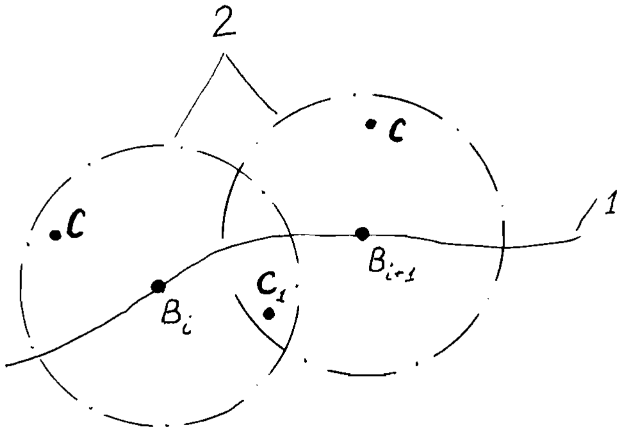


Fig. 5

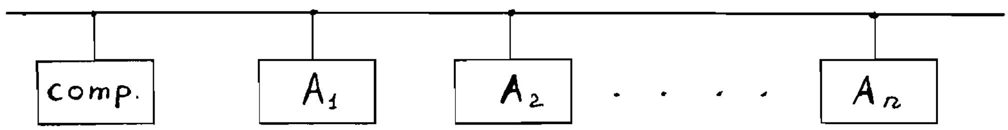


Fig. 6

Several exchanges may be initiated among subscribers without the computer. This exclusion turns the structure into a data exchange network between subscribers. Otherwise we can assume that some or all the subscribers are computers. This would be a computer network. It is of significant interest to create a DPC--based unifed interface using mini and micro computers.

The same approach may be applied to organize communication between computer blocks. Figure 7a displays a simplified computer structure. Computer blocks are linked as shown in Figure 6. The interpretation provides an overview of the conventional methods applied to define autonomity levels and cooperation mode. Figure 7b displays a second computer the same as the one in Figure 7a. By joining the channels of both computers the following may result. The flowchart may be viewed as a network consisting of two computers, or otherwise all computer blocks are linked to one another on equal terms, the structure presenting a joined four CPU computer. The next step may be achieved by introducing coalition control.

Let CPU 1 run a task, a part of which should be charged to another unoccupied CPU, in the system. The request for CPU time may be transmitted into the line according to its priority through a group request. Suppose CPU 3 and 4 are free and 2 busy with some other task. CPU 3 and 4 receive the request from CPU 1 and initiate an attempt to get an entry into the line. If CPU's priority is higher than 3's, CPU 4 occupies the line and transmits its name to CPU 1. Thus a decentralized distribution of system resources is achieved. The results hold for all the remaining system blocks, not only CPU.

Further let every block in Figure 7 have a medium-sized associative memory used to organize block communications. One associative register per block may suffice. As the block participates in the task of coalition with other units, the coalition name is registered in its associative memory in addition to its proper name. From now on, until the block is free, it is allowed to answer only those requests initiated by coalition units. Thus if, as shown in Figure 7, three tasks are processed, the resource distribution may be represented by triangles, squares and circles. This means that in the system no location may be specified where all the resources requested by a particular task are concentrated. Consequently, this control mode creates structures with a high degree of integration. It may prove to provide a simple mode of informing the system units of the arrival of new data. The informing process may be used to create an interaction in data processing, more active than the one provided by conventional interrupting systems.

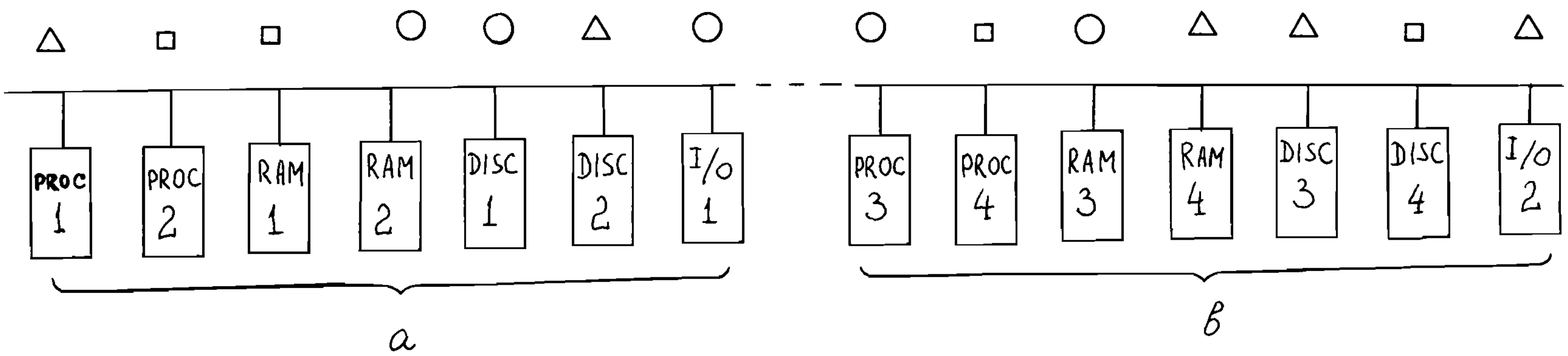

PROC 1
PROC 2
RAM 1
RAM 2
DISC 1
DISC 2
I/O 1
PROC 3
PROC 4
RAM 3
RAM 4
DISC 3
DISC 4
I/O 2
a
b

Fig. 7

In conclusion let us enumerate some problems which must be solved for DPC to be of practical use:

1. Analysis of concrete DPC and DPC/TWS applications. The solving of control algorithms and the evaluation of their efficiency.

2. Solving the decentralized support of reliability methods.

3. Analyzing the characteristics of integrated data communication and data processing systems similar to those in Figure 7 (e.g. it may be useful to investigate applications and modifications of minicomputers in relation to the structures in Figure 7).

4. Solving the electronic circuits of DPC control units.

DISCUSSION

Question: If there is a loop, would it be possible for more than one device to transmit at the same time, as long as each device does not transmit more than a fixed amount?

Answer: Yes. In the first system described, there is a line and some packets are moving along the line, packet 1, packet 2, etc., which is exactly what is desired.

Question: In the diagram shown, symbol A_1 was being read and symbol A_2 was being written, if that wire is a single electrical conductor how is the symbol A_1 removed from the line without interfering with A_2?

Answer: There are a number of ways to do this, the simplest one is to cut off the line.

An Adaptive Routing Technique for Channel Switching Networks*

D.E. Bell
A. Butrimenko

Abstract

The performance of a communication system depends greatly on the technique used for finding good routes for transmitting information. In order to improve the robustness and reliability of such systems, a decentralized or distributed technique to control information flows is proposed.

The routing technique to be described here is based on the principle of distributed control. We consider a network in which every node (exchange) receives control information only from its neighbouring nodes. In every node there is stored a special routing matrix and every node estimates continuously the probability of the trunks to each of its neighbours being blocked, that is, being unavailable for transmission, which will depend on the congestion in the system. The routing matrix at node i has entries W_{ij}^{k} which represent the estimated probability of a message reaching destination k from the neighbouring node j. Clearly, when routing a message to destination k the node j, to which it should be sent next, is that which maximizes W_{ij}^{k} among those nodes j for which trunk (i,j) is unblocked. If all trunks are blocked, the message is considered to be lost.

To summarize, then, an algorithm for such networks is presented which converges to the optimal routing policy, under the assumptions:

i) only local information is available; and

ii) the blocking probabilities remain constant.

*Published as an IIASA Research Memorandum RM-74-18, October, 1974, Laxenburg, Austria.

In practice these assumptions will be fulfilled if the traffic is light, in which case the blocking probabilities will be approximately constant. In other cases this policy may not be optimal since the blocking probabilities will depend partially on the routing policy. The behaviour of this algorithm in this case is investigated but it seems likely that it will yield a good if not optimal policy.

DISCUSSION

Question: Are the probabilities given for every node, or for the links?

Answer: There is a matrix, in which each node has a corresponding row with its respective probability.

Question: Does the algorithm converge fast enough to update the routing table every time?

Answer: If it is done too often, the capacity will be used for the updating, and blocking will occur. It has not been investigated sufficiently.

Comment: It is a classic result that the stability of the convergency of an iteration has a tremendous influence on how fast it converges, especially if a closely associated group of values exists. This can occur especially if one has a cluster of values. In your paper the P_{ij}'s have been spread to the range (0, 1). In theory, they are allowed to lie in that open interval, however, if they are all put near one end of the interval, they may not converge. What happens if the status of the P_{ij}'s changes very abruptly? If it changes smoothly, there would be no great changes in the probability vector. One wants to see how stable the thing is in the presence of an abrupt change from nearly 1 when the line is working, down to nearly 0 when it is broken. Because of the suspected clustered values the thing might oscillate. Perhaps in the first steps, one does not need the best path, but only a good approximation.

Question: If you have sent a message down a certain path, and it has broken, how many messages will be sent down that path before one notices that it is broken? Perhaps the transmission should be slowed down, and only one message should be sent to see if it has recovered.

Answer: We have not worked on that question yet, but have concentrated much more on the convergence of the algorithm itself.

Comment: The main target is to show that different algorithms converge, and to choose the most suitable.

Question: The "moral" basis of the routing algorithm is trying to minimize the probability of message loss, or maximize probability delivery. Under which circumstances will a message be lost? Only in the case where the destination has been cut off from the rest of the NW? Routing algorithms normally assume that the NW stays up, in order to maximize throughput. In your NW the assumption is that if falls apart.

Answer: We assume that a call gets lost when there is no free channel for it.

The Computer Conferencing System of the Institute for the Future

Jacques F. Vallee

The Institute for the Future began its investigation of teleconferencing in order to improve the methods by which experts from diverse fields address problems in social forecasting and technology assessment. In March 1971, the Institute proposed to the Advanced Research Projects Agency:

> "....to explore the applicability of on-line group conferencing for policy formulation via computer terminals. The key goal of such conferencing will be the effective use of judgemental data as input for forecasting, planning, and decision-making, where the participants are geographically separated."

To create an advanced form of teleconferencing, we have now implemented and tested a computer system called FORUM. Its basic objective is to allow unhampered interaction of participants under the guidance of an organizer who defines a topic of discussion, assembles a panel of participants, and presents the relevant material. Each participant establishes communication with the computer network via a portable terminal with a standard typewriter keyboard. FORUM is able to convey questions and answers, assemble group opinions, protect anonymous statements, and supply other information to, and within, the group while the organizer monitors the proceedings and intervenes as necessary.

In order to illustrate the interaction made possible by FORUM, imagine a hypothetical discussion among a group of experts on the projected availability of mineral and energy resources, 1980-1990. There are about 20 participants. Among them are planners, economists, geologists, and petroleum experts. Two are specialists in computerized data bases. In addition, there might be representatives from power and utility companies, and the president of a mining corporation. The conference organizer has experience in dealing with groups, and is familiar with various techniques used to draw out forecasts and intuitive judgements in technological areas.

This hypothetical conference differs from the usual workshop in that the participants are not meeting face-to-face. Instead, they are geographically separated and use a variety of communication media. Some are sitting around a terminal in a Washington D.C. office building. A geologist is in the computer room of the Branch of Computations of the U.S. Geological Survey in Denver. One of the economists is in his office at Stanford University. Another one may be sitting in his study at home in New Jersey or in London, for that matter. (These experts are in telephone communication with a central operator who can instantly advise them of the status of the conference, of the progress of work done in subcommittees, or of the reasons for any particular difficulty or delay.) The substantive part of the interaction takes place through entries typed on standard terminals. They are connected to the network and controlled by a computer.

This is the capability of FORUM. The research issues addressed in creating this fall into two categories. First, from a computer science viewpoint, the central problem reduces to that of identifying, defining, and implementing a range of structures under which the participants are able to share information and enter comments into a common computer-storage file. Second, from a human-factors viewpoint, our research has identified certain basic principles of computer communication, and created methods for displaying, in a meaningful way, both the contents and the dynamics of this communication. This task raises some unusual design problems: a group of experts, or decision-makers typically does not have much knowledge of, or interest in, computer technology per se. There is no opportunity to train them in the use of a text-oriented language before the conference. It is not feasible to ask them to interface with their peers through information specialists, because each participant has a unique awareness of the problem at hand and needs to experience direct contact with his data and with other participants in order to perform at the "cutting edge" of his thinking.

As a teleconferencing system, FORUM has unique characteristics not typical of other systems. Of primary concern in its development have been the techniques of translating the perceived flow of a face-to-face discussion into the medium provided by the computer. The information flow in a synchronous conference is now well displayed by FORUM. The ability of a participant to join an activity in an asynchronous manner created an unforeseen demand on the system: the need to review past entries, add new comments or ideas, or suggest changes, for example, plays a more significant role than had been anticipated. As a result, it has been necessary to expand the range of work styles available to users.

When a group communicates via FORUM, each participant uses a terminal which has been logged into a computer network. The user is presented with a list of discussions which he can attend (just as he would if he were to walk into the lobby of a convention center to review the day's program). Having selected an activity, the conferee is given a short background statement describing it. He is then free to observe the ongoing discussion, to review past comments, or to start typing his own remarks. At any point during the discussion, a conferee can send a private message to another participant or make an anonymous entry. All of these can be entered without the participant's having to learn a single command, thus avoiding a major problem of most interactive systems in existence; namely, that system commands get in the way of the participant and clutter the transcript with extraneous lines meaningful only for the machine.

An important facet of FORUM conferences lies in the ease with which the participants have access to services outside the discussion: they can, for instance, submit a prepared statement to the rest of the group or inset parts of the discussion into a personal file. They can also draw responses from a data-base system and enter them into the general discussion. Clearly, the level of interaction thus reached is one not found in face-to-face meetings where experts usually are cut off from their files and personal notes.

The initial tasks of the FORUM project included an analysis of available resources, and a review of existing terminal technology in terms of character set, plotting symbols, size of frame, speed of presentation, and interface standards. A decision involving the programming language to be used had to be made early; after exploring the languages available on the PDP-10 under the TENEX operating system, we reluctantly concluded that assembly language was the only suitable medium to gain access to shared files and to control terminal behavior, both functions being critical to our goal. Additional requirements were speed and low central-processor utilization.

Finally, a major requirement has been to master techniques to adapt the behavior of the system to the demonstrated skill of the user. This task has two aspects:

1. introducing an "intelligent" mechanism in FORUM to recognize the user's success at learning the functions of the system, with a provision for downgrading this skill when the functions are not used over a certain period of time; and

2. learning how to phrase information, as well as system prompts and responses, in a way that is adapted to the user's skill for a particular function and takes into account that this information may or may not have been presented previously.

Based on this mechanism, FORUM makes decisions that match the user's demonstrated expertise, and it adjusts the degree of verbosity of its response to suit the needs and experience of the human participant.

The FORUM system is still very primitive in several respects: first, the implications of synchronous (simultaneous) conferencing remain largely unknown, both in terms of computer support of thought processes and in terms of user behavior; second, the possibility of multimedia (audio and video) adjuncts to computer conferences has hardly been explored.

Much of our work to date has consisted of inventing data structures and access mechanisms suitable for the use of computers as a communication medium. We have encountered two major technical obstacles in this respect.

1. Ordinary computer architecture creates severe limitations for our conferencing needs. Teleconferencing will require a revision of the organization of various computer resources to make efficient communication among many users possible.

2. Ordinary concepts in file processing, that rely on separation of user files and access paths, do not apply in the conferencing environment. We have found that a single user, namely FORUM itself, had to have complete access control. This leads to an unusual situation in terms of privacy, protection, and accounting.

An extremely rich domain for further research has been identified in the course of this effort. The specific computer-science questions it addresses (leaving aside the entire field of social impact and psychological reaction to the medium) follow:

- How can conferencing be interfaced with other functions in the areas of text editing and fact retrieval?

- How can a mechanism be provided for the identifaction of semantic threads and subgoals in a discussion?

- How can the knowledge of man/machine interaction be adapted, or extended, to a nonclassical situation of computer-mediated group communication (man/machine/man interaction)?

- How can a conferencing program be offered at a very high level of reliability over a network where actual processing of the conference activity becomes machine-independent?

When computer communication becomes more widespread, we foresee a number of new problem areas that deserve careful exploration. One of these involves the coupling of conferencing activities to other functions, such as document preparation and publication, that are equally vital to the management task. Another is the definition of retrieval functions capable of operating on what is basically an unstructured, unindexed data base. Yet another area deals with the adaptation of the computer to an office environment.

In considering the last point, it is useful to remember that management information systems have largely failed because their assimilation in the framework of an executive's activities presents too high a threshold in terms of training, usage patterns, and restriction of information types. Reliability and privacy are the main requirements for a successful system, and these must be addressed immediately. It is unlikely, however, that certain constraints--for instance, the need to type on a keyboard--will be removed within the next ten years. Human aspects of software design have been generally neglected; under these conditions we feel that considerable attention should be devoted to psychological and social factors before introducing teleconferencing into a large, operating environment.

Another aspect that requires further examination is the trade-off between travel and communication. Writing in the February 1974 issue of Telecommications, Paul Polishuk has stated that:

> "The Office of Telecommunications believes, based on analyses conducted to date, that as much as 5 percent of the total annual petroleum consumption in the US can be saved by substituting telecommunications for the transport of people and goods."

While our own project intends to explore these substitution possibilities further, experience with the FORUM medium leads us to consider as equally exciting the opportunity to regard computer conferencing as a genuinely new way for groups to exchange ideas, to do planning, and to arrive at decisions. If a communication framework can be maintained, and even enhanced, while members of a research group or managment team are physically separated by large distances, then a major change in work styles and intellectual patterns seems to be possible. Such a change may be even more significant in the long run than the energy savings which teleconferencing may provide in the near future.

DISCUSSION

Question: What would have been the costs of this conference if it had been held in teleconferencing style? In this face-to-face modus it costs approximately AS 300,000.

Answer: This is difficult to answer, because technology cannot supply everything yet. We only have estimates. A Bell-Conference costs less than $ 5,000 but to bring people together costs $ 22,000. Besides, teleconferencing is not suitable for every purpose. In a problem solving conference a face-to-face situation might be better; when only information exchange is needed, teleconferencing might be preferred. But in general, one can say that the system costs $ 16 per hour, $ 6 of which are for the computer, and $ 10 for the communication (fixed). These costs may be reduced by a factor of 2 in the next few years.

Question: If one sets up a three week conference, how long will the participants be using the terminals?

Answer: It depends very much where the terminal is located, whether at home, at their desk, or if borrowed. It may be from a few minutes every day, to two hours per day. It also depends on the theme; whether you have to do transcripts or only answer questions. Usually there is some delay sometimes a whole day. Special arrangements must be made to have a simultaneous conference. But in teleconferencing the probability of getting real and correct answers is much higher.

Question: How much did the system design cost?

Answer: We had a preliminary system and rewrote it on time-sharing. It cost 4 man-months (first it was on a telex basis). The design of questionnaires took a long time.

Question: Are there any difficulties or complaints in the structural situation?

Answer: An electronic blackboard would be ideal, because until now we can only send the pictures through mail or keyboard. The input of curves and graphics is very limited and time consuming. Designs are at the edge of capability, and maps, as needed by geologists cannot be coped with at all.

Comment: An obvious addition would be a facsimile device. There is no problem at all in using packet switching to transmit facsimiles and have them recorded in the dialogue so that they can be referred to. Kirstein has been experimenting with facsimile devices in his laboratory.

Question: How long does it take to teach people how to use the system?

Answer: First commands are taught, then access to the system was made with keywords, passwords, etc. Then, at last, a trust level, between the person and the system had to be gained. For example, a one day conference was held where seven institutions took part. The preparation started one month before, to teach the participants how to use the system. When they are already in the system, the training time will take ten minutes. If artificial intelligence techniques are included and a retrieval function is used successfully, skill will be decremented.

Question: Which methods were used to connect the participants of the Bell Canada Conference?

Answer: The participants were connected through ARPA, the London participants were connected by Kirstein and the Canadians had a line to Boston.

Question: What were the communication costs of the conference?

Answer: It is not known as ARPA does not have any cost calculation. However, the total number of connected hours was calculated, in order to see what it would have cost in time sharing--about $ 60,000.

Question: There are several kinds of conferences where real time actions are very important, and time is vital, for example, negotiations. How is this solved in teleconferencing?

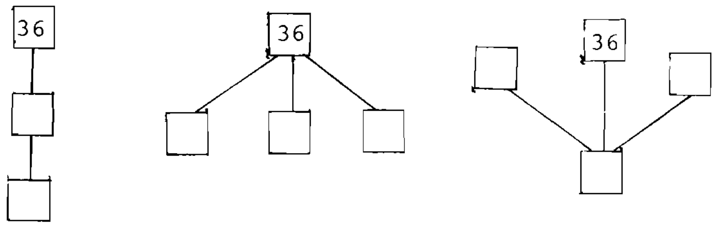
Answer: There is no answer to this because there has been only one conference until now. However all the entries made on Tuesday, for example, could be saved thus integrating this day. The chairman could edit them giving a summary of what was said. We are now exploring active synchronous discussions, and experimenting with social simulation games played on a computer NW. One great disadvantage of computer teleconferencing is that feelings could not be conveyed. But it is possible to read between the lines and learn from the person's reaction.

Comment: It is well to say that by analyzing the semantic content of the messages, a psychologist or similar professional can improve the emotional content. Surely this is very different from a face-to-face meeting in which not only the psychologist is aware of the feelings, but also the participants who can recognize emotional reactions from the manner of speech and faces, and thus modify their behavior. And that, it seems to me, that it has a great effect on the proceedings of a conference.

Answer: Yes, but in teleconferencing another aspect enters into it. In a face-to-face conference you are not aware of the semantics, in teleconferencing people get frustrated, they do not know why, it may depend on the content, or what is implied in the words.

Comment: The effect of the responses will depend, to some extent on their sequence and time of completion. When a provocative message comes, you will have a rush of answers.

Answer: If you look at these diffferent topologies, you see that the answer depends very much on them, that a whole answer system may be triggered.



Question: Is it possible to identify people in conferences--the analysts, who may provoke discussions, the synthesizers who may have a calming effect? It should give interesting results if one analyzes, first, the person's initial responses and then how they may have changed during the conference.

Answer: There may possibly be behavioral differences. One can put any idea on record during a teleconference, and pick it up later; whereas in a face-to-face conference you may perhaps not wish to disturb the speaker. Besides, anonymous entries are possible in teleconference, which can create new aspects.

KUIPNET: Inhouse Computer Network for Information Processing and Some Applications

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I. INTRODUCTION

The word "information" is used in this paper to mean any of the following:

1. Data, program in coded form;
2. Wave form, such as speech wave;
3. Line drawings, figures, and gray-scale or color photographs.

An information processing system handles these different types of information, therefore, transmission rate greatly varies ranging from 50 bps to several hundred Kbps.

Information processing activities in our laboratory may be better understood in relation to the history of digital computers in Japan. The first stage was from 1957 to 1966, when the input to computers was only by punched tape and card. These input media were exclusively prepared by humans. To improve this situation we tried to give the computer human-like perceptional abilities, which enabled it to accept and understand speech, characters, figures, and pictures without difficulty. These studies revealed not only the advantages but also the limits of the Von Neumann type digital computer.

The second stage was the period from 1967 to 1971, which corresponds to the period of development of TSS computers. Typewriters as TSS terminals were insufficient for the great range of information processing. Speech sounds, characters, figures, and pictures, all these different information media essential for human beings, must be accepted by an on-line computer. The system we thought most powerful was conversational information processing in on-line, real-time mode with powerful question-answering capability. We introduced minicomputers to control terminals for handling various kinds of information. Eventually a communication network among these distributed intelligent terminals was established.

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The third stage was started in 1972. Many computers are connected effectively by a telecommunication network, each component of which has its own autonomous function. Here, the study of the balanced harmonious behavior of each autonomous intelligent terminal and that of the total system involving human factors is most important. We have to make an effort to develop varieties of I/O devices and new concepts of CPU.

The main projects now in progress are as follows.

a) Automatic Speech Processing

Speech recognition is the major aim. To further this efficiently, an experimental system is developed which can both analyze and synthesize speech by utilizing the zero-crossing wave technique. This system can process a great amount of speech data with least human intervention. Not only models of speech-wave-form analysis and selection of effective parameters for recognition, but also factors of individual and linguistic properties of Japanese speech will be studied.

b) Pattern Recognition and Picture Processing

Pattern recognition and picture processing of handwritten characters, printed Chinese characters, line drawings, gray-scale photographs will be advanced further using a TV camera, a flying spot scanner and other on-line devices. An efficient total system of input and processing has been developed. The study also will be extended to include such problems as machine judgement, inference, and action by the information of input pictures.

c) Mechanical Processing of Natural Languages

The fundamental studies on natural languages will be advanced in every stage from statistics and syntax to semantics for the purpose of establishing a system to understand natural language. The study also included mechanical inference and deduction, and heuristic problem solving related to human activities.

These separate projects are now going to be combined to form an inhouse computer network: several computers are mutually connected to realize a distributed intelligent system. The hardware and software problems in such a network, especially for small-scale laboratory systems, are being studied. On the other hand, the traffic phenonema and queueing problems are studied theoretically and by simulation experiments.

Since the very early stages, we have been using computers for non-numerical computation. In the programming of numerical calculation one communicates with the computer by "characters." We have been interested in "information" processing: computer processing of not only characters but also speech waves and photographs. "Does the usual TSS have sufficient and appropriate ability to process speech waves and photographs?" This question has led us to the development of the computer complex and intelligent terminals.

The computer sytem in our laboratory includes almost all the computer terminals and I/O devices except for a graphic display unit with a light-pen. With the store-and-forward method for communication between a computer and terminals, the total cost of the terminals will be very large if each terminal is equipped with its dedicated buffer storage. "Is it possible for a group of terminals to share the memory storage of a minicomputer as the common buffer storage?"

If a certain job needs interactive usage of more than one terminal, it requires switching from one terminal to another at an electronic speed. In other words, the terminals are to be exchanged automatically as the program runs. This characteristic is also essential for dynamic and efficient man-machine interaction and is similar to the role of an electronic exchange device in the telecommunication system. "Is it possible for a minicomputer to be used as an exchange device for various terminals.

To answer the above two questions, we have developed a computer complex system for picture processing: it consists of a minicomputer combined with several graphical I/O devices; the minicomputer is connected with a medium-size general purpose computer as the main computer.

A special controller was designed and constructed to exchange information among I/O devices and control them with the instruction from the minicomputer. The high-speed capability of the minicomputer enables it both to control the devices in high-speed real-time, and to process the pictorial data using its own memory storage and processing unit. The minicomputer in our system handles buffering and processing. Thus, it satisfies the minimum requirements for an intelligent terminal.

In order to increase both the efficiency and cost performance, it may be necessary to distribute the power of processing from the central processor to the terminals. The concept has yielded the "intelligent terminal" system before appearance of the concept of computer networks. In this system the terminal itself has a processing unit and memory, thus intelligent tasks can be performed also in the front terminal. This type of system is expected to greatly improve the efficiency of the whole system, and to open a new field of computer application.

In the case of computer networks, such as ARPA, the power of the computer is to be increased through the intercomputer connection without affecting their own operating systems. In contrast, the above mentioned intelligent terminal has evolved from the research activities in information processing considering the following three principles:

1. on line use,
2. man-machine interactive use, and
3. resource sharing of computers.

We think that the prospective computer network can be used not only for keyboard communication but also for the transmission and processing of data from speech waves, drawings, and photographs. Furthermore, a computer network does not need to cover the entire nation. Indeed, we think a computer network limited to a building or campus also has merit similar to PBX in telephone networks, which is able to connect public computer networks.

II. OUR INHOUSE COMPUTER NETWORK FOR INFORMATION PROCESSING

For the reasons described above, an inhouse computer network is constructed. The specifications of the network depend very much on what kind of objectives are emphasized.

A. Requirements for an Inhouse Computer Network

1. It is easy to use and accessible practically at any time with quick response.
2. It offers economic advantages in both initial and operating costs for information processing research by sharing existing files and peripherals of the member computers in the network.
3. Messages made of characters or commands are transmitted within a reasonable time-delay by employing message switching (store-and-forward).
4. Furthermore, it effectively assures high-speed transmission of a large volume of raw data by line switching methods. This is essential to the research of speech and pictorial patterns.
5. It is open ended to the outside: not only to inhouse, but extendible to beyond the laboratory. A public communication line (48 Kbps) could be connected.
6. Interconnection of computers is possible without MODEM up to a distance of 300 meters.

B. Specifications of the Inhouse Computer Network

1. IMP (Interface Message Processor) is a minicomputer having the capability of store-and-forward message switching to handle character messages. Furthermore, real-time messages, e.g. raw data, are handled with higher priority over regular messages: the user of the real-time process is guaranteed effective data transmission at more than 200 Kbps.

2. The HOST-IMP transmission is bit-serial and half duplex. The bit rate of transmission is up to 1 Mbps. Four coaxial cables are used to connect a HOST and the IMP. The number of cables for the connection should be decided on the basis of costs of the communication control devices and cables.

3. For the moment, the IMP at the center of the star configuration serves five HOSTs.

4. A HOST computer is able to execute multi-processes at any time. Since each HOST is a minicomputer, not many users can time-share it, but it is able to serve at least two users; one communicates with remote processes through the network, and the other uses the HOST.

5. File, which is one of the most important resources shared by many users, is assigned and controlled by FCP (File Control Program) of the HOST which has the file.

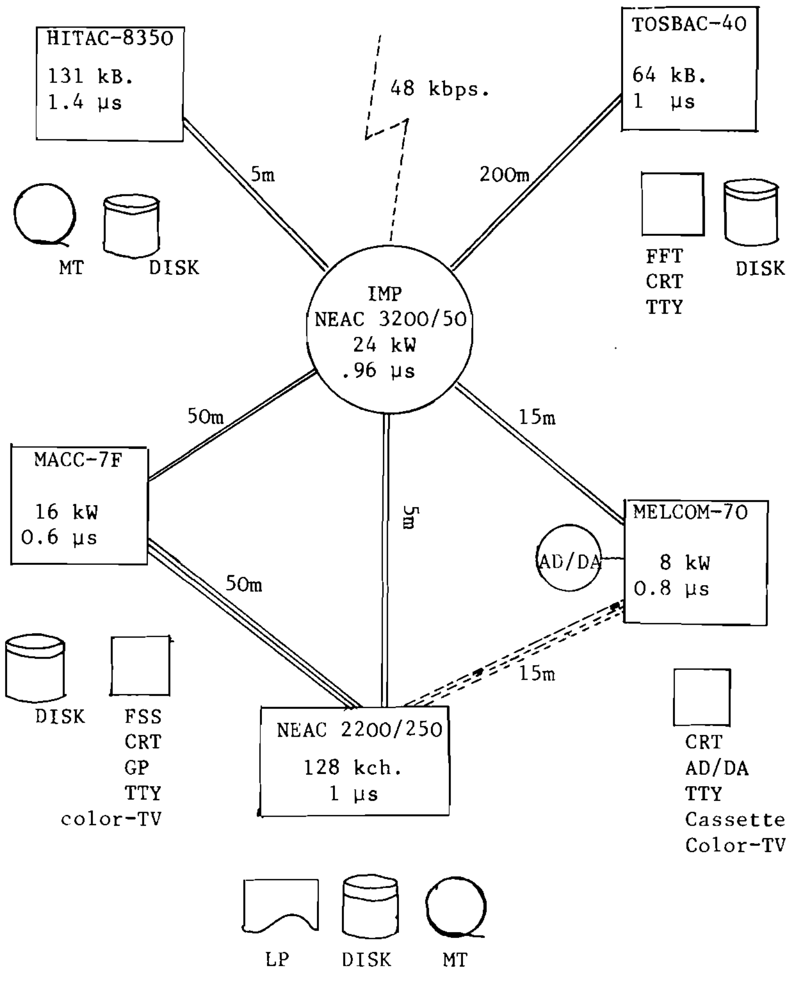
6. Peripherals (line printer, high-speed puncher, and special I/O devices) as well as files are controlled by the HOST monitor for common use.

7. Software resources, for example, a Fast Fourier Transform Program, LISP, are also available for use through the network, as well as special hardware devices such as a flying-spot scanner, speech synthesizer, etc.

C. Configuration of the Inhouse Computer Network

The configuration of the inhouse computer network is shown in Figure 1. Fundamentally, HOSTs are connected in a "star type" to a message-switching computer IMP. The maximum distance of direct connections between IMP and HOSTs is 300 meters. The types and characteristics of the computers in the network are as follows:

Inhouse Computer Network "KUIPNET"



4 Coaxial Cables

64 Coaxial Cables

FSS : Flying Spot Scanner
GP : Graf Pen
FFT : Fast Fourier Transform
TTY : Tele Type-Writer

Figure 1. Organization of KUIPNET (since April, 1974).

NEAC 3200/50: A message-switching computer IMP; 24 kw (16 bits/w) core memory, 0.9 μsec. memory cycle, Direct Mutliplexed Channel (DCM) with priority control to which the computer communication controllers are connected. Equivalent to Honeywell DDP 516.

NEAC 2200/250: A middle-scale character machine, which is equal to Honeywell 2200. DISK, magnetic tape, LP and shared file under file control program. Multi-programming OS for network is almost finished.

HITAC-8350: A micro-programming computer with multi-programming operating system. DISK, Magnetic tape, card reader.

MACC 7/F: A minicomputer for the pattern recognition and picture processing group, 16KW (16 bits/w) core memory 0.6 μsec. memory cycle, on-line control of FSS (Flying Spot Scanner), Graf/pen, CRT, Color TV display.

MELCOM-70: A minicomputer for the pattern recognition processing group; 8KW (16 bits/w) core memory, 0.8 μsec. memory cycle. A/D converter, D/A converter, CRT, Casset MTs.

TOSBAC-40: A minicomputer for the natural language processing group; 64 KB (8 bits/byte) core memory, 1 μsec. memory cycle. DISK, CRT, input and output equipment of picture.

The connection of HOSTs and IMP is half-duplex via four coaxial cables, and the maximum transmission rate is 1 Mbps. MACC 7/F and MELCOM-70 in this network deal with a large amount of picture and speech data, and may request to store the data in the file of NEAC 2200/250. In this case, continuous transmission of more than 200 Kbps is assured. This network may be connected to remote computers through a 48 Kbps data line of the Nippon Telephone and Telegraph Corporation.

D. The Software System of HOSTs

The Software System of the HOSTs should be constructed so that a user is not inconvenienced due to the network's mode of operation. HOST's software system has the following capabilities:

1. Time-sharing processing of multi-process;

2. In the case of raw data, a very quick response is to be assured. When a continuous speech sound is fed into the computer, it must respond every 50 μsec. So the real-time process dominates over normal processes. This preference, however, does not last long--typically 1 min for speech data, and several tens of seconds for one frame of picture data.

3. Nonreal-time processes that do not require such a quick response are slowed down while a real-time process is being executed, but this rarely occurs, so the user of a lower priority process is not often inconvenienced. In the case of a minicomputer HOST with a small main memory, it is more feasible for a small number of processes to time-share the processor: the number of users who can use the HOST concurrently is two or more.

4. At the minicomputer HOST, a user can write a program via the teletypewriter under the control of the time-sharing monitor. He can edit his source program, assemble it, and store the object program in his file in NEAC 2200/250. He can debug the program by means of debugging utilities

E. Message Transmission by IMP

We selected a NEAC 3200/50 (DDP-516) for the IMP processor. This machine has 24 kw of core memory (16 bit word, 0.96 μsec. memory cycle), 250 μsec. interval-timer, 16 multiplexed channels with priority control, and a memory protection mechanism. Information is transmitted in bit serial mode between the IMP and HOSTs. The unit of transmission is called a message of cira 8000 bits in length.

There are two different types of messages that pass the IMP--characters and raw data. Messages of characters include source programs, object machine codes, line printer output, etc. A message of raw data is data of video sampling or speech. Messages of raw data must be stored in real time into the file. The continuous speech data should be transmitted at more than 200 Kbps average rate (20 kHz sampling of 10 bits). Therefore, they are given a higher priority than the ordinary data character.

For this purpose, the priorities of data channels have to be dynamically changed by the demand from the arbitrary pair of HOSTs. But, priority control of the multiplexed channels of NEAC 3200 is the fixed one, that is, the level of priority is assigned to one of the sequential channel numbers.

Our home-made hardware, named "Variable Priority-Specifying Circuit," intervenes between the multiplexed channels of NEAC 3200 and the communication device to HOST, and it controls the priority of data transmission passes.

Our hardware keeps the priority in the interval between the initiation and close of one communication link, but the passes with lower priorities are never prohibited in that interval. Its behavior is not at all similar to that of the so-called mask-circuit used in the usual priority control. The channel given the top priority transmits data at the maximum bit rate, 1 Mbps.

III. SOME APPLICATIONS

Since December 1973 the KUIPNET (Kyoto University Information Processing Network) has been working as the resource-sharing network connecting mainly three HOST computers (see lower part in Figure 1) in our laboratory.

This network is intended to supply augmented facilities for information science research. The experiment of sending digitalized speech and picture data in real-time through the network has proved the high-speed data-throughput of the network.

The processing performance of the minicomputer HOSTs was greatly enhanced by sharing the resources (files, peripherals) with the other computers. The minicomputer HOSTs have been also provided with a new utility for efficient program development using the network.

A. Multi-Feature Display and Speech Response In On-Line Signature

An operator signs his name using a Graf-pen (sonic digitizer), and the successive coordinate data are fed into minicomputer MACC 7/F. It displays the signature pattern on the CRT, while calculating the number of strokes, the speed of the pen movement at every point, and the time spent for signing. These data are integrated as a message in MACC 7/F, which is then dispatched to NEAC 2200/250 through the network in order to make it edit into sufficient data for voice answering. MACC 7/F displays, besides the original signature, a color modified one according to the pen speed on a color TV display: bluish for slowly written sections and reddish for quickly written sections. Under another modification, each stroke is assigned with corresponding colors. Using not only the original figure of the signature, but also the above-mentioned emphasized pattern and audible ones, one can easily distinguish the individuals who signed on the on-line system.

NEAC 2200/250, after receiving the data of strokes and speed of the signature, compiles the digital data to voice-wave form retrieving the necessary data from the file of digital data series corresponding to phonetics.

MELCOM 70 receives the digital speech data from NEAC 2200/250 via IMP. These digits, after code conversion are synthesized into speech sound by a D/A converter.

The picture data are displayed on the CRT from the minidisk of MACC, are also transmitted through the network to be displayed on the MELCOM CRT.

The files of the HOSTs, except NEAC, are so small that a large amount of data of speech sentences for voice answering, which are high-quality A/D converted digital-voice, data, cannot be stored in the local file. NEAC 2200/250 supports not only the file medium for other HOSTs but also the job of editing the sentences for the response.

MELCOM 70 synthesizes speech sounds. It can reproduce natural female voices of good quality when the same D/A converting clock as A/D conversion is used: 10 kHz clock/10 bits. Besides, by changing the reproducing D/A converter clock interval, voices like that of a child or a man may be heard from the speaker even during the network digital transmission period.

The following minicom-mode job can be done concurrently with the network-mode job. The minicom-mode job is an editing of source assemble-language programs. A user at a teletpywriter types a program made up of a series of instructions and stores them into the computer. Then he may get the list of the program and correct some errors or add/delete instructions by giving simple commands from the typewriter.

B. Discussion on Real-Time Processing of Speech Data Using KUIPNET (Figure 2)

KUIPNET is intended to be convenient both for picture processing research, which handles a large volume of raw data, and for speech processing research, which imposes a requirement on real-time (constant delay or line-switching type exchange) data transfer. Therefore the subnet (IMP-HOST) assures the high-speed data transmission up to 1.6 Mbps.

The delay time in the real-time transmission of speech information and the processing time at each part of the network is discussed. For example, we consider the case that the speech data, stored serially in the disk file at the HOST computer, NEAC 2200/250, are sent to MELCOM through the IMP, while the D/A converter connected to MELCOM reproduces speech wave forms from the transmitted digital data. The important thing is that NEAC 2200/250 and MELCOM 70 have the different type of word length; the former is a 6-bit character machine and the latter is a 16-bit word machine. So the data, bit-serially transfered, need code conversion at either side of two computers. Here we send two characters per one sample to MELCOM and it converts one analog speech amplitude into one word.

Figure 3 illustrates the transmission path of the speech data. The all speech data in the NEAC 2200/250's file whose record size is equivalent to a message length (1280 chs) are accessed under the control of FCP (File Control Program), and the messages are immediately moved to a different area of the NCP (Network Control Program). Consequently the data flows 1, 2, 3 in Figure 3 take place concurrently. IMP usually reserves several buffering areas for the store-and-forward message switching; thus the inputs 3, 4 and outputs 5, 6 are processed at the same time.

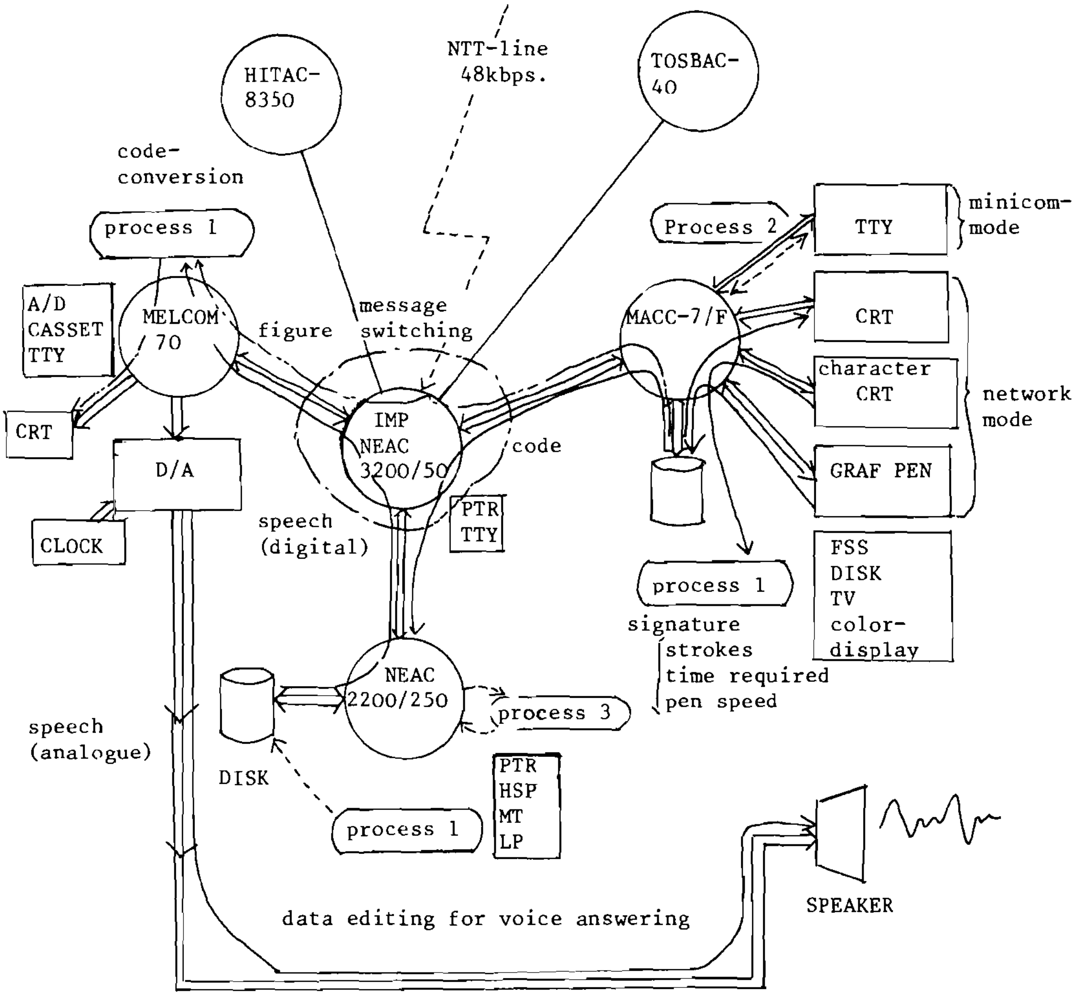


Figure 2. An Experimental Usage of KUIPNET.

The arriving data at the MELCOM NCP 7 are, after the code-transformation 8 mentioned above, fed alternately into the double-buffers connected to the D/A converter. 9 is the process supplying D/A with speech data.

At every 100 μsec. (normal clock interval) the computer transmits 1 word to D/A. Thus one message (640 Ws = 1280 chs) can reproduce a speech sound which lasts 64 msec.

Table 1 shows the transmission and processing time of a speech data message at every stage. The third column shows the ratio of the time required at each stage to the real-time speech duration (64 msec).

The processes [1], [2, 3, 4], and [5, 6, 7, 8] are supported by double-buffers; NEAC 2200/250's NCP, IMP's buffers and D/A's double-buffer respectively. Therefore they can take place concurrently.

In order to produce a speech sound without discontinuities or gaps, the sub-totals of the time required at individual processes must be smaller than 64 msec. In this case their ratios to the real-time are 0.36, 0.26, and 0.53 respectively. This fact shows the real-time transmission of speech data is assured by this system. Moreover this fact was practically confirmed by changing the D/A converter clock from 1 sample/100 μsec. up to 1 sample/60 μsec.

According to another experiment, the maximum throughput between the NEAC's NCP and MELCOM's NCP turned out to be 360 Kbps.

KUIPNET requires a new flow-control mechanism different from the ARPANET so as to allow the sharing of IMP's computational capability among several real-time and interactive communication over many HOST computers. The special priority-control logic, that satisfies the above requirement, has been introduced at the IMP side. This logic arranges the memory-access priority order of interface hardware to each HOST via programmable control pulse generated from user command and process.

The following experimental results shows that the channel-throughput of IMP is effectively controlled with this hardware. (Figure 4)

1. Using two pairs of half-duplex links, the memory-access priority of these links is alternately changed. The channel capacity for each HOST is balanced to be about 0.5 Mbps (one half of available maximum transmission bit-rate).

	process	time required for transmission or processing	ratio to real-time	comments
1	DISK ACCESS	23 msec.	0.36	max 37.5 msec.
		sub total	0.36	
2	N 2200 NCP	1.8 msec.	0.03	20 µsec.x 90 step
3	N 2200-IMP	15.4 msec.	0.24	500kbps,7.7kb/packet
4	IMP input	340. µsec.	0.005	1.7 µsec.x 200 step
		sub total	0.26	
5	IMP output	170 µsec.	0.003	1.7 µsec.x 100 step
6	IMP-MELCOM	4.8 msec.	0.075	1.6 Mbps
7	MELCOM NCP	3.2 msec.	0.05	1.6 µsec.x2000 step
8	Code transformation	25.6 msec.	0.40	1.6 µsec.x 16000 step
		sub total	0.53	

Table 1. Time required for transmission and processing.

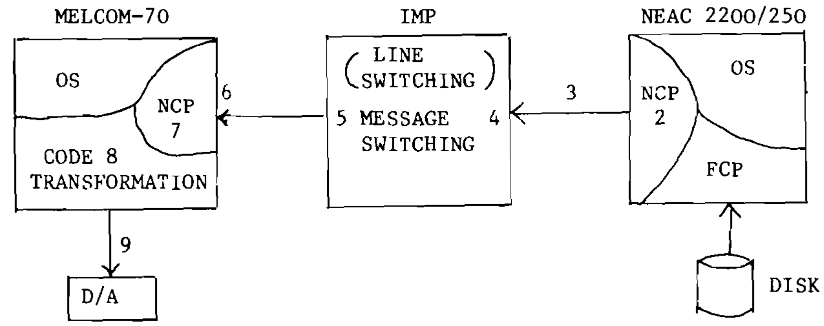


Figure 3. Transmission Path of Speech Data.

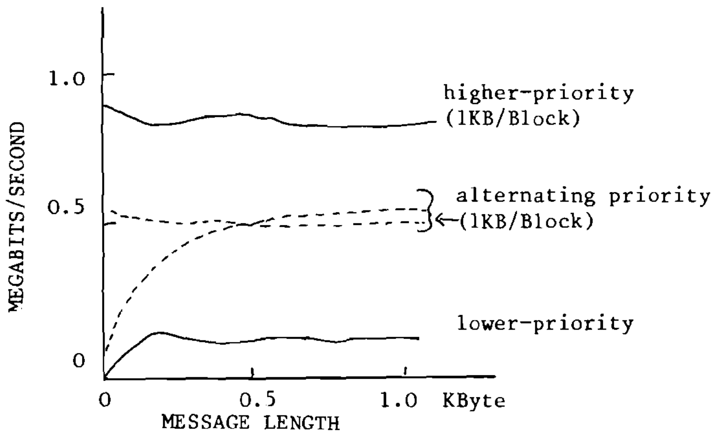


Figure 4. IMP Throughput Due to Channel Priority Control.

2. With the same links in (1), under the limitation that the maximum bit-rate is less than 1 Mbps, the throughput of higher priority links appears to be near the value of 800 Kbps.

C. <u>Some Examples of Resource Sharing</u>

a) Use of the Color-TV Display Unit

The data of on-line signature, which "decay" with time, are picked up by the Graf/pen. They are transmitted from MACC to NEAC 2200/250 through IMP, and stored in the file of NEAC for registration. When a new signature is written for verification, the registered signatures are drawn from the data file to be displayed on the color-TV unit attached to MACC. To help the identification, the signatures can be color-modified by those factors such as order-of-stroke, number-of-strokes, speed-of-pen at each segment etc. which take advantage of an on-line signature. This method of multi-feature display by color-modification has a wide variety of applications in man-machine interactive processing.

Another example of hardware-resource sharing is color representation of speech spectrum (Figure 5). An operator inputs a command via the key-board to get MELCOM to start the tape recorder and display on the CRT the waveform of the recorded speech. He then selects a segment to be analyzed, the spectrum data obtained by the 20-channel filter bank are transferred to MACC from MELCOM through IMP, and the color-TV display produces a representation of spectrum. This provides a much more informative on-line real-time research facility for analyzing the spectrum, compared with line-printer output which usually involves a time delay.

b) Speech Synthesis (Figure 6)

The older speech synthesis system using the NEAC 2200 series has been converted to a network version which is capable of true on-line real-time speech synthesis research of unlimited vocabulary.

Distributing the processing power over the network resulted in the reduction of the amount of data that flow through the transmission line. The input sentences are keyed-in at the MELCOM in Japanese phonetic string. They are sent to NEAC, and there, using the older program, transformed into the wave-element expressions (WEE) which then are sent back to MELCOM through the network. MELCOM along with a D/A converter functions both as a processor from WEE to a sequence of

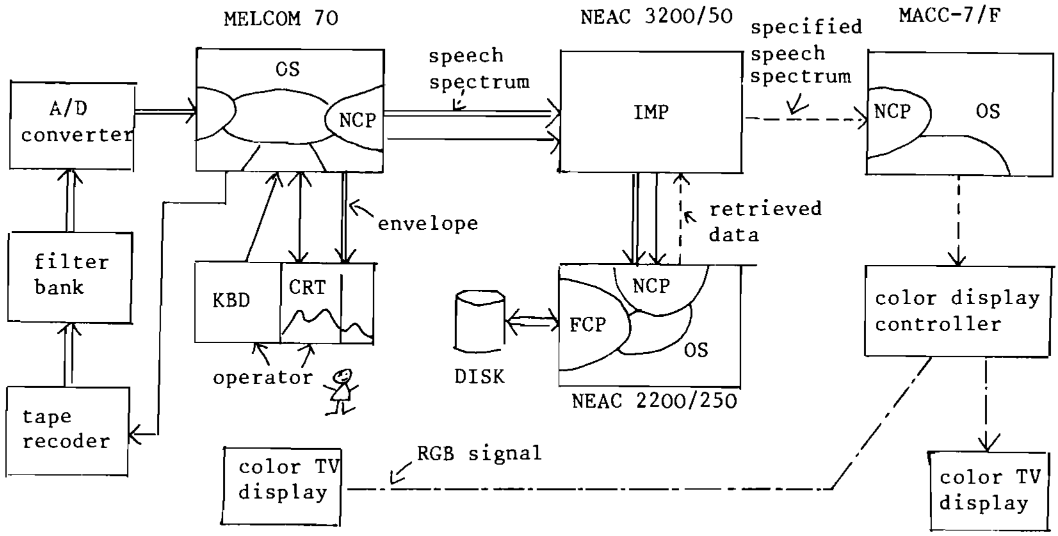


Figure 5 Color Representation of Speech Spectrum.

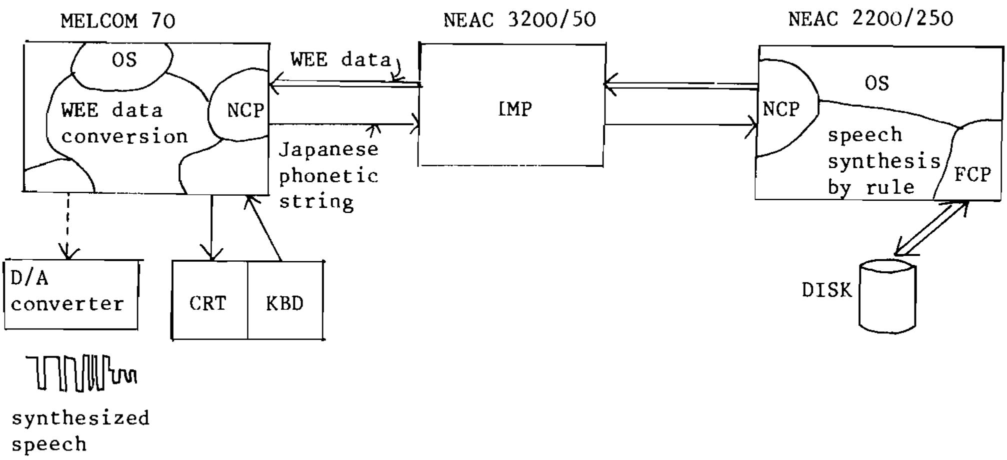


Figure 6. Speech Synthesis System by OX Wave .

zero crossing (OX) intervals (OXI) and as a speech synthesizer to reproduce analog speech. The amount of WEE data (3.6 Kbps) is much smaller than the OX wave data (12 Kbps), thus the load on the transmission speech corresponds to 200 Kbps (20 KHZ sampling with 10 bits/1 sample).

DISCUSSION

Question: Does KUIPNET have any connections to Tokyo University NW?

Answer: No, it is a different system.

Question: Throughput and delay analysis is not very difficult because there is only one node. In ARPA we measure a high speed packet switching. Our problem is, that there are only 50 Kb lines, but they are not fully used. There is a substantial delay because of the store and forward transmission. Have you made any computation to see how the delay in your net would increase if you had many more nodes; and would you still be able to get adequate speech transmissions?

Answer: There are no experiments or simulations on this at the moment.

Question: For the picture processing, e.g. in medical research, do your store, subtract, etc. information?

Answer: Information is stored in a host computer, however, in the mini-computer there is only room for a small part, and it then has to be split.

Question: Is there a special language and a high capacity memory?

Answer: For the picture processing, we have at the moment a very nice command language, but this command system may be changed into the plot core, because we are intending to connect to Osaka. As it is only 50 km away, the base line is not expensive. A large computer and their picture processing facilities will be connected.

Comment: There have been some experiments with doctors using computers at Stanford. There was a negative feedback, because they were used to their old things and methods, and could not adjust to and understand new things.

Answer: We have the same problem in Japan, and therefore have had discussions with doctors. But our system may help them, not in the traditional methods, but for example in an early cancer recognition, or heart malady by the perception of the heart beat, which otherwise would not have been heard. It is especially useful for the education of students.

Evaluation of Terminal Nets

J. Blatny*
Z. Benda*

The application of computers in large control and information systems has increased. It is evident that the computer controlling these systems cannot be managed by an off-line (computer--man-system) only. It is necessary to ensure direct interaction between them. In geographically spread systems the use of remote connections using data transmission is required. The optimal utilization and distribution of the equipment to ensure the desired functions at minimal cost becomes a problem. Many authors have solved system topology problems, and using these solutions it is possible to design a net with minimal costs.

The main problem is to be able to determine or estimate the basic functional parameters of the proposed net. We start from a given geographical distribution and from a set of technical equipment, from which a selection will be made. Originally we expect to solve only a terminal network connected with one computing center, because in the near future this will be more actual in the CSSR than the building of more complex computer networks.

Figure 1 illustrates the type of terminal networks with which we are concerned. All the technical means of the model are divided into the following categories: central computers; nodes; terminal nodes; and connections.

The central computer represents the actual computing capacity of the system. It receives the messages from the terminals, processes them and generates messages for the terminals. Nodes enable branching in the network and thus decrease the total cost. Multiplexers, message and channel switches are the technical branching equipment. Since multiplexers realize only physical (and not logical) branching, and influence only the cost and not the throughput of the system, we have replaced them by a set of equivalent connections in the net. Message switches, on the other hand require a more accurate design, especially in determining the buffers' number and size, and in input and output transmission rates. Design problems of the channel switches determine switching times--the building or releasing of a connection. The terminal node forms an interface between the network and one or several terminal devices. It must control these, perform data buffering and data editing, and connect each terminal to the network.

Attention must be devoted to the terminals because their type, parameters and usage determine input and output of data

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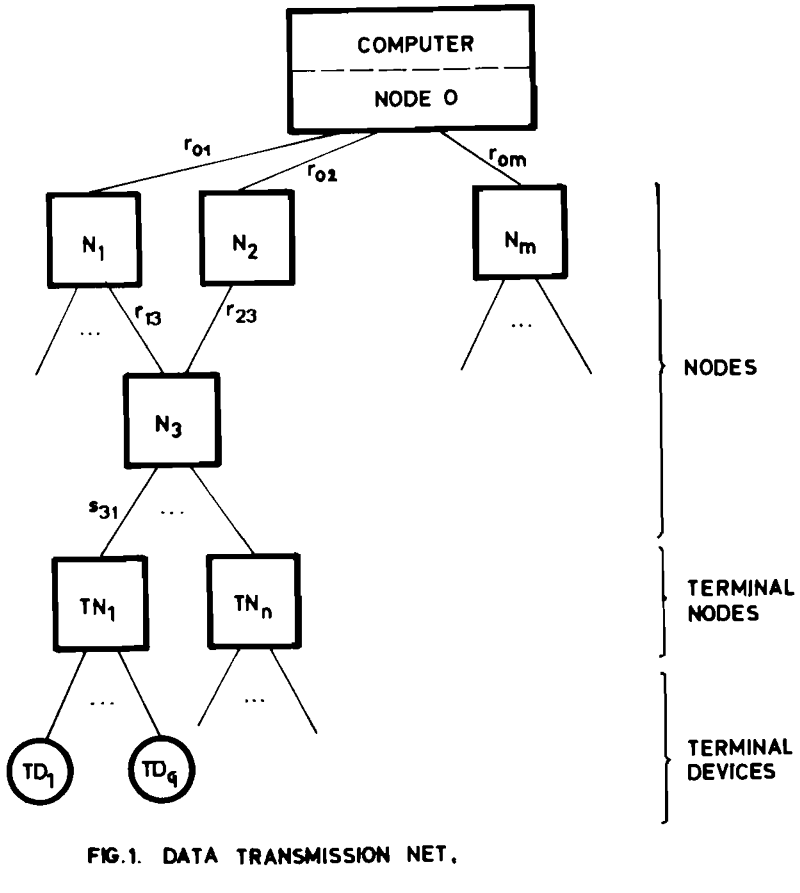

FIG.1. DATA TRANSMISSION NET.

flow, and thus the character of the network load. The majority of work so far solves this by choosing the most advantageous distribution of input and output flows (worst-case characteristics). Sometimes it is questionable whether the chosen distribution is really the worst; sometimes it may also lead to unnecessary overdimensioning of the network.

Connections can be divided into point-to-point and multidrop connections. Multidrop connection is presented as a modified node with a set of point-to-point connections. Basic descriptive connection parameters are: capacity (transmission rate), mode of operation and frequency of errors.

One of the fundamental functional network parameters is message throughput time. Indpendent of the system's requirements, it can have various forms. It can be the mean value, or its distribution.

One of the possible solutions is the application of mathematical methods and analytical design of a system. When dealing with a large system, it is necessary to work out a number of simplifications and to neglect all extraneous facts, even relevant facts have to be omitted in order to simplify the problem. A presupposition for a simple mathematical solution is the use of an idealized model. The results can be considered as the first approach to a solution that needs further refinement, e.g. by using a simulation model.

In the second part of the paper a rough idea of our simulation model is described.

SIMULATION MODEL OF A DATA TRANSMISSION NETWORK

A data transmission network is designed for the transmission of messages that are further divided into packets or blocks. Transmission is realized by packet switching. A message can contain (according to its length) various numbers of blocks of fixed length. Besides its own information, each block contains a header with service information; in our case this is the network address which determines the block's path through the network. Various paths are chosen according to the different network addresses of the messages; all the blocks of the same message have the same address.

The structure of the net need not be a tree as in Figure 1. Thus different paths between the computer and a terminal device are possible, however a particular path is unambiguously given by the address.

The network is created by a set of nodes whose number can vary and which can be connected by lines with suggested transmission rates. A node can model the described technical means, which are placed at the connection or branching of several transmission lines. This node is described as a queuing system (Figure 2) performing the following functions.

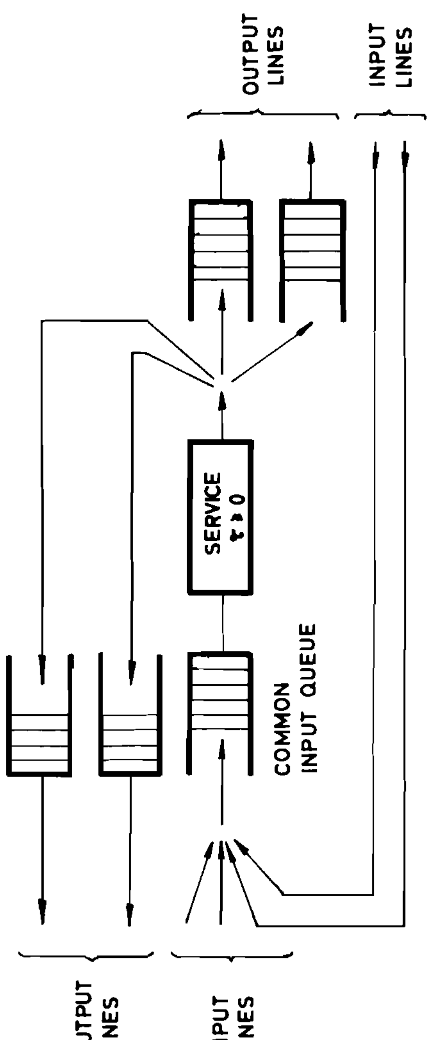

FIG. 2. MODEL OF A NODE.

Blocks coming from all input lines are stored in the node's memory. If several blocks arrive simultaneously, they can create an input queue in the memory. Blocks are then selected from the input queue according to their arrival times (first in, first out). Each block contains a network address which is then decoded; this is processed to determine the address of the next node through which the block should pass. After being processed the block is sent to the next node. If the desired line is occupied, the block must wait in the node's output queue. The time interval during which the block is transmitted and the line is occupied is given by the quotient of the block length and the transmission rate of this line.

These terminal nodes and devices, representing control units and input-output devices, serve for terminal modelling. The terminal node (Figure 3) has the following function; for message input it receives information from the terminal devices and stores it in its buffer. When the buffer is full, the information block is sent to the nearest node and then to further nodes according to the network address. If the line to the nearest node is occupied, the blocks must wait in the output queue of the terminal node. In the other direction the terminal node receives blocks from a node and transmits them to terminal devices. We assume that terminal devices can receive data at the transmission rate of the line: node-terminal-node. Thus input queues do not exist.

Terminal devices are divided into three groups:

1. Input devices--for data collection and transmission towards the central computer;

2. Output devices--for the output of messages transmitted by the central computer; and

3. Conversational devices--for transmitting messages to the computer and receiving answers from it.

Independent of these groups any number of types of devices may be described. Each type has its own activity, i.e. the probability distribution of message length (transmitted or received) and the times between the arrival of successive messages.

The subsystem: terminal node--terminal device can represent a simple terminal as well as more complex equipment (e.g. the 200 user terminal of the CDC).

The complete model of a network has the following parameters:

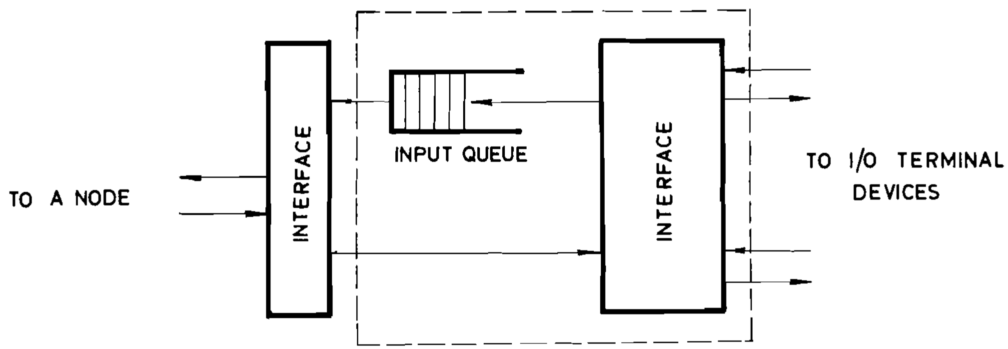

FIG. 3. MODEL OF A TERMINAL NODE.

- Number of nodes;
- Connection matrix with transmission rates between nodes;
- Memory capacity and processing times for each node;
- Number of terminal nodes connected to each node;
- Transmission rates of the lines: Terminal node--node;
- Number of terminal devices connected to each terminal node and type designation of each device;
- The above-mentioned probability distributions for each type of terminal device;
- Block length (fixed).

The generalized approach to the modelling of network elements allows us to simulate:

1. Terminal network of a computer (Figure 4);
2. Data transmission network with nodes equipped with communication mini-computers connected to a central computer (Figure 5).

Resulting parameters of each simulation experiment are:

- The number of transmitted or received blocks and messages for each terminal device;
- The mean value of block and message throughput times;
- The maximum value of block and message throughput times;
- The utilization of all particular lines;
- The queue lengths on nodes and terminal nodes.

The described model has been written in SIMULA 67 and used on the CDC 3300 computer at the Computing Research Center with the Uninted Nations Organization program in Bratislava.

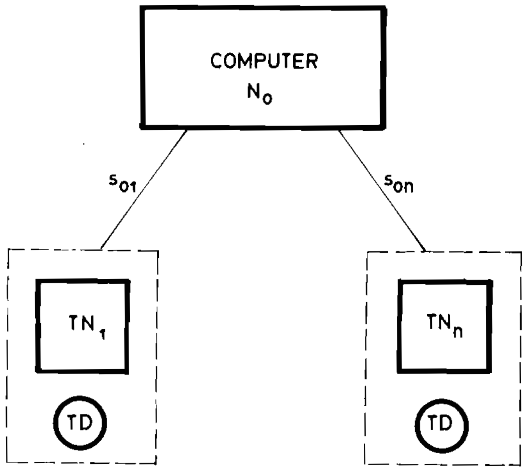

FIG. 4. TERMINAL NET OF A COMPUTER.

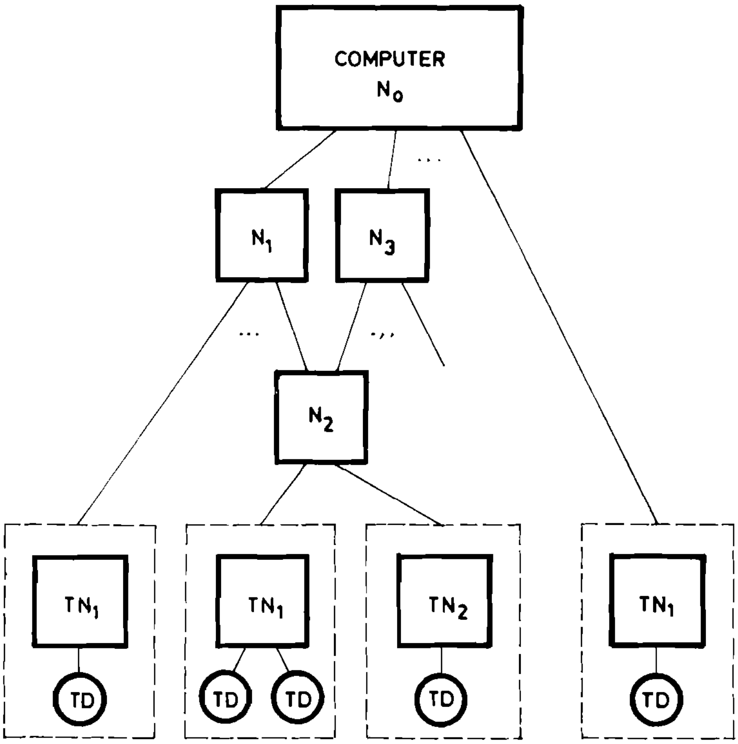

FIG.5. DATA TRANSMISSION NET WITH COMMUNICATION COMPUTERS CONNECTED TO A CENTRAL COMPUTER.

It has approximately 750 SIMULA statements. It has been debugged just recently. Its functional correctness and possibilities were proved on different test examples. Now it will be used for planning data transmission nets in cooperation with institutions responsible for the projects. Further refinements of the model are expected based on the experience gained from the practical use of the model.

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[3] Blatny, J., Benda, Z. and Linhart, M. "Pouziti simulacniho modelu pri vyhodnocovani terminalovych siti," Informacne systemy, 3, Alfa Bratislava, 1975.

[4] Dahl, O.J., Nygaard, K. and Myhrhand, K. "Simula 67 - Common Base Language," Norwegian Computing Center, Pub. No. V, 2, Oslo, 1968.

DISCUSSION

Question: How is the model evaluated, and was the model NW run against an actual NW?

Answer: We had no practical experience with NWs, and our model was written according to other papers.

Question: Now that a NW will be designed, what will you do next, and how will you change the parameters? Are you familiar with the analytical methods to do this?

Answer: We started four years ago with the simulation, not of the whole NW, but of the throughput of messages through a machine. We tried both ways, the analytical and the simulated. It would be best to cooperate with a real project and to compare the results.

Overview of an Experimental Loop System For Data Transmission*

A.R. Kaye**

I. INTRODUCTION

A loop or ring structure for the transmission of data by means of packet switching first was described by Newhall and Farmer [1]. Since then several authors have described and analyzed different control and signalling protocols for use in such systems [2-10]. This paper describes an experimental system whose control protocols are closely related to the Newhall and Farmer proposals. The system is based on a two level hierarchy of loops and will connect four different locations in three cities.

One of the objectives of the experiment is to identify those problems likely to be encountered in building and operating a packet switching system in which the intelligence necessary to operate the system is located mainly in the terminal interfaces, and is based on the use of integrated circuit logic and memory devices whose cost is rapidly decreasing. Another objective is to explore the use of such a system in a computer and terminal network. A single loop system of this type has already been successfully constructed and tested.

II. THE BASIC SYSTEM

The system will serve clusters of terminals and computers in each of the organizations participating in the experiment: Communications Research Center in Ottawa (CRC), Edmunde Newhall Associates Ltd. in Toronto (ENA), the University of Toronto (UOT) and the University of Waterloo (UOW). In each location the various machines are connected by a local loop. The four local loops are interconnected by a long-haul loop and a dial-up line. The basic structure is shown in figure 1. A local loop will be described in principle; more detailed descriptions of the various units, signal formats and protocols are given later

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*The development and design of the equipment being used in this experiment was the work of E. Newhall and T. Nagel of E. Newhall Associates Ltd.

**Department of Communications, Ottawa, Canada.

A. Local-Loops

The transmission medium in a local-loop is a pair of wires, and data transmission is at 9600 bps using the bipolar format (see figure 2). In normal operation transmission occurs in only one direction.

At each terminal, a repeater/multiplexer is connected in series with the loop (see figure 3). The repeater/ multiplexer is line powered and associated with a distributed switch. The user's business machine as shown in figure 4. When it is quiescent the repeater/multiplexer repeats the loop signal, however, when it is active, the data packets formed by the distributed-switch are multiplexed onto the loop under control of the distributed-switch. Timing information is extracted from the bipolar loop signal in both the repeater/multiplexer and the distributed-switch. If local power is removed from the distributed-switch, the repeater/multiplexer repeats all loop signals.

The distributed-switch formats data from the business machine into packets with address and control headers, to transmit them on the loop and to receive incoming packets from the loop according to the system protocols described below. The address of the terminal to be carried is entered into the distributed-switch by the user at the keyboard of the call initiation unit.

Each loop is equipped with a supervisor which maintains synchronism on the loop, detects fault conditions and provides a control center for remote identifications and isolation of faulty distributed switches or repeater/multiplexers and for start-up procedures.

B. The Long-Haul Loop

The transmission medium in the long-haul loop is a conditioned voice circuit operating at 2400 bps in binary format. One direction of the leased, full-duplex facility is used in normal operation, the other direction is used during fault diagnosis and recovery procedures. A binary signalling format allows for the use of slightly modified 201 equivalent modems on the voice-grade line. This format necessitates the use of framing signals to separate control information from data. (In the local-loops this is achieved by bipolar violation signals.) The loop supervisor is responsible for establishing and maintaining the frame structure. It also carries out all the functions performed by the local-loop supervisors.

C. The Nodes

Transfer of packets between the local- and long-haul loops is effected by nodes or gateways. A node is essentially a pair of distributed-switches connected back-to-back with a repeater/multiplexer on each loop. Packets on a local-loop, with addresses not on that loop are transferred by the node to the long-haul loop. When they reach the node for the local-loop to which the addressed terminal is connected, they are transmitted via that node.

Normally a node is a single piece of hardware to which both the local- and long-haul loops are connected. However, in the connection to the CRC loop a different structure is being experimented with. The two halves of the node will be connected by a dial-up, four-wire, voice circuit. This connection will be limited to a speed of 300 bps so that only low-speed terminals will be able to use it. This is a limitation of our available modems, however, and could eventually be up-graded.

III. PROTOCOLS

Protocols have been implemented on three distinct levels. Their functions are: to establish a virtual circuit between a pair of business machines; to control data transfer between the business machine and its associated distributed-switch, and between the loop and the distributed-switch. All protocols are implemented by micro-programming in the distributed-switches.

A. Virtual Circuits

The described virtual circuit protocol has been implemented in the experiment system because it is well suited to the users the system will serve. It is, however, an optional feature of the basic loop system which could be implemented equally well without it.

When a user wishes to call another terminal or computer he first enters the address of that machine and presses a "Hello" key on the call initiation unit. The distributed-switch then sends an "are you busy?" (RYB) message to the called terminal which responds with either a busy (BSY) or a not busy (NBY) message. If the called terminal is not busy, then both distributed-switches enter the data mode in which a data transparent, full-duplex, virtual circuit is established between the two business machines. Until the call is terminated the two distributed-switches will return a busy message (BSY) to any other terminal trying to establish a connection. A virtual circuit is taken down when the appropriate key on the call initiation unit is pressed, causing the END-message to be transmitted to the other distributed-switch

B. Business Machine to Distributed-Switch Protocol

The distributed-switch is connected to its associated business machine by a standard interface (e.g. EIA RS 232C); thus the machine has access to a "virtual circuit" which appears to it to be a standard physical, full-duplex circuit.

The distributed-switch is programmed to recognize the code set of the particular machine, including all its control signals (e.g. breaks). All character and control signals entering the distributed-switch are assembled in a transmit buffer 32 or 128 characters in length. The buffer contents are transmitted onto the loop in packets according to the protocol described in the next section.

C. Loop Transmission Protocol

The transmission on local- and long-haul loops are functionally identical although they differ in implementation to allow for various signalling formats on the two loops. Only a functional description is given here.

The distributed-switch transmits a packet on the loop through the repeater/multiplexer when: it has control information, such as RYB, BSY, END to send; OR* its transmit buffer fills to a pre-set level; OR* when a carriage return or equivalent control character is received from its associated machine AND* it acquires control of the loop.*

Control is achieved by monitoring the signals passing around the loop until and end-of-message (EOM) code followed by a pass-control-bit (PC) set to logical 1 is observed. Then, if the distributed-switch wishes to transmit, it sets PC = 0 to notify subsequent stations on the loop that it has taken control. The one bit delay between the receiver and the transmitter allows this by changing the repeat/ multiplex switch to multiplex (see figure 4). It then transmits a start-of-message (SOM) code followed by a packet as shown in figure 5. It contains either control information or all the data currently in the transmit buffer. Control is relinquished at the end of a packet by transmitting an EOM followed by PC = 1. The virtual circuit, however, remains intact.

The distributed-switch receives packets by monitoring the loop signals until it observes its own address. It copies into its receive buffer only those packets addressed to it,

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*The terms OR, AND, and (•) are used here in the logical sense of a Boolean function.

and which come from the terminal with which a virtual circuit has been established. It then sets a "mark bit" in the packet which continues to propagate around the loop, and should it come by again, it will not be copied. The distributed-switch at all times copies those packets addressed to it containing the RYB code, or certain diagnostic and fault recovery instructions issued by the loop supervisor, into a special buffer, and takes the necessary action.

IV. SPECIAL FEATURES

A. Remote Port Selector

In many computing applications, connecting a subset of terminals to a smaller number of available ports becomes a problem. In the experimental system a special remote port selector has been developed to avoid this, and it is used in conjunction with a time-sharing computer at UOW. It has only a single interface to the loop, and is designated by both a master and specific address. A user wishing to access the computer keys the master address into his call-initiation unit, presses the "Hello" key in the normal way and the distributed-switch transmits an RYB. If a computer port is free the selector responds with a control packet which enters the specific address into the where-to address register of the calling distributed-switch and establishes a virtual circuit between the terminal and the specific port. If all the computer ports are busy then the selector returns a BSY packet.

B. Automatic Dialling

The system incorporates two automatic dialling units connected to the Direct Distance Dialling Network. Each is pre-programmed to dial a single telephone number, usually that of a service bureau computer. They are interfaced to the loop through a standard distributed-switch. A user wishing to be connected to one of the telephone numbers keys the address of the automatic dialling unit and presses his "Hello" key. On receiving the RYB packet, the unit starts dialling; the virtual circuit is established only when the call has been connected.

C. Automatic Call-Initiation

An automatic call-initiation unit is provided to allow computers rather than a human operator to begin calls. It carries out the same functions as the standard call-initiation unit but accepts inputs directly from a computer.

V. RELATIONSHIP TO OTHER LOOP SYSTEMS AND WAITING TIME

Several authors have postulated and analyzed different transmission protocols for loop systems [2, 3, 4, 5] based on a slot format which allows terminals to fill empty transmission slots defined by frames circulating around the loop. This may appear to be a rather minor difference from the protocol described here but in fact it causes significant behavourial differences.

One of these protocols [4] is designed for use in star-shaped systems, i.e. all messages are addressed to, or emanate from, a single point on the loop, such as a computer or CPU serving terminals or peripherals. The central point creates empty slots and fills them when it has output data to send or transmits them empty for use by terminals sending input data. This system is not directly comparable to the one we use except for identical traffic patterns. In this case the mean waiting time varies according to the position of the terminal on the loop and the ratio of input to output traffic. In our system it is the same for all terminals.

Another slot-based protocol [2, 3, 5] allows the receiving terminal to delete data from a slot. The empty slot may then be re-used by the same or another terminal. In this system also, waiting time varies.

Waiting time analysis relevant to the transmission protocols described here appears in [6, 7, 8, 10]; different protocol performance is compared in [9]. The main conclusion is that the present protocol achieves waiting times, averaged over all terminals, which fall between the values of the mean waiting time of the slot based protocols.

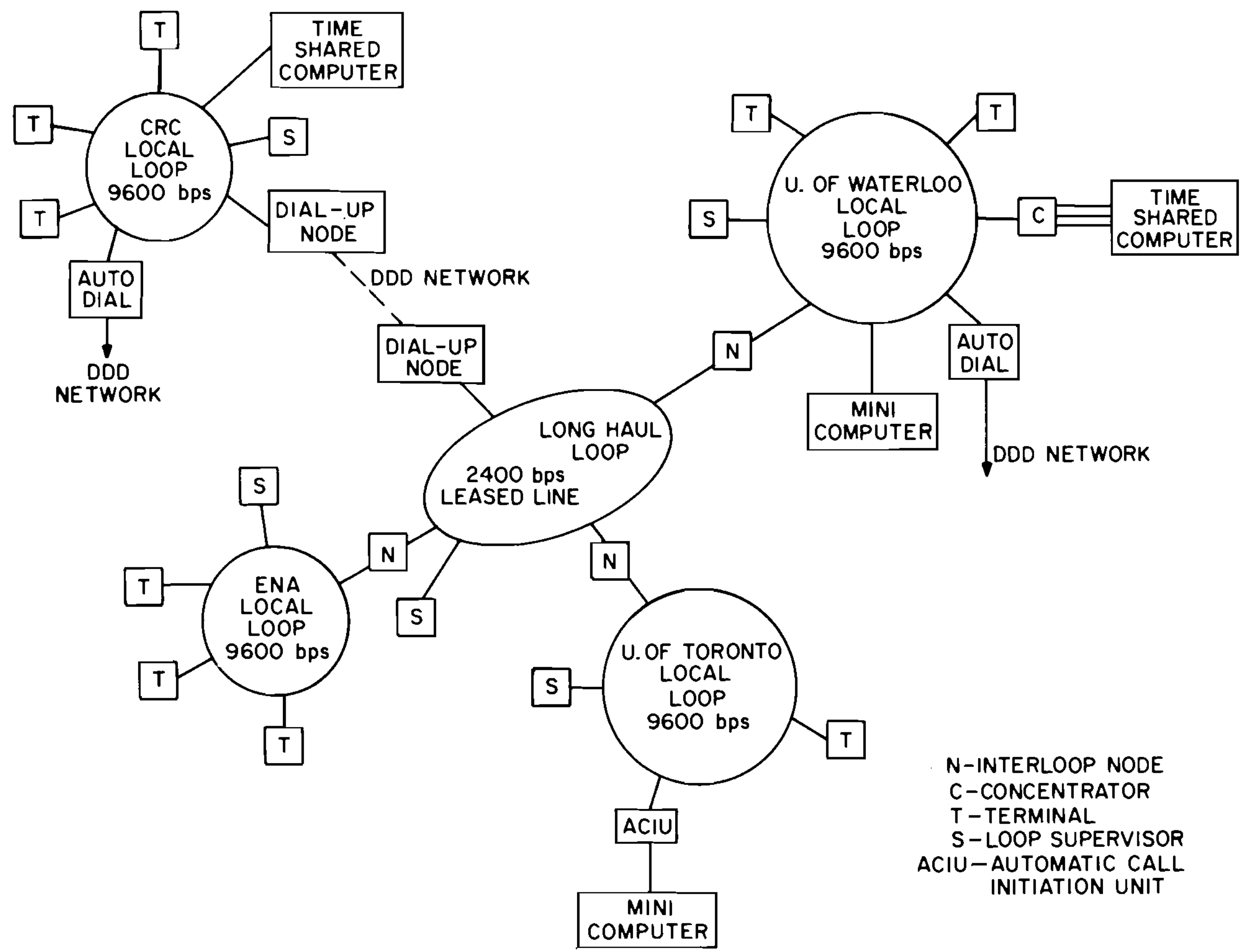
VI. PERFORMANCE MONITORING

Each supervisor continuously monitors the operation of its loop as described earlier. The long-haul loop may have a higher fault rate than the local-loops. Accordingly the long-haul loop supervisor will be equipped with a printer to record errors caused by loop transmission problems.

To monitor reliability of the overall system a mini-computer is being programmed to initiate and log continuously the results of calls to all the terminals on the system in turn. It will be located on the UOT local-loop, and will interface the system through a standard distributed-switch equipped with an automatic call-initiation unit. On each call-initiation it will receive a busy response or set up a virtual circuit or, if a fault exists, no response will be received. By calling all terminals in this way it will be possible to detect a wide variety of functional problems.

Captions for Figures

FIGURE 1


CRC LOCAL LOOP 9600 bps
T
T
T
TIME SHARED COMPUTER
S
DIAL-UP NODE
AUTO DIAL
DDD NETWORK
DDD NETWORK
DIAL-UP NODE
LONG HAUL LOOP 2400 bps LEASED LINE
N
S
N
U. OF WATERLOO LOCAL LOOP 9600 bps
T
T
S
C
TIME SHARED COMPUTER
MINI COMPUTER
AUTO DIAL
DDD NETWORK
ENA LOCAL LOOP 9600 bps
S
T
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N
U. OF TORONTO LOCAL LOOP 9600 bps
S
T
ACIU
MINI COMPUTER
N-INTERLOOP NODE
C-CONCENTRATOR
T-TERMINAL
S-LOOP SUPERVISOR
ACIU-AUTOMATIC CALL INITIATION UNIT

FIGURE 2

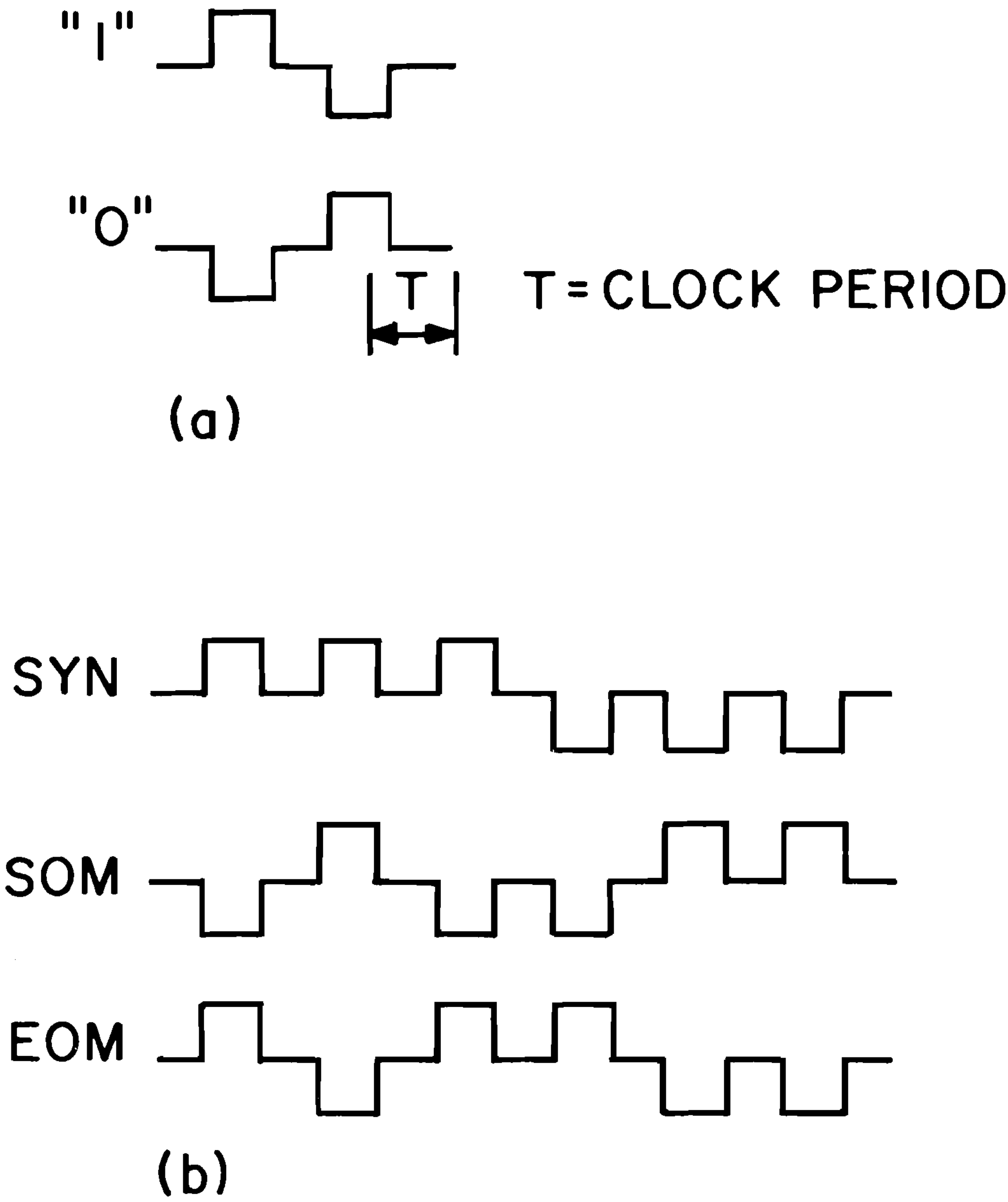


FIGURE 3

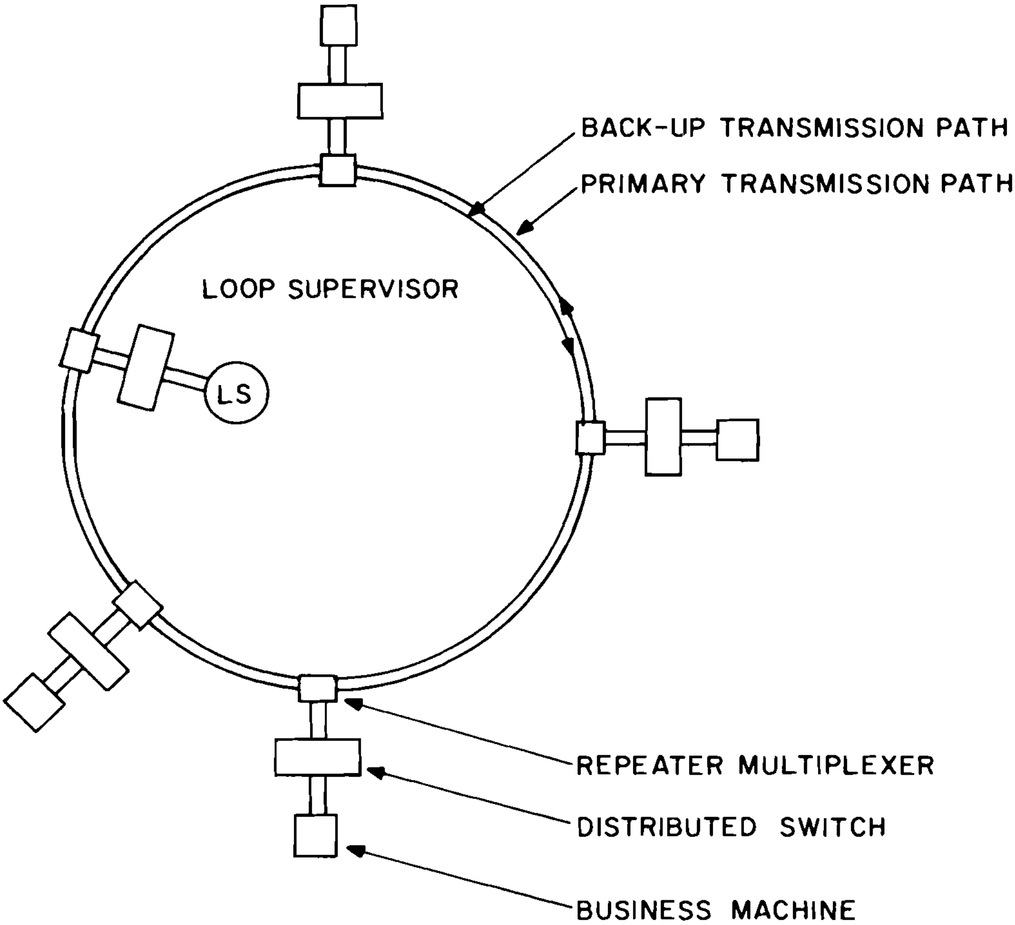

BACK-UP TRANSMISSION PATH
PRIMARY TRANSMISSION PATH
LOOP SUPERVISOR
LS
REPEATER MULTIPLEXER
DISTRIBUTED SWITCH
BUSINESS MACHINE

FIGURE 4

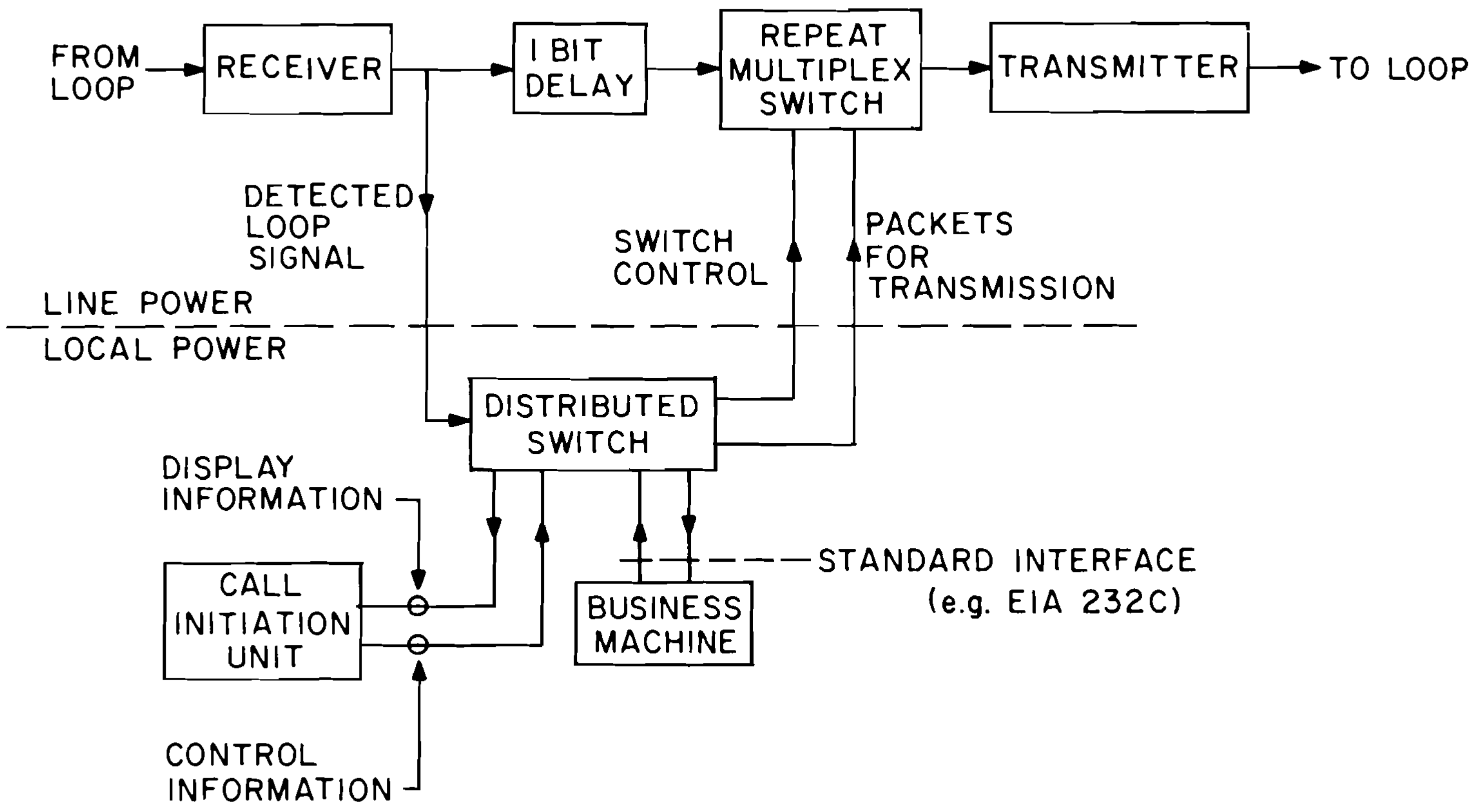
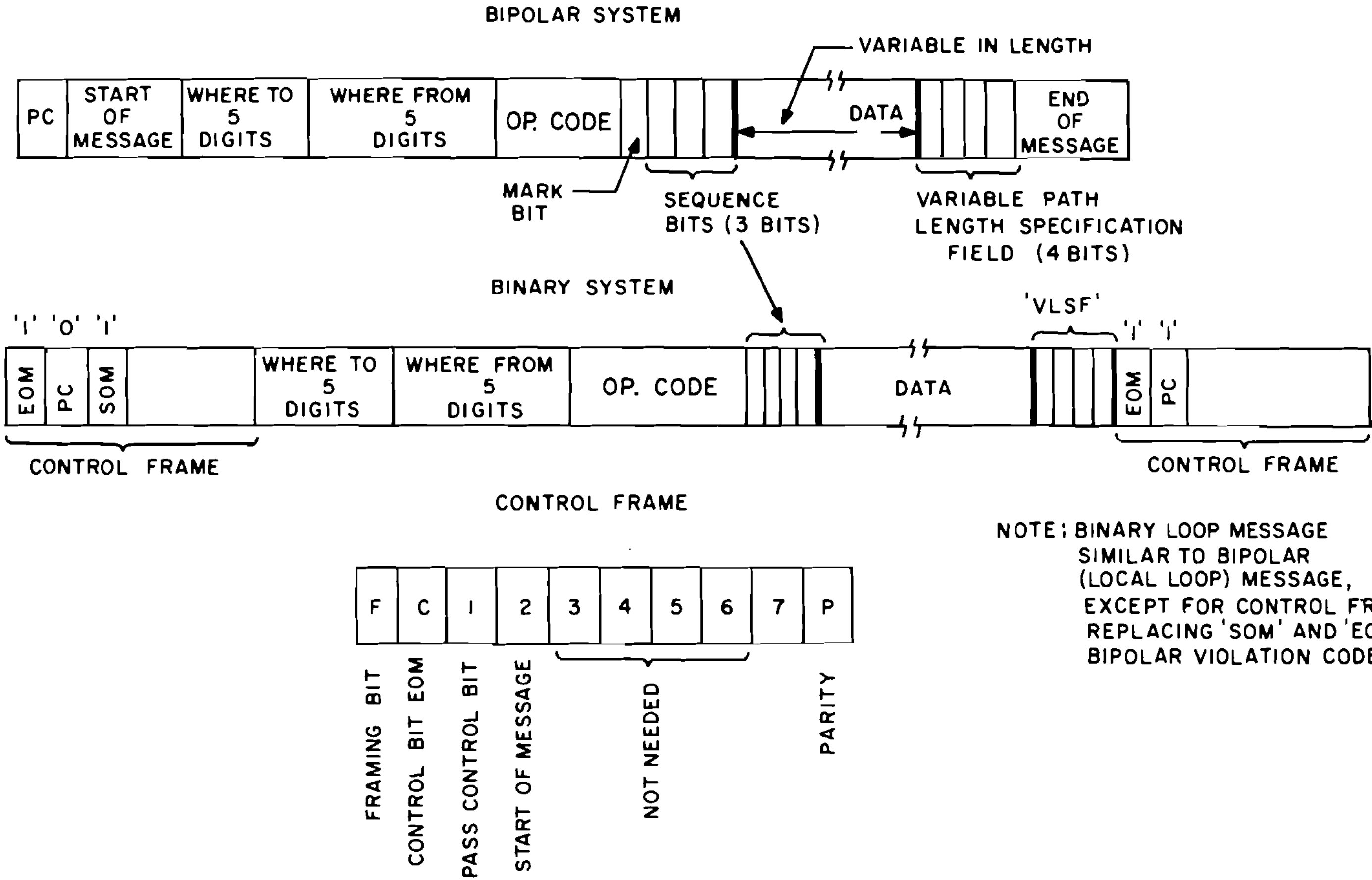

FROM LOOP
RECEIVER
1 BIT DELAY
REPEAT MULTIPLEX SWITCH
TRANSMITTER
TO LOOP
DETECTED LOOP SIGNAL
SWITCH CONTROL
PACKETS FOR TRANSMISSION
LINE POWER
LOCAL POWER
DISTRIBUTED SWITCH
DISPLAY INFORMATION
CALL INITIATION UNIT
BUSINESS MACHINE
STANDARD INTERFACE (e.g. EIA 232C)
CONTROL INFORMATION

FIGURE 5



NOTE: BINARY LOOP MESSAGE SIMILAR TO BIPOLAR (LOCAL LOOP) MESSAGE, EXCEPT FOR CONTROL FRAMES REPLACING 'SOM' AND 'EOM' BIPOLAR VIOLATION CODES

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[10] Yuen, M.L.T., Black, B.A., Newhall, E.E., and Venetsanopoulos, A.N. "Traffic Flow in a Distributed Loop Switching System." Computer-Communications and Teletraffic, Polytechnic Press, New York, 1972, pp. 29-46.

DISCUSSION

Question: Is there any defined or designed protocol to set up virtual calls?

Answer: It is very simple. At first you key in the address you want. The circuit switch then sends out a packet addressed to the wanted terminal, which sends an: "are you busy?" That terminal replies either: "yes, I am busy, go away," in which case your call attempt failed; or else it replies: "no, I am not busy, but I am now; and let us go ahead."

Question: The state described happens when a user is on a terminal; what happens if the user is in the computer?

Answer: There is a method which connects a computer to the system: there is a second interface which accepts the address information, logical information, instead of pressing buttons. The procedure is the same, it is just done in a different way.

Question: How does the multi-level addressing work? What happens when more than one stream goes into one computer, does it have a multiple virtual circuit?

Answer: The virtual circuit protocol should be left to the machine. You can construct switches that accept packets from any source and transfer them to the machine. It has to check them, and make a host-to-host protocol. In one system there is a mixture of terminals, but in the loop all the messages look the same. The switching and uncoding is the same one used in a telephone system. Terminals and nodes have no routing information, they only know which message they have to pick up. There then exists a possibility of interlinks, you do not need to connect only to loops, you can establish connections to anything else.

Comment: This is a hierarchical structure. Any loop system covering a wide geographical territory may consist of many loops and can still continue to be tree-like in its topology.

Answer: It is like that. People tend to speak of loops as if no other possibilities exist. This is not true, there are mainly mixtures--inside loops, outside trees, etc.

Question: If you connect to the PTT, does the NW traffic go both ways? Charging problems can arise if you connect from the PTT to the data NW.

Answer: Until now the only possible connection is from the loop (data NW) to the public NW, but it is at present only in an experimental stage. Therefore there are no charging and political problems.

Question: How is the transparency when you go into the public telephone NW? Is the bit sequence independent? Will the disconnection be done from the data NW?

Answer: You disconnect from the loop (data NW) not from the public telephone.

Comment: If everything is under the control of the data NW you will be okay, but this cannot be a general solution.

Answer: This is an early experimental system initiated from the research point of view, not from an economical one.

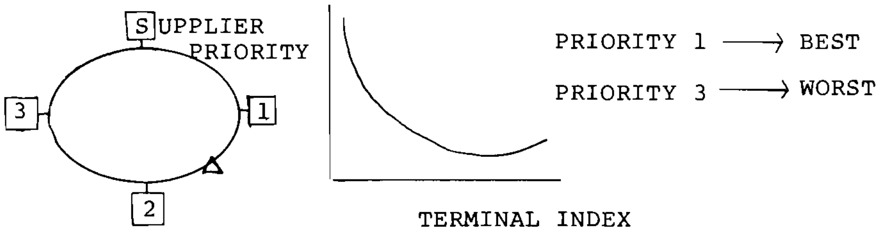
Question: What are different areas of application? Which loop system is where?

Answer: There are two kinds of slot systems:

1) One by Pierce, where you open the slot, empty it, and then you get a free one;

2) One where you only copy the contents.

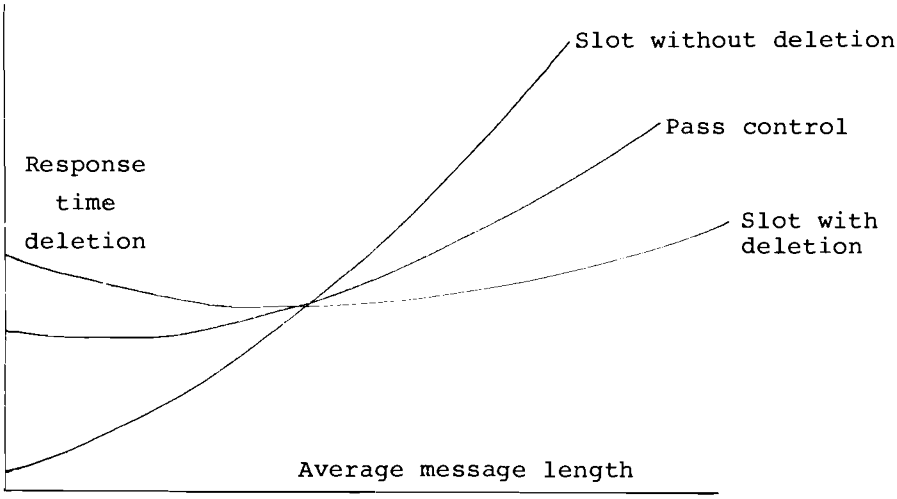
The central control delivers slots in a star pattern. You have a transmission loop, where terminals get priorities, but then the distribution of the slots will not be even.



Out of the computer it is the other way round. In Pierce's model, there is a slot supplier and every terminal is a supplier also. When a message arrives, it empties the slot, and uses it again. Priority then depends entirely on the traffic pattern.

Question: How closely is this packet switching related to Farmer's?

Answer: It is closely related. We also have pass control protocol; Farmer also considers slot with and without deletion protocol. Our curves are the following:



Question: Can the ALOHA type protocol be used?

Answer: It is purely a collision protocol, whereas ours is a control protocol.

Views on the IIASA Program in Computer Communications and a Review of some Canadian Projects

A.R. Kaye
D.D. Cowan
J.L. Hanna

I. INTRODUCTION

Our purpose in preparing this paper is to focus attention on those aspects of computer communication networks which seem to be appropriate for study within IIASA. Our first recourse, therefore, was to the various statements made by the Director regarding the objectives of IIASA and guidelines for the selection of projects. The main objective has been stated as "the investigation of problems of modern societies arising from scientific and technological development." We also found the following guidelines:

> "In order for IIASA to become involved in a particular research area, the problem must be of either global or universal importance."

> "The output of IIASA should be of such a nature that it is useful to those who make or implement policy."

IIASA will conduct work in two categories:

a) Specific problem areas;

b) The methodology which can be harnessed to solve specific problems using systems analysis.

Dr. Butrimenko has stated that the objective of the computer networks project is: "The development of a system analysis approach to the elaboration of criteria and methods of designing and management of large scale, geographically distributed, computer communications systems." Thus the project clearly is in category b), methodology, mentioned above and not category a). In addition to this there is the immediate problem of providing computer services to IIASA itself by means of communication links of various lengths.

Bearing these factors in mind we decided to divide our paper into three sections: some views on the pragmatic approach that needs to be taken to the immediate computer/communications problems of IIASA, some suggestions for topics which might be included in the IIASA program and a review of some related work going on in Canada.

II. THE IMMEDIATE PROBLEM OF THE IIASA NETWORK

The requirements to be met by the IIASA network fall into three categories:

a) To provide access to programs and data developed as part of the ongoing research at IIASA;

b) To provide access to data and programs available elsewhere, which are required to support the ongoing research at IIASA;

c) To provide access to data and programs available elsewhere which are required by visiting research workers for short periods of time.

In view of the very considerable amount of computer power available within the area surrounding Laxenburg we support the general thesis that IIASA should not purchase a large inhouse computer to serve its needs. Such a move would clearly impose a serious burden on IIASA funding both to purchase and support the machine. It is equally true that any attempt to build for IIASA an international network of any appreciable size would also impose a very large burden in terms of direct expenditures for rental of facilities and also for the engineering effort needed to support and maintain the system. In view of these comments we suggest that programs developed by IIASA for use in its ongoing research programs, category a), should wherever possible be developed on computers in the local area. Similarly it would be worthwhile giving very serious consideration to transferring to computers in the local area, programs, and perhaps data, required in category b) for support of ongoing research at IIASA. Even in category c), programs developed elsewhere and required by visiting research workers, we suggest that a careful analysis of each case to determine the relative costs of transferring the programs or establishing communication facilities, in terms of direct expenditures and man months, would be well worthwhile.

In order to serve the needs of those research workers who do require communication facilities to remote computers, it will probably be necessary for IIASA to maintain a small engineering staff to plan, negotiate and monitor the performance of the required facilities. Such a staff could take steps to avail itself of the extensive experience which is now being built up throughout the world in establishing computer networks.

Since most of the facilities established will be used to serve the short term requirements of visiting scientists, it is clear that most of them will have to be established at short notice. For this reason we recommend that the simplest possible approach be used for each case rather than attempting to build sophisticated networking facilities. We would like to offer a few comments on the approach to be taken based on our own limited experience.

In each case, it will be necessary to determine as far as possible the nature of the traffic to be carried. Is it bursty? Is the flow the same in both directions? What are the holding times and the required access times? Since each of the facilities will probably be directed to a different part of the world it will also be necessary to determine the characteristics of the transmission facilities available from the various carriers. This will require monitoring the line to determine the characteristics of impulsive interference, the frequency regularity and duration of any drop-outs. It is unlikely that the carriers will be able to supply accurate information in this respect or that they will themselves monitor the lines to ensure adequate conditions. An analysis of these factors will assist in making decisions as to what communication protocol, what type of store and forward facilities, what type of error control, block or package length and flow control techniques are to be used. Models already exist for the analysis of optimum block length, given the noise and interference characteristics of lines.

Because of the potentially very high cost of communication services it would be advisable to give consideration in each case, based on the forecast traffic characteristics, to the use of dial-up four-wire data circuits, if they are available, or standard, switched telephone circuits accessed by acoustic couplers or telex facilities where these are appropriate.

In order to minimize the effort required to establish communication facilities it would clearly be advantageous to use the facilities of existing computer networks wherever possible. For instance, if the desired remote computer is attached to, or close to ARPANET or the CYCLADES networks then the shortest possible hook up to the appropriate network might be the best solution. Even though the equipment costs might be high, the design is well established and the network is maintained. If usage of the facilities of an established network becomes substantial then IIASA might well consider a permanent connection to that network.

III. THE IIASA COMPUTER COMMUNICATIONS PROGRAM

Many projects are well under way in various parts of the world which are concerned with designing and managing large-scale computer communication networks. Notable examples are ARPANET, SITA, CYCLADES and COST 11 as well as several commercial service bureau networks. In addition, there are many organizations throughout the world which are conducting research in computer/communications. These projects are covering, and have covered, a very broad range of basic problems. The opportunity therefore exists for IIASA to draw upon all this basic expertise and seek to find common factors as well as to identify different projects which would be consistent with IIASA's objectives.

The International Network Working Group (INWG), a subcommittee of TC-6, one of the technical committees of IFIP, has many interests in the area of networks, both technical and social, and includes among its membership many representatives of the world's expertise in computer networks. INWG is an excellent forum for discussing many of the social, political and management problems which are going to arise as networks become truly international. IIASA might consider participating in the activities of this group. In the design area they are now looking at the problems in establishing protocols among the many different data networks which already exist or are being designed and implemented.

If computer networks are to be more easily assembled in the future, it will be necessary to introduce standards. This must eventually be done by the various standards organizations but they will need a great deal of information and background research before they can undertake such work. This preparatory work should not be left solely to manufacturers. As an international organization IIASA could identify areas where compatibility will be required and contribute to the development of guidelines for future generations of computer and communication systems.

A. Some Suggested Topics

The above comments lead us to suggest the following problems which might be investigated by IIASA:

1. What communication and software structures and procedures are most suited to present day communication problems?

2. What structures and procedures are most suited for present day computer operating systems?

3. What changes in the operating system, computer architecture and communications environments are likely to occur in the near future which could change the structures and protocols required in computer communications networks? How do these compare with the changes which are desirable? What changes are desirable?

4. Which application areas have the greatest potential need for computers to be connected together in networks? Of these, which cause the most problems? Definition of these "maximum payoff" areas would serve to focus the activities carried on in the projects. Are the maximum payoff areas uniform around the world or are there national differences?

5. Many computer networks are likely to consist of non-homogeneous computers. This is the situation now, and it is likely to be the situation in the foreseeable future. What can be done to reduce the incompatibility of these computers as far as communications and networking is concerned?

The suggestions we have made so far are very technology oriented. The following societal interests, requiring interdisciplinary work are also suggested:

6. What problems are likely to arise in the global society as the more developed nations introduce computer communications networks and data bases of increasing complexity to which underdeveloped nations have only limited or no access? What policies could be adopted in order to avoid detrimental effects of such a development? Are there any techniques which could be incorporated in the development of computer communications networks to facilitate the participation on an equal basis of countries or agencies which do not themselves own or operate large computing systems?

7. What effect will the introduction of efficient computer communications have on the relative growth of developed and less developed regions? Will development of low-cost, high capacity communication corridors encourage an associated increased concentration of urban populations? What effect would location-independent cost structures for communications have on development?

8. The international management problems for computer networks require an international body to develop a world view of developments in this area. It is possible that a form of management similar to the one presently used by the worldwide telephone network will have to be developed. A system analysis of this organization and management problem is desirable and might be tackled as an IIASA project. Combined with items 1-6 above this would lead to a scenario both for the organizational structure and for some of the required technological input.

B. Topics Which Have Already Been Proposed by IIASA

In the research proposal dated February 1974 the following items were proposed as elements of the program:

1. Develpment of a unified appraisal function for computer communication networks;

2. Development and preparation of simulation programs of data and communication networks;

3. Elaboration and improvement of evaluation methods for computers joined to networks;

4. Development of approaches for the distribution of tasks in computer communication networks which optimize the function of the complete network as a single system;

5. Provision of a program library on this subject.

In the notes dated June 14, 1974 papers on the following subjects are listed as having been produced or being in preparation:

6. Distribution of information flow in communication networks;

7. Adaptive control of vehicular traffic streams in urban areas;

8. An algorithm for flow distribution in channel switching networks.

9. Delivery and loading times for store and forward and channel switching networks.

We would like to offer some comments on the above items for the consideration of the project personnel.

Comments on Item 1

For the purposes of comparing different proposed solutions to any specific computer communication network problem, an appraisal function or criterion is exceedingly desirable. It is most unlikely, however, that any single appraisal function would be valid for a wide variety of applications of computer communication networks or even for all the classes of users within a given application. Before attempting any work on defining such an appraisal function, it would be most desirable to select several specific types of application and examine them for common features. This is the approach mentioned by Professor Danzig in a discussion of the research program on methodology relating to forest management.

Comments on Items 2 and 5

In our experience programs and simulation languages etc., designed to cover a broad range of applications, are generally much less efficient, less effective and more difficult to write than simulation programs devised for a specific application. They are also frequently difficult to use unless the user has been intimately involved with their preparation. Our experience has been that the majority of researchers would much rather go to the trouble of writing their own specific simulation programs rather than use a general tool whose structure and behavioral characteristics they may not fully understand. We suggest, therefore, that it might be best to leave the development of simulation programs or languages until a specific problem has been defined. On the other hand a study of simulation and performance evaluation methodology, drawing on world wide experience, might be a very useful undertaking

Comments on Item 4

The most efficient approach to the distribution of tasks in a network is also likely to be very application dependent.

Summary of Comments

The common factor in the comments given above is that many of the topics are very application dependent so that work might be most productive if it were tied initially to a specific application. One approach to finding a suitable application might be to identify a specific problem area being investigated elsewhere in IIASA in which the real world operations involve networks. Water resources management and other ecological problems are possibilities.

IV. A REVIEW OF SOME CANADIAN PROJECTS

A. Transaction Oriented Network

The Communications Research Center of the Department of Communications, in conjunction with Edmunde Newhall Associates Ltd. of Toronto and the Universities of Waterloo and Toronto, is carrying out research on switched computer networks based on distributed intelligence implemented by means of medium scale integrated circuits and microprogrammed logic. Switching is achieved by setting up virtual circuits achieved by the time division multiplexing of packets. The emphasis is on techniques and problems related to the use of computer networks for business transaction processing although some aspects of the project also relate to scientific computing. In a network of the former type it is more likely that the various computing facilities would be selected as a package or that, at least, they would have operating systems and communications processors which would be mutually compatible than is the case in networks of general purpose scientific computers. The project commenced with a study and a hardware development program of a basic communications system which would be suitable for application with great flexibility in many diverse parts of such a network including: the interconnection of terminals, the connection of terminals to computers, and the interconnection of the various elements of a distributed computing system. Thus the communication system has been designed to:

- interface with a wide variety of terminals and computer system components;
- accommodate a wide variety of transmission speeds;
- provide switched interconnection by means of virtual circuits;
- have low per unit cost;
- grow gracefully;
- eliminate the necessity for centralized control of switching;
- provide buffering capability at the communications interface;
- allow efficient use of a shared communication line by devices having widely varying traffic characteristics.

The basic communications structure selected was a loop system in which various devices are connected in series on a closed transmission loop. Transmission is by means of packets carrying address and control headers. The various devices are interfaced to the loop by a "distributed-switch" unit which contains all the necessary intelligence for interfacing to the specific characteristics of the device, forming and multiplexing packets and setting up virtual circuits. Multiplexing is achieved by the use of a pass-control signal which must be detected on and deleted from the loop by the distributed-switch before a packet can be transmitted. The pass-control signal is then retransmitted so that a subsequent device can transmit. A supervisory unit is located on the loop. It monitors the loop to detect, diagnose and effect recovery from correct fault conditions. It has no responsibility for switching. Loops can be interconnected in a hierarchy by means of nodes or gateways.

A single-loop system has already been built and tested. Building on this experience we are now constructing a two-level hierarchy of five local and regional loops which will connect four different locations in three cities. This will be used to investigate the problems of combining loops based on local distribution and long-haul transmission facilities. The local loop facility at the University of Waterloo will eventually be used in an experimental data-sharing network of homogeneous processors and peripherals. Initially, the system will interconnect standard terminals, time-sharing computers and some mini-computers.

All hardware in the system is based on micro-program controlled, integrated circuit logic. By using this technique we obtain the very flexible approach we need, together with the advantage of potential for large batch production which is essential if costs are to be low.

The experimental system is described in the following paper:

Reference: Newhall, E.E., Kaye, A.R., Nagel, T., and Manning, E.G. "An Experimental Loop System for Data Transmission" to be published.

B. Computer Networks Information System

The University of Waterloo is building up a Computer Networks Information System. This is a computer based index of literature relevant to computer networks which currently contains about 500 entries indexed by keywords, author, source, date and project names. A current listing is supplied as a background paper. The University is prepared to consider requests for assistance or proposals for cooperation. The contact is Dr. E.G. Manning, Computer Communications Network Group, University of Waterloo, Waterloo, Ontario, Canada.

Reference: Valantin, R., "The Computer Networks Information System Users Guide," Department of A.A. & C.S., University of Waterloo, Waterloo, Ontario.

C. Measurement of Computer Networks

Personnel at the University of Waterloo are developing a hardware and software system for measuring the performance of a computer network. The measurement system is intended to:

(a) allow easy and accurate measurement of performance;

(b) allow network accounting systems to be developed;

(c) allow the network to be adaptive and change its policies to improve its performance.

The network measurement system consists of a set of program-controlled hardware monitors which are attached to the network undergoing measurement. Each monitor contains probes which are attached to each node of the network and which gather information about the node, such as instruction counter, operand addresses and status indicators. The monitor will be controlled from, and transmit measurement data to, a measurement processor via communication lines.

A monitor and measurement processor have been constructed and are now being tested on a small network in the laboratory.

Reference: Morgan, D.E., "Performance Measurement in Computer Networks," Computer Communications Network Group Report, University of Waterloo, April 1973.

D. Reliable Data Transmission

This study being conducted at the University of Waterloo aims at producing a transmission link which offers high data rate and low error rate performance. This performance is to be achieved by introducing sophisticated signal conditioning and signal processing schemes at both ends of present physical communications channels to combat channel impairments.

Extensive work has been done on signal processing techniques and some of these techniques are now being tested in the laboratory. Many of these techniques will also be tried in satellite transmission when the Communications Technology Satellite is launched.

Reference: Mark, Jon W., "An Innovations Approach to Adaptive Data Compression," in Data Transmission Computer Communications Network Group, External Report 17, University of Waterloo.

E. Programmable Time Division Multiplexing (PTDM)

There is a broad spectrum of types of data traffic which have to be carried by the communication traffic. At one end of the spectrum is transaction traffic which consists of short messages evoking rapid short responses. At the other end of the spectrum is file traffic which consists of long messages with no constraint on response time. It would be helpful to develop switches and switching discipline which could handle both types of traffic, and intermediate types as well.

The key concept here is that the user should be able to negotiate with the network to obtain a data rate required for a specific call and the rate should be variable as the call progresses. These concepts suggest a generalization of Time Division Multiplexing (TDM) so that the number of time slots per call can be varied dynamically. Such a switching discipline is called Programmable Time Division Multiplexing. Research in this discipline is presently being conducted at the University of Waterloo. A preliminary design of PTDM switch has now been completed and simulation studies of the design are underway.

References: Manning, E.G., "Newhall Loops and Programmable TDM Two Facets of Canadian Research in Computer Communications." Proceedings, International Conference on Computer Communications, October 1972, pp. 338-42.

Manning, E.G. et al "Switching Matrices for Programmable Time Division Multiplexing," Computer Communications Network Group Report University of Waterloo, November 1972.

F. Bell Northern Research Activity

A great deal of work is going on within Bell Northern Research in the general area of data networks. Generally speaking this work falls into two categories:

(a) The area of theoretical analysis and studies. This work also includes simulations leading to probem definition and solution.

(b) Work of an experimental nature. This work also includes the development of working systems based upon theoretical considerations mentioned above, and upon other parameters which may not have been available at the time of the theoretical analysis.

Dealing with the second area first, two activities are worthy of mention. The first of these is in the area of transmitting data over what we term subscriber loops, that is, over the actual wire pairs that are in place between the users premises and the switching center. Much work has been conducted on the parameters of actual loops. In general these loops are noisy, the noise being impulsive in nature rather than having a Gaussian distribution, which, in turn, leads to rather different approaches to the design of systems and hardware than might be expected otherwise. Some of this work has been published [1]. A second activity in this experimental area is the internal three city packet switched network which has been installed within Bell Northern Research, to allow access to the computer from widely dispersed geographic locations [2].

Turning now to the work of a more theoretical nature, two aspects of a major activity here should be mentioned. First, the determination of how a Packet Switched Data Network can make best use of the transmission facilities available to it. Normally, long haul facilities are designed for voice transmission. Voice signals can, by virtue of their redundant nature, stand a much higher level of Gaussian and impulsive noise, and indeed drop-outs of short duration, than can data signals. Because of drop-outs and outages in high speed links, flow control, including alternate routing, assumes tremendous importance in such areas as buffer size requirements in the nodes of packet networks. The probability of successful transmission of a block or packet of data depends upon the noise characteristics of the transmission facility. This study is concerned therefore, with the optimum packet length, maximum permissible error conditions on trunks, buffer size and occupancy, and packet delays. Three techniques are being used in this work. These are examinations of published work, theoretical analysis, and simulations using actual error statistics gathered from working networks. No published results are as yet available. Secondly, under the auspices of Bell Northern Research--Institut National de la Recherche Scientifique, work is being conducted of flow control [3]. Monte Carlo simulation techniques are being used in this work.

The foregoing is not intended to be an exhaustive list of Data Network being conducted within Bell Northern Research, but merely an indication of the general scope and depth of this work.

References

[1] Hanna, J.L., Kirk, B.C., and Moore, E.J. "Digital Data Loops Using Diphase Transmission." International Conference on Communications Philadelphia, U.S.A., June 1972.

[2] Martel, C.C. "The Bell Northern Research Network, Canadian Experience with Packet Switching Technology." International Federation for Information Processing Conference, Stockholm, Sweden, August 1974.

[3] Girard, A. "Simple Flow Control for Small Store and Forward Networks." Canadian Communications and Power Conference, Montreal, Canada, November 1974.

GENERAL DISCUSSIONS

Comment: It is a real question if IIASA can have an impact on standards by building up its own NW; raising a lot of smoke and dust to say we need these and what kind of standards to make it work. I do not think that investing a lot of money to build a NW is necessary, but it is necessary to invest a lot of brainpower to think up standards.

Comment: The last thing IIASA should do, if it were interested in making input to standard recommendations, is to get involved in its own live NW. Because there are many institutions working on NW's and building new ones. Any person who comes from them will not be unbiased. What we need is unbiased people, who are able to think of the next generation of the computer and its communications and connections. One must consider what makes it not possible to connect TYMNET and ARPA, what are the basic differences of both, and why do both work as they do?

Comment: The existing NW's should be used by more people, and IIASA especially should support them. I think it is a waste of time to build another NW incompatible with the others, one should worry about future users. It should not be difficult for IIASA to get on ARPA or TYMNET if it played its cards right. Coming to the projected IIASA NW, PDP 11/45 is the ideal machine to act as a terminal for several NWs.

Comment: You are never going to get a critical number of people to make any impact on NW design. It is just not encouraging to do any real NW research. Very rarely does anybody sit back and say: why did I have those problems; why did we go through this long dry problem of getting on a NW and getting our installation to work? What were the incompatibilities of our procedures? You should not get involved in standards, but make positive contributions to the studies.

Comment: It is not useful to build up a new NW. It is reasonable to think what protocol would have solved a specific problem or made it easier, or what are the real problems with this protocol. Very little has been done in this area. There are no outward looking people comparing one NW's philosophy with another. People working closely with a NW cannot see the forest for the trees. You need people who know the innerpart of one NW looking at the outside of another.

Comment: IIASA's principle aims in the Computer Science Group now are:

1) To create an IIASA NW, not because we are especially interested in it, but because we need to have a linkage to the NMO's;
2) The other is research on a broader scale.

Comment: NWs take a lot of money. IIASA should be in a very strong position so as to make the links to NMO's. They will only take your budget and tear it apart.

Comment: The general policy at IIASA will be to charge the NMOs for its use and linkage.

Comment: The cost is not only in the communication, but also in the manpower used.

Comment: If you need a link to a NMO, do this in the quickest, simplest manner, with the PDP 11/45 to simulate whatever the other wants to look like, but in doing that take care to note what it is that prevents you. What we really need today is a world-wide gathering of information of what stops people from connecting, using standards etc. Preparatory work is not being done at all. Yet the knowledge exists, because there must be thousands of people who have experience in putting together computer NWs. IIASA is supposed to apply systems analysis, not just do free research.

Comment: It would be a mistake to separate computer sciences from computer services.

Comment: User communities are usually just involved with their NW, and their problems have nothing to do with the NW design problems. Very often NWs are designed without the knowledge of what the user's problems actually are, besides, nobody has published a characterization of users for the last eight years. Here at IIASA the main problem is: getting reliable ordinary telephone line connections to anywhere. Once you have that, the next problem is what sort of connection to what kind of NW is going to be comfortable for individual classes of users? The question is, where should the NW's be user oriented: in standards, economics, technology?

Comment: We do not know if people like using NWs. It would be interesting to know why so many users give up using NW's, a problem which is also related to the characterization.

Comment: The question is whether one wants to offer a service or do computer science research. If one wants to offer a service, complexity minimization is more important than system integration.

DISCUSSION NOW CONTINUED ON COMPUTER TELECONFERENCING

Question: When will the next teleconferencing take place?

Answer: There are three planned conferences:

a) One at AIC, which has started already on sulphur pollution in humans;

b) One that will link the centers of our research NSF, universities and Bell Canada;

C) A Bell-Conference about travel-communication.

Question: What steps would be necessary for IIASA to participate in a teleconference?

Answer: IIASA would need a link to TYMNET; or possibly through ARPA, but it is more expensive and the negotiations more complicated. Concerning the timing of activities at IIASA, it seems as though a one year budget were allocated. Perhaps IIASA should look at a three year budget, and at the scheduled activities taking place in this time. Perhaps IIASA could plan something with EIN.

Comment: It is not clear at this moment how the resources of the various EIN centers will be allocated. Initially it may be free, but that will not last forever. The physical connection cost will vary depending on how it is done. If only a terminal is attached to one of the centers, it is the cost of a telephone line rather than a terminal, that is great. But if it is to have one node then maybe IIASA or Austria will become a member of COST 11, which has applications for grants to support the whole project. It will surely not be completely at IIASA's expenses but it could be shared by the Austrian Government or NMOs.

Question: When will EIN be supported?

Answer: It is planned to be operational from 1973-1978.

Comment: There are not many good research possibilities in the computer sciences. Teleconferencing studies did not start with the users, for example. Studies have been done on human factors, not in time-sharing systems, but in systems used by the general public, which are much more important in the future. One must distinguish, when one talks about the users of computers, between those who simply want access to the nearest scientific computer, and the end user of the computer information systems of the future. IIASA, while providing good computer services to other groups should also take an interest in the future of computer users.

CLOSING COMMENTS BY DR. A. BUTRIMENKO

It is clear that IIASA should not only concentrate on installations, but should extend activities. It should investigate, for example, the impacts, criterion, and economics of NW's.

IIASA's Computer Science Group should aim to:

a) Improve computer services;

b) Cope with new demands, like the need for large data banks for the different projects. These data banks could be located either here or elsewhere.

People come here only for a short time. They then have to work intensively, and the Computer Science Group should be very dynamic.

The conference participants could make propaganda for the Institute and look for appointments to the Computer Science and Service Group.

One of the participants then expressed in the name of all conferees his thanks for the conference, for the organization, and its extras, and he thought that everybody benefited from it. Thank you.