CREATING STANDARDS FOR INTERCONNECT SYSTEMS

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The following article addresses issues of concern to all engineers whose areas of technical interest are in one or more of the many communications transmission technologies. These technologies are converging as a result of the maturing of interactive data retrieval systems. Cable television systems are but one of the more important of these technologies. This article offers the reader an update on the current state of standardization in the field of hybrid interactive data networks. It also introduces system modelling which includes, as an important physical subnode, a two-way CATV transmission system interfacing in a larger hybrid interconnected data-based network.

Hybrid interconnect systems, or systems handling various forms of data from multiple-input ports using multiple transmission media, pose special problems for designers who are concerned about channel utilization, bandwidth efficiency and related control, routing and congestion problems. Global solutions to these design issues remain elusive primarily because of a lack of recognized standards for interfacing such systems. This article will summarize some of the more salient aspects of the standardization problem for a broadband coaxial based interconnect system from a practical viewpoint. It will also relate the current efforts of various international committees that are working toward a solution to the standards problem, describe the components of the problem and the attendant difficulties and conclude by citing recent experience at Communications Technology Management, Inc., in developing an interactive data services network (IDSN).

A DUBIOUS TASK

In communication systems recoupment of original investment and healthy return on investment are important up front considerations. Low-channel utilization, inefficient use of bandwidth and faulty network routing and control schemes are costly. The nonexistence of standards for the complex interconnect systems make predictions for successful system implementation a highly dubious task. Because standards are not readily available, machine-to-machine incompatabilities and non-transportability of system software usually result in inefficiencies in bandwidth utilization and a slowdown in thruput between nodes. Nonstandard interfaces to "foreign" devices must be made transparent to the system software which provides a protocol or handshaking feature for interprocessor communication. In network theory terminology, this amounts to a gateway processor which is an added expense for interconnection. Each buffering operation costs not only dollars but time in terms of thruput. If thruput is diminished, response times suffer and customers begin walking away or reneging from service. So from a dollar and cents standpoint, a universally accepted standard would (a) improve compatability and transportability of hardware interface and system software; (b) minimize the number of intermediate devices, or gateways, which perform speed and code conversion; (c) minimize the reliance on a particular vendor; and (d) provide a common language for data consumers and design personnel.

What form should the standard take? The format should contain a consistent set of rules which govern data communication between any two points in a network. This set of rules often referred to as a protocol has as many facets as it needs to define the procedures for a virtual encyclopedia of data communication related functions. There are various attempts being made at adding some structure to the encycolpedia.

The International Standards Organisation (ISO) has undertaken this monumental task and defined a general protocol structure termed the "Open Systems Interconnection" (OSI) reference model. It used fairly general rules to distinguish the protocol functions performed by each layer using a hierarchical structure. The definition of the protocol layers was developed applying a MINIMAX strategy. The concept is to localize functions to a single layer where significant overlap among the functions existed and thereby minimize both the number and complexity of the interfaces between layers. New layers were created only when either (i) a different "level of abstraction" was needed to distinguish between functions being performed or (ii) the information flow could be minimized. The resulting model contained seven layers as seen in Figure 1. The basic concept is that a user on machine "A" has a "message" to exchange with a user on machine "B". The message may be a task executing in machine "A" which needs to fetch data from machine "B" to complete its transaction. The sender's message unit is decomposed into successively smaller units (packet-frame-bit) as the level of communication service gets closer to the physical transmission level of a binary bit stream and then is reassembled up the line to the receiving application task where it is finally serviced. An example of this process is provided in Figure 2.

A LANDMARK IN INTERFACING

The International Telephone and Telegraph Consultative Committee (CCITT) also is chartered to make recommendations on data communications interfaces. Its work in packet switching standards has helped it pioneer this field. Some of the standards have gained international acceptance. A good example is CCITT V.24, also known as EIA RS232. As far as interconnected broadband data service networks are concerned, the X.25 standard is a landmark for computer to computer interfacing on packet switched public data networks.

Packet switching is an extremely important technology in the data communications field. The traffic characteristics associated with most information services tend to be bursty and the duty cycle for any particular subscriber on the network is extremely low. The implication with regard to network design needs to be fully understood. An analysis of the characteristics of the three basic forms of switching (circuit, message and packet) in view of the traffic characteristics associated with information services clearly favor packet switching. Packet switching will facilitate high transmission facility utilization; a more flexible form of network routing; flexible message handling independent of message type; minimal network transit delay; and adaptive flow control. Most of these advantages are intuitively obvious, as a comparison with circuit switching and message switching will show. Getting back to the CCITT standards work, we cite the X.25 recommendation once again. X.25 consists of three protocol levels: the physical, line and packet protocols. These are conceptually similar to the three lowest OSI model layers. Table 1 summarizes the CCITT recommendations for packet switched networks that are important to large-scale interconnect systems relying on packet switch techniques.

Another important development in the standards arena that has relevance to this article is the work on the IEEE Local Network Standards Committee. This committee is taking on the ISO model one step at a time. Its primary focus thus far has been on layers 1 and 2. The IEEE functional requirements for the IEEE 802 standard addresse transmission line lengths, data rates, media independence features, reliability and freedom from dependence on intermediary devices.

At the physical layer, the IEEE has defined the Media Access Unit (MAU) as the device which will interface to a particular kind of transmission line. Physical layer functions (i.e. coding/decoding, synchronization, and related handshaking signal procedures) will be performed on the data terminal equipment side of the MAU interface.

The layer 2 standard, the link layer, takes the binary bit stream from the physical layer and forms frames. It prefixes the frame with source and destination addressing bits, places additional bits and frame synchronization bits into a control segment of the frame. It also appends the frame with a code that permits the receiving station to authenticate the correctness of the transmitted bit stream.

The IEEE 802 standard is also attempting to address the method of access rights to the transmission line. It is currently investigating contention access and token passing. A significant amount of information is available on these schemes in the literature, and both have their merits. We make no pretense as to which is better. Each network designer should evaluate his network model to select the most appropriate strategy, based upon his traffic statistics and related parameters.

RELATED PROBLEMS

While these committees have addressed the universal need for a set of rules governing intra- and internetwork data communications and have published recommendations for same, a set of related problems confront network design engineers which are even less likely to be solved by a standard. We have summarized the principal technical issues requiring resolution for complex interconnect systems in Table 2 using the OSI reference model layers. Each issue will be defined and related to a generic interactive data services network.

Prior to this discussion we introduce the network topology of our generic IDSN (reference is made in Figure 3). Two primary nodes, A and B. form the backbone of this multi-star network. Two levels of subnodes are shown. The level one subnode A_i and B_i are the primary interconnects to the nodes A and B. The second level's sub-nodes are shown with small letters and are double subscripted, a_{ij} and b_{ij}. The distinction between nodes and sub-nodes is made primarily because of the bandwidth and processing properties at each level. Each nodal entity has the capability to communicate with any other nodal entity in the network. An example of this using the OSI reference model protocol will be shown later.

Let's assume that all second level subnodes are CATV subscribers who are vying for one of the interactive data services shown in Table 3 (the contents of this table will be described later) and, therefore, have some processing capability. Furthermore, assume that the first level subnodes are cable headends (A_3-A_6) when connected to a second level sub-node (a_{31} . . . a_{3n} . . . a_{4n} . . . a_{6n}), or are foreign networks (A_1 & A_2) vying for service through the primary node (e.g., a DEMS system or a TELCO trunk). The physical interconnect involves coaxial cable, digital microwave, and satellite. The generic interfaces are summarized at the physical level in Table 4.

PHYSICAL LAYER PROTOCOL

In order to address issues at the physical layer protocol, we introduce the generic services that we postulate for the IDSN. Table 3 summarizes 19 interactive services and bounds each with the important technical parameters which matter in analyzing the bandwidth problem. Parameters "a-c" and "d-f" are used to compute the upstream and downstream data rates per service and per subscriber. The demand factor, or utilization factor for a channel is computed using parameters "g-j". The utilization factor is then multiplied by the respective upstream and downstream data rate to obtain the average sustained bit transmission rate during some "peak-period" interval on a per subscriber basis. It is clear from an analysis of Table 3 that the upstream data bandwidth consumers are the games, telephony, telewriting and videophone services. This holds true for the downstream data rate as well.

How then is this information used to compute a bandwidth requirement for a system as complex as is shown in Figure 3 ? The answer involves modeling of the network as a whole, at single step intervals. Initially, we need to examine as realistically as possible the true demand placed on the cable headend (A_3 for example). This can only be done by estimating the traffic parameters during the peak period for a nominal subscribership. If we assume the majority of the information of interest is resident in the database maintained at the headend and that one broadband cable channel is used per each direction (say 24-30 MH_z upstream and 220-226 MH_z downstream), we can begin budgeting the channels for a maximum subscribership. Let's assume that headend A_3 is being modeled. We know, given "n" subscribers on that channel, that the total demand for service placed on the headend is the sum of the individual arrivals from the "n" subscribers to the headend plus the arrivals for service coming from A and the locally originated material which requires processing support. If we assume no contention on the channel, or a discipline which effectively orthogonalizes the channel into a set of fixed slots with some sort of token passing, we can compute a maximum subscribership for a processor of size "M" where the average response time is to be no greater than "k" seconds. This is not necessarily the most efficient use of the channel and will certainly limit the ultimate subscribership. Analyzing the channel with contention involves the use of a more dynamic channel allocation model, but this will generally permit a larger subscribership and more efficient use of an otherwise scarce resource.

MODEL PROTOCOL LAYERS

Before expanding the complexity of the network model to analyze the intricacies of the other interfaces and arrival/service rates, it is instructive to illustrate the interoperability of the OSI reference model protocol layers. Figure 2 overviews this process.

Given a message "m" produced by a task running in the application layer, an information originator in the primary node has prepared a new product data set that he wants inserted into the database of one or more headends (A_3-A_5, B_1-B_2) and is requesting that this change be broadcast to subscribers downstream (a_{31}-a_{3n}, a_{41}, a_{51}-a_{5n}, b_{11}-b_{1n}, b_{21}-b_{2n}). The application layer task passes this data set into a compressed form via a suitable technique to use more efficiently bandwidth and digital storage at each headend and passes a new message "M" to the session layer. The session layer does not modify "M" but simply regulates the direction of flow of messages between the presentation layer and the transport layer.

The transport layer takes the variable length message "M" formed by the presentation layer and decomposes it into a set of smaller fixed length messages (or packets) and prefixes each with a header. The header will include control information, such as sequence numbers, to allow the transport layer on the destination machine to reassemble the data frames (i.e., data frames may be transmitted out of order as a result of retransmission or some other congestion anomoly).

The network layer is responsible for determining which physical line to the destination is to be used as the transmission path. It converts the logical line suggested by the application layer message "m" to the physical line via a communication routing table which lists all physical paths from source to destination. The paths actually selected will depend upon circuit status and traffic statistics, since in a fully connected multistar topology, multiple logical lines between two communicating entities will exist.

In this example, node B routes the packet stream to primary node A and to B_1 and B_2. The network layer will also attach its own header and pass the data units to the link layer. The link layer adds a header and trailer to each packet. The trailer is appended to the packet for purposes of error detection and correction. It passes the augmented packet to the physical level protocol for transmission at the receiving machine (A_1, B_1, and B_2); the packets are serviced; their corrections checked; headers stripped-off; and the message is reassembled by performing a sequence of operations which are the mirror image of those used in going from layer 7 to 1. The servicing of this message will tell the communications tasks executing in A, B_1 and B_2 to retransmit the newly acquired data set after storing. A dispatches this data set to A_3-A_5. The foreign networks A_1 and A_2 will be told that a change to the database has been made when a request by the foreign network is made for some service. The respective cable headends (A_3-A_5 and B_1-B_2) will update their respective databases and dispatch the message to the cable subscribers currently logged on.

An analysis of the network model between A_i and B_i and B requires that soulutions to the congestion, routing, flow control and buffering problems be found. Routing and flow control are often cited as the two most important factors in determining the performance of a network. Inefficient control schemes chew up CPU time and network bandwidth, often resulting in congested networks and deadlock states in buffer pools. Ideally, a routing and flow control mechanism will not consume resources. Today's modern data communications networks all use some form of adaptive routing. That is they use information on the current state of the network to base their routing decisions on. Our generic IDSN is no different. It should include some form of adaptive routing.

An adaptive routing scheme which has good efficiency for a generic IDSN such as we have described was originally proposed by Boorstyn and Livne. It involves the use of two-step heuristic. Step one requires the solution of an assignment problem where there is a search for an assignment of paths. This is made for each pair of nodes that need to communicate, that are "good" in some sense. Rather than employing a possibly exhaustive search for the "best" path, the path selection might be based strictly upon use of low utilization circuits, the minimum number of hops between two nodes, or some other decision criteria. Ties found during this selection process are handed-off to the next step of the heuristic where the departure of a packet from a node is modeled as a multiple choice, chance-constrained queuing problem. A comparison of this method with nonadaptive routing methods indicates that it is possible to reduce the time delay in packet processing by a factor which is roughly the equivalent of the average degree (# of outgoing links from a node) of the nodes in the network.

ROUTING TECHNIQUES

Efficient routing techniques may not be sufficient in a "store and forward" environment since blocking, a particularly disastrous form of contention, can still occur and idle network resources. Kaufman, Gopinath and Wunderlich have proposed the use of a "structured buffer pool with reservation" to eliminate the possibility of node-to-node blocking. In this procedure, packets enter a nodal facility and are placed into the "inboard" queue of the overall buffer pool existing at that node. Each packet is then out processed by the routing and switching processor (perhaps the adaptive routing scheme suggested by Boorstyn). Packets destined for other nodes are placed into an "outbound" queue in the buffer pool. Packets are sent when there is space in the inbound queue of the receiving packet switch and other related network processing (e.g., acknowledgements) has been completed. The analysis by Kaufman et al shows how to set-up the structured buffer pool with reservation, assuming that arrival and service rates are known a priori for the network.

It is hoped that this article has imparted some flavor for the difficulty of fielding complex interconnect systems a problem made worse by the virtual nonexistence of "recognized" standards of relevance. However, based upon the work of the various committees and the lessons being learned by the numerous experimenters today, future endeavors are anticipated to be much less cumbersome and more efficient in general. The more successful ventures will have "solved" the types of problems discussed here.

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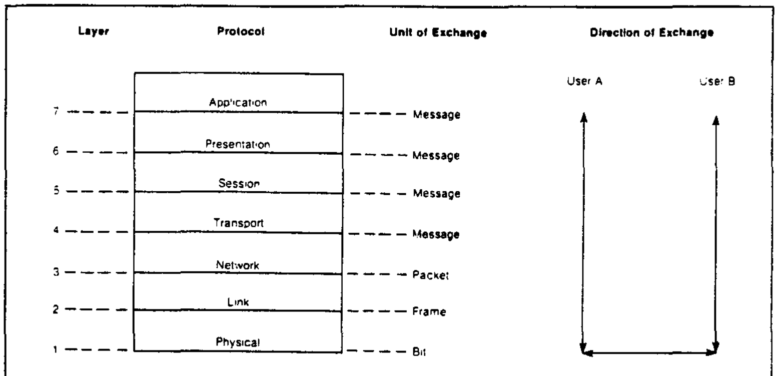

Figure 1 **The Seven Layers of the ISO-OSI Model**

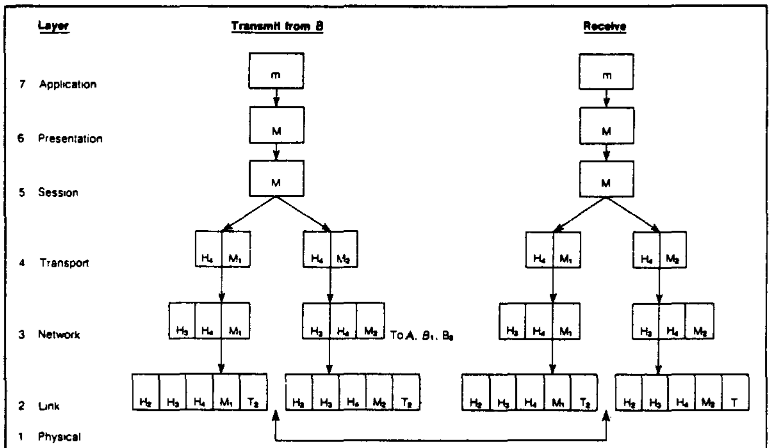

Figure 2 **Tracing the Multi-layer Protocol for a Sample Transaction in ISDN**

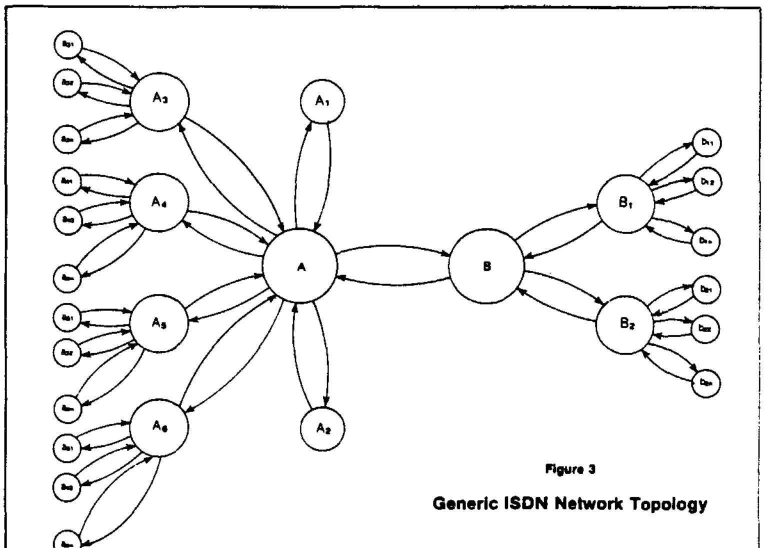

Figure 3

Generic ISDN Network Topology

Table 1

CCITT Recommendations for Packet Switching Networks

X.1	User classes of service
X.92	Hypothetical reference connections
X.95	Network parameters
X.121	International numbering plan
X.2	User facilities
X.3	Packet assembler-disassembler (PAD)
X.96	Call progress signals
X.9x	Architecture model
X.25	Data terminal equipment/data circuit-terminating equipment interface for packet mode terminals
X.75	International inter-change signaling for packet-switched networks
X.28	DTE/DCE interface of start-stop DTE accessing the PAD
X.29	Interworking between a PAD and a packet mode DTE

Table 2:

Key Design Issues in Complex Interconnect System Protocol Layers

Protocol Layers Design Factor	1 Physical	2 Link	3 Network	4 Transport	5 Session	6 Present.	7 Applic.
Bandwidth	X						
BER	X						
Signal levels	X						
Modulation	X						
Framing		X					
Synchronization		X					
Transmission line sharing		X					
Error detection		X	X				
Error correction		X					
Connection establishment		X					
Congestion			X				
Routing			X				
Formation/deformation of packets			X				
Connection to foreign devices				X			
Flow control				X			
Buffering				X			
Network security and privacy						X	
Encoding/decoding						X	
Distribution of data							X

Table 3:

General Services and Key Technical Parameters For an IDSN

	a	b	c	d	e	f	g	h	i	j
Advertising	L	L	L	H	L	L	L	L	B	L
Info retrieval	L	L	M	M	L	M	H	M	B	L
Interest matching	L	L	L	M	L	L	M	L	B	L
Messaging (short)	L	L	L	H	L	L	L	M	B	L
Electronic mail	H	L	L	L	L	L	H	L	B	L
Commerc. trnsctns	L	M	M	L	M	M	M	M	B	L
Questionnaires	L	M	M	H	M	M	M	L	B	L
Auction bidding	L	L	M	M	L	M	M	L	B	L
Pers. database	L	M	M	L	M	M	H	M	B	L
Computation	L	M	H	L	M	H	H	L	B	L
Games	L	H	H	L	H	H	H	L	C	M
Education	L	M	H	H	M	H	H	L	B	L
Telephony	H	H	H	H	H	H	M	H	C	L
Videophone	H	H	H	H	H	H	M	L	C	L
Facsimile	H	L	L	M	L	L	L	M	B	L
Telewriting	M	L	L	L	L	L	H	M	B	L
Home security	L	M	L	L	M	L	L	H	B	L
Remote meter rdg.	L	M	L	L	M	L	L	L	B	H
Energy mgmt	L	L	L	L	L	L	L	H	B	L

Legend:
a— upstream bits/interaction (a≤1K=L; 1K<a≤50K=M; a>50K=H)
b— upstream interactions/second (b≤0.1=L; 0.1≤b<0.5=M; b>0.5=H)
c— upstream interactions/call (c≤10=L; 10<c≤100=M; c>100=H)
d— downstream bits/interaction (d≤1K=L; 1K<d≤50K=M; c>100=H)
e— downstream interactions/second (e≤ 0.1=L, 0.1<e≤0.5=M; e>0.5=H)
f— downstream interactions/call (f≤10=L; 10<f≤100=M; f>100=H)
g— call duration (seconds) (g≤100=L; 100<g≤500=M; g>500=H)
h— call frequency (calls/month) (h≤30=L; 30<h≤100=M; h>100=H)
i— transmission type (B — bursty; C — continuous)
j— penetration (%) (j≤25=L; 25<j≤50=M; j>50=H)

Table 4:

General Physical Level Interface Summary

Interconnect	Example	Transmission
Primary node-primary node	A — B	Satellite
Primary node-1st level subnode	$A - A_1$	Microwave
1st level subnode-2nd level subnode	$A_3 - a_{31}$	Coaxial cable