

A PROTOCOL FOR INTEGRATING VOICE AND DATA ON FDDI/ETHERNET INTERCONNECTED NETWORKS

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ABSTRACT

FDDI is a high performance, high bandwidth network that can be used as a backbone for LANs located within a large area (e.g., a campus, a city, etc.). In this paper we design an FDDI/Ethernet interconnected network which can provide integrated voice and data services. The system configuration is described and a protocol for providing the services on this interconnected network is proposed. Furthermore, we formulate the maximum number of the concurrent telephone calls that our FDDI/Ethernet interconnected network can support in a mathematical expression.

Theoretical analysis shows that for our model there can be up to 281 voice calls on each Ethernet-like LAN concurrently. In the model, each Ethernet has also a capacity limit on the number of remote calls. By reducing this limit from 281 to 140, we can increase the maximum number of FDDI bridges from 10 to 21. In fact, by reducing this limit to be 281 divided by n , we can increase the number of FDDI bridges to be about n times as large as 10.

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I. INTRODUCTION

Technological advances in the design of networks, specifically local area networks, have continually increased the number of different services that can be offered over these networks. Combining voice traffic and data traffic on a single network has the potential for reducing overall communication costs and simplifying the design and implementation of services that depend on voice and data communication (e.g., electronic voice mail).

Reference [5] describes a simulation study of a class of carrier sense multiaccess buses with collision detection, similar to the Xerox Ethernet. Various backoff algorithms are considered, to investigate the suitability of the approach and to suggest an appropriate family of architectures for communication within this medium.

In [6], a technique for local-area communication among a large number of digital speech terminals is presented. Access to the medium is controlled by a carrier-sense multiple-access protocol with collision detection. Several distributed access control algorithms are described, and a technique for predicting the performance of these algorithms is presented. Computer simulations were performed to verify the feasibility of these ideas.

Reference [7] reports on experimental implementations of packet voice communication systems over two types of personal-computer-based local area networks. The first is a Proteon proNET token-passing ring network, and the second is an Ethernet network. Models of network performance to estimate the maximum allowable number of active voice stations without incurring intolerable packet loss are presented for each system.

The advances in Fiber Optic Technology have dramatically increased the number of new services that can be provided by communication networks. In addition to interactive computer users, a network can now support voice conversations, video images, or real time control packets. Voice, video and control packets are real time applications. Not only must they be given priority

in transmission but also they must guarantee delivery within a fixed amount of time. It is well understood that voice traffic requires not only real-time deliveries, but also synchronous deliveries (i.e., the time between any two consecutive deliveries has a bound, too), whereas longer delay may be tolerable by data traffic.

In this paper, we proposed a voice-data integrated high performance LAN which has Ethernet-like LANs interconnected by an FDDI backbone through bridges; each Ethernet-like LAN transmits the voice packets by virtual circuits and the data packets by a modified CSMA/CD protocol. In section 2, we give an overview of the FDDI standard. The PMD, PHY and MAC sub-layers and the SMT standard are described. In section 3, we present the configuration of our network model and the protocol principle. The system operations are described and a protocol for providing the call setup/release, holding and reconnection services on this interconnected network is proposed. In section 4, we analyze the theoretical bound on the number of voice calls on this network. Theoretical analysis shows that for our model there can be up to 281 voice calls on each Ethernet-like LAN concurrently. In the model, each Ethernet-like LAN has also a capacity limit on the number of remote calls. When we set the capacity limit of remote calls to 281, there can be up to 10 bridges attached to the FDDI backbone. By reducing this limit from 281 down to 140, we can increase the maximum number of FDDI bridges from 10 to 21. In fact, by reducing this limit to be 281 divided by n , we can increase the number of FDDI bridges to be about n times as large as 10. Furthermore, we formulate the maximum number of the concurrent telephone calls that our FDDI/Ethernet interconnected network can support in a mathematical expression. For further study, in the last section, we propose an extended network model which can be used to provide integrated services for metropolitan areas.

II. OVERVIEW OF FDDI

FDDI (Fiber Distributed Data Interface) is a high performance fiber optic token ring LAN running at 100 Mbps over distance up to 200 km with up to 1000 stations connected [1, 2, 3, 4]. It can be used in the same way as any of the 802 LANs, but with its high bandwidth, another common use is as a backbone to connect copper LANs, as shown in Fig. 1. The FDDI design specification calls for no more than 1 error in 2.5×10^{10} bits.

Fig. 2 shows the FDDI layers specified in the ANSI X3T9.5 standard. Physical Medium Dependent (PMD) sublayer defines: (1) Configuration, (2) Wavelength used in the optical transmission, (3) FO (Fiber Optic) connectors, (4) FO transmitters/receivers and (5) FO bypass. An FDDI network consists of a set of stations logically connected in a ring. Physically, the FDDI network consists of a ring of trees. Information is sent sequentially from one active station to the next one. Each station generally regenerates and repeats the information and provides the means for attaching one or more devices to the network.

Physical (PHY) sublayer defines: (1) Encoding scheme (4B/5B) and (2) Method for retiming transmission. Information is organized in 4-bit symbols. There are 3 types of symbols: (1) line status symbols – Quiet, Halt, Idle, (2) control symbols – starting delimiter, ending delimiter, control indicators and (3) data symbols. The physical layer does not use Manchester encoding because 100 Mbps Manchester encoding requires 200 megabaud, which was accounted to be too expensive. Instead a scheme called 4 out of 5 encoding is used.

Medium Access Control (MAC) sublayer defines: (1) Token protocol, (2) Packet format, (3) Addressing, (4) CRC and (5) Recovery mechanism. Token is the mean by which the right to transmit is passed from one station to another (as opposed to the normal process of repeating). Frame is used for transmitting messages between stations. The basic FDDI protocols are

closely modeled on the 802.5 protocols. To transmit data, a station must first capture the token. Then it transmits a frame and removes it when it comes around again. One difference between FDDI and 802.5 is that in 802.5, a station may not generate a new token until its frame has gone all the way around and come back. In FDDI, with potentially 1000 stations and 200 km of fiber, the amount of time wasted waiting for the frame to circumnavigate the ring could be substantial. For this reason, it was decided to allow a station to put a new token back onto the ring as soon as it is done transmitting its frames.

TRT (Token Rotation Timer) monitors the time elapsed from last token. TTRT (Target Token Rotation Timer) is the average time elapsed from token to token as seen by a station. TTRT value is negotiated among the stations when the ring is being initialized (claim token). The standard guarantees a maximum waiting time from token to token of $(2 \times \text{TTRT})$. Each station knows (from its upper layer) what is its maximum allowed waiting time. This waiting time divided by 2 gives the station's entry (T_{req}) in the bid. The station bidding the lowest T_{req} value wins the bid, and its T_{req} value is remembered by all the stations on the ring as the operative TTRT (T_{opr}).

When a token arrives, the actual value of the TRT (Token Rotation Timer) is compared with the negotiated value of the TTRT ($=T_{\text{opr}}$). If the token arrived before expiration of T_{opr} then it is an early token (i.e., the ring is not loaded) and the station is allowed to transmit both asynchronous frames (low priority) and synchronous frames (high priority). If the token arrived after expiration of T_{opr} then it is a late token (i.e., the ring is loaded) and the station is allowed to transmit only synchronous frames (high priority).

The time allocated to all the stations on the ring for synchronous transmission is equal to the negotiated value for TTRT ($=T_{\text{opr}}$) minus propagation delay D_{max} ($=\text{stations delay} + \text{cable delay}$). Each station is allowed to transmit synchronous frames for a specific percentage of T_{opr} , which is negotiated among the stations.

The time allocated to a station for asynchronous transmission is the difference between T_{opr} and the actual time elapsed from last token, as monitored by TRT (Token Rotation Timer). The MAC protocol requires each station to have a token rotation timer to keep track of how long it was since the last token was seen. A priority algorithm similar to 802.4 is used to determine which priority classes may transmit on a given token pass. If the token is ahead of schedule, all priorities may transmit, but if it is behind schedule, only the highest ones may send.

Station Management (SMT) standard defines: (1) Connection management and (2) Configuration of the station. It provides the control necessary at the station level to manage the processes in the various FDDI layers such that a station may work cooperatively on a ring. SMT shall provide services such as (1) connection management, which includes station insertion and removal, station initialization, configuration management and fault isolation and (2) recovery, which includes error detection, station isolation, configuration partitioning, communications protocol for external authority, scheduling policies, collection of station statistics, address administration and downline load and upline dump requirements.

III. DESCRIPTION OF THE PROTOCOL

A. system configuration

Fig. 3 shows the physical configuration of our system. The Ethernet-like LANs are connected to the FDDI backbone through the bridges. First, we describe the configurations of the Ethernet-like LANs. Each Ethernet-like LAN consists of a monitor station and several user stations. Each monitor station's jobs are to setup and control conversations between user-pairs and to provide the users with services necessary to maintain the conversations. Each monitor station has a clock and will intervene the activity of the Ethernet-like LAN when necessary, and at the rest of the time, allows the net-

work to operate according to a modified CSMA/CD protocol, to be described later. Each user station has a timer to monitor the time for voice or data transmission and an on/off flag, CONFLAG (CONtention FLAG, initially off), to know when it can contend for the transmission medium. More than one voice sources can be attached to each user station, i.e., the user station plays the role of a voice concentrator.

The data traffic is transmitted in variable-sized packets as shown in Fig. 7 (b), which can be independent messages or segments of longer messages. The voice traffic, from each voice source, consists of fixed-sized packets as shown in Fig. 7 (a), which are formed by packetization of the incoming bit stream created by the digitization and encoding of an analog voice signal. During the conversation, packets from a voice source arrive deterministically, i.e., one packet for every packetization period P_v second.

The network time of each Ethernet-like LAN consists of a series of cycle times. Each cycle time, as shown in Fig. 4, consists of several (monitor time, voice time) pairs and one (monitor time, data time) pair. When a conversation is terminated, the (monitor time, voice time) pair corresponding to the conversation is replaced with a (monitor time, data time) pair. The adjacent (monitor time, data time) pairs will be merged into one larger (monitor, data time) pair. The length of each cycle time is equal to the packetization period P_v second.

The monitor times are for the control purpose. Each monitor time is reserved for the monitor station to broadcast a monitor command, as shown in Fig. 5, to all the user stations on the same Ethernet-like LAN. The monitor station uses the monitor command to specify who can use the transmission medium and how long they can use. Each caller or callee address of the monitor command consists of 6-byte Ethernet station address (assigned by IEEE and unique in the world) and 1-byte voice source number of the Ethernet station (i.e., each caller or callee address corresponds to a subscriber of the interconnected network). If both caller address and callee address of the

monitor command are special values (e. g., the value of 0xFFFFFFFFFFFFFFF) then all the user stations can contend for the transmission medium using modified CSMA/CD for the time specified in time allowed field. Otherwise, the two subscribers specified in caller address field and callee address field get the rights to transmit their voice packets for the time specified in time allowed field.

The voice times are for the voice communications. Each voice time is divided into two equal halves and reserved for a pair of subscribers in conversation (caller and callee). Each conversation is allocated a (monitor time, voice time) pair in every cycle time according to the time of the connection request. The conversation with earlier connection request will be allocated earlier in the cycle time. At the end of the monitor time, the timers of the two subscribers have already been loaded with the value specified in time allowed field of the monitor command. The voice time just before the timers come to one half of their original values is defined to be the former half, and the rest is defined to be the latter half. The former half of the voice time is reserved for the caller to transmit a voice packet, and the latter half is reserved for the callee. We assume that silence detection is used with the coder. Instead of transmitting the silence voice packets, the station transmits UUSIs, as shown in Fig. 7(c), to the destination voice sources to indicate silence periods and transmits its data packets for the rest of the usable time.

The data times are for the data communications, when the CONFLAG is on. Each data time is for all the stations to contend for the transmission medium using a modified CSMA/CD protocol. The modified CSMA/CD protocol. The modified CSMA/CD protocol is as follows: the binary exponential algorithm backoffs are limited to exist in the data times only. If the remainder of the data time is not enough for a contending station to

transmit its data packet then the attempt on contending for the transmission medium will be delayed until the next data time when the CONFLAG is on.

We now describe the configuration of the FDDI backbone. Each bridge station on FDDI contains the values of two parameters. The value of the first parameter is T_{v_i} which controls the maximum allowed transmission time for synchronous traffic (voice packets) by bridge station i per token arrival. Each bridge station has a known allocation of synchronous bandwidth, which is established by SMT. Initially, each bridge station has a zero allocation, and it shall exercise the SMT protocol to change its allocation.

The second is the operational time parameter T_{opr} which is common to all stations and controls the available time for data transmission. The value of $(2 \times T_{opr})$ is the maximum delay time between two consecutive token arrivals at a given station, hence it guarantees the delay bound of the voice packets on each bridge station [9]. We want that the maximum transmission time for the voice frames waiting on the bridge must not be greater than the synchronous bandwidth allocation for the bridge. That is, we want that every time the token arrives at a bridge station the synchronous traffic is first transmitted until the voice buffer becomes empty. Let P_{max} be the maximum number of the voice packets that can be transmitted from each Ethernet-like LAN to its connected FDDI bridge station in one cycle time. For remote calls, each bridge has P_{max} buffers each of which may be assigned to a remote call. The bridge will transmit the voice frames in a First Come First Serve (FCFS) manner. For each remote call, the outgoing voice frame on the buffer will be replaced when the next one arrives, and the undelivered incoming voice frames will be buffered before transmitting. Asynchronous transmission is initiated by a bridge station only if the immediately preceding token cycle is less than T_{opr} in duration. If u_0 is the time of a complete token rotation which started and ended at the same station, then $(T_{opr} - u_0)$ is the maximum allowed time for data transmissions from this station.

We derive now for FDDI the condition that the operational time parameter T_{opr} must satisfy. Suppose that the number of FDDI bridge stations is N_v . Fig. 6 shows the most conservative maximum waiting time, W_{max} , that a voice packet might encounter for a FDDI bridge station (station N_v in Fig. 6). This worst case appears when:

- a) There are no voice or data packet transmissions during the first cycle.
- b) Bridge station N_v is the last among all bridge stations on the ring.
- c) The Ethernet-like LAN connected to bridge station N_v transmits its voice packets to bridge station N_v immediately after the departure of the token during the first cycle.
- d) Without loss of generality, we assume that the maximum possible amount of asynchronous traffic is transmitted by the first bridge station in the second cycle. This amount is equal to $(T_{opr} - u_0)$.
- e) Each of the bridge stations except station N_v transmits P_{max} voice frames onto the backbone during the second cycle.

Let H_v be the time for any bridge station to transmit a voice frame. The most conservative maximum waiting time that a voice packet might encounter is (See Fig. 6)

$$\begin{aligned} W_{max} &= u_0 + (T_{opr} - u_0) + (N_v - 1) \times P_{max} \times H_v \\ &= T_{opr} + (N_v - 1) \times P_{max} \times H_v \end{aligned} \quad (0)$$

According to the FDDI standard, the sum of the current synchronous bandwidth allocations for all the bridge stations shall not exceed the maximum usable synchronous bandwidth of the ring, which is equal to $[T_{opr} - (D_{max} + F_{max} + T_{tok})]$ [9]. F_{max} consists of the time required to transmit a maximum length frame (9000 symbols) and its preamble (16 symbols). T_{tok} consists

of the time required to transmit a token (6 symbols) and its preamble (16 symbols). Assume that $T_{v_i} = [T_{opr} - (u_0 + F_{max} + T_{tok})]/N_v$, for all i . That is, the value of T_{v_i} is the same for each FDDI bridge station.

We want the operation of our protocol to guarantee that for each remote call the outgoing voice frame in the bridge buffer will be transmitted before another one arrives at the bridge from the Ethernet-like LAN. In this way the maximum delay (i.e., maximum waiting time + transmission time) of a voice packet must not be greater than one packetization period P_v second. Using the assumption described above, we have (See Fig. 6)

$$W_{max} + H_v \leq P_v \quad (1)$$

Next, since the time from token to token must not be greater than $(2 \times T_{opr})$ [9], we have

$$W_{max} \leq 2 \times T_{opr} \quad (2)$$

Moreover, since we want that the maximum transmission time for the voice frames waiting on the bridge must not be greater than the synchronous bandwidth allocation for the bridge, we have

$$P_{max} \times H_v \leq [T_{opr} - (u_0 + F_{max} + T_{tok})]/N_v \quad (3)$$

From inequality (1) we have

$$T_{opr} \leq P_v - [(N_v - 1) \times P_{max} + 1] \times H_v. \quad (4)$$

Next, from inequality (2) we have

$$(N_v - 1) \times P_{max} \times H_v \leq T_{opr}. \quad (5)$$

Moreover, from inequality (3) we have

$$N_v \times P_{\max} \times H_v + u_0 + F_{\max} + T_{\text{tok}} \leq T_{\text{opr}}. \quad (6)$$

Finally by inequalities (4), (5) and (6), we conclude

$$N_v \times P_{\max} \times H_v + u_0 + F_{\max} + T_{\text{tok}} \leq T_{\text{opr}} \leq P_v - [(N_v - 1) \times P_{\max} + 1] \times H_v. \quad (7)$$

B. system operation

Communications between the users is achieved by the Protocol Data Units (PDUs), as shown in Fig. 7. The following is a brief description of the PDUs that are used for control purpose:

- UMCR – User station to Monitor station Connection Request
- UMDR – User station to Monitor station Disconnection Request
- UMHR – User station to Monitor station Holding Request
- UMRR – User station to Monitor station Reconnection Request
- MUDR – Monitor station to User station Disconnection Request
- MUCR – Monitor station to User station Connection Response
- UUCR – User station to User station Connection Request
- UUCC – User station to User station Connection Confirmation
- UUDR – User station to User station Disconnection Request
- UUDC – User station to User station Disconnection Confirmation
- UUCF – User station to User station Called entity Free
- UUHR – User station to User station Holding Request
- UUHC – User station to User station Holding Confirmation
- UURR – User station to User station Reconnection Request
- UURC – User station to User station Reconnection Confirmation
- UUSI – User station to User station Silence Indication

Our protocol provides services for call setup, call release, holding and reconnection.

(a) *call setup*

Suppose A2 calls A'4 as shown in Fig. 3. To establish a call, A2 must first request monitor M to allocate a (monitor time, voice time) pair for A2 and F (A2 sends a UMCR to M with caller address A2 and callee address F). If resource is not enough (i.e., the number of active and holding calls is not less than the theoretical maximum number of the concurrent telephone calls on each Ethernet-like LAN), monitor M will refuse to allocate for them (M sends an MUDR to A2 with caller address A2 and callee address F). If, on the other hand, resource is enough, monitor M will allocate for them (M sends an MUCR to A2 with caller address A2 and callee address F). A2 then requests A'4 to connect with it (A2 sends a UUCR to A'4 with caller address A2 and callee address A'4), and will be refused if A'4 is busy (A'4 sends a UUDR to A2 with caller address A2 and callee address A'4). If, on the other hand, A'4 is not busy, it will request monitor M' to allocate for A'4 and F' (A'4 sends a UMCR to M' with caller address A'4 and callee address F'). If resource is not enough, monitor M' will refuse to allocate for them (M' sends an MUDR to A'4 with caller address A'4 and callee address F'). A2 will be informed of this (A'4 sends a UUDR to A2 with caller address A2 and callee address A'4) and request monitor M to deallocate for A2 and F (A2 sends a UMDR to M with caller address A2 and callee address F). Otherwise, A'4 continuously sends ringback signals to A2 (A'4 sends UUCFs to A2 at intervals with caller address A2 and callee address A'4) and after A'4 user answers the call A'4 informs A2 of connection success (A'4 sends a UUCC to A2 with caller address A2 and callee address A'4) and both of them can start to transmit their voice packets.

(b) *call release*

Suppose A2 finishes the call first. A2 informs A'4 of disconnection of the call (A2 sends a UUDR to A'4 with caller address A2 and callee address A'4). As soon as receiving the disconnection information, A'4 informs A2 of the disconnection confirmation (A'4 sends a UUDC to A2 with caller

address A2 and callee address A'4) and requests monitor M' to deallocate for A'4 and F' (A'4 sends a UMDR to M' with caller address A'4 and callee address F'). On receiving the disconnection confirmation, A2 requests monitor M to deallocate for A2 and F (A2 sends a UMDR to M with caller address A2 and callee address F), and the call is terminated.

(c) *holding*

Suppose A2 is in talk with A'4 but A2 wants to hold A'4 and initiate a new call. A2 informs A'4 of the holding of the call (A2 sends a UUHR to A'4 with caller address A2 and callee address A'4). As soon as receiving the holding information, A'4 informs A2 of the holding confirmation (A'4 sends a UUHC to A2 with caller address A2 and callee address A'4) and requests monitor M' to hold for A'4 and F' (A'4 sends a UMHR to M' with caller address A'4 and callee address F' and M will stop allocating for them). On receiving the holding confirmation, A2 requests monitor M to hold for A2 and F (A2 sends a UMHR to M with caller address A2 and callee address F). Now A2 can initiate a new call.

(d) *reconnection*

Suppose A2 wants to reconnect with A'4. If A'4 has disconnected the call, A2 stops. Otherwise, A2 requests monitor M to reconnect for A2 and F (A2 sends a UMRR to M with caller address A2 and callee address F) and informs A'4 of the reconnection of the call (A2 sends a UURR to A'4 with caller address A2 and callee address A'4). If A'4 wishes to reconnect with A2, it requests monitor M' to reconnect for A'4 and F' (A'4 sends a UMRR to M' with caller address A'4 and callee address F') and informs A2 of reconnection confirmation (A'4 sends a UURC to A2 with caller address A2 and callee address A'4). If, on the other hand, it doesn't want to reconnect with A2, it informs A2 of disconnection of the holding (A'4 sends a UUDR to A2 with caller address A2 and callee address A'4) and A2 requests monitor M to deallocate for A2 and F (A2 sends a UMDR to M with caller address A2 and callee address F).

IV. PERFORMANCE ANALYSIS

A. theoretical analysis

To analyze the theoretical performance of an FDDI/Ethernet interconnected network, we assume that the FDDI backbone's circumference is equal to the maximum distance, i.e., 200 km. The FDDI backbone runs at 100 Mbps and each of its Ethernet-like LANs runs at 10 Mbps. The ordinary delay for each FDDI station is 600 ns and the total delay for the 200 km long ring cable is 1.017 ms [9].

The coding scheme we use to convert voice from its analog form to a digital representation is a difference-coding waveform coder whose coding rate is 16 kbps. This yields highly acceptable speech and since the algorithm for the coding is relatively simple, the coder can be implemented with low cost hardware [8].

First, we want to compute the theoretical maximum number of the concurrent telephone calls on each Ethernet-like LAN. The maximum number of the (monitor time, voice time) pairs that each Ethernet-like LAN contains is equal to P_v seconds divided by the sum of the monitor time and the voice time. The sum of the monitor time and the voice time is equal to the time required to transmit a monitor command and two voice packets at 10 Mbps. Because the size of any Ethernet packet is limited to be greater than or equal to 72 bytes, the size of the monitor command is 72 bytes. The voice packet consists of 2-byte control data, fixed-sized voice data and 26-byte overhead. The size of the voice data is equal to the coding rate (16 kbps) multiplied by the packetization period P_v . But we must reserve at least the time for transmitting a monitor command and a maximum data packet in order that data can be transmitted successfully. Thus, the theoretical maximum number of the concurrent telephone calls on each Ethernet-like LAN, expressed as a function of P_v , is

$$\begin{aligned}
f(P_v) &= \left[\frac{P_v - \frac{72 \times 8 + 1526 \times 8}{10^7}}{\frac{72 \times 8 + 2 \times (2 \times 8 + 16000 \times P_v + 26 \times 8)}{10^7}} \right] \\
&= \left[\frac{(P_v - \frac{12784}{10^7}) \times 10^7}{1024 + 32000 \times P_v} \right] \\
&= \left[\frac{P_v \times 10^7 - 12784}{1024 + 32000 \times P_v} \right] \\
&= \left[\frac{10^7 - \frac{12784}{P_v}}{\frac{1024}{P_v} + 32000} \right] \tag{8}.
\end{aligned}$$

Notice that in our model the maximum delay of a voice packet is about $(2 \times P_v)$ second and the transmission delay tolerable by a telephone circuit is about 0.6 second [10], we have

$$2 \times P_v \leq 0.6$$

$$P_v \leq 0.3.$$

Obviously, $f(P_v)$ reaches its maximum value (281) when P_v is equal to 0.3. That is, there can be up to 281 local or remote calls on each Ethernet-like LAN concurrently. The bandwidth for each conversation (in bps) is

$$\begin{aligned}
b(P_v) &= \frac{1024+32000 \times P_v}{P_v} \\
&= \frac{1024}{P_v} + 32000.
\end{aligned}$$

For example, in the case P_v equals 0.3 the bandwidth for each conversation is 35.4 kbps.

Next, we want to compute the maximum number of the Ethernet-like LANs that can be connected to the FDDI backbone through bridges. Recall inequality (7), because the upper bound must be greater than or equal to the lower bound, we have

$$\begin{aligned}
P_v - [(N_v - 1) \times P_{\max} + 1] \times H_v &\geq N_v \times P_{\max} \times H_v + u_0 + F_{\max} + T_{\text{tok}} \\
P_v &\geq [(2 \times N_v - 1) \times P_{\max} + 1] \times H_v + u_0 + F_{\max} + T_{\text{tok}} \\
P_v &\geq [(2 \times N_v - 1) \times P_{\max} + 1] \times H_v + (N_v \times 600 \times 10^{-9} + 1.017 \times 10^{-3}) + \\
&\quad F_{\max} + T_{\text{tok}} \\
P_v + (P_{\max} - 1) \times H_v - 1.017 \times 10^{-3} - F_{\max} - T_{\text{tok}} \\
&\geq 2 \times N_v \times P_{\max} \times H_v + N_v \times 600 \times 10^{-9} \\
P_v + (P_{\max} - 1) \times H_v - 1.017 \times 10^{-3} - F_{\max} - T_{\text{tok}} \\
&\geq (2 \times P_{\max} \times H_v + 600 \times 10^{-9}) \times N_v \\
\frac{P_v + (P_{\max} - 1) \times H_v - 1.017 \times 10^{-3} - F_{\max} - T_{\text{tok}}}{2 \times P_{\max} \times H_v + 600 \times 10^{-9}} &\geq N_v \tag{9}.
\end{aligned}$$

Recall that the speed of FDDI is 100 Mbps and the overhead for an FDDI voice frame is 160 bits, H_v equals $[(2 \times 8 + 16000 \times P_v + 26 \times 8) + 160] / 10^8$. F_{\max} equals 3.61×10^{-4} (second) and T_{tok} equals 8.8×10^{-7} (second)[9].

$$\text{Let } h(P_v, P_{\max}) = \left\lceil \frac{P_v + (P_{\max} - 1) \times H_v - 1.017 \times 10^{-3} - F_{\max} - T_{\text{tok}}}{2 \times P_{\max} \times H_v + 600 \times 10^{-9}} \right\rceil,$$

$P_v \leq 0.3$, $P_{\max} \leq f(P_v)$. The theoretical maximum number of the Ethernet-like LANs that can be connected to the FDDI backbone through bridges is

$$g(P_v, P_{\max}) = \min\{h(P_v, P_{\max}), 1000\}, \quad P_v \leq 0.3, P_{\max} \leq f(P_v).$$

Notice that for the case P_v is 0.3, $g(P_v, P_{\max})$ grows from 10 to 21 with P_{\max} reduced from 281 to 140. In fact, by reducing P_{\max} to be 281 divided by n , we can increase $g(P_v, P_{\max})$ to be about n times as large as 10.

Consequently, the theoretical maximum number of the concurrent telephone calls that our FDDI/Ethernet interconnected network can support is

$$\max\{f(P_v) \times g(P_v, P_{\max})\}, \quad P_v \leq 0.3, P_{\max} \leq f(P_v).$$

B. simulation results

In the following, we establish a simulation model and give the computational results. In our simulations, each Ethernet-like LAN is modeled as a LAN with 30 stations (including a monitor and a bridge) and a slot time of 10 microseconds. Data packet arrivals at each station are modeled by a Poisson process, and data arrivals at different stations are independent of each other. The length of the data field of each data packet is assumed to be uniformly distributed between 1 and 1500 bytes.

The simulation model assumes the length of time for a colliding station to jam is half of the slot time. The simulator detects a collision when a second station attempts to send a packet within one slot time after some other station has initiated transmission. Upon collision detection, the LAN is jammed,

and each of the colliding stations is backed off by the backoff time specified by the binary exponential algorithm [5]. A station will not try to transmit unless there is enough time for it to transmit.

The simulation results are shown in Fig. 8. The variable names and their brief descriptions are given below.

avgdelay	— average delay of data packet
pacperiod	— packetization period
connumber	— conversation number of the Ethernet-like LAN
lambda	— data packet arrival rate of each station

Fig. 8 shows the average delays of data packets (avgdelay) as a function of the number of voice calls on each Ethernet-like LAN (connumber). In the figure, the data packet arrival rate of each station (lambda) is equal to 10. Fig. 8 shows that, with the increase of the number of voice calls on each Ethernet-like LAN, the average data delays of the case pacperiod equals 0.1 first rise suddenly. The average data delays of the case pacperiod equals 0.2 rise second and those of the case pacperiod equals 0.3 rise last. This suggests that the average data delay of the case with smaller packetization period rise suddenly at a smaller number of voice calls on each Ethernet-like LAN. The reason for their sudden rising is that their data times in each cycle time are almost exhausted, so the collisions rapidly increase. For example, the maximum number of voice calls supported by the case packetization period equals 0.1 is 233. So, when the number of the voice calls on each Ethernet-like LAN is 200 the average data delays increase rapidly as expected.

For connumber less than 175, the Ethernet-like LAN with longer packetization period has larger average data delay. On the other hand, theoretical performance analysis shows that the Ethernet-like LAN with longer packetization period supports a larger number of voice calls. Thus, it should be careful to choose packetization period so as to minimize the delay effect on data users and to maximize the number of voice calls.

V. CONCLUDING REMARKS

FDDI is a high performance, high bandwidth network that can be used as a backbone for LANs located within a large area (e.g., a campus, a city, etc.). In this paper we design an FDDI/Ethernet interconnected network which can provide integrated voice and data services. The system configuration is described and a protocol for providing the services on this interconnected network is proposed.

The 16 kbps difference-coding waveform coder used in this paper may be replaced with a vocoder to increase the theoretical maximum number of the concurrent telephone calls on each Ethernet-like LAN (but this may degrade the quality of speech) [11].

For further study, we can use a global FDDI backbone to connect several FDDI/Ethernet interconnected networks. As shown in Fig. 9, the local FDDI backbone in each FDDI/Ethernet interconnected network may be used to connect a group of post offices, a group of hospitals, a group of departments in a university, etc. Consequently, it is possible to use FDDI to build Metropolitan Area Networks (MANs) that provide integrated services.

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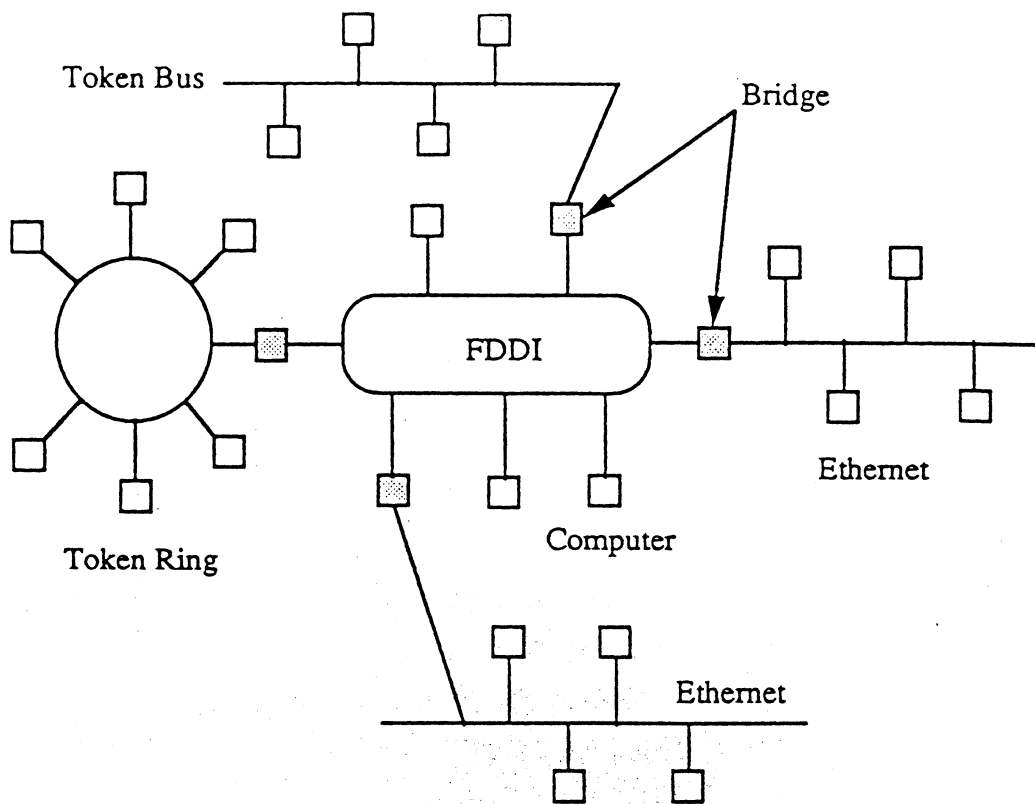


Fig. 1 Using FDDI as a backbone to connect the copper LANs.

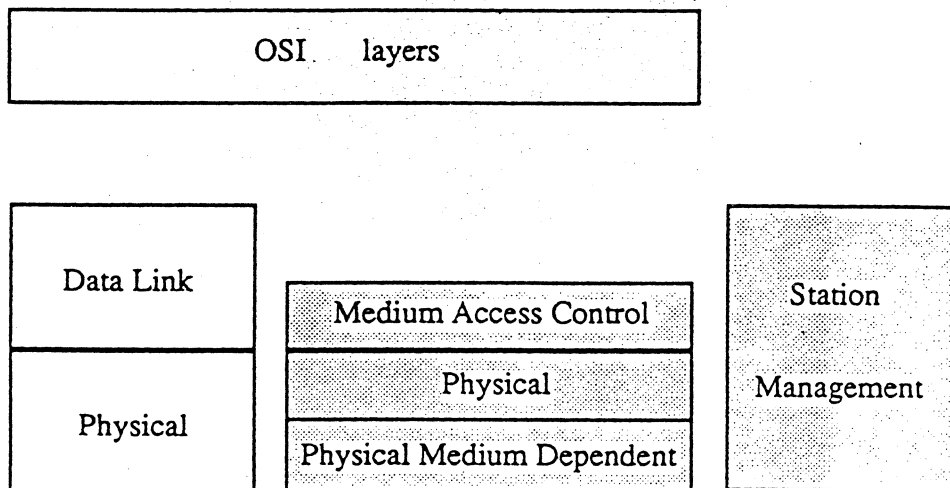


Fig. 2 The three FDDI layers specified in the ANSI X3T9.5 standard.

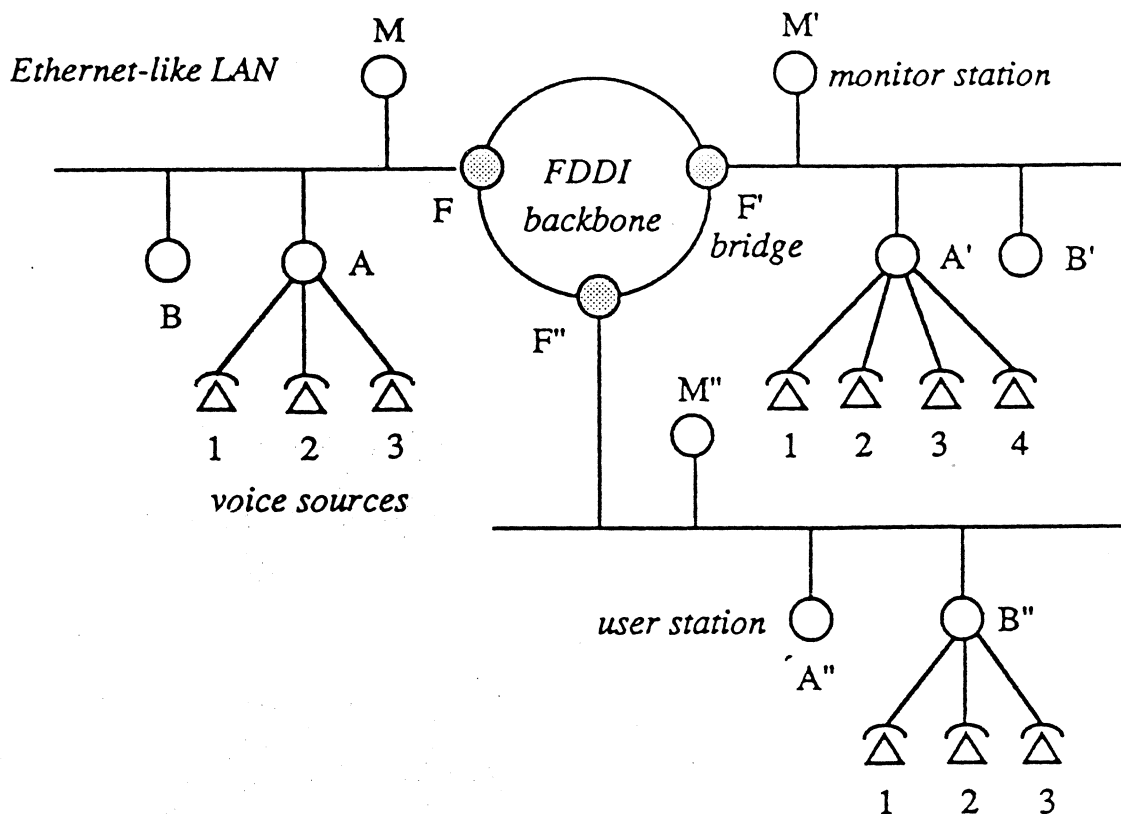


Fig. 3 The physical configuration of our system.

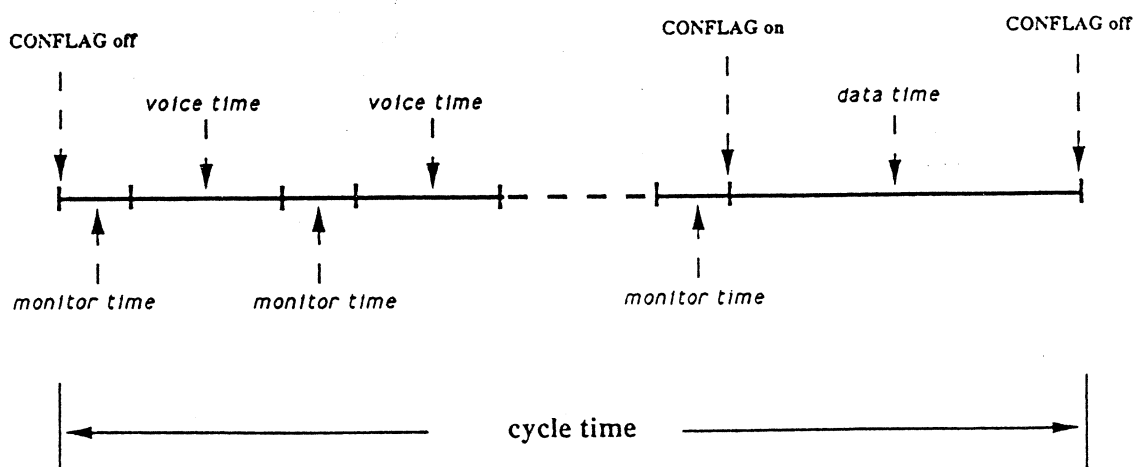


Fig. 4 The components of each cycle time.

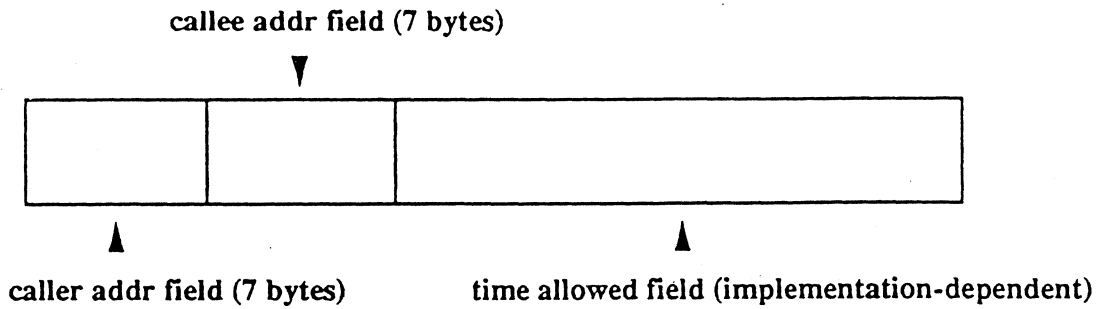


Fig. 5 The monitor command format. The monitor command is transmitted as the data field of the Ethernet packet.

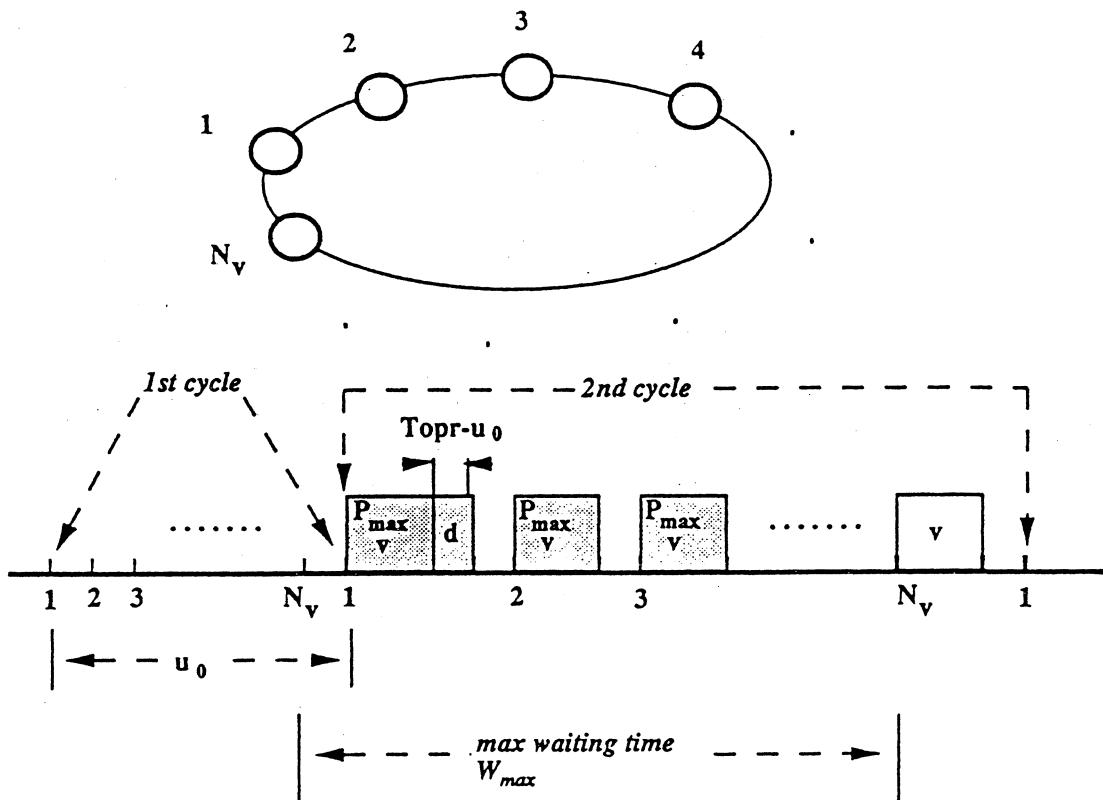
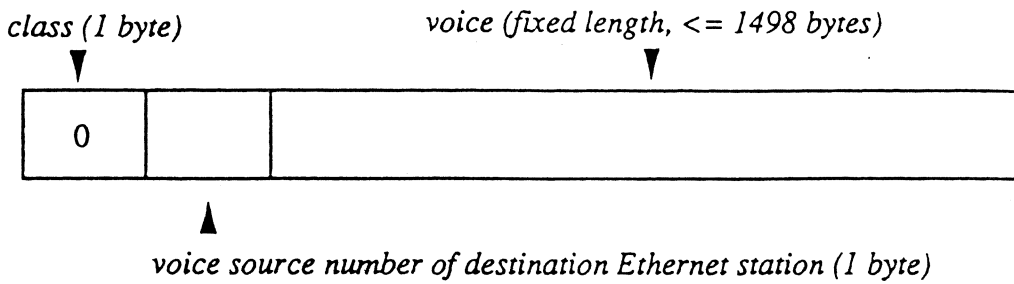
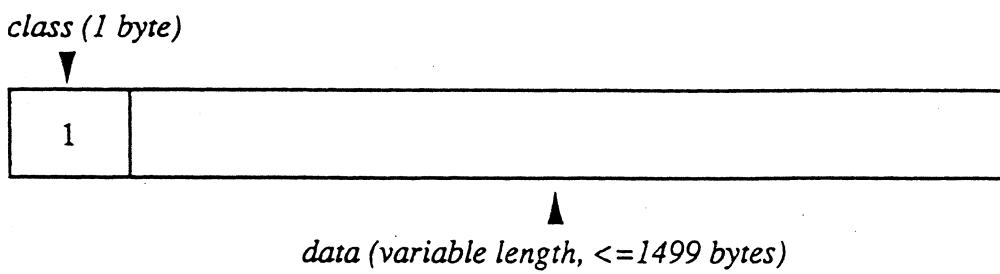


Fig. 6 The most conservative maximum waiting time, W_{max} , that a voice packet might encounter for an FDDI bridge station (bridge station N_v in this figure).

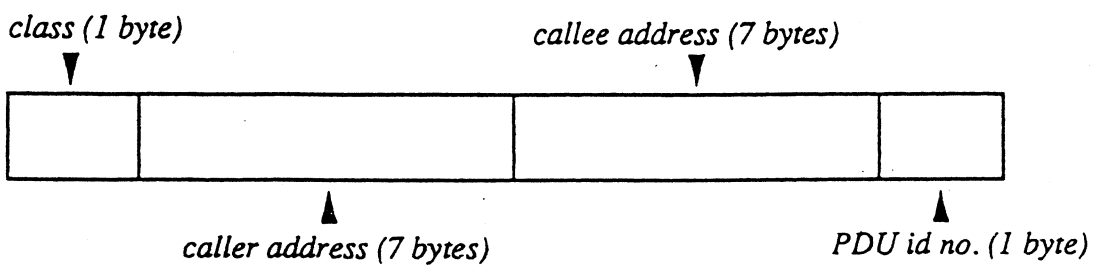
(a) voice packet format



(b) data packet format



(c) control packet format



(continued)

	class	PDU id no.
UMCR	2	0
UMDR	2	1
UMHR	2	2
UMRR	2	3
MUDR	3	0
MUCR	3	1
UUCR	4	0
UUCC	4	1
UUDR	4	2
UUDC	4	3
UUCF	4	4
UUHR	4	5
UUHC	4	6
UURR	4	7
UURC	4	8
UUSI	5	0

Fig. 7 Communications between users is achieved by the Protocol Data Units (PDUs), which are transmitted as the data field of the Ethernet packet.

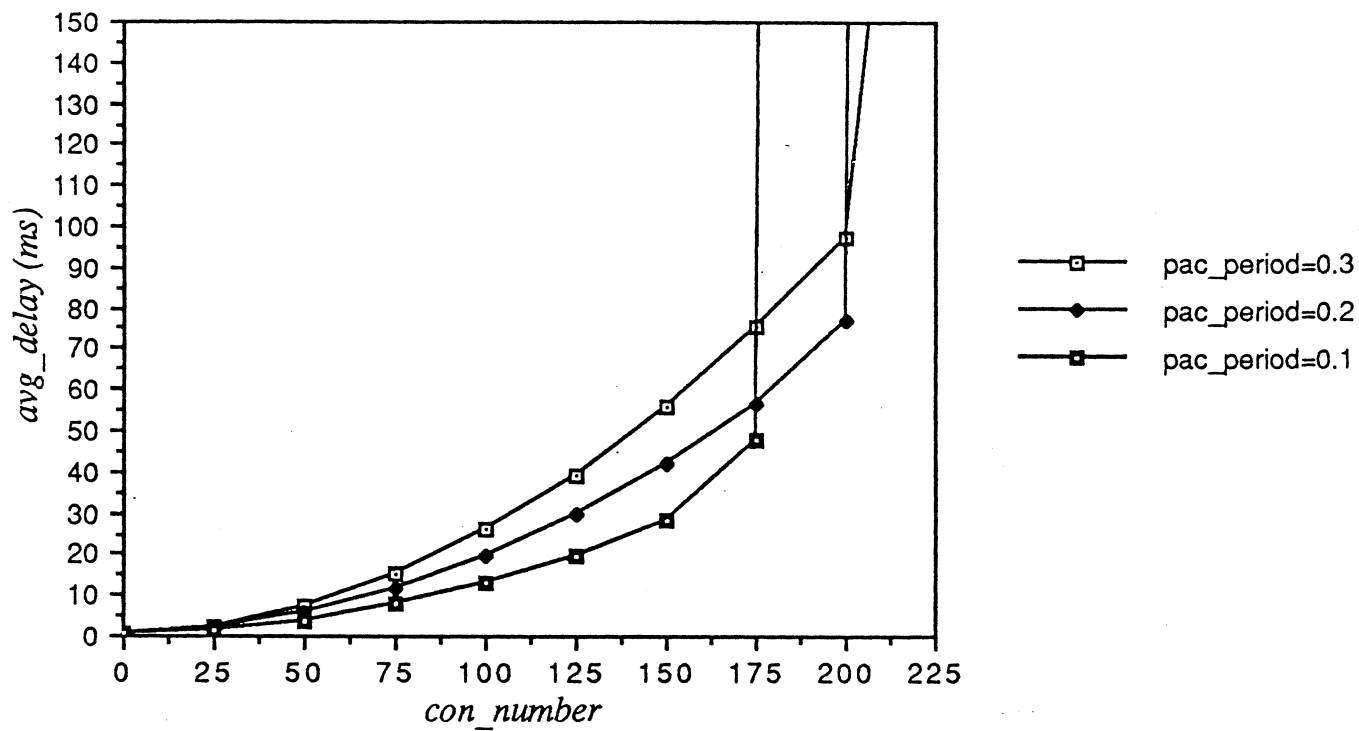


Fig. 8 Average data delay as a function of the number of voice calls ($\lambda = 10$).

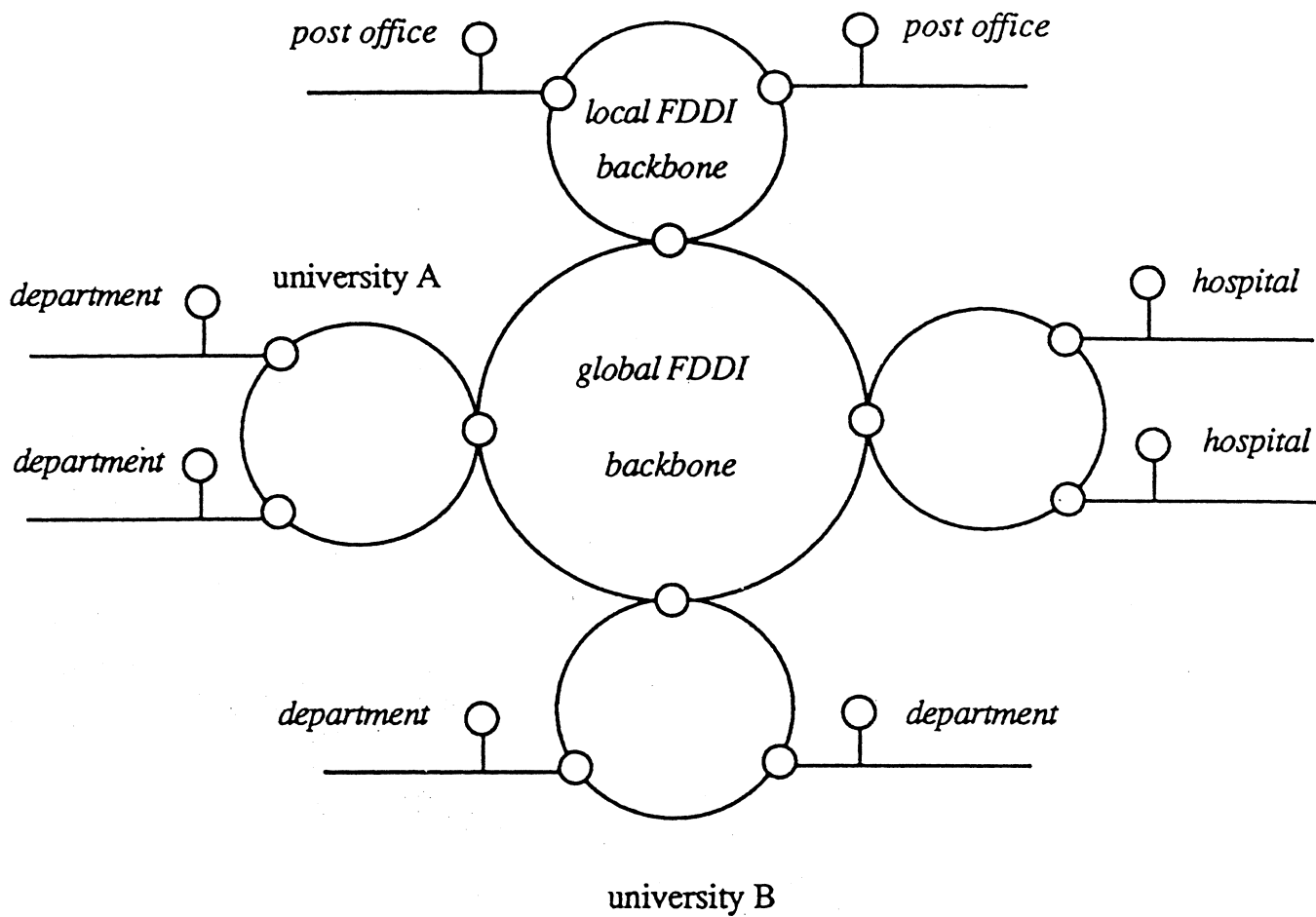


Fig. 9 A example of MAN that provides integrated services. This network model is for further study.

光纖與以太連結網路上整合語音 與數據通訊協定之設計

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摘 要

光纖分散式數據介面網路(Fiber Distributed Data Interface)是一個能使用於大區域(例如：一個校園、一個城市等等)當成數個區域網路(Local Area Networks)主幹(backbone)的高性能、高頻寬(bandwidth)的網路。在本篇論文裡我們設計了一個以光纖分散式數據介面網路為主幹經由橋接器(bridges)連接各似以太網路(Ethernet-linked LANs)的高性能語音－數據整合區域網路。每個似以太網路以虛擬線路(Virtual Circuits)傳送語音包(Voice Packets)以及使用一個修改過的碰撞偵測載波感知多重門徑通訊協定(Carrier Sense Multiple Access with Collision Detection Protocol)來傳送數據包(Data Packets)。

在第一章中，我們介紹一些在區域網路上整合語音數據的參考論文並且對本篇論文其他各章內容作簡短的描述。在第二章中，我們概略介紹光纖分散式數據介面網路的標準，其中包括實體媒介相關(Physical Medium Dependent)、實體(Physical)及媒介存取控制(Medium Access Control)等副層(sublayers)以及工作站管理標準(Station Management Standard)。在第三章中，我們說明我們的網路模型的环境(configuration)以及語音－數據整合通訊協定的原理。除了描述整個系統操作之外，我們也描述了一個在這整合網路上提供打／掛電話(call setup / release)、保留(holding)以及重接(reconnection)服務的通訊協定。在第四章中，我們分析求出這整合網路上電話通數的理論限制。理論分析顯示對於我們的網路模型來說，各個似以太網路上同時最多能有 281 通電話。此外，每個似以太網路有一個外線電話通數的限制。當我們設定外線電話通數的限制是 281 通時，在光纖主幹(FDDI backbone)上至多能安裝 10 台橋接器。如果把外線電話通數限制的數目從 281 通減少至 140 通，則光纖主幹上至多能安裝的橋接器數目從 10 台增加至 21 台。事實上，如果把這個外線電話通數限制的數目減少為 281 的 n 分之一我們能把光纖主幹上至多能安裝的橋接器數目增為大約 10 倍的 n 倍。再者，我們把整合網路上同時最多能有的電話通數以一個數學式子表達出來。在最後一章中我們提出一個能在大都會區域(Metropolitan Areas)提供整合服務的擴展網路模型，作為未來更進一步研究的對象。