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802 LAN Access Method for Wireless Physical Medium

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TITLE: **THE NECESSARY CONDITIONS FOR SUPPORT OF CONNECTION-TYPE AND CONNECTIONLESS SERVICES ON WIRELESS LAN**

SUMMARY AND CONCLUSIONS

From among the conclusions drawn and shown in the concluding section, some of the most important are as follows.

a) To provide competent connection-type and segmented packet type services, it is necessary at setup time to reserve future capacity in the medium for following segment transfers.

b) The scheduling algorithm for capacity allotment must consider unreserved capacity available, priority, type, bandwidth, length and other descriptors of the requested service before enabling a segmented transfer.

e) An intelligent scheduler must be positioned at the intersection of the combined access-point data flow and the data from the interconnected isochronous and packet networks to have immediate access to the information necessary for its function. As part of the infrastructure, *the vital scheduling function alone justifies and requires the presence of an intelligent hub.*

f) The need for reserved channel time or space precludes enabling stations to transmit except within constraints placed by a central control. Spontaneous, autonomous groups may not operate in this mode within the reliable coverage of the organized infrastructure.

j) The provision of these services requires anticipation of the necessary information and control functions in the design of the MAC and PHY layers.

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Table of Contents | Page

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OVERVIEW

Within the wireless LAN context, the ability to render a connection-type service on an asynchronous packet medium is dependent on reservation of future transmission space/time in the medium to assure timely delivery of each octet bundle. This space need not be at regular intervals but within a maximum acceptable value. These requirements are similar for any long or sustained communication which must be transmitted in bursts or segments.

There is no reservation algorithm unless the results of it can be used to control and schedule use of the transmission medium. The resulting requirement on the PHY and MAC is the ability to transfer:

1) the information on which access decisions are made, and
2) the enabling instructions for station originated transmissions.

The scheduling algorithm must achieve some performance parameters infrequently used in specifying packet LAN:

3) worst-case access delay. and
4) transfer delay which is the sum of access, quantizing and propagation delays.

Historically, space has been reserved in a regularly time slotted medium by a setup negotiation, the result of which is agreement between the ends that a particular time or frequency space is allotted for use until disconnect originates from one of the ends. This is a two step process with a secondary addressing scheme to define the allotted space. Further, the negotiation is conducted at a data speed which is a fraction of the medium transfer rate and may be less than that of one of the allotted slots (16 Kbps D-channel).

Later integrated voice-data systems have been defined by IEEE 802.6 for metropolitan areas and by IEEE 802.9 for a workstation interface. In both cases, an isochronous physical medium is multiplexed for independent and parallel use by isochronous and connectionless protocols. LAN packets are mapped into designated available slot space. When the partition between isochronous and packet services is adaptive, the difficulties of rearrangement (using slow signaling) of slot assignments as connections are setup and released became quite evident.

It has been recognized that a class of access methods can provide the connection-type service when a reservation function is implemented.[1] What is now considered is a time division medium in which there is no rigid timing structure implemented in the physical layer, but in which each user has consecutive access at the full medium transfer rate. This definition would include the bus type 802 LANs: 802.3 and 802.4. Instead, any necessary timing constraint is maintained in an intelligent hub and conveyed to stations by enabling messages. The general intent is to use a connectionless medium in which connection-type services are also properly supported. In this context. the following assertion is made:

> Delay objectives cannot be met unless the medium capacity used by any one connection is a small fraction of the total medium transfer rate, and the aggregate capacity used by all active connections is not the major use of the medium capacity.

It is not difficult to provide a system which operates when transmission capacity is lightly loaded. It is much more complex to provide a system plan that uses most of the transmission potential and which behaves in a specified and predictable way when the offered load exceeds the carryable load. It is also necessary to assure a minimum quality of service in a reference environment that is common if not universal.

SERVICES

This paper addresses the requirements for a high utilization and high service quality type of system.

IEEE P802.11 Project Authorization[2] states: "*Supported Service -- The Wireless MAC shall support both connectionless service as defined in the MAC Service definition at rates between 1 an 20 Mbit/sec as well as a service supporting packetized voice.*" It is possible to dispute whether packetized voice means all connection-type services, however the narrow interpretation precludes support of "Multi-media" services as described by T. Kwok.[3]

A major precedent for definition of an IS (integrated services) LAN is in the work of IEEE 802.9 draft Standard[4] version 20. Consideration of transfer delay is avoided in 802.9 because the 125 μsec frame length and octet(s) per frame transfer keeps it small.

In the IEEE 802.11 context, connection-type services shouid be characterized by bandwidth (average transfer rate) and by transfer delay. Connectionlessless services should be characterized by access delay and instantaneous transfer rate. Both services include peak aggregate capacity dimensions and some degree of time averaging of aggregate traffic.

Bandwidth and Quality for Connection-type Services

A minimal connection type service is BRI (Basic Rate Interface) consisting of 2 B and one D channels totaling 144 Kbps in up and down directions. It could be argued that the minimum is one B channel at 64 Kbps. While one B-channel could be userful in private systems, it is not readily used through the public switched network without the setup facilities provided by the D-channel at 16 Kbps.

The service needs for connection type services, including "multi-media," may be taken from T. Kwok's paper[1] on this subject. In Table II, he shows video bandwidths of N x 64 Kbps and 1.5 Mbps (PRI--Primary Rate Interface). A general solution would permit multiples of 64 Kbps up to 30 (to include international practices). A sufficient range of video services would require connection bandwidths at least up to 384 Kbps and probably higher.

If particular highly-used numbers for channel size were selected, the values would be: 64, 128, 192, 256, 384, 768 and 1,536 Kbps for North America.

The limiting parameters for quality of service could be tied to worst-case public network -- a very low quality. Limits on echo for international calls are not a fixed value. More transmission delay is allowable for local and intra-city connections than for greater distances.

Delay

Delay is limited by echo from reflective 2/4 wire junctions or acoustic coupling in 4-wire handsets. Tolerable added delay from a local distribution system is an argument on how to share about 30-40 milliseconds of tolerable delay with other parts of the public network. The important sources of delay are:

a) quantizing delay -- e.g.; 6 milliseconds are required to accumulate 48 octets of samples for a 64 Kbps channel. This example corresponds to B-ISDN (ATM) voice transmission.

b) coding delay -- codecs for speech compression generally require 10-60 milliseconds between input and ouput. Avoidance of this delay is inducement for not using highly compressed voice.

c) access delay -- the worst-case delay before transmission on the medium can start after the assembled octets are ready for transfer.

d) transfer delay -- the sum of the quantizing, coding (if used) and access delays and the added delay between the start of transfer at the originating end and the start of output of transferred octets at the receiving end. (On cross-continent and overseas connections, the propagation delay is significant.)

In general, the sum of these delays and those of the tandem telecom facilities must be less than 30-40 milliseconds to avoid unacceptable degradation on telephone connections. This leaves about 10 milliseconds for local distribution which is sometimes claimed by a digital PBX. Bellcore UPCS designs used 2 milliseconds as the allowable delay contribution from the radio link.

Circuit Interruption

Circuit interruptions of short duration, such as occur during hand-off in "cellular," are tolerable for plain voice. PBX vendors have difficulty accepting the circuit interruptions permitted in radio telephone. For compressed voice, the answer is less clear because of algorithm dependence.

For modems, a short circuit interruption is a lost transfer unless the modem is practicing a burst mode with ARQ (automatic repetition) protocol such as "Xmodem."

Error Detection and Failed Transfers

The 802 Standard "Functional Requirements" specify a minimum "Hamming distance" of 4 for error detection. This parameter defines the CRC necessary to provide infallible detection of up to three errors. It is this requirment that precludes use of HDLC bit stuffing to provide delimiters in an 802 MAC. A further requirement is an undetected bit-error-rate of 1 in 10^{-8}.

With burst mode transfer, error performance is better expressed as lost or failed segment transfers largely resulting from detection of errors in the received transfer. If there is an embedded FEC or channel coding, a successful transfer is possible even though bit errors are present.

The 802 Standard "Functional Requirements" for packet loss are reproduced in the 802.11 PAR: *"5.6.1 The MAC Service Data Unit (MSDU) loss rate shall be less than 4*10E-5 for an MSDU length of 512 octets." ..."99.9% of the time on a daily basis..."* If this requirement is met, the performance would be fully adequate or better than expected for connection-type services. However, the time required for recovery from lost packets that is allowable in 802 might not be acceptable for connection-type services where a delayed segment is a lost segment.

Required Medium Transfer Rate

The delays required by telecom are fixed, but the transfer rate is a designer's choice. It would be useful to relate these two parameters. This relationship is complicated by dependence on details of the access method, particularly those functions involved with multi-site frequency reuse and retransmission mechanisms.

Two of many access methods will be considered. One is the access method of the Author for single channel systems, and the other is that of IBM using regular frames with adaptively located partitions between random access, uplink and down link time slots. In general, the two methods seek similar goals, but have material difference in methods.

Single Channel Asynchronous Sequential Access

Using a previously described access method[5] for a single channel at a 16 Mbps transfer rate, the following is assumed:

Reuse factor:	4
One site frame length:	600 octets or 300 µsec
Four site frame length:	1200 µsec

In this case failure to access and transfer in one frame will incur a delay of 1200 µsec before another opportunity occurs. This delay would frequently apply to requested retransmissions.

If the transfer rate were halved, there is a choice of halving the frame length in octets lowering efficiency, or doubling the frame length in µsec increasing access delay. Since overhead message and segments up to 300 octets will be fitted into the 600 octet frame, the incentives for limiting maximum segment length are apparent. The smaller the pieces, the larger the utilization of a fixed amount of space. It is desirable but nor required that all transfers be completed within the same site frame.

Other improved algorithms dynamically dividing the use of the multi-site frame period between access-points within a group are also known. In these, the frame end boundary is elastic and may be extended if there is not limiting load for a following site.

It is concluded that at 16 Mbps there is margin, but not a lot.

Frequency division channelization into four 4 Mbps channels could be used. The frame boundary would not physically exist but would exist by definition. *Then there would be no way to shift capacity between sites.*

If a reuse number of 9 were required, the same choice would have to be made after accepting the resulting capacity reduction.

Regular Frame Adaptively Partitioned

First proposed, and occasionally updated, by K. S. Natarajan[6] is plan using a regular periodic frame adaptively partitioned between uplink, downlink and random access services. This approach was further analyzed by R. O. LaMaire[7] who gave a fair dimensional description of the plan.

As analyzed and proportioned, the plan applies only to LAN. "Mean" rather than "worst case" delay without separating contention mode delay from the request function.

This plan does recognize the importance of segmentation, of tradeoff between parameters optimized for random-length transfers and for maximum channel efficiency. It does not recognize, as given, the difference between allotting space in a frame and reserving future capacity without associating the reservation with a particular instant of time.

As presented, the plan is based on a frequency hopping PHY with a maximum likely transfer rate of 1 Mbps within present 802.11 constraints. This is adverse compared with the 2 Mbps assumptions made in the analysis.

A further assumption is that the channel is clear and unimpaired by other stations. With many FH stations operating independently in the same frequence space, there will be a loss mostly impacting worst case access or transfer delay.

Nonetheless, this is one of the better prepared proposals base on time slots submitted to 802.11. Therefore a more detailed description and commentary will be provided in a separate contribution.

Interworking with 802 LAN Services

The IEEE 802 LAN Standards Committee is by definition confined to layers 1 and 2 and excluded from network and transport functions at the next two higher layers. This exclusion stresses the ability to interconnect two different types of networks, but allows bridging between separate but like-type networks. It is necessary to deal with this matter when there are multiple access points each of which may be considered as an externally addressable network or in the alternative a number of them as a group is an externally addressable entity. For the latter case, a common function to route externally originated communication to the approriate access-point is essential. Requiring external entities to track the minor movements of a wireless station within a building is a possible higher layer function, however invoking it would distress the system with largely meaningless traffic.

With a constant and given BER (bit-error-rate), the IEEE 802 lost MSDU rate will decrease to better than 3.2 x 10^{-6} failed transfer rate for average packet lengths of 64 octets. To achieve this rate, repetition and FEC are permissible as long as the result is timely achieved. The less channel time that is invested in any one transfer, the less time is required to recover. The shorter the packet, less frequent will packets be received for which a retransmission is required.

A number of important parameters are unspecified by IEEE 802, but are important to the usefulness of the system. IEEE 802 requires a minimum medium transfer rate of 1 MBps, but is silent on the net useful transfer rate. Speed and utilization are competitive factors between suppliers provided that interworking in a multiple vendor market is maintained.

It is true that completion of assembly of a packet to be transferred occurs at unspecified and possibly random time intervals. *It is not true that asynchronous access to the physical medium is needed for this traffic.* However, there is a necessary limit on access and transfer delay. One of the ways in which this may be achieved includes, but is not limited to, the use of an isochronous frame (see IEEE 802.6 and 802.9).

> Assertion: Marketability of any new LAN (wireless included) requires at least a 10 Mbps transfer rate and a *worst case* access delay under 5 milliseconds associated with error probabilities little worse than those now available on wire.

The radio medium requires transfers in short bursts to avoid excessive investment in a transfer in which the probability of error increases with length. A more subtle reason for keeping the burst short is that the worst case access delay is proportional to the average channel occupancy time of these bursts.

The approximate range of usable burst lengths is 32-256 octets. Since LAN packet sizes are commonly limited to 500-1500 octets as determined in higher layer functions, it is inevitable and necessary that there be further segmentation for transmission of long packets on the wireless medium. It is possible that segmentation will be done at higher layers at some future date.

Integrated Services

For similar reasons, both isochronous and packet services must be converted to short bursts for transmission with guaranteed access to the transmission medium for continuing segments of any connection or segmented packet accepted for carriage. A near deterministic system is required.

In this context, there is a further justification for a high medium transfer rate.

Classification of Services by Delay Category

Accuracy, delay and time efficiency of the use of medium are tradeoffs that cannot all be maximized simultaneously. There is a need to classify this tradeoff as a specification parameter in a service request. The necessary delay parameters are mostly absolute, however the attainable values depend upon a high medium transfer rate and the access method.

Example combinations are shown in Table I.

TABLE I -- CLASSIFICATION OF SERVICES BY PRIORITY AND DELAY REQUIREMENT

SERVICE DELAY CATEGORY	CONN/ PACKET	APPLICATION	MAX DELAY MILLISEC	REPEAT ENABLE	PAYLOAD SIZE octets
LOW DELAY	Connection Priority 1	International and priority telecom 64 or 128 Kbps	6 total 4 quantizing 2 access	no	32 or 64
LOW DELAY	Packet Priority 2	Short interactive and overhead messages	2 access	yes	variable to 64 max
NORMAL	Connection Priority 3	General voice, data or video using 1 to 6 B-channels	9 total 6 quantizing 3 access	no	48, 96, 144, 196, 240, 288
DELAY TOLERANT	Connection Priority 4	Local and/or compressed voice and intercom	19 total 16 quantiz'g 3 access	yes	128 or 256
NORMAL	Packet Priority 5	Local LAN packet segmented when LEN over 256 octets	3 access	yes	variable up to 256 maximum
DELAY TOLERANT	Packet Priority 6	File transfer, longer than 256 octets, segmented only	16 access	yes	256 maximum

The priority marking shown is indicative of the precedence of messages concurrently processed. Six possible levels are shown with separate sets for connections and packets and with each set having a normal, low and tolerant delay requirement. Priority does not affect whether a service is continued but only the sequence in which the service is provided to various users. Truncation, if it occurs in a scheduling mechanism, can only affect priority 4 transfers by delay until a later frame.

The example values shown in Table I above assume a particular message-based medium access method and a minimum medium transfer rate of 16 Mbps.

The above Table is a basis for asserting that a 4-bit field is necessary to define the type of service required in a REQUEST message.

FUNCTIONS SUPPORTING FUTURE CAPACITY RESERVATION

The essential questions that must be answered by an access scheduling function within a central function are:

a) Is the capacity available to grant the requested service now and in future frames?

b) If capacity is not now available, is there prospect of availability for which queing of the request for a future grant is appropriate?

c) If the current request is high priority, and if capacity is not immediately available; is there any low priority traffic that may be delayed to make capacity immediately available?

To answer these questions, the central control algorithm needs a set of input facts and the means to convey the results of the answers to the affected stations.

Access Method Assumptions

The intent is to define necessary functions rather than an access method, however these functions are restrictive on possible access protocols. It is assumed that all functions are accomplished by exchange of messages the content of which includes the necessary information for the operation of the system. The message types assumed are the relevant subset of a larger plan, and are named: Invitation, Request, Grant, Transfer and Acknowledge. Contention after Invitation from multiple Requests is assumed possible. No contention occurs on other message types.

The content of the messages used must include the necessary information for future space reservation to be implemented. More specifically, service requests must contain a complete description of the requested service including type, length, bandwidth, priority and service type as well as the necessary addressing.

For the *station-originate (SO)* case, transfers occur only after Grant. Continuing segments after setup, may be transferred either at a specified time or after an automatically generated Grant.

For the *station-terminate (ST)* case, the Access-point may send a complete packet or a segment at an internally selected time. The station must be always ready except when a mute condition is allowed by sleep mode algorithms.

Frame Priority Description

The word "frame" is used more generally as the interval in which a set of communication possibilities occurs which are repeated in subsequent frames. As now used, frame does not imply a constant length, identical functions, or that all possibilities are exercised. Different frames for different purposes may have varied priority and processing.

As an example, the ordering of service opportunities by priority might be used in the order shown in Table I.

At each priority level, ST transfers take priority over SO. Requests for new service are solicited but not necessarily granted at priority 2 (overhead messages). Other overhead messages may be transferred at priority 5. The bulk of the traffic is expected to transfer at the normal levels which are used as the defaults.

Processing Algorithms

1) neither packet or connection-type services are allowed to use more than a specified fraction of available capacity set as a user configurable parameter (70% each default).
2) if available capacity is fully committed, no further SO requests will be granted, and outside ST requests of priority 3 or higher may delay lower priority transfers in progress to obtain space.
3) no one station may use more than a specified fraction of the capacity of one access-point within limits on time or transfer rate except when channel time is available after reserves for other users. This limit is a user configurable parameter (50% default).
4) short packets are higher priority than longer packets.

Future Load Prediction

The ability to handle connections and segmented packets depends upon predictability of future load. System management to avoid loss of capacity when offered more load than can be carried also depends on predictability.. It is inherent that a compilation of connection and segmented packet traffic accepted and not yet terminated is the basic predictor valid at least for the current and next frames.

Packet traffic can be handled as fast as it can be accessed to the medium. The LENgth field is indispensable for prediction of when a particular packet transfer will be concluded. If the LEN field is not available from a particular interface, then the whole packet will have to be buffered until the end delimiter is received before it can be processed over the wireless medium. The packet reservation is dimensioned in octets to be transferred at the earliest opportunity.

Connection traffic is handled on a bandwidth basis where the required bandwidth is specified at setup. The processing must assume that this bandwidth will be used indefinitely into the future at a steady rate. It is always possible that the current burst will contain a disconnect indication making its occupied space available for allottment in the next frame.

To the traffic in progress may be added that which is in queue. Newly available space can be used immediately. The queue can be managed with service in-order-of-arrival for requests of like priority. For the queue to be maintained at high levels of offered traffic, it is necessary for the service request function to have a higher priority than traffic transfer.

Overload Processing

For SO traffic, it is possible for the system to reject or delay both types of traffic when offered in excess of capacity. With extreme overload, it is possible to temporarily suspend invitation messages.

For ST connections, it may be possible to return the equivalent of "trunks busy" when the request is unservable for lack of capacity. This implies that D channel signaling is terminated within the hub where the medium access is scheduled.

For ST packet traffic originating outside of the system, there may not be a way to delay or refuse traffic for valid addresses. There is no limit to how much traffic may be offered to a particular address or set of addresses. This makes it difficult and inaccurate to associate a lost packet/cell probability with a buffer size. It is known that ATM switches dealing with this same problem extimate 100-300 buffers of 48 octets for near zero loss by public switching standards.

ARCHITECTURE AND TOPOLOGY

The future transmission space reservation and scheduling algorithm must have a place of residence. That place is defined by commonality to a group of user stations for one or serveral access-points. It must reside at a common point to which all station and external requests for access are addressed. This is the only place where all of the necessary input information is available without retransmission.

It follows that the service cannot be provided with wholly distributed logic. There is a possibility that this infrastructure function could be called a "Scheduling Server" to suggest that non-connection type services could be provided without supplying this function and while using a medium access method which was wholly distributed apart from connection-type service support. While probably possible, the desirability and cost avoided are arguable. The following is asserted:

> In a system where autonomous direct peer-to-peer transfer is supported, it is probably impossible to provide a valid connection-type service without infra-stucture including an intelligent central controller to manage reservation and scheduling of future capacity.

There are many other functions required that also need to be centrally located. Particularly important, are those functions that deal with overlapping coverage from a number of access-points and which convert that redundancy to a transmission advantage. In addition, the facilities that follow variable location of stations also needs the central function.

Without a connection-type service requirement, a distributed logic access method still could not be implemented. It is also ineffective for high capacity systems. What may be overlooked is that the inevitable segmented packet needs a higher degree of assurance of transmission of the following segments than does a packet transfer not yet started.

These requirements favor a hub in a wiring closet which is the intersection of links to many access-points, bridges to other 802 LANs and to a higher level interconnection to the public network or to a PBX for connection type services. This hub could and should complete connections autonomously when all parties are local.

SUMMARY AND CONCLUSIONS

a) To provide competent connection-type and segmented packet type services, it is necessary at setup time to reserve future capacity in the medium for each transfer.

b) The scheduling algorithm for capacity allotment must consider unreserved capacity available, priority, type, bandwidth, length and other descriptors of the requested service before enabling a segmented transfer.

c) The estimate of capacity available requires a summing of capacity claimed by transfers already setup which also using the service descriptor information in the initial Request messages.

d) To maximize channel utilization, requests for service must be classified and queued awaiting availability and served in order-of-arrival for each priority level. This function requires separation and priority of request messages relative to information transfers.

e) The intelligent scheduler must reside at the intersection of the combined access-point data flow and the data from the interconnected isochronous and packet networks to have immediate access to the information necessary for its function. As part of the infrastructure, *the vital scheduling function alone justifies and requires the presence of an intelligent hub.*

f) The availability of a particular port on an intelligent hub may be contingent on the existing or pending use of other ports, when the ports are used for unconfined wireless transmission mediums.

g) The need for reserved channel time or space precludes enabling stations to transmit except within constraints placed by a central control. Spontaneous, autonomous groups may not operate in this mode within the coverage of the organized infrastructure.

h) The medium access protocol must be capable of conveying from the central controller to the station the time and type of transmission permitted from the station, and stations must not be allowed to or be capable of transmitting outside of this protocol.

i) To attain a two millisecond worst-case access delay (or any other defined value) for existing telecom services, there is a minimum medium transfer rate that must be high enough to enable many average length bursts to be transferred in this interval.

j) The provision of these services requires anticipation of the necessary information and control functions in the design of the MAC and PHY layers.

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