PERFORMANCE OF LAMS-DLC: A PROTOCOL FOR
LOW ALTITUDE SATELLITE NETWORKS

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Christopher Ward
Department of Computer Science and Engineering
Auburn University, AL 36849-5347

Cheong H. Choi
Samsung Inc.
Kwachon, South Korea

Thomas F. Hain
School of Computer and Information Sciences
University of South Alabama
Mobile, AL 36688
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CHRISTOPHER WARD
Department of Computer Science and Engineering, Auburn University, AL 36849, U.S.A.

CHEONG H. CHOI
Samsung Inc., Kwachon, South Korea

THOMAS F. HAIN
School of Computer and Information Sciences, University of South Alabama, Mobile, AL 36688, U.S.A.

SUMMARY

LAMS-DLC, a data link control protocol for low altitude satellite networks, was developed to overcome throughput limitations inherent in event-based positive acknowledgment (POS-ACK) automatic repeat request (ARQ) class protocols by relaxing certain reliability constraints, in particular, the in-sequence delivery constraint. This relaxation results in a new class of link layer service, that of reliable datagram, and permits a fresh approach to link layer protocol design. In this paper we discuss the motivation for such a protocol, describe one such protocol, LAMS-DLC, and compare throughput efficiency with results obtained from a simulation model. Results suggest that LAMS-DLC provides near optimal throughput efficiencies in the target environment, while using significantly less buffer space than is required for conventional POS-ACK protocols.

KEY WORDS LAMS-DLC Link layer protocol NAK-based protocol Satellite networks

1 INTRODUCTION

The term satellite network usually implies the following properties: one or more satellites in geosynchronous orbits; satellites used as communications relays; point-to-point uplinks with common channels shared by several earth stations using FDMA or TDMA; broadcast downlinks; and long propagation delays (in the order of 250 ms). The properties of these networks differ significantly from a new class of satellite network which is currently gaining attention, that of the
low altitude multiple satellite (LAMS) network (consisting of low earth orbit (LEO) satellites). These networks exhibit very distinct characteristics from conventional satellite networks and invite a fresh look at the link layer protocol assumptions and requirements.

LAMS networks are described in detail elsewhere [8, 11, 12], but they typically have the following features: multiple satellites in predetermined LEO paths functioning as store-and-forward packet switched nodes; point-to-point laser communication with very high bandwidth (100 Mbps to 1 Gbps); long distance communication links (1,000 - 10,000 Km with relatively high bit error rates (BER) $10^{-5}$ to $10^{-7}$); a limited number of communication links per satellite due to size, weight, and power (SWAP) constraints; and high satellite mobility.

There have been many data link control protocols (DLCP) suggested in the literature to accommodate communication between LEO satellites using variations on the HDLC automatic repeat request (ARQ) protocol [1, 3, 4] and hybrid FEC/ARQ protocols [6, 5, 10, 2]. However, the present long propagation delays and high error rates dictate large buffer requirements in order for them to achieve high throughput efficiency. The reasons for this are clear if data are to be delivered in sequence and without error; an ARQ protocol will be required to buffer a sequence of frames at either the sender (in the case of go-back-N (GBN) and its variants) or the receiver (in the case of selective repeat (SR) and its variants) until the correctly sequenced packet is received. While large buffer requirements are acceptable for terrestrial links they are clearly undesirable in LAMS links due to SWAP limitations. The design of a robust protocol with reasonable throughput efficiency and modest buffer requirements was therefore our goal.

In Section 2 we examine in more detail some of the underlying assumptions related to LAMS networks. In particular, we note that LAMS links are likely to be of short duration and it is therefore important that the channel bandwidth be used as efficiently as possible. We also explore the relationship between the fundamental reliability requirements (FIFO packet delivery, without loss or duplication) and their impact on protocol performance. We note that the FIFO
delivery requirement restricts packet flow, and causes either the maximum sender queueing time (MSQT) or the maximum receiver holding time (MRHT) to be unbounded. We take a fresh approach to this topic by relaxing the FIFO delivery requirement. This introduces a new class of link layer protocol suited to the LAMS environment which we call reliable-datagram which has characteristics of both the datagram service and the virtual circuit service. The relaxation also permits packet numbering to be limited with respect to both MSQT and MRHT and allows the notion of continuous packet flow (or transparent buffering) which is impossible otherwise. In Section 3 we discuss how context evolution (the difference between the sender's and receiver's state information) is influenced by relaxing the FIFO requirement and how this in turn affects MSQT and MRHT. We establish that, provided a number of conditions are satisfied, a negative acknowledgment (NAK) timer-based check point positive acknowledgment ARQ protocol, is sufficient to provide the reliable datagram service without the requirement for event-based positive acknowledgment (POS-ACK) ARQ. In Section 4 we describe the protocol-LAMS-DLC—which is uses these concepts to provide a link-layer reliable-datagram service for satellite point-to-point links. The frame format for the protocol is described, along with an informal description of protocol operation. Section 5 provides an analytical model for the expected throughput efficiency and buffer requirements for LAMS-DLC as a function of BER for a representative set of parameters. Some measure of protocol correctness is established in Section 6 by running a variety of test cases on a LAMS-DLC simulation. Results from this simulation model using the same parameter settings as in Section 5 verify the theoretical analysis.

2 MOTIVATION FOR LAMS-DLC

2.1 Characteristics of LAMS networks

Satellite mobility has a profound impact upon the way in which a link is viewed. Links are available for only a relatively short period of time due to occultation by the earth. Targeting a
laser cross-link takes a relatively long time period due to increased laser synchronization as a result of SWAP limitations [7] and precise targeting requirements [8]. Thus active link time is short and is considered a highly valuable resource.

The high bandwidths and long propagation delays in LAMS networks also influences protocol design. If high link traffic (meaning data transmission is continuous) is assumed, the combination of high bandwidths and long propagation delays tends to significantly increase the number of in-transit packets. In high traffic, the maximum number of in-transit packets are \( \frac{t_R}{2p} \), where \( d, t_R, p \) are the data rate, the round-trip time, and the mean packet size respectively. If the GBN error recovery scheme is used and a single bit is corrupted the next \( \frac{t_R d}{p} \) packets following the corrupted one must be retransmitted (i.e., must be stored in the sender buffer). In the case of the SR scheme, the same situation implies that \( \frac{t_R d}{p} \) packets must be stored in the receiver buffer for sequencing. In LAMS networks, the combined value \( t_R d \) will be significant. For example, if \( d \) is 100 Mbps and \( t_R \) is 100 ms, the value \( t_R d \) is \( 10^7 \) bits, that is, 1.25 Mbytes. This means that in the case of SR, at least 1.25 Mbytes of buffer space is necessary for high traffic, and in the case of GBN, a bit error makes 1.25 Mbytes of transmission useless. Higher data rates magnify this problem.

The combination of long burst errors and high mobility in LAMS networks leads to another problem when using conventional DLCPs. The long burst errors (3 - 6 ms) may simulate an unexpected link failure and impede reliable communication. Recovery from such a simulated link failure is accomplished using a time-out mechanism, so that if either protocol agent receives no response for a predetermined period of time, the link is regarded as having failed. However, such a mechanism works poorly with LAMS networks due to the changing link distance. Usually the time-out value is \( t_R + 2\sigma_R \), where \( \sigma_R \) is the standard deviation of \( t_R \). In LAMS networks, \( t_R \) and \( \sigma_R \) will either vary over time (reflecting changing link distances) or the time-out must be set large enough to cover the maximum round-trip time. Thus, protocols that make use
of a time-out mechanism and resolving period will either experience significant performance
deterioration and resource under-utilization, or must provide a dynamic mechanism to select a
time-out value appropriate to the link length.

The problems of large buffer requirements and/or large wasted transmissions are compounded
in a high BER environment. The packet retransmission probability in the SR scheme is \(1 - P_f P_c\),
where \(P_f\) is the probability that an information packet arrives without error and \(P_c\) is the
probability that the corresponding ACK also arrives without error. If piggybacking is used
to reduce redundancy caused by positive acknowledgment, \(P_c\) is equal to \(P_f\), and the packet
retransmission probability is \(1 - P_f^2\). Suppose that the packet length, \(n\) is 1,000 bytes, the ACK
length, \(m\) is 100 bytes, and the BER, \(P_b\) is \(10^{-5}\). Then \(P_f\) is \((1 - P_b)^n \approx 1 - n P_b = 0.92\), and
\(P_c\) is \((1 - P_b)^n \approx 1 - m P_b = 0.992\). In the piggybacking SR scheme, \(P_f^2\) is 0.846 showing
about 15% waste of link utilization, whereas in a non-piggybacking SR scheme, \(P_f P_c\) is 0.913
showing about 9% waste. As can be seen, high BERs make the conventional error recovery
schemes inherently inefficient.

The high BER resulting from combinations of long burst errors and random errors also has
the effect of increasing packet queueing delays and, therefore, sender and/or receiver buffer
requirements. In event-based POS-ACK protocols senders are required to keep copies of
transmitted packets until they are successfully resolved (acknowledged). This resolving period
will be typically one round-trip time; however, if an ACK is lost or corrupted, the corresponding
packet must remain buffered for another round-trip time once the time-out timer has expired, etc.

2.2 Relaxation of Reliability Constraints

Reliability at the data-link and network levels in conventional networks is composed of the
three requirements of (1) no packet loss, (2) no duplicates, and (3) first-in-first-out (FIFO)
delivery. This strict reliability derives from the notion of a virtual circuit at the network
layer, and has as its underlying goal the reduction of the burden of reliability assurance at
the transport layer. Strict reliability constraints do not impose an undue performance reduction
on conventional networks, where propagation delays are relatively small and links are fixed.
However, in LAMS networks it is observed that strict reliability is one of the primary barriers
to performance improvement, by imposing an unnecessary overhead on queuing delays. In the
LAMS-DLC design we consider relaxed reliability, in which we reduce the set to only the first
two requirements in strict reliability. By removing the FIFO delivery requirement at the DLC
level, we shall show overall performance gains. The associated network service will be called
reliable datagram.

Consider the relations between constituents of the strict reliability constraints. Let \( I \), \( D \), and
\( L \) be FIFO delivery, no duplicate, and no loss respectively, with violations of these constraints
being denoted \( \bar{I} \), \( \bar{D} \), and \( \bar{L} \). It can be seen that \( I \Rightarrow D \) and \( I \Rightarrow L \), but the converse relations
are not satisfied. Similarly \( \bar{D} \Rightarrow \bar{I} \) and \( \bar{L} \Rightarrow \bar{I} \), but again, not conversely. This relation shows
that the effect of \( \bar{I} \) is minimal; that is, a relaxation of \( \bar{I} \) gives rise to flexibility, yet has no impact on \( D \) and \( L \).

If \( I \) is relaxed, the receiver no longer has to hold correctly received out-of-sequence packets,
and may forward them immediately, thereby eliminating corresponding packet queueing delays.
Also, with \( \bar{I} \) it is unnecessary for a packet be identified uniquely at the DLCP level when initially
transmitted, or retransmitted, because the receiver does not need to check packet order. This
relaxation provides the basis for a transparent numbering size and is also the basis to bound
packet acknowledgment numbers.

Notice also that in the strict reliability constraints, \( I \Rightarrow D \) and \( I \Rightarrow L \). However, at the
transport layer \( L_{DLCP} \not\Rightarrow L_{TL} \) because of end-to-end time-out mechanisms, and \( I_{DLCP} \not\Rightarrow L_{TL} \)
because of unbounded packet delays. Therefore, packet sequencing at the transport layer is
required regardless of the DLCP.
3 LAMS-DLC DESIGN

In terms of protocol synchronization, an active link period in a LAMS network is composed of three distinct phases: link (re)initialization (connection establishment), context evolution (data transfer), and link closure (connection closure and physical link closure). Protocol synchronization of these three phases is managed by two cooperating entities which exchange state information and reach an agreement on the state information for the association.

While the term protocol synchronization could refer to the management of state information during all these phases, we shall use it only in reference to the context evolution phase. For correct context evolution, both ends of the link must maintain consistent state (context) information, e.g., association, message numbers, credits, delays, erroneous packet numbers. Note that, since contexts evolve asynchronously (due to transmission delays over the communication links), context inconsistency must be allowed for the sake of efficiency. If an active end (sender) has no evolution, this inconsistency will eventually disappear provided that the passive end (receiver) keeps responding and not all responses are lost. We define the time period (measured in seconds) required to eliminate the inconsistency the inconsistency gap \( I_g \). Another measure of inconsistency, which we call the inconsistency degree \( I_d \), is defined as the difference between contexts at both ends expressed as the number of unresolved packets. Figure 1 shows how \( I_d \) might change with time.

POS-ACK is used by all conventional ARQ-based DLCPs for notifying the sender of the status of received packets. The FIFO packet delivery requirement demands that each packet have a unique sequence number until the packet is acknowledged. Therefore, in the case of continuous transmission in a noisy medium, \( I_g \) of POS-ACK-based ARQ schemes is necessarily unbounded. To see this, notice that the sender can do nothing about a packet if the corresponding POS-ACK is not received, resulting in an ever-increasing \( I_g \). This is not to suggest however,
that \( I_d \) is also unbounded, in the case of the sliding window POS-ACK protocols, every window demands a synchronization phase (or resolving phase) in an effort to ensure that both contexts are simultaneously in a well-defined state at some points in time. Thus, \( I_d \) is limited by the window size.

Limiting \( I_g \) requires placing bounds on two time factors: the sender queuing time (SQT) and the receiver holding time (RHT). The SQT is the time for which the sender must keep unresolved packets in its buffer. The RHT is the time required to allow packets lost by corruption or buffer overflow to be resolved, during which the receiver must hold good packets in its buffer before they can be delivered in FIFO order. If these times could be bounded to maximum SQT (MSQT) and maximum RHT (MRHT) respectively, then \( I_g \) will also be bounded, with \( I_g = \max(\text{MSQT}, \text{MRHT}) \). Provided the sender sends no new packets during \( I_g \), no unresolved packets will remain in the sender or receiver buffers. Furthermore, the bound on \( I_g \) implies a bound on the queuing delay, with sequence numbering and buffer requirements reaching a steady state (which we call transparent numbering and transparent buffering respectively) determined by the BER.

For example, suppose a packet with sequence number \( n \) has yet to be acknowledged and is therefore kept in the sender buffer. This packet contributes to \( I_d \) by one. If \( I_g \) is bounded, \( I_d \) should be zero after a delay of \( I_g \) provided the active end refrains from transmitting new packets. The packet \( n \) could therefore be safely discarded from the sender after \( I_g \) seconds without protocol violation. A similar case may be made for the receiver.

3.2 Mechanism for providing bounds MSQT, MRHT in LAMS-DLC

In this section, we discuss how to impose the time constraints, MSQT and MRHT in LAMS-DLC. Clearly the SQT of an individual packet cannot be bounded, since the possibility of repeated corruption of a packet always exists, and therefore our considerations on protocol synchronization
are concerned with limiting $I_d$ and yet still provide the possibility of continuous transmission. Limiting SQT with respect to $I_d$, can be achieved by meeting the following conditions:

**Condition 1:** A packet should be removed from the sender buffer, or retransmitted, a within finite time of initial transmission.

**Condition 2:** Each packet transmission, including retransmissions, should be identified differently.

**Condition 3:** A link failure should be detected within a finite time.

The first condition is the definition of the limited SQT in terms of $I_d$. If a packet can be removed within an MSQT, the corresponding sequence number can be reused. Since the FIFO delivery requirement is not required in LAMS-DLC, a *new sequence number* may be assigned to the retransmitted packet, and $I_d$ is reduced by 1. Erroneous packets can be discarded immediately and this implies that MRHT is zero, i.e. bounded. In order to prevent the numbering scheme from blocking continuous transmission, the numbering range must be greater than the maximum number of packets to be transmitted during MSQT. The second condition, associated with transparent numbering, is a basic requirement for limiting SQT. If a packet is identified with a particular sequence number, $I_g$ cannot be bounded even though SQT may be limited since loss of this packet will prevent others from being accepted. The third condition is concerned with an exceptional case which prevents limiting SQT. Recall that in LAMS networks, many simulated link failures are likely to occur because of long burst errors. The design must therefore take into account these failures since they will cause SQT to be increased indefinitely. If real link failures occur, $I_g$ cannot be bounded and LAMS-DLC fails (as would any POS-ACK protocol). There is no way to avoid link layer protocol errors due to real link failures.

To satisfy the conditions for limiting SQT in terms of $I_d$, LAMS-DLC is based on a strategy involving NAKs and periodic positive acknowledgment with cumulative selective repeat, called
check-pointing acknowledgment.

The basic idea of event-based POS-ACK protocols is that the sender will remove a packet from the send buffer, and change its associated state information, only at the corresponding arrival event. Conversely, a pure NAK-based protocol relies on the notion that "no news is good news"; that is, the receiver sends a NAK only in the case of an erroneous packet. However, pure NAK-based protocols have the potential of packet loss. In LAMS-DLC this problem is solved by acknowledging with cumulative selection, in which the receiver sends periodic responses called CP commands, as illustrated in Figure 3. In the case of an erroneous packet, this information is repeatedly reported to the sender by NAKs—as many as specified by a cumulation depth (C_{depth}). The probability of a packet being lost this way is extremely low since all these C_{depth} NAKs would have to be lost which triggers link failure recovery.

Thus, LAMS-DLC ensures that a packet is acknowledged (either implicitly or explicitly) within a time period bounded by the sum of d_{normal} and the cumulation time C_{depth}W_{cp}, which satisfies condition 1. One potential problem would be link failures simulated by burst errors. This is solved by ensuring that the cumulation time is longer than the burst error duration. If the sender fails to receive any response within a cumulation time, it suspects the link as having failed. This notion guarantees that the LAMS-DLC can detect link failures within the cumulation time and also allows it to satisfy condition 3. If the sender stops sending packets immediately after a link failure detection, \( I_g \) is still bounded since both ends do not evolve contexts. In LAMS-DLC, the \( I_d \) does not exceed the maximum number of packets to be sent during the cumulation time. In the example in Figure 4, the maximum \( I_d \) is at most 9.

4 PROTOCOL DESCRIPTION

This section discusses LAMS-DLC from the perspective of frame format and error control; a flow control mechanism is not provided in this paper. The error control mechanism will be
divided into two distinct recovery modes: *check point recovery* and *enforced recovery*.

### 4.1 Frames Format

In LAMS-DLC two types of frame format are defined: information frame (I-frame) and control frame (C-frame) as illustrated in Figure 5. The first field (1 bit) of both frames indicates the type of frame. LAMS-DLC does not piggyback control fields with data packets (even though this appears wasteful) since this increases the probability that the control information is lost. An I-frame is composed of 4 fields: (1) frame type field, used to indicate the frame type; (2) sequence number field (SN field), to contain its identifier; (3) information field, to contain the actual data; and (4) FCS field for frame error detection. In C-frames, there are two frame formats: the command format and the NAK format. Their common fields are: (1) frame type field; (2) command type field, used to indicate command types; (3) enforced bit field, for the enforced recovery of LAMS-DLC; and (4) SN field, to contain the sequence number of the most recent I-frame, allowing for fast resolution of the sender buffer. There are three command types which use the command format: (1) check point command (CP command); (2) resolving command (RES command); and (3) request-negative acknowledgment command (REQ-NAK command). The NAK formats are used for acknowledging erroneous I-frames by the receiver. The NAK format results from attaching the error list field. However, REQ-NAK does not have the error list field since it is issued by a sender only to request information. Only two command types, CP command and RES command, have associated NAK forms.

### 4.2 Error Recovery

This section is concerned with the details of how the error recovery mechanism of LAMS-DLC operates. This mechanism is divided into two recovery modes: check point recovery mode for erroneous I-frames and enforced recovery mode for simulated link failures.
In check point recovery the receiver periodically sends CP commands (CP-NAKs) at every check point interval denoted by $I_{cp}$. In the absence of errors, the CP command functions as an implicit POS-ACK. For example, CP(0,(none)) in Figure 6, acts as an implicit POS-ACK, while CP(2,(1)) indicates a NAK for the I-frame numbered by SN=1 and implicit POS-ACKs for I-frames with SN=0, 2. As stated before, since each CP-NAK contains information on the erroneous I-frames that previously occurred during a cumulation time period, the sender will be notified of an erroneous I-frame as many as $C_{depth}$ times. However, the sender responds with retransmission of the I-frame only once after the first recognized notification. For example, the I-frame numbered as SN=6 is the retransmission of the erroneous I-frame with original SN=1, since LAMS-DLC allocates new SNs to retransmissions. From that time, there is no I-frame with SN=1 in the sender buffer. Thus the sender ignores subsequent CP-NAK retransmission requests regarding the I-frame with SN=1.

We now consider enforced recovery. If the sender suspects a link failure because there is no response during a cumulation time denoted by $W_{cp}$, the sender sends REQ-NAKs to the receiver, provided the sender anticipates that the expected response will arrive within the remaining active link time (recoverable link failure). See Figure 7. The sender then starts the failure timer, stops sending new I-frames, and sends REQ-NAKs at every check point interval, during the cumulation time period of $W_{cp}$. On the other hand, the receiver keeps sending CP commands regardless of link failures. If the sender receives at least one of the CP commands (CP-NAKs), the sender starts to send I-frames again. However, the sender still expects to receive an ENF-NAK as a response to the REQ-NAK because the sender buffer has some unacknowledged I-frames which can be resolved. In the example in Figure 7 the suspected link failure turned out to be a burst error and recovery mode returns to the check point recovery mode. When the receiver receives REQ-NAKs, the receiver sets the enforce-bit field of CP commands to 1 (ON), and attaches an error list containing information on all the erroneous I-frames in the error information table to the
CP-NAKs. This mechanism is similar to P/F bit check-pointing recovery of HDLC. When the sender receives the appropriate ENF-NAK as a response to its REQ-NAK, the sender resolves the unacknowledged I-frames and then returns to check point recovery mode.

When the sender receives a CP-NAK prior to an ENF-NAK in the enforced recovery mode, and cannot find any unacknowledged I-frame in the sender buffer, it immediately returns to the check point recovery mode. In this case, the sender may receive the ENF-NAKs in a check point recovery mode; at this time, the REQ-NAKs can be dealt with simply as CP-NAKs. Since limited SQT also guarantees that the sender cannot receive a REQ-NAK emanated from a previous enforced recovery mode, upon receiving a REQ-NAK, the receiver must respond immediately with ENF-NAK. When the sender receives a ENF-NAK within the expected normal response time, the sender stops the failure timer and sends all the I-frames indicated in the error list field of the ENF-NAK. If the receiver has no erroneous I-frames, it simply sends RES commands for synchronization. If the sender receives neither ENF-NAK nor RES commands within the expected normal response time plus the cumulation time, the sender regards the receiver as having failed. This is justified, as indicated previously, by noting that the probability of successive RES commands (ENF-NAKs) all failing is negligible, since \((P_c)^{C_{depth}}\) is extremely small. Once the sender determines a link failure has occurred, it stops transmitting I-frames and informs the network layer.

5 PERFORMANCE

An analytical model for LAMS-DLC has been described in [11] and is included here for comparison with the experimental results. The model is based on counting transmission and retransmission tails for the successful delivery of \(N\) packets. The time required to deliver these packets divided by \(N\) yields the throughput. The model provides an insight into the throughput
efficiency and buffer requirements of LAMS-DLC when compared to the well known event-based POS-ACK ARQ protocol Selective Reject HDLC (SR-HDLC).

Let

\[ t_R = \text{Round trip time between two connected nodes} \]

\[ t_{cp} = \text{Time interval between two consecutive I-frames} \]

\[ W = \text{The window size for HDLC} \]

\[ t_f, t_c = \text{Transmission times of an I-frame and a control command respectively} \]

\[ t_{proc} = \text{The maximum time required to process an I-frame or control command} \]

\[ t_{out} = \text{The timeout defined in HDLC} \]

\[ \overline{n}_{cp} = \text{The mean number of CP-commands needed to ensure I-frames delivery} \]

\[ = \frac{1}{1 - P_c} \]

\[ C_{depth} = \text{The number of consecutive CP-commands covering an I-frame} \]

\[ \overline{S} = \text{Mean number of transmission attempts for an I-frame} \]

\[ D(N) = \text{Mean delay for successful delivery of N I-frames} \]

\[ N_{total}(N) = \text{Total number of transmissions and retransmissions to transmit} \]

\[ N \text{ I-frames} \]

The mean number \( (\overline{S}) \) of periods for successful transmission of an I-frame in LAMS-DLC and SR-HDLC are given by:

\[ \overline{S}_{LAMS} = \frac{1}{1 - P_f} \]

\[ \overline{S}_{HDLC} = \frac{1}{1 - (P_f + P_c - P_f P_c)} \]

In low traffic, and given an infinite buffer, the mean delays for the successful delivery of N I-frames in LAMS-DLC and SR-HDLC are:

\[ D_{low}^{LAMS}(N) \approx N t_f + \overline{S}_{LAMS} t_R + \overline{S}_{LAMS}(t_{cp} - 1/2) t_{cp} \]

\[ D_{low}^{HDLC}(N) \approx N t_f + \overline{S}_{HDLC} t_R + ((\overline{S}_{HDLC} - 1)(1 - P_f - P_c + P_f P_c) - P_c) \alpha \]
where \( \alpha = t_{out} - t_R \). See [12].

While in high traffic, if we let \( m = \left\lfloor \frac{N}{W} \right\rfloor \), \( r = N \mod W \), \( N_{\text{win}}^{HDLC} = N_{\text{total}}(W) \), and \( N_{\text{total}}^{LAMS} = N_{\text{total}}(N) \), the mean total times are given by:

\[
D_{\text{high}}^{HDLC}(N) = m D_{\text{low}}^{HDLC}(N_{\text{win}}^{HDLC}) + D_{\text{low}}^{HDLC}(r_w)
\]

\[
D_{\text{high}}^{LAMS}(N) = D_{\text{low}}^{LAMS}(N_{\text{total}}^{LAMS})
\]

In high traffic, LAMS-DLC shows the following throughput efficiency \( (\eta_{LAMS}) \) with the transparent buffer size \( B_{LAMS} \):

\[
\eta_{LAMS} = \frac{N}{D_{\text{high}}^{LAMS}(N)}
\]

\[
= \frac{N}{N_{\text{total}}^{LAMS} t_f + \bar{s}_{LAMS} t_R + \delta_{LAMS}}
\]

where \( B_{LAMS} = \frac{1}{t_f} \bar{s}_{LAMS} (t_R + (\bar{n}_{cp} - 1/2) I_{cp}) \) and \( \delta_{LAMS} = \bar{s}_{LAMS} (\bar{n}_{cp} - 1/2) I_{cp} \). In low traffic the equation remains the same.

For SR-HDLC, if \( W = B_{LAMS} \) and \( N_{\text{total}}^{HDLC} = m N_{\text{win}}^{HDLC} + r_w \), the throughput efficiency \( (\eta_{HDLC}) \) with buffer size \( B_{HDLC} = 2B_{LAMS} \) is:

\[
\eta_{HDLC} = \frac{N}{D_{\text{high}}^{HDLC}(N)}
\]

\[
= \frac{N}{m D_{\text{low}}^{HDLC}(N_{\text{total}}^{HDLC}) + D_{\text{low}}^{HDLC}(r_w)}
\]

\[
= \frac{N}{N_{\text{total}}^{HDLC} t_f + (M+1)(\bar{s}_{HDLC} t_R + \delta_{HDLC})}
\]

where \( \delta_{HDLC} = ((\bar{s}_{HDLC} - 1)(1 - P_F - P_C - P_F P_C) - P_C) \alpha \). In low traffic \( m = 0 \). We see from the above equations that as channel traffic increases, the throughput efficiency of LAMS-DLC will be significantly higher than that of SR-HDLC.

The equations described above were plotted as a function of BER using parameters shown in Table 1. An appropriate cumulation depth for LAMS-DLC was selected according to Table 2. In the above parameter setting, we set Link Distance and Data Transmission Rate to 300,000
Km and 10 Mbps respectively (different from usual values in LAMS networks). The reason for this is to provide a direct comparison with the results from the simulation model. The simulation is unable to model the transmission rates of 100 Mbps. However, recall that LAMS-DLC is interested in the combination of Link Distance and Data Transmission Rate. Therefore, we can obtain equivalent results as in the a LAMS environment by increasing the Link Distance and proportionally decreasing the Data Transmission Rate. The above parameter settings are equivalent to a Link Distance of 30,000 Km and a Data Transmission Rate of 100 Mbps. Notice also that according to the analysis, the parameter Traffic is dictated by the total number of packets to be transmitted.

<table>
<thead>
<tr>
<th>Link Property</th>
<th>BER (Bit Error Rate) = $10^{-5}$</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Link Distance = 300,000 Km</td>
</tr>
<tr>
<td></td>
<td>Propagation Delay = 1 second</td>
</tr>
<tr>
<td></td>
<td>Round Trip Time = 2 Seconds</td>
</tr>
<tr>
<td>Channel Property</td>
<td>Data Transmission Rate = 10 Mbps</td>
</tr>
<tr>
<td>Data Property</td>
<td>Mean Packet Size = 10,000 Bits</td>
</tr>
<tr>
<td></td>
<td>Mean Command Size = 1,000 Bits</td>
</tr>
<tr>
<td>Traffic</td>
<td>Traffic is determined by the number of packets to be transmitted as follows: very high traffic corresponds to 20 round trip times (RTTs); high traffic to 5 RTTs; medium traffic to one and half RTTs; low traffic to less than one RTT.</td>
</tr>
</tbody>
</table>

Table 1 Parameters for mathematical performance analysis

<table>
<thead>
<tr>
<th>Bit Error Rate</th>
<th>Cumulation Depth $(C_{depth})$</th>
<th>Probability of $C_{depth}$ Successive Checkpoints Lost</th>
</tr>
</thead>
<tbody>
<tr>
<td>$10^{-4}$</td>
<td>36</td>
<td>$6.746 \times 10^{-8}$</td>
</tr>
<tr>
<td>$10^{-5}$</td>
<td>7</td>
<td>$7.067 \times 10^{-8}$</td>
</tr>
<tr>
<td>$10^{-6}$</td>
<td>4</td>
<td>$0.980 \times 10^{-8}$</td>
</tr>
</tbody>
</table>

Table 2 Maximum Cumulation Depth (Continued ... )
| $10^{-7}$ | 3 | $0.099 \times 10^8$ |
| $10^{-8}$ | 2 | $9.999 \times 10^8$ |

Threshold Value = $1.0 \times 10^{-7}$, Command Size = 1,000 Bits

Table 2 Maximum Cumulation Depth

Figures 8, 9, 10, and 11 shows the throughput efficiencies of LAMS-DLC and SR-HDLC as a function of BER for various traffic intensities. Figures 8, 9, 10, and 11 demonstrate that LAMS-DLC is nearly optimal. Figure 12 shows that buffer requirements are stable until a BER of $10^{-5}$ is reached.

6 EXPERIMENTAL RESULTS

6.1 Overview

The LAMS-DLC protocol was implemented using Opnet, an event driven simulation package\(^1\). With the exception of the inter-arrival distribution (which was exponential) the parameters used in the simulation were set to be the same as those used in the mathematical analysis. This model also played the role of verifying that the LAMS-DLC protocol works correctly by ensuring that all packets are received according to the relaxed reliability requirements (i.e. zero packet loss and no duplicates).

6.2 Time-based Analysis

The time-based analysis shows how typical performance measures (link utilization, throughput efficiency, and buffer size) change as communication time elapses. In this analysis, the inter-arrival time of packets is 0.0023 second (434 packets are entering the sending node per second) which corresponds to very high traffic. This rate is optimal for LAMS-DLC given the packet size and BER range and is the maximum sustainable data rate (i.e. a packet interarrival rate greater than this will cause buffer requirements to grow without limit). At this optimal rate LAMS-DLC performs continuous transmission, the throughput efficiency of LAMS-DLC converges to

\(^1\) Opnet version 2.31, Mil-3 Incorporated, Washington, DC 20008.
the upper limit which is determined by BER as illustrated in Figure 13. We have included the
throughput efficiencies derived from conventional analysis [9] (denoted by □) and retransmission
tail analysis (denoted by ◦) for comparison. Buffer behavior and mean queuing delay are
provided in Figures 14, and 15 respectively. As expected, the buffer requirement for continuous
transmission is definitely bounded and the packet queueing delay is limited.

The optimal rate was obtained by considering the throughput efficiency, link utilization, and
buffer requirements of LAMS-DLC at the maximum BER of 10^{-5} as packet arrival rate is
increased (see Figures 16, 17, and 18).

6.3 Bit Error Rate based Analysis

The BER analysis shows how the performance measures (throughput efficiency, buffer re-
quirement and lost packets) change as the BER increases. Figure 19 shows that throughput
efficiency begins to drop rapidly at BERs higher than 10^{-5} while Figure 20 shows a stable
buffer requirement up to a data rate of 10^{-5}, above which the buffer requirements increase
rapidly. The figures agree with the mathematical analysis in Figures 8 and 9 of Section 5.

7 CONCLUSIONS

This paper has demonstrated that a relaxation of the strict reliability constraints (FIFO packet
delivery, zero packet loss and zero duplicates) provides the opportunity to specify a new class
of link layer service, namely the reliable datagram. This service has characteristics of highly
reliable services as found in the event-based POS-ACK ARQ protocols such as Selective Reject
HDLC (SR-HDLC) but offers performance and buffer advantages over them. In particular, we
have shown that by relaxing the FIFO delivery requirement the sender queuing time (SQT) and
receiver holding time (RHT) may be bounded from the perspective of packet numbering, which
is impossible for event-based POS-ACK protocols. This permits continuous packet transmission
without the requirement for a resolution period.
We have described a new timer-based protocol, LAMS-DLC, with these characteristics based on check points and negative ACK (NAK) error control. We have demonstrated that the performance of the protocol is nearly optimal in its target environment and requires only half the buffer space of a typical event-based POS-ACK protocol such as SR-HDLC. LAMS-DLC has been implemented using the Opnet simulation package to verify correct operation and provides results that agree closely with theoretical analysis.

8 BIBLIOGRAPHY


Figure 1 Inconsistency degree over time
Figure 2 Queuing delay and the inconsistency gap
Figure 3 Check point acknowledgment
Figure 4: LAMS-DLC protocol operation and inconsistency degree over time
1. Information Frame Format

2. Control Frame Format

<table>
<thead>
<tr>
<th>General Form</th>
<th>CP types</th>
<th>0</th>
<th>SN</th>
<th>Error List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Command format</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NAK format</td>
<td></td>
<td></td>
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<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>CP Command or CP-NAK</th>
<th>CP types</th>
<th>SN</th>
<th>Error List</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>RES Command or ENF-NAK</th>
<th>CP types</th>
<th>SN</th>
<th>Error List</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>REQ-NAK</th>
<th>CP types</th>
<th>SN</th>
</tr>
</thead>
</table>

Figure 5 LAMS-DLC frame format
I-frames | CP commands
---|---
0 | CP(0,(none))
1 | CP(2,(1))
2 | CP(5,(1,4,5))

Check Point Interval

I-frame 6 is a retransmission of I-frame 1.

This range is a cumulation time of CP(5,(1,4,5)). So its error list contains 1, 4, 5.

Note: 1. CP((i,(j,k,...))) means that i is the most recent SN and (j,k,...) is the error list.
2. CP(0,(none)) is a CP command since it does not have an error list.
3. CP(5,(1,4,5)) is a CP NAK because of its error list.

Figure 6 Protocol operation for check point recovery
Cumulation Depth = 4

During this cumulation time, No response.
The sender suspects a link failure
REQ-NAK 1
REQ-NAK 2
REQ-NAK 3
REQ-NAK 4

From this point, everything is resolved. Thus the sender starts sending I-frames again.

If CPs after corruption period are lost, from this point, the sender goes back to normal.

At this point, the receiver gets REQ-NAK, so it starts sending ENF-NAK

Figure 7 Protocol operation for enforced recovery
Figure 8: Comparison of throughput efficiencies of LAMS-DLC and HDLC using retransmission tail analysis at very high traffic.
Figure 9  Comparison of throughput efficiencies of LAMS-DLC and HDLC using retransmission tail analysis at high traffic
Figure 10: Comparison of LAMS-DLC and HDLC using retransmission tail analysis at medium traffic.
Figure 11  Comparison of throughput efficiencies of LAMS-DLC and HDLC using retransmission tail analysis at low traffic
Figure 12 LAMS-DLC buffer requirements
Link Utilization and Throughput Efficiency

- Link Utilization
- Retx-Tailed Thruput Eff.
- Tradition Ideal Thruput Eff.

Figure 13 Performance of LAMS-DLC in terms of throughput efficiency
Buffer Behavior at Optimal Incoming Rate

- Send Eff Rates (x1e-07)
-Recv Eff Rates (x1e-07)
-Buffer Size (x1e-07)

1. Incoming Packet Inter-Arrival Time = 0.0023 (sec)
   Effective Incoming Data Rate = 434 (packets per sec)
   Max Outgoing Packet Inter-Departure Time = 0.0010 (sec)
   Effective Outgoing Packet Inter-Departure = 0.0023 (sec)
   Optimal Outgoing Data Rate = 434 (packets per sec)

Figure 14 LAMS-DLC buffer convergence in the case of optimal incoming rate
The Mean Queueing Delay in Physical Channel Buffer
at Inter-Arrival Time = 0.0023

Figure 15 The mean delay that a packet waits for a channel in the case of optimal incoming rate.
Throughput Efficiency at Packet Size = 10,000

Figure 16 LAMS-DLC throughput efficiency versus inter-arrival time at mean packet size = 10,000 bits
Figure 17 LAMS-DLC link utilization versus inter-arrival time at mean packet size = 10,000 bits.
Figure 18 LAMS-DLC buffer requirement versus inter-arrival time at mean packet size = 10,000 bits
Throughput Efficiency Versus BER (Retransmission Tail)

Figure 19 LAMS-DLC throughput efficiency using retransmission tail analysis versus BER
Figure 20 LAMS-DLC buffer requirement versus BER