ADAPTIVE TIMEOUT FOR VIDEO DELIVERY OVER A BLUETOOTH WIRELESS NETWORK

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ABSTRACT
Bluetooth is a short-range wireless interconnect that in its 2.0 version has sufficient capacity to support encoded video streaming, possibly from an Internet access point. The default setting of the Automatic Repeat Request (ARQ) facility leads to poor quality video, as streaming is delay-sensitive. The paper introduces an adaptive algorithm that alters the number of ARQ retransmissions according to buffer fullness, which is used to indicate both buffer congestion and radio frequency channel conditions. This is a necessary but not sufficient requirement for quality video at the receiver. In an extension to the cross-layer approach, the number of retransmissions is made sensitive to picture importance. Delivered video quality is demonstrated to be consistently high.

KEYWORDS
Bluetooth, ARQ, video streaming, wireless network

1. INTRODUCTION

IEEE 802.15.1, Bluetooth (B/T) version (v.) 1.0 [6] in the 2.4 GHz ISM band, has received comparatively limited investigation as a way of streaming video across a wireless interconnect. The Enhanced Data Rate (EDR) of B/T version 2.0 [10] now has a peak user payload of 2.2 Mb/s, which is the same rate offered by current implementations of interactive IP-TV. Therefore, a bottleneck free way exists of streaming encoded video clips from a server across an IP network to a B/T master node, and, thence, over a B/T interconnect. Moreover, many cellular phones are also equipped with a B/T transceiver and larger resolution screens of CIF (352×288) and QCIF (176×144) pixel size.

B/T employs variable-sized packets up to a maximum of five frequency-hopping slots of 625 μs duration. Due to the risk of radio-frequency (RF) noise, by default, Bluetooth provides repeat packet transmissions, if a packet is not acknowledged within a pre-set time. Every Bluetooth frame consists of a packet transmitted from the master node over 1, 3 or 5 timeslots, while a slave replies with a packet occupying at least one slot, so that each frame has an even number of slots. Therefore, a single-slot packet serves for a link layer stop-and-go Automatic Repeat Request (ARQ) message whenever a corrupted packet payload is detected. The timeout or retransmission limit value by default is set to an infinite number of retransmissions. On general grounds, this is unwise in conditions of fast fading causing by multi-path echoes, as error bursts occur.

Work in [8] appears to have first introduced priority-based re-transmission for video frames, though that paper went no further than a *static* scheme favouring anchor pictures at the link layer, based on similar application layer techniques such as [11]. The need to adjust *adaptively* the default re-transmission limit for multimedia applications was established in [1]. Our paper's contribution is an *adaptive* scheme according to channel conditions, as reported by buffer fullness, with priority given to differing picture types and not just anchor pictures.

Real-time delivery of video is delay-sensitive and a frame cannot be displayed if it arrives after the decode deadline. In practice, a play-out buffer exists on a mobile device to account for start-up delay. Additionally, pictures are not necessarily decoded in order, as some (bi-predictive or B-pictures) rely on predictions from subsequent pictures before the frame can be displayed. Therefore, the maximum delay permissible corresponds to the tolerable start-up delay. (Ideally, start-up delay should be imperceptible but typically a device will also contain a receive buffer, while memory has an impact on cost and power usage [13].) Equally, simply dropping all packets corrupted by RF noise is not an option, as it is likely that the display quality, the luma (Y) Peak Signal-to-Noise Ratio (PSNR) will deteriorate, though for encoded video the actual impact is dependent on the type of picture dropped and scene content characteristics. For mobile applications a PSNR in the range 34-38 dB is desirable [4].

Therefore, some compromise value for the ARQ timeout value should be sought and in this paper that value is made adaptive to the Bluetooth network conditions. Though various options are available for judging the adapted value such as packet delay recorded at the receiver, packet loss, and master send buffer fullness we adopted buffer fullness (see Section 6). Buffer fullness is available to an application via the Host Computer Interface (HCI) presented by a Bluetooth hardware module to the upper layer software protocol stack. The need to retransmit arises from RF frequency interference, which has a volatile distribution over time. However, retransmissions also cause the master's send buffer queue to grow, with the possibility of packet loss from buffer overflow. (The receive buffer is assumed to have the same capacity as the send buffer and is thus unaffected by congestion.) There are two possible sources of congestion in the master send buffer: 1) from the streaming video source making a final single-hop across the B/T network to the destination slave; and 2) packets arriving from another slave and passing via the master send buffer to the destination slave. Consequently, one advantage of monitoring buffer fullness is that it can be expanded to B/T network operation (or more precisely a B/T piconet formed by a master and up to seven slaves [6]). Piconet-wide congestion is not considered further in this paper.

It turns out that adaptation to channel conditions and buffer congestion, as advocated by this paper, though improving video quality, is insufficient by itself to achieve acceptable quality. Therefore, this paper contributes additional adaptation to picture type, therefore combining the merits of various ideas. For reasons of error resilience, encoded video is transmitted as a repeating sequence of Group of Pictures (GOP) [5], with the start of each GOP formed by an Intra-coded or I-picture. An I-picture is the basis for prediction of all other pictures in the GOP (usually 12 to 15 pictures in all) and, hence, its loss has drastic consequences for all other pictures in the GOP. Other predictive pictures or P-pictures also form the basis for predictions but are not essential for the reconstruction of other pictures within the GOP (as other I- or P- anchor pictures retained in the decode buffer can be applied). Lastly, the third type of picture, the B-picture, has no predictive value. The picture type is identifiable through the bit-stream header without decoding. By adapting the timeout value to the type of picture, the received video is indeed raised to a desirable quality level, as the paper demonstrates.

2. METHODOLOGY

This research employed the University of Cincinatti B/T (UCB/T) extension (download is available from `http://www.ececs.uc.edu/~cdmc/ucB`) to the well-known ns-2 network simulator (v. 2.28 used). The UCB/T extension supports B/T EDR but is also built on the air models of previous B/T extensions such as BlueHoc from IBM and Blueware. All links were set at the maximum EDR 3.0 Mbps gross air rate. The send and receive buffer sizes were set to fifty packets. Simulation runs were each repeated ten times and the results averaged to produce summary statistics.

A Gilbert-Elliot [2] two state discrete-time, ergodic Markov chain modeled the wireless channel error characteristics between a B/T master and slave node. By adopting this model it was possible to simulate burst errors of the kind that cause problems to an ARQ mechanism. The mean duration of a good state, T_g, was set at 2 s and in a bad state, T_b, was set to 0.25 s. In units of 625 μs (the B/T time slot duration), T_g= 3200 and T_b= 400, which implies from:

$$T_g = \frac{1}{1 - Pgg}, T_b = \frac{1}{1 - Pbb} \quad (1)$$

that, given the current state is g, Pgg, the probability that the next state is also g , is 0.9996875 and Pbb, given the current state is b, the probability that the next state is also b, is 0.9975. At 3.0 Mb/s, the Bit Error Rate (BER) during a good state was set to 10^{-5} and during a bad state to 10^{-3}. The transition probabilities, Pgg and Pbb, as well as the BER, are approximately similar to those in [3], but the mean state durations are adapted to B/T. The two states result in Carrier-to-Noise Ratios (CNRs) of respectively 15.97 and 13.02 dB.

The simulations were carried out with input from an MPEG-2 encoded bitstream at a mean rate of 1.2 Mbit/s for a 40 s video clip with moderate motion. PSNR was found by reconstructing with a reference MPEG-2 decoder. The display rate was 25 frame/s, 1000 frames in all. The source video was CIF sized (366×288 pixels) with a GOP structure of N = 12, and M = 3 (M is the number of pictures from the I-picture to the first P-picture, i.e. including two B-pictures). In [9], fully-filled B/T packets were formed using maximal bandwidth five time slot packets, regardless of slice boundaries. While this results in some loss in error resilience, as each MPEG-2 slice contains a decoder synchronization marker, in [9] it is shown that the overall video performance is superior.

3. IMPACT OF ARQ

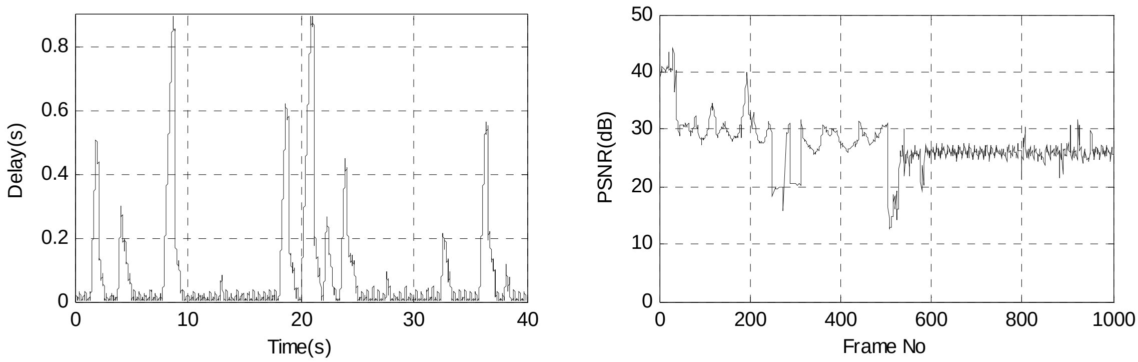
An ARQ may occur in the following circumstances [12]: a) failure to synchronize on the access header code; b) header corruption detected by a triple redundancy code; c) payload corruption detected by CRC; d) failure to synchronize with the return packet header; e) header corruption of the return packet. Notice that a faulty ARQ packet can itself cause retransmission. Though not investigated further herein, there is a power usage implication from excessive re-transmissions. The default value of the ARQ retransmission timeout (RTO) in most B/T chipsets is set to infinity. In [1], a fixed RTO and an adaptive RTO were considered. The disadvantage of a fixed RTO is that it is difficult to arrive at a value that avoids either excessive delay or excessive packet drops in *all* circumstances. The adaptive RTO, which was upper and lower- bounded, was based in [1] on a smoothed round-trip time (srtt). The RTO was adapted downwards or upwards if the new rtt respectively is less than or more than the previous srtt.

In a first set of simulations, the ARQ RTO was set to its default value, i.e. infinite repeat retransmissions. In Fig. 1 (a), owing to channel noise and buffer congestion, there are delay spikes during error bursts. Packet discard occur after 0.4 s at the receiver application, though the packet is delivered by the B/T data-link sub-system. However, delay is not the only cause of packet loss, as overflow of the fifty packet send buffer also leads to packet loss. In Fig. 1 (b), the PSNR is mostly below the target range. A brief period of higher quality video occurs at the start of the sequence, before the send buffer has had time to fill up. The two phases of PSNR, discernible before and after about 500 s, are caused by a change in the amount of motion present, with a possible scene change marked by a dip in quality around the transition point. In Table 1, because ARQ is applied, there are no losses from RF noise but as a result losses from buffer overflow occur at a faster rate than those from late arrival.

Table 1. Packet loss rates (packets lost/total packets sent) and resulting mean PSNR, with ARQ

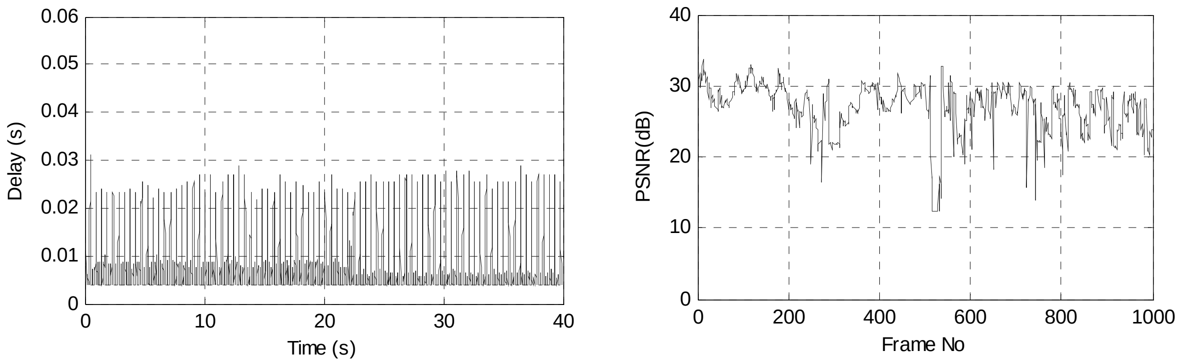
Total packet loss rate	0.1735
Loss rate due to RF noise	0.0
Loss rate due to buffer overflow	0.0990
Loss rate due to late arrival	0.0744
Mean PSNR (dB)	26.99

In a second set of experiments, ARQ retransmission was disabled, but otherwise the configuration was the same as in the previous experiment. In Fig. 2 (a), as might naturally be expected delay is always well below the discard threshold at the receiver. However, the sample run recorded in Fig. 2 (b) shows that the video quality remains below an acceptable level. In Table 2, the sole cause of packet loss is due to RF noise and packets are chiefly affected by propagation delay.



(a) (b)

Figure 1. Default ARQ (a) Packet delay (b) PSNR with receiver packet discard after 0.4 s



(a) (b)

Figure 2. ARQ disabled (a) Packet delay (b) PSNR with received packet discard after 0.4 s

Table 2. Packet loss rates (packets lost/total packets sent) and resulting mean PSNR, without ARQ

Total packet loss rate	0.1504
Loss rate due to RF noise	0.1504
Loss rate due to buffer overflow	0.0
Loss rate due to late arrival	0.0
Mean PSNR (dB)	27.39

4. ADAPTIVE RETRANSMISSION

The ARQ RTO can be adaptively selected in terms of number of retransmissions allowed, to avoid further delay after the packet enters the tail of the FIFO send buffer. A threshold is set that is the maximum number of retransmissions allowed when the buffer is empty. The maximum number of retransmissions is subsequently changed by a factor depending on the buffer fullness reported by the B/T module. The formula employed is summarized as

$$N = round\left(\frac{m.(c-f)}{c}\right), \quad (2)$$

where N is the maximum number of retransmissions allowed -- the RTO, m is the maximum number of retransmissions allowed when the buffer is empty, f is the number of packets buffered in the send buffer (buffer fullness), and c is the buffer capacity (set to 50 in the simulations). The operator *round* returns the nearest integer. According to (2), the maximum number of retransmissions allowed is a function of buffer fullness. Figure 3 plots this function when $m = 2,3,5$ and $c = 50$ packets. When the buffer is empty, $f = 0$, then the maximum number of retransmissions occurs, whereas when the buffer approaches full occupation then no retransmissions may occur. The smaller the value of m the sooner this latter event occurs.

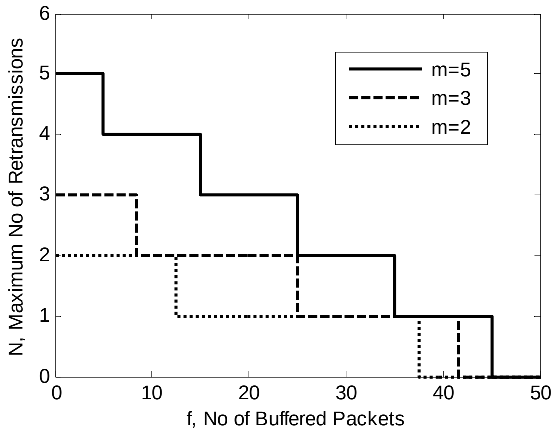


Figure 3. Maximum number of retransmissions, RTO according to sender buffer fullness

Figure 4 shows results with c set to 50 packets and m to 3. Compared to Figure 1, packet delay in Figure 4 (b) is considerably reduced, although there are still some peaks in the delay, occurring when the transmitter repeatedly tries to retransmit packets, at a time when the channel mode is bad. The overall video quality is improved considerably over time, Figure 4 (b), as the mean PSNR is now 32.34 dB, Table 3. The adaptive ARQ algorithm avoids pushing the buffer to its limit, Figure 4 (c). In fact, buffer fullness closely follows the pattern of delays in Figure 4 (a), indicating the appropriateness of choosing buffer fullness as a gauge. Therefore, no buffer overflow, Table 3, was observed. However, apart from the initial period during which the buffer gradually fills up, the quality still falls below an optimal level.

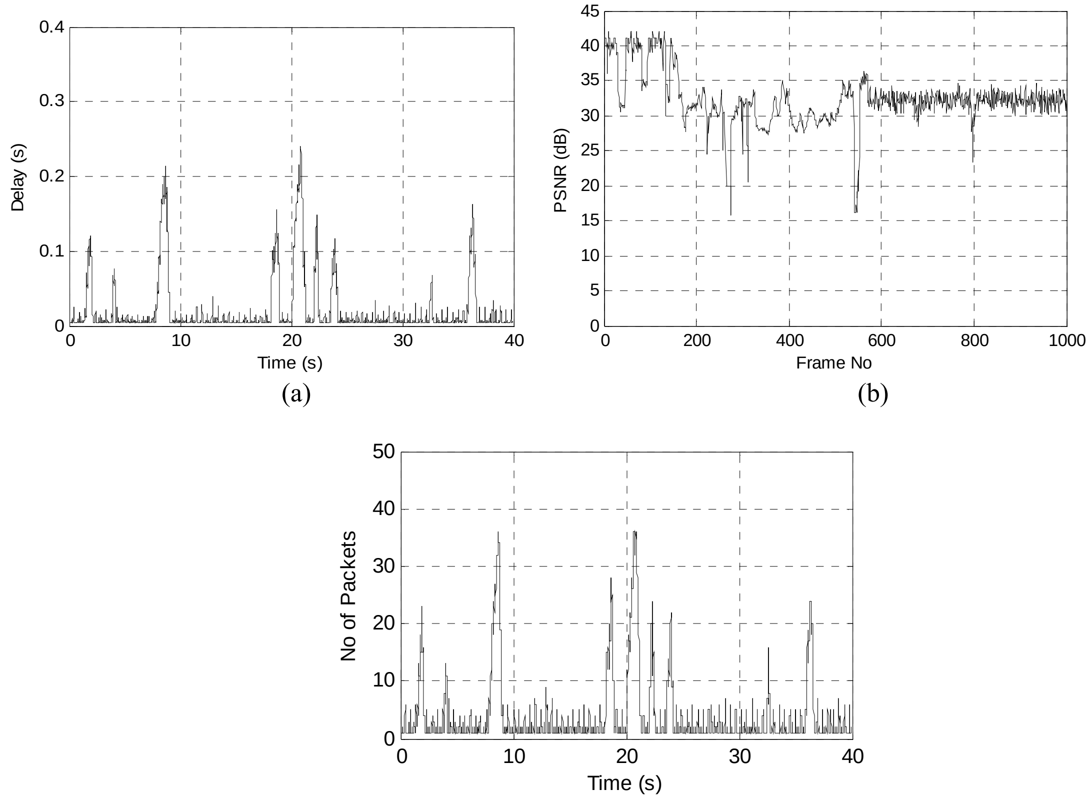
Table 3. Packet loss rates (packets lost/total packets sent) and resulting mean PSNR, with adaptive ARQ

Total packet loss rate	0.0923
Loss rate due to RF noise	0.0923
Loss rate due to buffer overflow	0.0
Loss rate due to late arrival	0.0
Mean PSNR (dB)	32.34

5. PRIORITY-BASED ADAPTIVE ARQ

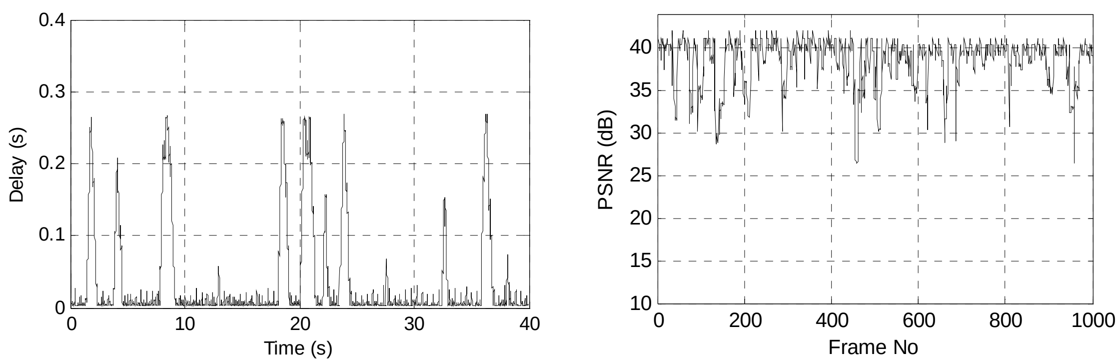
A further extension is to adapt the value of m to the type of picture being transmitted. If the picture is of I-type, upon which all other pictures in the GOP depend, then m is set to five. Similarly for P- and B-pictures, m is respectively set to three and two (in [8] bandwidth was distributed randomly to P- and B-pictures).

Comparing Figure 5 (a) with Figure 4 (a), there are actually more events when delay reaches above 0.2 s and visual inspection establishes that delay is greater. However, the delay never exceeds the discard threshold, 0.4 s, because of the adaptive algorithm. Turning to the PSNR, Figure 5 (b), it can be seen that the PSNR is now consistently very high, with mean 38.54 dB from Table 4, and even exceeds the quality normally acceptable for mobile devices.



(a) (b) (c)

Figure 4. Adaptive ARQ (a) Packet delay (b) PSNR with received packet discard (c) buffer fullness



(a) (b)

Figure 5. Priority-based adaptive ARQ (a) Packet delay (b) PSNR with receiver packet discard

Table 4. Packet loss rates (packets lost/total packets sent) and resulting mean PSNR, with adaptive ARQ

Total packet loss rate	0.0876
Loss rate due to RF noise	0.0876
Loss rate due to buffer overflow	0.0
Loss rate due to late arrival	0.0
Mean PSNR (dB)	38.54

6. BUFFER FULLNESS

In [7], it is suggested that for congestion control the input packet rate should be increased (decreased) rate when the loss rate is below 5% (higher than 15%), based on periodic feedback from the receiver. (ARQ appears to have been turned off in the simulations.) Unfortunately, apart from the possible delay in reporting packet losses, for our video clip example these levels cannot be recommended, as loss rates of 10% or more already lead to a poor PSNR. Figure 6 shows this for a Uniform pdf of packet losses.

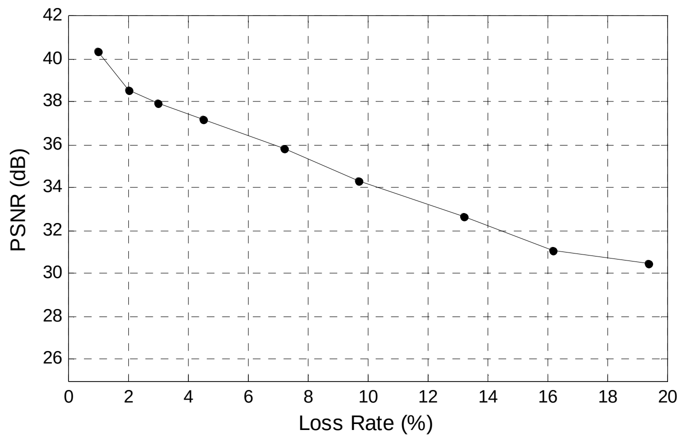


Figure 6: The effect of packet loss on the received video clip quality

In [9], packet delay recorded at the receiver was found to be a better indicator of congestion than packet loss but resulted in oscillations in both video quality and delay in packet delivery. In Figure 7, it is apparent that the mean buffer fullness is dependent on the input rate (from the same video clip in previous Sections' simulations encoded with MPEG-2 at differing target rates). Similarly, Figure 8 shows that there is also a correspondence between the bit error rate (BER) of the RF channel and buffer fullness when retransmission is in use, because packets are held back in the buffer while a retransmission completes. (Bit errors are taken from a Uniform distribution.) Therefore, we conclude that for B/T interconnects buffer fullness is a superior metric to both packet loss and delay.

7. CONCLUSION

The paper has demonstrated that by effectively managing Bluetooth's ARQ mechanism, high quality video can be delivered across the wireless link. Though Bluetooth has an obvious application in audio streaming, it has been though that video streaming was not appropriate even with EDR. The main additional requirement of the management algorithm is monitoring of encoded packet headers and buffer fullness so as to change the retransmission limit. Encoded packet headers are available to an Internet access point as the IP packets arrive from a remote multimedia server, while the buffer fullness and retransmission limit can be set through the existing cross-layer HCI. In future work, the algorithms can be refined further by monitoring the number of intra-, inter, and SKIP macro-blocks in a P-picture and progressively weighting its ARQ retransmission priority accordingly, at a cost in keeping a frame buffer at the sender.

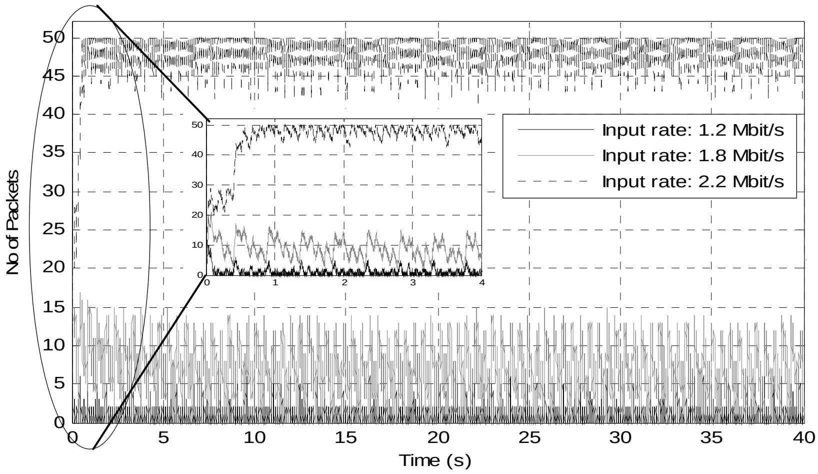


Figure 7: The effect of changes in the input rate on the sender's buffer fullness

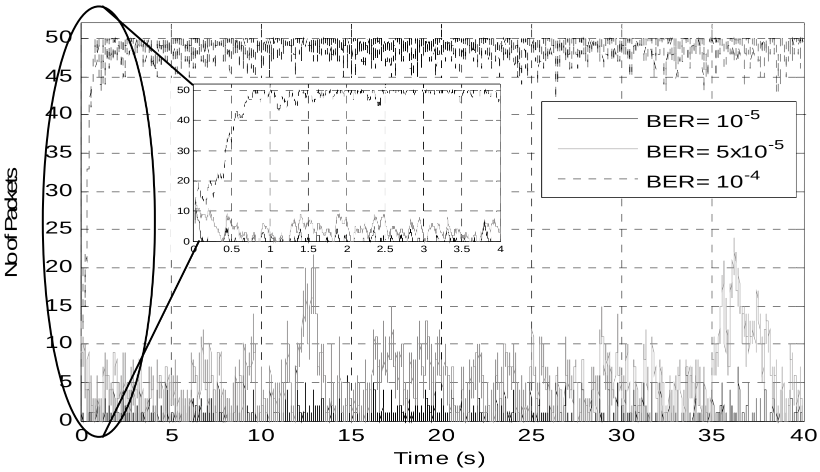


Figure 8: The effect of changes in the RF channel on the sender's buffer fullness

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