

Achieving reliable UDP transmission at 10 Gb/s using BSD socket for data acquisition systems

M.J. Christensen,^{a,1} T. Richter^b

^a*European Spallation Source, Data Management and Software Centre, Ole Maaløes vej 3, 2200 Copenhagen N, Denmark*

^b*European Spallation Source ERIC, P.O. Box 176, 221 00 Lund, Sweden*

E-mail: mortenchristensen@esss.se

ABSTRACT: User Datagram Protocol (UDP) is a commonly used protocol for data transmission in small embedded systems. UDP as such is unreliable and packet losses can occur. The achievable data rates can suffer if optimal packet sizes are not used. The alternative, Transmission Control Protocol (TCP) guarantees the ordered delivery of data and automatically adjusts transmission to match the capability of the transmission link. Nevertheless UDP is often favored over TCP due to its simplicity, small memory and instruction footprints. Both UDP and TCP are implemented in all larger operating systems and commercial embedded frameworks. In addition UDP also supported on a variety of small hardware platforms such as Digital Signal Processors (DSP) Field Programmable Gate Arrays (FPGA). This is not so common for TCP. This paper describes how high speed UDP based data transmission with very low packet error ratios was achieved. The near-reliable communications link is used in a data acquisition (DAQ) system for the next generation of extremely intense neutron source, European Spallation Source. This paper presents measurements of UDP performance and reliability as achieved by employing several optimizations. The measurements were performed on Xeon E5 based CentOS (Linux) servers. The measured data rates are very close to the 10 Gb/s line rate, and zero packet loss was achieved. The performance was obtained utilizing a single processor core as transmitter and a single core as receiver. The results show that support for transmitting large data packets is a key parameter for good performance.

Optimizations for throughput are: MTU, packet sizes, tuning Linux kernel parameters, thread affinity, core locality and efficient timers.

KEYWORDS: Computing (architecture, farms, GRID for recording, storage, archiving, and distribution of data), Data acquisition concepts, Software architectures (event data models, frameworks and databases)

¹Corresponding author.

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1 Introduction

European Spallation Source [1] is a next generation neutron source currently being developed in Lund, Sweden. The facility will initially support about 16 different instruments for neutron scattering. In addition to the instrument infrastructure, the ESS Data Management and Software Centre (DMSC), located in Copenhagen, provides infrastructure and computational support for the acquisition, event formation and long term storage of the experimental data. At the heart of each instrument is a neutron detector and its associated readout system. Both detectors and readout systems are currently in the design phase and various prototypes have already been produced [2–5]. During experiments data is being produced at high rates: Detector data is read out by custom electronics and the readings are converted into UDP packets by the readout system and sent to event formation servers over 10 Gb/s optical Ethernet links. The event formation servers are based on

general purpose CPUs and it is anticipated that most if not all data reduction at ESS is done in software. This includes reception of raw readout data, threshold rejection, clustering and event formation. For a detailed description of the software architecture and early detector processing implementations see [6]. UDP is a simple protocol for connectionless data transmission [7] and packet loss can occur during transmission. Nevertheless UDP is widely used, for example in the RD51 Scalable Readout System [8], or the CMS trigger readout [9], both using 1 Gb/s Ethernet. The two central components are the readout system and the event formation system. The readout system is a hybrid of analog and digital electronics. The electronics convert deposited charges into electric signals which are digitized and timestamped. In the digital domain simple data reduction such as zero suppression and threshold based rejection can be performed. The event formation system receives these timestamped digital readouts and performs the necessary steps to determine the position of the neutron. These processing steps are different for each detector type. The performance of UDP over 10G Ethernet has been the subject of previous studies [10] [11], which measured TCP and UDP performance and CPU usages on Linux using commodity hardware. Both studies use a certain set of optimizations but otherwise using standard Linux. In [10] the transmitting process is found to be a bottleneck in terms of CPU usage, whereas a comparison between Ethernet and InfiniBand [11] reinforces the earlier results and concludes that Ethernet is a serious contender for use in a readout system. This study is aimed at characterizing the performance of a prototype data acquisition system based on UDP. The study is not so much concerned with transmitter performance as we expect to receive data from a FPGA based platform capable of transmitting at wire speed at all packet sizes. Instead comparisons between the measured and theoretically possible throughput and measurements of packet error ratios are presented. Finally, this paper presents strategies for optimizing the performance of data transmission between the readout system and the event formation system.

2 TCP and UDP pros and cons

Since TCP is reliable and has good performance whereas UDP is unreliable why not always just use TCP? The pros and cons for this will be discussed in the following. Both TCP and UDP are designed to provide end-to-end communications between hosts connected over a network of packet forwarders. Originally these forwarders were routers but today the group of forwarders include firewalls, load balancers, switches, Network Address Translator (NAT) devices etc. TCP is connection oriented, whereas UDP is connectionless. This means that TCP requires that a connection is setup before data can be transmitted. It also implies that TCP data can only be sent from a single transmitter to a single receiver. In contrast UDP does not have a connection concept and UDP data can be transmitted as either Internet Protocol (IP) broadcast or IP multicast. As mentioned earlier the main argument for UDP is that it is often supported on smaller systems where TCP is not. A notable example are FPGA based systems (see [12] for one case). For a brief overview of efforts for providing TCP/IP support in FPGAs see [13]. But some of the TCP features are not actually improving the performance and reliability in the case of special network topologies as explained below.

2.1 Congestion

Any forwarder is potentially subject to congestion and can drop packets when unable to cope with the traffic load. TCP was designed to react to this congestion. Firstly TCP has a slow start algorithm whereby the data rate is ramped up gradually in order not to contribute to the network congestion itself. Secondly TCP will back off and reduce its transmission rate when congestion is detected. In a readout system such as ours the network only consists of a data sender and a data receiver with an optional switch connecting them. Thus the only places where congestion occurs are at the sender or receiver. The readout system will typically produce data at near constant rates during measurements so congestion at the receiver will result in reduced data rates by the transmitter when using TCP. This first causes buffering at the transmitting application until the buffer is full and eventually packets are lost.

For some detector readout it is not even evident that guaranteed delivery is necessary. In one detector prototype we discarded around 24% of the data due to threshold suppression, so spending extra time making an occasional retransmission may not be worth the added complexity.

2.2 Connections

Since TCP requires the establishment of a connection, both the receiving and transmitting applications must implement additional state to detect the possible loss of a connection. For example upon reset of the readout system after a software upgrade or a parameter change. With UDP the receiver will just 'listen' on a specified UDP port whenever it is ready and receive data when it arrives. Correspondingly the transmitter can send data whenever it is ready. UDP reception supports many-to-one communication, supporting for example two or more readout systems in a single receiver. For TCP to support this would require handling multiple TCP connections.

2.3 Addressing

UDP can be transmitted over IP as multicast. This means that a single sender can reach multiple receivers without any additional programming effort. This can be used for seamless switchovers, redundancy, load distribution, monitoring, etc.. Implementing this in TCP would add complexity to the transmitter.

In summary: For our purposes UDP appears to have more relevant features than TCP. Thus it is preferred provided we can achieve the desired performance and reliability.

3 Performance optimizations

This section explains the factors that contribute to limiting performance, reproducibility or accuracy of the measurements. Here we also discuss the optimization strategies used to achieve the results.

3.1 Transmission of data

An Ethernet frame consists of a fixed 14 bytes header the Ethernet payload, padding and a 4 byte checksum field. Padding is applied to ensure a minimum Ethernet packet size of 64 bytes. There is a minimum gap between Ethernet frames of 20 bytes. This is called the Inter Frame Gap (IFG).

Standard Ethernet supports ethernet payloads from 1 to 1500 bytes. Ethernet frames with payload sizes above 1500 bytes are called jumbo frames. Some Ethernet hardware support payload sizes of 9000 bytes corresponding to Ethernet frame sizes of 9018 when including the header and checksum fields. This is shown in Figure 1 (top). The Ethernet payload consists of IP and UDP headers as well as user data. This is illustrated in Figure 1 (bottom). For any data to be transmitted over Ethernet, the factors influencing the packet and data rates are: The link speed, IFG and the payload size. The largest supported Ethernet payload is called the Maximum Transmission Unit (MTU). For further information see [14] and [15].

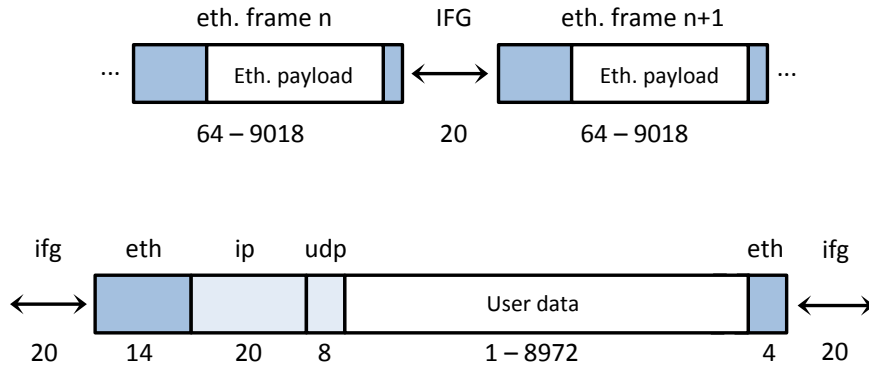


Figure 1. (top) Ethernet frames are separated by a 20 byte inter frame gap. (bottom) The Ethernet, IP and UDP headers take up 46 bytes. The largest UDP user data size is 1472 bytes on most Ethernet interfaces due to a default MTU of 1500. This can be extended on some equipment to 8972 bytes by the use of jumbo frames.

Sending data larger than the MTU will result in the data being split in chunks of size MTU before transmission. Given a specific link speed and packet size, the packet rate is given by

$$\text{rate}[\text{packets per second}] = \frac{ls}{8 \cdot (ps + ifg)}$$

where ls is the link speed in b/s, ps the packet size and ifg the inter frame gap. Thus for a 10 Gb/s Ethernet link, the packet rate for 64 byte packets is 14.88 M packets per second (pps) as is shown in Table 1.

Table 1. Packet rates as function of packet sizes for 10 Gb/s Ethernet

User data size [B]	1	18	82	210	466	978	1472	8972
Packet size [B]	64	64	128	256	512	1024	1518	9018
Overhead [%]	98.8	78.6	44.6	23.9	12.4	5.5	4.3	0.7
Frame rate [Mpps]	14.88	14.88	8.45	4.53	2.35	1.20	0.81	0.14

Packets arriving at a data acquisition system are subject to a nearly constant per-packet processing overhead. This is due to interrupt handling, context switching, checksum validations and header processing. At almost 15 M packets per second this processing alone can consume most of the available CPU resources. In order to achieve maximum performance, data from the electronics

readout should be bundled into jumbo frames if at all possible. Using the maximum Ethernet packet size of 9018 bytes reduces the per-packet overhead by a factor of 100. This does, however, come at the cost of larger latency. For example the transmission time of 64 bytes + IFG is 67 ns, whereas for 9018 + IFG it is 902 ns. For applications sensitive to latency a tradeoff must be made between low packet rates and low latency.

Not all transmitted data are of interest for the receiver and can be considered as overhead. Packet headers is such an example. The Ethernet, IP and UDP headers are always present and takes up a total of 46 bytes as shown in Figure 1 (bottom). The utilization of an Ethernet link can be calculated as

$$U = \frac{d}{d + 46 + ifg + pad}$$

where U is the link utilization, d the user data size, ifg the inter frame gap and pad is the padding mentioned earlier. For user data larger than 18 bytes no padding is applied. This means that for small user payloads the overhead can be significant, making it impossible to achieve high throughput. For example transmitting a 32 bit counter over UDP will take up 84 bytes on the wire (20 bytes IFG + 64 byte for a minimum Ethernet frame) and the overhead will account for approx. 95% of the available bandwidth. In contrast when sending 8972 byte user data the overhead is as low as 0.73%.

3.2 Network buffers and packet loss

A UDP packet can be dropped in any part of the communications chain: The sender, the receiver, intermediate systems such as routers, firewalls, switches, load balancers, etc. This makes it difficult in general to rely on UDP for high speed communications. However for simple network topologies such as the ones found in detector readout systems it is possible to achieve very reliable UDP communications. When for example the system comprise two hosts (sender and receiver) connected via a switch of high quality, the packet loss is mainly caused by the Ethernet Network Interface Card (NIC) transmit queue and the socket receive buffer size. Fortunately these can be optimized. The main parameters for controlling socket buffers are `rmem_max` and `wmem_max`. The former is the size of the UDP socket receive buffer, whereas the latter is the size of the UDP socket transmit buffer. To change these values from an application use `setsockopt()`, for example

```
int buffer = 4000000;
setsockopt(s, SOL_SOCKET, SO_SNDBUF, buffer, sizeof(buffer));
setsockopt(s, SOL_SOCKET, SO_RCVBUF, buffer, sizeof(buffer));
```

In addition there is an internal queue for packet reception whose size (in packets) is named `netdev_max_backlog`, and a network interface parameter, `txqueuelen` which were also adjusted.

The default value of these parameters on Linux are not optimized for high speed data links such as 10 Gb/s Ethernet, so for this investigation the following parameters were used.

```
net.core.rmem_max=12582912
net.core.wmem_max=12582912
net.core.netdev_max_backlog=5000
txqueuelen 10000
```

These values have largely been determined by experimentation although a detailed description of these and other optimisations can be found in [16]. We also configured the ethernet adaptors with an MTU of 9000 allowing user payloads up to 8972 bytes when taking into account that IP and UDP headers are also transmitted.

3.3 Core locality

Modern CPUs rely heavily on cache memories to achieve performance. This holds for both instructions and data access. For Xeon E5 processors there are three levels of cache. Some is shared between instructions and data, some is dedicated. The L3 cache is shared across all cores and hyperthreads, whereas the L1 cache is only shared between two hyperthreads. The way to ensure that the transmit and receive applications always uses the same cache is to 'lock' the applications to specific cores. For this we use the Linux command `taskset`.

This prevents the application processes to be moved to other cores and thereby causing interrupts in the data processing, but it does not prevent other processes to be swapped onto the same core.

3.4 Timers

The transmitter and receiver applications for this investigation periodically prints out the measured data speed, PER and other parameters. Initially the standard C++ chrono class timer was used (version: libstdc++.so.6). But profiling showed that significant time was spent here, enough to affect the measurements at high loads. Instead we decided to use the CPU's hardware based Time Stamp Counter (TSC). TSC is a 64 bit counter running at CPU clock frequency. Since processor speeds are subject to throttling, the TSC cannot be directly relied upon to measure time. In this investigation time checking is a two-step process: First we estimate when it is time to do the periodic update based on the inaccurate TSC value. Then we use the more expensive C++ chrono functions to calculate the elapsed time used in the rate calculations. An example of this is shown in the source code which is publicly available. See Section A for instructions on how to obtain the source code.

4 Testbed for the experiments

The experimental configuration is shown in Figure 2. It consists of two hosts, one acting as a UDP data generator and the other as a UDP receiver. The hosts are HPE ProLiant DL360 Gen9 servers connected to a 10 Gb/s Ethernet switch using short (2 m) single mode fiber cables. The switch is a HP E5406 switch equipped with a J9538A 8-port SFP+ module. The server specifications are shown in table 2. Except for processor internals the servers are equipped with identical hardware.

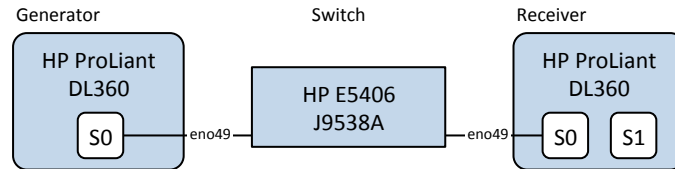


Figure 2. Experimental setup.

Table 2. Hardware components for the testbed

Motherboard	HPE ProLiant DL360 Gen9
Processor type (receiver)	Two 10-core Intel Xeon E5-2650v3 CPU @ 2.30GHz
Processor type (generator)	One 6-core Intel Xeon E5-2620v3 CPU @ 2.40GHz
RAM	64 GB (DDR4) - 4 x 16 GB DIMM - 2133MHz
NIC	dual port Broadcom NetXtreme II BCM57810 10 Gigabit Ethernet
Hard Disk	Internal SSD drive (120GB) for local installation of CentOS 7.1.1503
Linux kernel	3.10.0-229.7.2.el7.x86_64

The data generator is a small C++ program using BSD socket, specifically the `sendto()` system call for transmission of UDP data. The data receiver is based on a DAQ and event formation system developed at ESS as a prototype. The system, named the Event Formation Unit (EFU), supports loadable processing pipelines. A special UDP 'instrument' pipeline was created for the purpose of these tests. Both the generator and receiver uses `setsockopt()` to adjust transmit and receive buffer sizes. Sequence numbers are embedded in the user payload by the transmitter allowing the receiver to detect packet loss and hence to calculate packet error ratios. Both the transmitting and receiving applications were locked to a specific processor core using the `taskset` command and `pthread_setaffinity_np()` function. The measured user payload data-rates were calculated using a combination of fast timestamp counters and microsecond counters from the C++ `chrono` class. Care was taken not to run other programs that might adversely affect performance while performing the experiments. CPU usages were calculated from the `/proc/stat` pseudofile as also used in [10].

A measurement series typically consisted of the following steps:

1. Start receiver
2. Start transmitter with specified packet size
3. Record packet error ratios (PER) and data rates
4. Stop transmitter and receiver after 400 GB

The above steps were then repeated for measurements of CPU usage using `/proc/stat` averaged over 10 second intervals.

A series of measurements of speed, packet error ratios and CPU usage were made as a function of user data size for reasons discussed in Section 3.1.

4.1 Experimental limitations

The current experiments are subject to some limitations. We do not however believe that these pose any significant problems in the evaluation of the results. The main limitations are described below.

Multi user issues: The servers used for the tests are multi user systems in a shared integration laboratory. Care was taken to ensure that other users were not running applications at the same time to avoid competition for CPU, memory and network resources. However a number of standard demon processes were running in the background, some of which triggers the transmission of data and some of which are triggered by packet reception.

Measuring affects performance: Several configuration, performance and debugging tools need access to kernel or driver data structures. Examples we encountered are `netstat`, `ethtool` and `dropwatch`. However the use of these tools can cause additional packet drops when running at high system loads. These tools were not run while measuring packet losses.

Packet reordering: The test application is unable to detect misordered packets. Packet reordering however is highly unlikely in the current setup, but would be falsely reported as packet loss.

Packet checksum errors: The NICs perform checksums of Ethernet and IP in hardware. Thus packets with wrong checksums will not be delivered to the application and subsequently be falsely reported as packet loss. For the purpose of this study this is the desired behavior.

5 Performance

The performance results covers user data speed, packet error ratios and CPU load. These topics will be covered in the following sections.

5.1 Data Speed

The result of the measurements of achievable user data speeds is shown in Figure 3 (a). The figure shows both the measured and the theoretical maximum speed. For packets with user data sizes larger than 2000 bytes the achieved rates match the theoretical maximum. However at smaller data sizes the performance gap increases rapidly. It is clear that either the transmitter or the receiver is unable to cope with the increasing load. This is mainly due to the higher packet arrival rates occurring at smaller packet sizes. The higher rates increases the per-packet overhead and also the number of interrupts and system calls. At the maximum data size of 8972 bytes the CPU load on the receiver was 20%.

5.2 Packet error ratios

The achieved packet error ratios in this experiment are shown in Figure 3 (b), which also shows the corresponding values obtained using the default system parameters. The raw measurements for the achieved values are listed in Table 3. It is observed that the packet error ratios depends on the size of transmitted data. This dependency is mainly caused by the per-packet overhead introduced by increasing packet rates with decreasing size. The onset of packet loss coincides with the onset of deviation of observed speed from the theoretical maximum speed suggesting a common cause. No packet loss was observed for data larger than 2200 bytes. When packet loss sets in at lower data sizes, the performance degrades rapidly: In the region from 2000 to 1700 bytes the PER increases by more than four orders of magnitude from $1.3 \cdot 10^{-6}$ to $7.1 \cdot 10^{-2}$.

Table 3. Packet error ratios as function of user data size

size [B]	64	128	256	472	772	1000	1472	1700
PER	$4.0 \cdot 10^{-1}$	$4.0 \cdot 10^{-1}$	$4.1 \cdot 10^{-1}$	$3.9 \cdot 10^{-1}$	$3.8 \cdot 10^{-1}$	$3.8 \cdot 10^{-1}$	$2.0 \cdot 10^{-1}$	$7.1 \cdot 10^{-2}$
size [B]	1800	1900	2000	2200	2972	4472	5972	8972
PER	$3.2 \cdot 10^{-3}$	$6.1 \cdot 10^{-6}$	$1.3 \cdot 10^{-6}$	0	0	0	0	0

5.3 CPU load

The CPU load as a function of user data size is shown in Figure 3 (c). The observation for both transmitter and receiver is that the CPU load increases with decreasing user data size. When the transmitter reaches 100% the receiver is slightly less busy at 84%. There is a clear cut-off value corresponding to packet loss and deviations from theoretical maximum speed around user data sizes of 2000 bytes. The measured CPU loads indicate that the transmission is the bottle neck at small data sizes (high packet rates), and that most CPU cycles are spent as System load as also reported by [10]. But the comparisons differ both qualitatively and quantitatively upon closer scrutiny. For example in this study we find the total CPU load for the receiver (system + user) to be 20% for user data sizes of 8972 bytes. This is much lower than reported earlier. On the other hand we observe a sharp increase CPU usage in soft IRQ from 0% to 100% over a narrow region which was not observed previously. We also observe a local minimum in Tx CPU load around 2000 bytes followed by a rapid increase at lower data sizes.

6 Conclusion

Measurements of data rates and packet error ratios for UDP based communications at 10 Gb/s have been presented. The data rates were achieved using standard hardware and software. No modifications were made to the kernel network stack but some standard Linux commands were used to optimise the behaviour of the system. The main change was increasing network buffers for UDP communications from a small default value of 212 kB to 12 MB. In addition packet error ratios were measured. The measurements shows that it is possible to achieve zero packet error ratios at 10 Gb/s, but that this requires the use of large Ethernet packets (jumbo frames), preferably as large as 9018 bytes. Thus the experiments have shown that it is feasible to create a reliable UDP based data acquisition system supporting readout data at 10 Gb/s.

This study supplements independent measurements done earlier [10] and reveals differences in performance across different platforms. The observed differences are likely to be caused by differences in CPU generations, Ethernet NIC capabilities and Linux kernel versions. These differences were not the focus of our study and have not been investigated further. But they do indicate that some performance numbers are difficult to compare directly across different setups. They also provide a strong hint to DAQ developers: When upgrading hardware or kernel versions in a (Linux based) DAQ system, performance tests should be done to ensure that specifications are still met.

There are several ways to improve performance to achieve 10 Gb/s with smaller packet sizes, but the complexity increases. For example it is possible to send and receive multiple messages using a single system call such as `sendmmsg()` and `recvmsg()` which will reduce the number of system calls and should improve performance. It is also possible to use multiple cores for the receiver instead of only one as we did in this test. This adds some complexity that has to handle distributing packets across cores in case it cannot be done automatically. One method for automatic load distribution is to use Receive Side Scaling (RSS). However this requires the transmitter to use several different source ports in the UDP packet when transmitting instead of one currently used. This may require changes to the readout system. It is also possible to move network processing

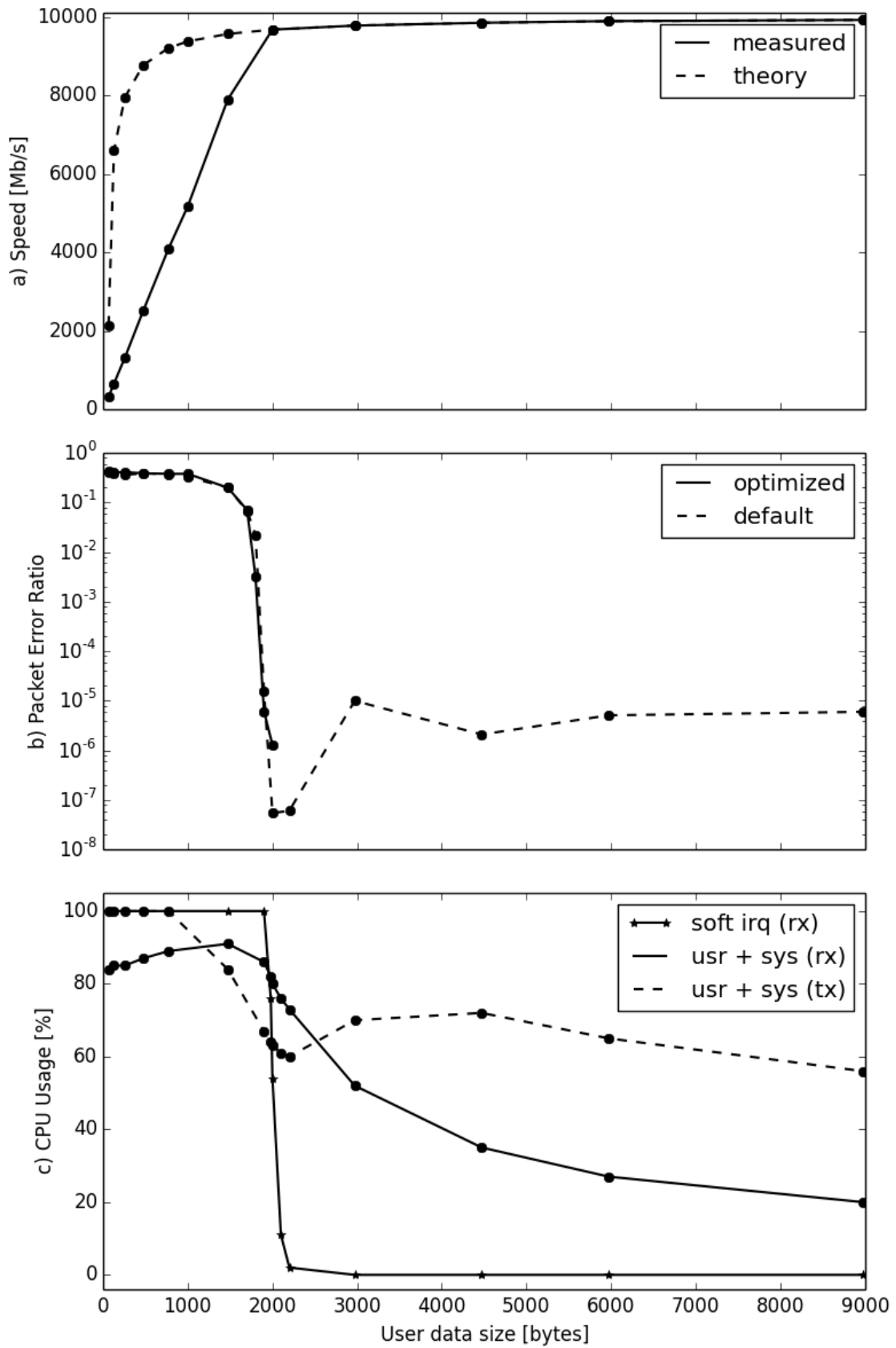


Figure 3. Performance measurements. a) User data speed. b) Packet Error Ratio. c) CPU Load. Note that for the optimised values PER is zero for user data larger than or equal to 2200 bytes (solid line).

away from the kernel and into user space avoiding context switches, and to change from interrupt driven reception to polling. These approaches are used in the Intel Data Plane Development Kit (DPDK) software packet processing framework [17] with impressive performance.

A Source code

The software for this project is released under a BSD license and is freely available at GitHub <https://github.com/mortenjc/udpperf>. To build the programs used for these experiments complete the following steps. To build and start the transmitter and receivers:

```
> git clone https://github.com/mortenjc/udpperf
> cd udpperf
> cmake -DCMAKE_BUILD_TYPE=Release ..
> make
> taskset -c coreid ./udptx -i ipaddress
> taskset -c coreid ./udprx
```

The programs have been demonstrated to build and run on macOS (High Sierra, Catalina), Ubuntu 16 and CentOS 7.1, however the `taskset` command is specific to Linux.

B System configuration

The following commands were used (performed as superuser) to change the system parameters on CentOS. Some of these have slightly different names on Ubuntu and generally are not available on macOS. The examples below modifies network interface `eno49`. This should be changed to match the name of the interface on the actual system.

```
> sysctl -w net.core.rmem_max=12582912
> sysctl -w net.core.wmem_max=12582912
> sysctl -w net.core.netdev_max_backlog=5000
> ifconfig eno49 mtu 9000 txqueuelen 10000 up
```

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