Communication Performance Evaluation of Bilateral Teleoperation Systems Implemented over WLAN

Piroska Haller[1], Tamás Vajda[2], Zoltán Szánto[3], Lőrinc Márton[2]

[1] Dept. of Informatics, 'Petru Maior' University, Nicolae Iorga 1, Tg. Mures, 540088, Romania, Email: phaller@upm.ro

[2] Dept. of Electrical Engineering, Sapientia Hungarian University of Transylvania, Sos. Sighisoarei 1C, Tg. Mures/Corunca, 547367, Romania, Email: martonl@ms.sapientia.ro, vajdat@ms.sapientia.ro

[3]Fac. of Automation and Computer Science, Technical University of Cluj Napoca, G. Baritiu nr. 26-28, Cluj Napoca, 400027, Romania, Email: zoltan.szanto@cs.utcluj.ro

Abstract: This study focuses on experimental measurements based performance evaluation of peer-to-peer applications, developed for supporting bilateral teleoperation over wireless local area networks (WLANs). Firstly, the characteristics of bilateral teleoperation systems are presented from communication point of view. The considerations and possible problems that should be taken into consideration during the design of the communication module of bilateral teleoperation software, implemented over WLAN, are also enumerated. Detailed experimental investigations were performed to show the effect of the WLAN communication medium on communication performances.

Keywords: IP Networks, Jitter, Networked control systems, Teleoperators, Wireless LAN

1 Introduction

Bilateral teleoperation systems allow the execution of tasks in environments that are dangerous or hardly accessible by humans. The working principle of these systems can be summarized as follows: the human operator generated force initiates the motion of a master robot. The motion of the master robot (position, velocity signals) is transmitted through a communication medium to the slave robot that tracks the motion of the master. During the task execution, the slave robot may be in contact with its environment. The slave transmits back to the

master the reaction forces from the task being performed, allowing the human operator to sense the environmental reaction forces that occur during the task execution.

Beside the force and position signals, the human operator should also follow the motion of the slave robot through video information. The captured video information on the slave side is sent to the master side in order to support the actions and decisions of the human operator.

In this view in a teleoperation system the following communication channels are necessary (see Fig. 1):

- Position and velocity signal channel (PCh) from the master side to the slave side
- Force signal channel (FCh) from the slave side to the master side
- Video signal channel (VCh) from the slave side to the master side

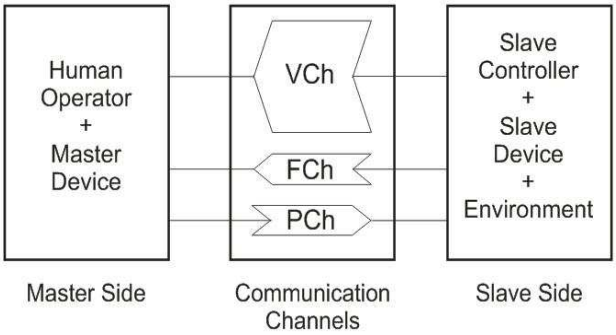

Figure 1
Communication channels in a bilateral teleoperation system

The sent data packet sizes in the first two channels are small (under 1 Kbyte). However, due to control requirements, the fast, periodical and timely delivery of the data packets is critical. The sending period of the data packet is in order of milliseconds. Since the signals sent through these channels are the inputs of discrete time control loops, which work with constant control period, the periodicity of the received signals through these channels is also important.

In the video channel the size of the packets are larger (in order of 10 Kbyte). The requirements here are related to the quality of the video transmission: such video frame sending rate and video frame size should be achieved that support the safe and reliable actions of the human operator.

From control engineering point of view the major challenge in the Internet based bilateral teleoperation is to guarantee the stability in any operating condition and meanwhile to preserve a good transparency of the system (precise position tracking and reliable force reflection) between the human operator and the

environment for a wide range of data transmission quality [1]. Various approaches were proposed to guarantee the stability of bilateral teleoperation systems. In the study [2] a communication strategy for teleoperation systems is described, according to which the measurements are sent over the communication channels only if the difference between the current measurement and the most recently sent value exceeds a certain threshold. The wave variable based teleoperation applies a transformation on the force and velocity signals to obtain the wave signals that are transmitted through the communication channel. In [3] it was shown that, by applying the wave transformation, the passivity of the teleoperation system can be guaranteed in the presence of time varying delays if the rate of the delay in the communication channel is upper-bounded. The problem of bilateral teleoperation was analyzed in robotic systems with constant communication delay in [4] and [5] respectively. It was shown that PD type controllers implemented both on the master and the slave side can ensure the passivity of the teleoperation systems.

The time domain passivity approach is a popular framework for guaranteeing the stability of teleoperation systems. The method is based on observing the energy of the teleoperator using a so-called Passivity Observer. When the observer shows that the teleoperation system looses its passivity, a Passivity Controller is switched on to dissipate the extra energy excess of the system. The method was originally introduced for haptic interfaces to achieve a stable interaction with virtual environments [6]. The applicability of the method for bilateral teleoperation systems was demonstrated in the work [7]. In the study [8] a method was proposed to extend the concept for multi-DOF robotic systems. In [9] the time domain passivity framework was applied for such cases when there is a considerable communication delay between the master and the slave.

The papers enumerated above mainly focus on the controller design in the presence of time varying communication delay without taking into consideration the characteristics of the communication channels. Moreover, only a few studies take into consideration the effect of the video channel on the performance of the teleoperation system. For example, in the paper [10] the influence of the delays in the video channel and the force and position channels on the performance of human operator was analyzed.

The proposed transport and application layer protocols for networked control systems are designed for wired local area networks or Internet, where the packet loss results from congestion. The paper [11] presents a survey of the above protocols.

The increased flexibility, mobility and reduced installation cost made the use of wireless local area networks (WLANs) widespread but they are not designed for real-time control applications. The requirements related to the periodic data transfer and data frames with large size (such as video data) simultaneously can not be satisfied with the existing protocols. The problems identified in the wired networks as delay variation due to network overload are aggravated by the packet

losses of the burst wireless channels and interference. The proposed bandwidth and congestion control algorithms in wired networks are based on explicit congestion notification received from the routers when the queue is full, or by the lack of the acknowledge packet (see e.g. [12-13]). As the packet loss characteristics of WLANs greatly differ from their wired counterparts, these protocols should be adapted for bilateral teleoperation over WLAN to obtain the optimal performance, based on measurements in real, noisy environment.

Today two different research directions in the use of wireless technologies in real time application with periodic traffic can be identified: Wireless Network Control Systems and multimedia communication over WLAN.

The studies related to Wireless Networked Control Systems mainly focus on the control system stability analysis under variable delay condition and in the presence of packet loss. The interactions between the network and the control system are considered and general networking and control co-design solutions are proposed for wireless control applications [14]. To achieve real-time requirements for networked control systems, some authors propose cross-layer designs, featuring application-layer admission control [15-16]. Mathematical models were developed to describe the dynamics of DCF based on MAC layer measurements and were validated using NS2 simulation environment [17], [18].

The analytical modeling of the MAC delay for IEEE 802.11-based networks receives more and more research attentions starting from the Markov chain model proposed by Bianchi [18]. The model uses the MAC layer parameters that are not available for the senders in the application layer in a multi-hop network. In [19] the authors propose a call admission control algorithm for real-time streaming traffic and a rate control algorithm to control the transmission rate of best effort traffic at the MAC layer, based on the channel busyness ratio. The throughput, the delay and delay variation in function of the channel busyness ratio was determined analytically based on MAC layer parameters. All admission decisions are made by a coordinating node, which can record the current number of admitted real-time flows and the bandwidth of the used channels in the network. The theoretical results and the simulations also show a severe performance degradation when the channel busyness ratio exceeds some threshold. It is important to find the critical turning point, with the maximum throughput, but this depends on the network characteristics. The proposed solution cannot be applied without the existence of a central coordinating node which periodically has to be updated with the state information of the whole system.

A major attention was paid to solve real-time multimedia communication on the existing wireless infrastructures. Most of the works estimate the throughput performance in IEEE 802.11 WLANs by comparing the UDP and TCP transport protocols and conclude that UDP traffic negatively affects the throughput of the TCP traffic, inducing severe unfairness [20]. For multimedia applications mixed wired-wireless network architectures can be used, extending the protocols with

priority information and increasing the number of simultaneous channels. Combining call admission and rate control for TCP stream and routing the packets in WLAN, the network can work at an optimal point [19]. Other authors propose new media access schemes to improve the capacity for Voice over IP (VoIP) traffic and to reduce end-to-end delay with mixed VoIP and data traffic [21]. A video transmission method was proposed in [22].

The paper [23] presents an experimental measurements based performance evaluation of a peer-to-peer application for supporting real-time force feedback operation in distributed haptic virtual environments. In the study we evaluated [24] communication strategies, using UDP and TCP, from the viewpoint of bilateral teleoperation systems. However, in the aforementioned works the video communication channel was not taken into consideration during performance analysis.

The goal of this work is to analyze the possibility of providing real-time services for bilateral teleoperation systems on existing wireless infrastructures using only application layer measurements without any information from network layer. As it was shown e.g. in the previous works [1, 3, 10, 23] the decreased transmission quality (increased delay and jitter) has a negative influence on the stability and performance of the teleoperation systems. Since almost all teleoperation systems should contain a video feedback channel, the influence of the video transmission (communication channel with large data frames and high transfer rate) on the transmission quality in the other two periodical channels (position channel and force channel respectively) has to be analyzed. Based on the delay variation measurements of the two opposite direction periodical data channels we try to identify the WLAN problems as congestion or collisions without the use of explicit acknowledge packets. The experiments were made in real environments using wirelessly connected routers in the presence of heavy independent traffics transferred by other wireless routers (campus network).

2 Considerations for the Design of Peer-to-Peer Architecture for Bilateral Teleoperation

To design a proper bilateral teleoperation application over wireless network it should be noted that WLANs suffer from various performance pathologies, such as low signal strength, significant noise, collision with hidden terminals from devices in the same WLAN or in nearby WLANs or congestion. These pathologies can result in throughput degradation, significant jitter and packet losses. In the teleoperation systems the position and force communication channels and the video communication channel have different delay requirements but it is important to preserve the synchronization between them. At the same time

the video transmission rate can affect the delay and the loss rate in the position and force channels, since they are using the same network layer.

For transport layer protocol the TCP or UDP can be chosen. TCP guarantees the reliability of the communication using transport layer acknowledges messages, but the control algorithms of this protocol degrade its real-time property. Since in the teleoperation applications large delays are not tolerable and the retransmission of lost packets at the application layer is not useful, the UDP protocol was chosen to transmit real-time traffic.

Using blocking function at application layer to send packets, it will either block the application until the packet was put into the sending queue or fails if a network error encountered. Accordingly, the application layer should not be allowed to send faster than the network layer. The queuing time at the application layer for UDP protocol depends on the processing time at different protocol layers, including the fragmentation and packet reassembling. Fragmenting means that the IP protocol splits the incoming packets if it is necessary, and sends them as a fragment with a predefined maximum size MTU (Maximum Transmission Unit). In addition, the end system must be able to put these fragments together to rebuild the original packet. The fragments are stored in the fragment cache, until either all fragments of a datagram have arrived and the packet was put into the input queue, or the maximum wait time for the fragments of a datagram has expired, which means that the datagram will be discarded.

By using two different endpoints at application layer (force/position channels and video channel), the packets in the force/position channels do not encounter delay at the application layer that could appear in a video channel due to the large video frame size. They only compete with the video fragment having the maximal size of MTU at the network layer.

The delay experienced by a packet consists of MAC delay (wireless access delay) and queuing time at every node traversed by the packet including the sending and receiving nodes. Most MAC protocols use a stop and go mechanism: they transmit the next packet of the queue only if the current packet has been properly acknowledged. The wireless access delay [25] is defined as the time interval between the time instant, when the packet becomes the first packet in the queue, and the time instant when the packet is successfully received and acknowledged. It captures the following delay components that a packet encounters at every wireless node: a) waiting until the channel becomes available, b) a variable back off window before its transmission, c) the transmission delay of potential retransmissions and d) certain constant delays defined and introduced by the applied protocol.

The newest 802.11 standards [26] describe two improvements for reducing MAC layer delay, but not all the wireless equipment implements this recommendation. One way is to provide differentiated channel access to data frames with different priorities. As a result, the high-priority real-time traffic gets smaller delay

variation, whereas the low-priority traffic encounters higher delay variation. However, precise timing constrains are not prescribed by the standards for the different priority classes. The second way is to reduce the delay introduced by channel contention. There is a provision that allows a station to transmit multiple MAC data frames consecutively after the channel become available as long as the whole transmission time does not exceed the transmission opportunity (TXOP) limit. It can considerably increase the throughput but at the same time the jitter for the periodic traffic also increases.

The packets can accumulate delay in intermediate nodes (access points and end-points) or eventually they can be dropped due to input queue overflow and packet loss can appear. In the queuing system the packet arrival process is determined by the aggregate traffic behavior of all applications that emit packets to the MAC layer. In the MAC layer a packet may be lost due to queue overflow or MAC collisions. It is known that the probability of collision in wireless transmissions is low as long as the network traffic is not heavy. In the presence of heavy traffic the collision probability is increasing and accordingly the delay and the jitter increase dramatically, the communication suffers from large loss rate. At the same time there is a packet loss caused by channel fading especially when the nodes are moving. Practically, it is difficult to distinguish the causes of the loss and, in most cases, by reducing the transfer rate the packet loss can be reduce.

For the reliability of the teleoperation the consecutive data frame loss in position and force channels should be maintained low and end to end delay near the constant sending period value. At the same time, to have good video perception quality, the throughput should be increased without achieving node saturation. This means that the network parameters at the application layer have to be monitored continuously.

Based on the system characteristics, namely the existence of the two periodic, fixed dimension data stream (the position and force channels), at every received packet the difference between two consecutive sending time and receiving time is calculated and it is compared to the constant sending period. The packet loss, duplicate packets and reverse order packets are also counted. In the same time for every received data frame the jitter and the data frame size was recorded. The blocking function usage at the application layer, the lack of the control algorithms in UDP protocol ensures, that the variation of network layer parameters (delay, jitter, packet loss) may be observed at application layer even if the packet traverse multiple access points. By analyzing the variation pattern of the parameters calculated above we can identify WLAN pathologies as congestion noise or low signal strength.

3 Implementation

The proposed software architecture for bilateral teleoperation uses UDP transport layer and creates two different endpoints for the position, force, and video communication channels. In the application layer the Boost.Asio cross-platform C++ library was used for network programming.

Over the position and force channels fixed size packets are sent periodically. According to the need of interoperability between heterogeneous, independently developed teleoperation systems [27, 11] the communication packet size, containing only position data, has an order of 100 bytes and they are sent with rates between 10Hz and 1kHz. Since for advanced control algorithms the velocity data also needs to be sent over the communication channels, the communication packet size is set to 500 bytes, which allows us to control higher order robotic systems. The packets are sent with 66 Hz.

The software captures the video frames with a desired rate by invoking asynchronously a function when the capture was finished. The video frame is split into smaller chunks that sizes are modifiable. The video chunk size and sending rate are modifiable to study their effects on the teleoperation application.

From the user point of view, the sending operation is implemented as a chain of asynchronous operations. A separate buffer is required for each endpoint to avoid the waiting for video frame send. By following the asynchronous approach, a new data frame can be send at each channel only when the pervious one was already sent. Since UDP transport layer is used, it only means that the data frame was moved into the sending queue.

The sent data frames or a part of it (MTU fragment) can be lost everywhere in the path from the source to the receiver, but the MAC layer retransmission was activated to reduce the number of loss packet at the application layer. At the application layer only complete data frames are received, hence the network layer bandwidth consumption should be higher than the obtained application layer transfer rate.

Using real timestamps instead of a simple counter gives the possibility to identify the correlation between the different communication problems (increasing delay, data frame loss). The computation does not need periodic clock synchronization between the sender and receiver and uses performance counters to obtain the desired time granularity.

Every sent data frame (position, force and video) contains an ID, unique for a given channel (*fid*) and the frame size (*framesize*). This ID is incremented for every sent data frame. The send timestamps (*s_time*) and the receive timestamps (*r_time*) are also recorded. The send timestamp is updated when the asynchronous send function is invoked for the current data frame. From the application point of view, the duration of the sending operation is related to the sending rate when the actual send rate

approaches the available bandwidth. Otherwise, the rate of putting the data into the sending queue is lower then the rate by which the data is delivered to the remote peer. When other data streams consume the whole available bandwidth in the current node or in any intermediate node, the difference between two send timestamps can be considerably larger than the data frame generating interval (i.e. the sending period - *T*) indicating communication problems.

The distributions of the sending timestamp's jitter (*s_jitter*) and receiving timestamp's jitter (*r_jitter*) for a periodic stream gives an indication about the network behavior. As some packets can be lost, the two jitter values were calculated based on the difference between two consecutive received packet divided by the number of expected packets (difference between the two frame IDs). The data frame ID is also used to measure the number of packet loss (*loss*) in each communication channel and to detect the number of the packet reordering, when the frame IDs are received in reverse order (*reverse*). As the large video frames are fragmented, it is hard to compare the jitter values for different frame sizes. It is why for the video channels the delay variation was also divided with the frame size.

Hence for every received force and position and video data frame the following parameters were computed:

$$
\begin{aligned}
&if\,(fid[i] > fid[i-1]) \\
&\quad s\_jitter[i] = \frac{s\_time[i] - s\_time[i-1]}{fid[i] - fid[i-1]} \\
&\quad r\_jitter[i] = \frac{r\_time[i] - r\_time[i-1]}{fid[i] - fid[i-1]} \\
&\quad v\_jitter[i] = \frac{r\_jitter[i]}{framesize[i]} \\
&\quad loss += fid[i] - fid[i-1] - 1 \\
&else \\
&\quad reverse++
\end{aligned}
$$

The statistics from the system are collected periodically containing the sending jitter, receiving jitter, the number of irregular packets (reverse order and loss) and the average sending rate.

4 Experimental Measurements

In this section experimental results are presented to study the effect of the following settings on the communication performances: chunk size, video frame generating period, distance between the endpoints and APs (signal strength) and the number of routers.

During the experiments, as endpoints (computers that control the master and the slave devices) two HP Pavilion g6 laptops were applied with Broadcom 4313GN type wireless network cards. As access points (AP) for wireless communication two TPLINK TL-WR941ND were used.

Both on the master and on the slave side Sensable Phantom Omni haptic devices generate periodically the position velocity signals on the master side and the force signals on the slave side. Microsoft USB camera device captures the video frames with a desired rate on the slave side.

The whole experiment was made in a real environment, where the same access points were used by another station for browsing and where the signals of other 15 independent routers interfere with the signals of the access points applied for the measurements.

The following experiments were performed:

M1 - 1AP, no VCh, static endpoints
M2 - 2AP, no VCh, static endpoints
M3 - 1AP, VCh with different chunk size, static endpoints
M4 - 2AP, VCh with different chunk size, static endpoints
M5 - 1AP, no VCh, moving endpoints
M6 - 1AP, with VCh and moving endpoints
M7 - 2AP, no VCh, moving endpoints
M8 - 2AP, with VCh and moving endpoints

During the *M1 experiment* the r_jitter and s_jitter were investigated on the sender and receiver side separately in the force channel (FCh) and position channel (PCh) without any other own generated network load (i.e. without video signal transmission). In M1 one AP was used. Fig. 2 presents the r_jitter for the PCh and FCh channels.

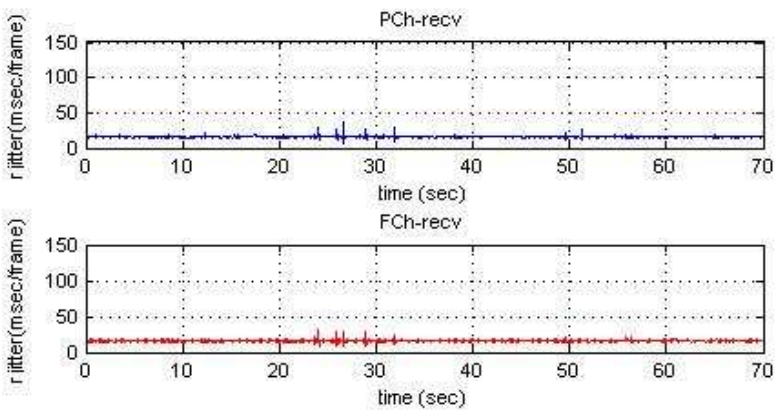

Figure 2
r_jitter observed in PCh and FCh (1AP and static device)

The timers that generate the sending rates are set to 15 milliseconds in both endpoints but the measured mean delay between two data frame is 15.59 ms. The s_jitter between the two data frame during sending is influenced by the operating system (Windows) some peaks can appear. Usually these peaks appear in pairs representing a higher and a lower s_jitter value between the data frames. If one data frame sends with lateness, the s_jitter for the next data frame will be shorter to maintain the sending period.

The effect of the s_jitter peak from the sending side cannot be observed in the received data's r_jitter. All of the communication channels, regardless of their direction, use the same access point in which they share the same queue. If the current sending packet encounters an interference or collision, the waiting time for every packet in the queue increases. In this situation, the received data has some peaks in all channels close in time. (see Fig. 2). Single peaks appear due the instantaneous disturbance such as interference and can be observed in a single channel if the waiting queue is empty.

The *M2 experiment* is similar to M1 experiment but it applies two APs connected through Wireless Distribution System (WDS) also with static devices. By comparing, the results from M1 and M2 experiments conclude that the use of one or two hop between devices does not influence significantly the behavior of the jitter.

In the *M3 experiment* we investigated the effects of the video channel (VCh) on the PCh and the FCh. During the experiment, the packet size was also modified (5Kbyte, 10Kbyte, 20Kbyte, 40Kbyte) to verify the effect of the data frame size on the communication performances. In the VCh 40Kbyte video frame is generated in every 30 ms. If the sending queue has more than 50 chunks from the previous frames, the newly captured video frame is discarded. The above parameters were chosen to avoid the saturation of the wireless AP and packet drop at the AP as a consequence of the full AP output queue. During this measurement, one AP was used.

The presence of VCh communication, parallel to PCh and FCh, increases the standard deviation in the PCh and FCh from 1.71 ms to 10.82 ms and the maximum deviation from 44 ms to 134.41 ms. Group of peaks in the jitter of both periodical channels are encountered when high amount of video packets are sent through the same AP and some of them are retransmitted. According to the measurements, if in the VCh the frames are split to 5 Kbyte chunks, the standard deviation will be 9.68 ms (the lowest value with VCh active) and the maximum deviation will be 80 ms. However, the maximum transfer rate that we can achieve is only 300 Kbyte/s. If the frames are split to 10 Kbyte, the standard deviation increases to 10.82 ms, the maximum deviation to 134.41 ms and the maximum transfer rate increases to 525 Kbyte/s. By increasing forward the packet size, the standard deviation and the maximum deviation will remain around the same value but the maximum transfer rate increases (see Table 1).

Table 1
Obtained maximum bandwidth using different chunk sizes and 1 AP

Frame Size	**Data transferred (Kbyte/sec)**
5 Kbyte	300
10 Kbyte	525
20 Kbyte	850
40 Kbyte	1270
80 Kbyte	1320

The Fig. 3 (*M3 experiment*) presents the r_jitter for the PCh, FCh and VCh data frames. The chunk size in the VCh was modified as follows: between 0-14 s - 5Kbyte, 14-33 s - 10Kbyte, 33-49 - 20Kbyte and 49-83 - 40Kbyte.

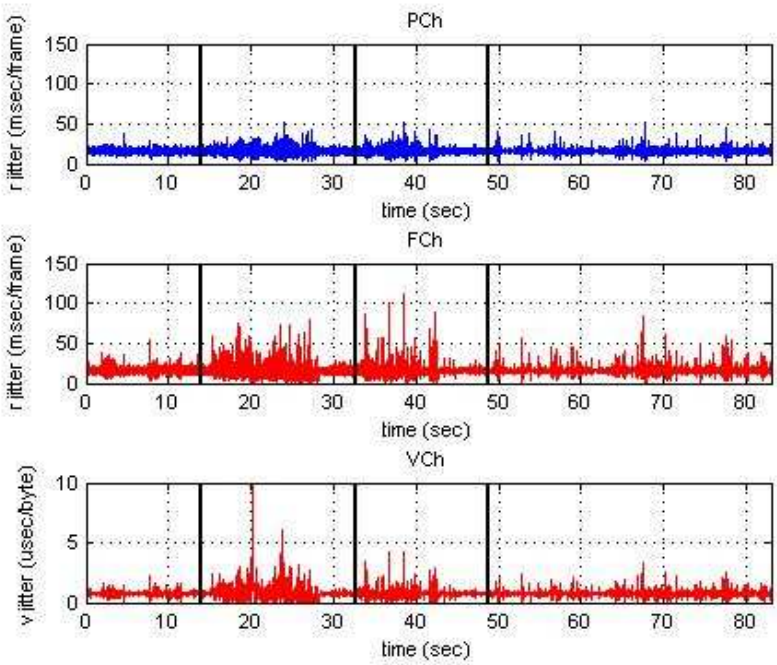

Figure 3
PCh, FCh and the VCh with variable chunk size using 1AP

The chunk size influences the communication even before sending. By breaking up the frame into small chunks, the Boost.Asio based communication module is forced to call the operating system frequently by introducing a delay even before

sending the data frame. When we increase the video chunk size over 20 Kbyte, the sending queue contains only the chunks of the current frame and it could be emptied before the next captured frame arrives. The system can transfer 1250 Kbyte without reaching the saturation.

During this experiment, the packet loss was 0.22% for PCh and FCh and 5 packets from 5000 were coming in reverse order in FCh. In the video channel, the packet loss was 1% but never more than 1 consecutive frame.

The *M4 experiment* is similar to M3 but we used 2 APs. By comparing the M4 and M3 experiments, we observed that a second hop does not break down the communication performance in the channels. The packet loss was 0.22% in PCh and 0.24% in FCh.

We have extended this experiment by sending 80 Kbyte video frames in every 30 ms. By modifying the chunk size from 5 Kbyte to 40 Kbyte, we obtain 1700 Kbyte/s transfer rate. In this case the system reached the saturation and the packet loss in the video channel increases considerably to 28%.

In the *experiment M5* moving devices were used to measure the effect of the signal strengths on the data transmission in PCh and FCh without VCh (Fig. 4). By using moving the endpoints away from the AP, the signal strength decreases continuously.

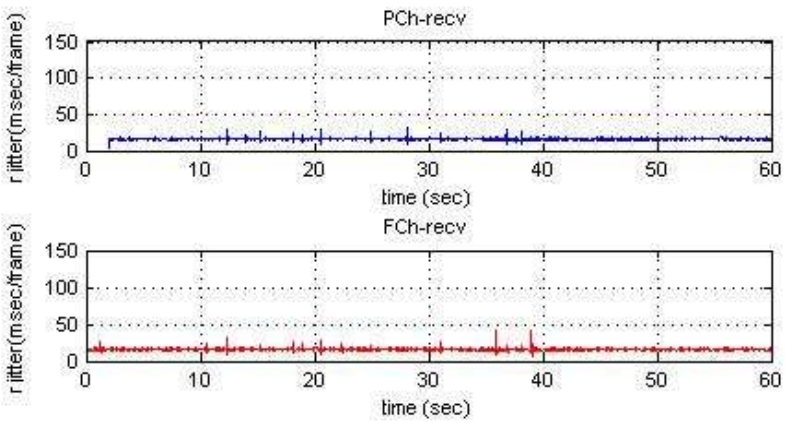

Figure 4

PCh and FCh, changing signal strength (moving device), 1AP

The test was performed using one AP and only the PCh and FCh channels were active. The r_jitter (mean value – 15.59ms, the standard deviation - 3 ms and the maximum deviation – 33.66 ms) is the same as with static endpoints. We haven't observed any significant influence of the strength variation on the r_jitter or

packet loss if the transfer rate is lower than a rate granted by the WLAN rate adaptation algorithm.

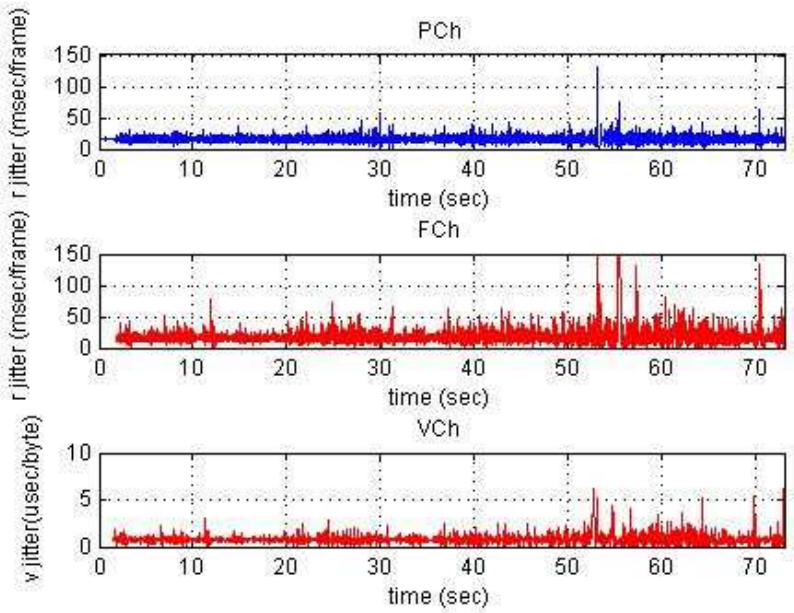

Figure 5
PCh, FCh and VCh, changing signal strength (moving device), 1AP

We repeated the same test during the *M6 experiment* with the VCh channel active (see Fig. 5). The 40 Kbyte size video frames were sent in every 30 ms and the frames were cut to 16 Kbyte size chunks. Reversed data frames appear if the signal strength decreases. The increase of the standard deviation in video frame stream's r_jitter is inverse proportional to the signal strength.

During the *M7 experiment* the wireless range was extended by connecting a second AP in WDS repeating mode. The behavior of the PCh and FCh channels were tested if the endpoints are moving away from each other. In the first part of the measurement we are in the range of the same AP, in the second part of the measurements the two devices are in the range of different APs. During the M7 experiment we measured the data frames r_jitter variation on the PCh and FCh without VCh (see Fig. 6). The r_jitter between consecutive data frames in PCh and FCh increase significantly during the AP reconnection (change). The packet loss was 0.24% in the PCh, 0.26% in the FCh channels and 0.2% of the packets were received in reverse order during the reconnection. This number is highly dependent on the reconnection time.

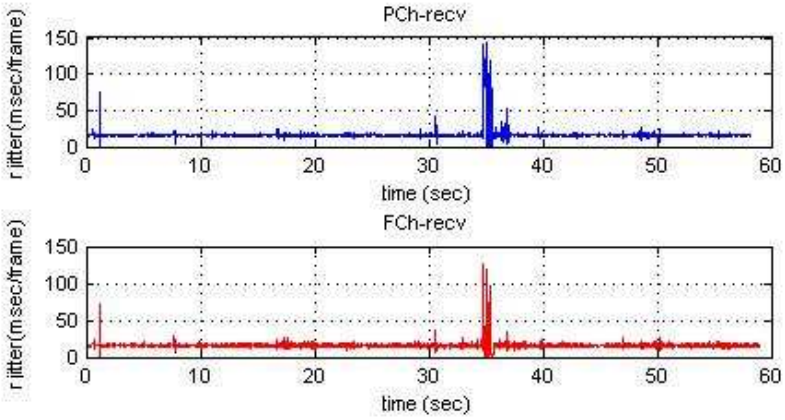

Figure 6
PCh and FCh, changing signal strength (moving device), 2 AP

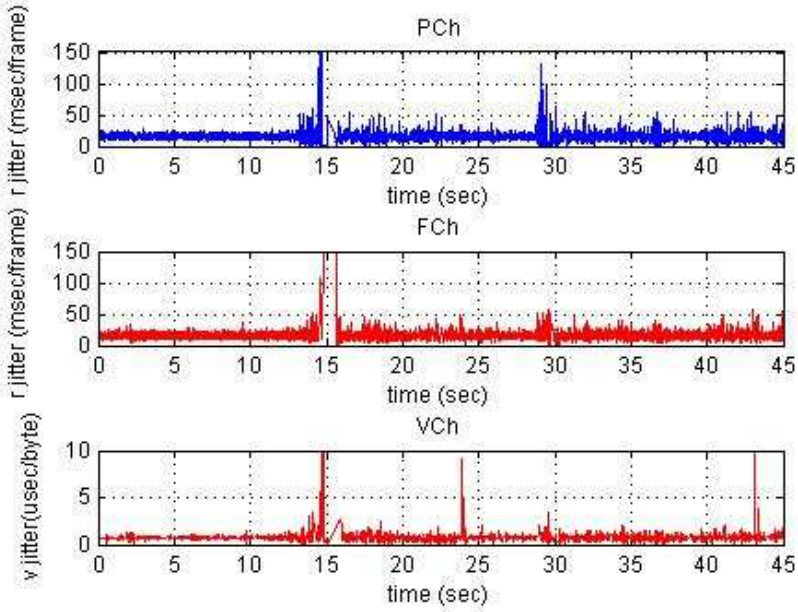

Figure 7
PCh, FCh and VCh, changing signal strength (moving device), 2 AP

The *M8 experiment* is similar to M7 but with VCh active (16Kbyte video frame chunks were sent in every 30 ms) (see Fig. 7). During this experiment 0.6% of the packets in PCh and 0.58% of the packets in FCh were lost during the AP change

(around the time instant 15 s) in both position and force data streams. By departing from the second AP, around the time instant 29 s the jitter values increased due to the decreasing WLAN signal strength. The instantaneous jitter variation happens due to the fact that, if the WLAN signal strength decreases, the WLAN APs halve automatically the available bandwidth for communication. After returning from this point in the direction of the AP, the signals strength increases and the network parameters are restored to the previous values.

The measurements related to the r_jitter (mean values, standard deviations, maximum values) are presented in the Table 2.

Table 2
Channel behavior (*r_jitter*) during different measurements

	PCh					**FCh**					**VCh**
	Mean (ms)	**Std** (ms)	**Max** (ms)	**Min** (ms)	**Loss %**	**Mean** (ms)	**Std** (ms)	**Max** (ms)	**Min** (ms)	**Loss %**	**Mean (μs/ byte)**
Static, 1AP	15.59	1.71	44.04	0.02	0	15.59	1.89	27.70	0.02	0	-
Static, 2AP	15.59	1.67	37.77	0.91	0	15.59	2.05	28.70	0.03	0	-
Static,1AP with video	15.63	10.82	134.41	0.42	0.22	15.58	11.14	77.17	1.52	0.22	0.76
Static, 2AP with video	15.60	5.83	80.93	0.05	0.22	15.60	5.96	46.62	0.83	0.24	0.76
Moving, 1AP	15.59	3.00	33.66	0.03	0	15.60	3.67	50.90	0.02	0	-
Moving, 1AP with video	15.71	17.30	278.92	0.01	0.22	15.64	10.67	187.70	0.02	0.22	0.78
Moving, 2AP	15.61	11.59	223.60	0.14	0.24	15.61	7.71	183.41	0.14	0.26	-
Moving, 2AP with video	15.82	19.57	856.81	0.03	0.6	15.78	19.44	920.54	0.90	0.58	0.80

The correlation coefficient was computed between the r_jitter signals in the PCh and FCh channels, as the covariance of the signals divided by the product of their standard deviations. As Table 3 shows, the measured jitter signals in the PCh and FCh channels correlate.

To obtain a fine grain control over asynchronous transfer mode we have modified the socket options (the send and receive buffer size) but based on measurement the size of the buffers did not influence the channel's behavior.

Conclusions and future work

We have proposed a method to evaluate the communication performance of bilateral teleoperation systems that include network parameters monitoring and adjustment. By using only application layer measurements without any administrative privileges or information from the wireless card, we can identify some communication related problems as congestion, collision and signal strength variation.

Table 3
Correlation coefficient during different measurements

	Correlation coefficient
Static, 1AP	0.3830
Static, 2AP	0.3511
Static,1AP with video	0.2533
Static, 2AP with video	0.2936
Moving, 1AP	0.2599
Moving, 1AP with video	0.2529
Moving, 2AP	0.5472
Moving, 2AP with video	0.5730

A large number of experiments were performed to select the appropriate settings for a given network architecture.

Based on the experiments we can conclude that sending position, velocity and force information with a constant 15ms period is possible under a light network load with a maximal deviation of 40 ms, even if the packets traverse two hops or the signal strength is variable due to the motion of the device.

In teleoperation systems good quality video feedback information is also important for the human operator. Good video transmission quality means high resolution and high frame rate, i.e. the video transfer rate should be increased, but without achieving saturation.

During the measurements, it was observed that in the presence of video communication in the position and force channels the change of the mean deviation of the receiving interval is not significant even if the network is congested, but the jitter and the maximal deviation increases considerably and seriously affects the control loop. By setting a proper video frame rate and chunk size, we can avoid the congestion and assure a good video quality.

WLAN based teleoperation systems have a promising future if the network parameters are continuously monitored and the video transfer rate can be controlled. In the future an application layer network control protocol will be designed that assures bounded jitter for the position and force channels, by modifying the video channel transfer rate. Using networking and control co-design solutions, high precision bilateral teleoperation system can be implemented.

Acknowledgement

This work was supported by a grant of the Romanian National Authority for Scientific National Authority for Scientific Research, CNCS – UEFISCDI, project number PN-II-RU-TE-2011-3-0005.

References

[1] P. Hokajem, M. Spong: Bilateral teleoperation: An historical survey, Automatica, vol. 42, 2006, pp. 2025 – 2057.

[2] N. Chopra, M. Spong, S. Hirche, M. Buss: Bilateral teleoperation over the internet: The time varying delay problem, in Proc. of American Control Conference, vol.1, 2003, pp. 155-160.

[3] N. Chopra, P. Beretsky, M. Spong: Bilateral teleoperation over unreliable communication networks, IEEE Transactions on Control Systems Technology, vol. 16, no. 2, 2008, pp. 304 – 313.

[4] E. Nuno, R. Ortega, N. Barabanov, L. Basanez: A globally stable PD controller for bilateral teleoperators, IEEE Transactions on Robotics, vol. 24, no. 3, 2008, pp. 753 – 758.

[5] D. Lee, M. Spong: Passive bilateral teleoperation with constant time delay, IEEE Transactions on Robotics, vol. 22, no. 2, 2006, pp. 269 – 281.

[6] B. Hannaford, J.-H. Ryu: Time-domain passivity control of haptic interfaces, IEEE Transactions on Robotics and Automation, vol. 18, no. 1, 2002, pp. 1 – 10.

[7] J.-H. Ryu, D.-S. Kwon, B. Hannaford: Stable teleoperation with time-domain passivity control, IEEE Transactions on Robotics and Automation, vol. 20, no. 2, 2004, pp. 365 – 373.

[8] C. Ott, J. Artigas, C. Preusche: Subspace-oriented energy distribution for the time domain passivity approach, in Proc. of IEEE/RSJ International Conference on Intelligent Robots and Systems, 2011, pp. 665 – 671.

[9] J.-H. Ryu, J. Artigas, C. Preusche: A passive bilateral control scheme for a teleoperator with time-varying communication delay, Mechatronics, vol. 20, 2010, pp. 812 – 823.

[10] Y.-S. Kim, J.-H. Ryu: Performance analysis of Teleoperation systems with different Haptic and Video time-delay, in Proc. of ICROS-SICE International Joint Conference, 2009, pp. 3371-3375.

[11] G. Kokkonis, K. Psannis, M. Roumeliotis, S. Kontogiannis, Y. Ishibashi: Evaluating Transport and Application Layer Protocols for Haptic Applications, in Proc of. IEEE International Symposium on Haptic Audio-Visual Environments and Games, 2012, pp. 66-71.

[12] R. Wirz: End-to-end congestion control protocols for remote programming of robots, using heterogeneous networks: A comparative analysis, Robotics and Autonomous Systems, vol. 56, no. 10, 2008, pp. 865 – 874.

[13] Y.C. Huang, C.H. Chu, E. H.-K. Wu: A novel congestion control mechanism on tfrc for streaming applications over wired-wireless

networks, in Proc. of 6th ACM workshop on Wireless multimedia networking and computing, 2011, pp. 9-16.

[14] M. Bjorkom, S. Netji, L. Eriksson, R. Jantti: Wireless control system design and co-simulation, Control Engineering Practice, vol. 19, 2011, pp. 1075–1086.

[15] L. Li, F. Wang: Control and communication synthesis in networked control systems, International Journal of Intelligent Control and Systems, vol. 13, 2008, pp. 81- 88.

[16] M. Trivellato, N. Benvenuto: Cross - layer design of networked control systems, in Proc. of IEEE Intern. Conf. on Communication, 2009, pp.1-5.

[17] G. Tian., Y-C. Tian: Modelling and performance evaluation of the IEEE 802.11 DCF for real-time control, Computer Networks, vol. 56, 2012, pp. 435–447.

[18] G. Bianchi: Performance analysis of the IEEE 802.11 distributed coordination function, IEEE Journal on Selected Areas in Communication, vol. 18, 2000, pp. 535 – 547.

[19] H. Zhai, X. Chen, Y. Fang: A call admission and rate control scheme for multimedia support over IEEE 802.11 wireless LANs, ACM Journal of Wireless Networks, vol. 12, 2006, pp. 451-463.

[20] R. Bruno, M. Conti, E. Gregori: Throughput analysis and measurements in IEEE 802.11 WLANs with TCP and UDP traffic flows, IEEE Trans. Mobile Computing, vol. 7, no. 2, 2008, pp. 171-186.

[21] T. Kawata, S. Shin, A. Forte, H. Schulzrinne: Using dynamic PCF to improve the capacity for VoIP traffic in IEEE 802.11 networks, in Proc. of IEEE Wireless Communications and Networking Conference, vol. 3, 2005, pp. 1589-1595.

[22] E. Puschita, P. Kantor, G. Manuliac, T. Palade, J. Bito: Enabling Frame-Based Adaptive Video Transmission in a Multilink Environment, Advances in Electrical and Computer Engineering vol. 12, no. 2, 2012, pp. 9-14.

[23] K. Yap, T.-H. Lee: Performance Evaluation of Wireless Networks for Supporting Real Time Collaborative Interactions in Distributed Haptic Virtual Environments, onlinepresent.org, vol. 7, 2012, pp. 194-201.

[24] N. Suzuki, S. Katsura: Investigation of transport layer protocols for wireless haptic communication system, in Proc. of IEEE/SICE International Symposium on System Integration, 2011, pp. 509-514.

[25] A Song, S. Wang, H. Zhao, J. Wei, M. Lai: An Accurate Delay Model with Consideration of Coupled Queues Effect for Nonsaturated IEEE 802.11 Wireless Networks, International Journal of Information Processing and Management, vol. 2, no. 3, 2011, pp. 52-64.

[26] IEEE P802.11n, Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification: Enhancements for Higher Throughput. 2009.

[27] H. H. King, B. Hannaford, K.W. Kwok, G. Z. Yang, P. Griffiths, A. Okamura: Plugfest 2009: global interoperability in telerobotics and telemedicine, in IEEE International Conference on Robotics and Automation (ICRA), 2010, pp. 1733-1738.