Performance evaluation of live video streaming service in 802.11b WLAN environment under different load conditions

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Abstract. Live video streaming service, which is common nowadays in the Internet, is supposed to be very challenging and demanding service in next-generation wireless networks. However, both limited quality of service (QoS) support and unstable quality of the air interface can restrain its wide deployment. In this paper we consider live video streaming over IEEE 802.11b wireless local area network (WLAN), which is claimed to be used as a part of layered infrastructure of next-generation mobile systems to provide coverage in highly populated areas . We performed our experiments under different signal-to-noise ratios (SNRs) and different competing TCP and UDP traffic volumes. The main conclusion of our paper is that despite a common belief live streaming multimedia services are not ready for wide implementation in hot-spot areas where both high traffic volume and relatively weak signal strength (less than 30 dB) may deny the service easily.

1 Introduction

Nowadays, a considerable part of research activities in telecommunications are switching towards development of next-generation IP-based wireless networks. The major motivation behind that is to introduce a common service platform and transport facilities for future composite mobile Internet.

In addition to broadband wireless access to the Internet, next-generation mobile systems should be able to provide quality of service (QoS) to various applications. Nowadays, new multimedia services attracted to these networks are striving hard towards commercial market. Both limited QoS support and unstable quality of the air interface can restrain their wide deployment.

The most crucial layers for QoS support in next-generation wireless networks are physical, data link and transport. In this paper we restrict our attention to transport layer only. The reason behind that is to judge whether the particular multimedia services may already be smoothly implemented on commercial basis over currently available wireless local area networks (WLANs), which are claimed to be a part of layered infrastructure of next-generation mobile systems. In this paper we consider most popular IEEE 802.11b WLAN.

In next-generation wireless networks there will be a clear separation between wireless part (Radio Access Network, RAN) and fixed one (Core Network, CN). RAN is used to hide all access specific peculiarities from the CN. Therefore, CN has a little impact on introduction of new RANs and can evolve independently.

A strict separation between RAN and CN will lead mobile systems to multi-access environment. This environment introduces an additional conceptual notion of next-generation wireless networks – an Always Best Connected (ABC) concept [1]. The ABC should allow users to choose the most suitable RAN at any instant of time during duration of a call. Particularly, this feature is claimed to be very attractive for users with complex mobility patterns. Technical implementation of ABC is to be based on intersystem (vertical) handover that should be implemented in a seamless way between any types of access networks [2, 3].

In addition to multi-access environment, it is becoming clear that next-generation networks will have a layered infrastructure with at least two hierarchical levels. In accordance with layered network infrastructure, there should be cells of different size (picocells, microcells, macrocells) each of which serves users in areas with different population densities. Layers with picocells or microcells are able to provide a high capacity with high bandwidth in hot-spot areas. They can serve slow mobility users with high traffic demands. It is assumed that in next-generation mobile systems this role will be assigned to WLANs. Therefore, we can state that WLANs and 3G RANs are not competitors, but complement each other to allow coverage in areas with different population densities.

Live video streaming service is widely spread nowadays in the Internet and QoS perceived by end-users varies in average from good up to excellent. For example, some of live streaming videos are already available in 300Kbps and can be streamed continuously without brake in service via well-provisioned parts of the Internet. It is supposed that live video streaming service will be very challenging and demanding in next-generation mobile systems.

In our study we have implemented series of performance tests of live video streaming service in 802.11b WLAN environment under different signal-to-noise ratios (SNR) and different load conditions. We concentrate our attention on tracing real traffic. Indeed, it is anticipated [4] that testing of real implementations can bring better understanding and new knowledge in the area.

To date only few studies on this topic are available. Authors in [5] considered the multimedia streaming service over IEEE 802.11b WLAN. They defined a number of SNR ranges and evaluated a perceived QoS provided to the user. In [6] the perceived QoS of monomedia applications under heterogenous wireless environment was evaluated. Among other conclusions, it can be found that bandwidth-greedy applications can already be implemented over current IEEE 802.11b networks. However, it may be the case that unstable nature of the wireless link along with high bandwidth competing traffic can deny the service easily. In this paper, we extend the previous results to the case of multimedia applications, different SNR ranges and different competing traffic loads.

The rest of the paper is organized as follows. Testing prerequisites are considered in Section 2. Testbed configuration is given in Section 3. Carried measurements and corresponding results are outlined in Section 4. Conclusions are drawn in last section.

2 Testing prerequisites

2.1 IEEE 802.11b WLAN

The IEEE 802.11 set of specifications are wireless standards that specify an 'over-the-air' interface between a wireless client and a base station (access point), as well as among wireless clients. The IEEE 802.11 specifications address both the physical and media access control (MAC) layers and are targeted to resolve compatibility issues between manufacturers of WLAN equipment.

Approved in 1997 by the IEEE 802 committee, IEEE 802.11 uses the 2.4-GHz microwave and defines two different (and mutually incompatible) methods of encoding: FHSS (Frequency Hopping Spread Spectrum) and DSSS (Direct Sequence Spread Spectrum). FHSS spreads the transmission across 75-MHz sub-channels, continuously skipping between them, while DSSS breaks the band into 14 overlapping 22-MHz channels and uses one at a time.

Two basic operating modes are defined: *infrastructure* and *ad-hoc*. Most dedicated hardware provides a basic service set that builds the wireless 'infrastructure'. It allows clients to roam between access points while roaming across routers is prohibited. The ad-hoc mode allows individual nodes to participate in a peer-to-peer communication without an access point.

The major problem with 802.11 was its relatively low throughput compared to wired networking and the mutual incompatibility of FHSS and DSSS equipment. In 1999, the IEEE 802 committee extended the specification, deciding to concentrate on DSSS. This extension, known as 802.11b, allowed more complicated encoding techniques which increased the throughput up to 5.5 Mbps.

2.2 Multimedia traffic

Multimedia applications are continuously growing in popularity. The availability of high-speed fixed access networks is the primary reason behind that. Today, it is necessary to support these services over wireless access networks.

Basically, real-time multimedia traffic consists of one or more media streams and can be characterized by strict delay requirements while can tolerate some losses. It is supposed that applications emerging from the Internet will become capable of defining the required QoS level soon. However, currently in almost all networks multimedia traffic is treated similar to ordinary best effort traffic, which do not often require strict delay guarantees. Therefore, it is crucial to predict the QoS degradation that may be experienced by multimedia applications over wireless access networks.

In our paper we consider live video streaming service, which consists of both video and audio medias. Note that from the user point of view the service can be described by two phases: prefetching phase and playing phase. While in prefetching phase the application stores data and then turns into playing phase. When application is in prefetching phase it uses all available bandwidth to prefetch data. When playing back, it restricts itself to the certain average bandwidth of combined stream (target rate).

3 Testbed

3.1 Client-server streaming implementation

In our testbed we used commercial implementation of client-server streaming service. Combination of RealNetworks' Helix server and RealNetworks' RealOne player was chosen. We used free distribution of Helix server available at [7]. It should be noted that compared to commercial distribution there are several limitations of free one. Particularly, the number of simultaneous connections should not exceed 10 while the bandwidth should be less than 1Mbps. However, all these limitations do not add bottlenecks in our testbed since it was not necessary to stream more than one video at a time.

Helix server can stream a lot of well known medias including both proprietary and standard-based ones. We have chosen real media streaming format because of the following reasons. At first, free distribution of Helix server allows all server-side capabilities only when real media format is used. Secondly, the real media format is currently very popular in the Internet because of relatively good quality of low bit rate videos. Additionally, when real media format is used free distribution of Helix server allows to serve clients with different bandwidth capabilities. Moreover, the bandwidth at which the client is served can also be changed dynamically during connection. In order to achieve that the video should be coded at different target rates each of which is specific for a certain bandwidth capability of the client.

In our testbed we used live streaming service. In accordance with it the server continuously listens specific ports for a connection requests. When the request arrives server sets up RTSP connection, adds client into connection pool and then begins streaming at the rate which is the most appropriate for requesting client. However, if the bandwidth capability of the client changes the server can adapt connection by increasing or decreasing the target rate of video. The bandwidth capability of the client is indicated in the 'BANDWIDTH' field of RTSP protocol. The 'live streaming' means that the server transmits video from that actual point in time when RTSP connection has already been established.

In our live video streaming service was emulated. The usage of real live streaming adds unnecessary complexity to our testbed i.e. requires to consider codec-specific peculiarities (type of source, compression and coding delays). It may lead our focus away from network-specific issues which are the main topics of our paper. Note that the Helix server allows to emulate all features of live streaming service using the 'slta' utility, which is the part of server distribution.

Protocol configuration used by Helix server and RealOne player is shown in Fig. 1. In order to facilitate live video streaming both TCP and UDP protocols are employed. RTSP over TCP (the solid line) is used at the connection establishment phase when the client poses the request on specific video. The actual streaming is performed over UDP (the dashed line). However, sometimes when it is not possible to use RTSP/UDP combination, RTSP/TCP is employed instead.

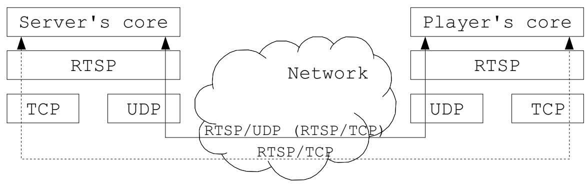

Fig. 1. Client-server protocol configuration.

3.2 Testbed configuration

Our testbed has been built in such way that we were able to test and compare performance of live video streaming service under different competing traffic loads and different signal-to-noise ratios (SNR).

Testbed configuration is presented in Fig. 2. In our environment we used several computers equipped with different operating systems (OS) and different access network devices. WLAN tests were carried out on a base of running implementation of 802.11b WLAN in campus area of Tampere University of Technology. To enable LAN access we used 100 Mbps Ethernet. Both WLAN and LAN are connected via 'broker-gw' edge router.

The mobile node called 'real-client' was IBM ThinkPad PIII laptop under Win2000 OS. It was equipped with exterior Cisco's Aironet 350 802.11b WLAN card. To ensure mobile node performance against OS-specific issues we validated all our tests with different mobile nodes. The fixed node called 'helix-server' was desktop PIV under Win2000 OS connected to 100 Mbps Ethernet LAN. The access point was Avaya's ORiNOCO range extender. To hide implementation-specific issues, all our test have been carried out with only one access point.

To generate competing traffic we used well-known 'iperf' client-server utility [8]. To maintain iperf server desktop PC PIII under Linux OS connected to 100 Mbps Ethernet LAN has been chosen. This node is called 'iperf-server' in Fig. 2. The mobile node called 'iperf-client' was Mac PowerBook G4 under Jaguar v.10.2 OS. It was equipped with interior Airport WLAN 802.11b adapter.

To evaluate performance of live steaming service the fragment of high motion pre-recoded video was chosen. Due to the fact that using the 'slta' utility we were able to simulate live streaming service, it was possible to carry out tests as long as required. To encode video to real media format Helix Producer v9.0 has been used. The resolution of video was set to 240×352 pixels, while the target rates were chosen to be 56Kbps, 150Kbps and 350Kbps.

Note that paths between both Helix server and iperf server and corresponding clients are stable and crosses only one router ('broker-gw'). We do not consider

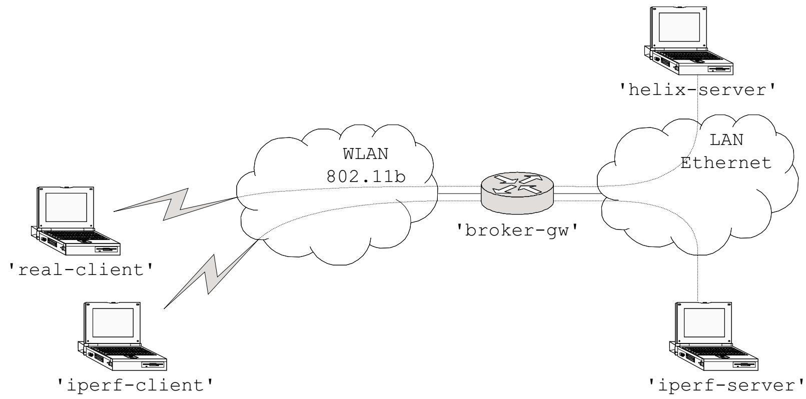

Fig. 2. Testbed configuration.

'broker-gw' as a bottleneck of our configuration since all tests were carried out when both LAN and WLAN were in totally unloaded conditions.

Additionally, we have to recall that both Helix server and iperf server were on the same Ethernet segment. Such condition cannot also be considered as the bottleneck, since the bandwidth of fixed LAN is substantially higher than that of 802.11b WLAN. Therefore, the only bottleneck in out testbed is the WLAN.

The bandwidth capability of the RealOne player was set to maximum allowable (10Mbps). The connection was assumed to be 'failed' after 30 seconds of unsuccessful attempts and the player was not allowed to prefetch data.

In this paper we propose to distinguish between five SNR levels. The ranges of SNR and corresponding user-friendly channel conditions were chosen as follows (note that the other partitioning of SNR is also possible): $\leq$ 10dB (very bad); 10 – 20dB (poor); 20 – 30dB (fair); 30 – 40dB (good); $\geq$ 40 (excellent).

Additionally to different SNR ranges, we performed our tests under different competing traffic volumes. We used both UDP and TCP competing traffic, which were generated by 'iperf-sever' and received by 'iperf-client'. Traffic characteristics are presented in Table 1. One can note that in certain cases the traffic volume increases the maximum theoretical throughput of the WLAN. It was done to be able to test the performance of live video streaming service in highly overloaded network conditions. The initial window size of TCP connections was chosen to be 60Kbps. We found out that with such choice TCP connections can potentially achieve the maximum throughput.

Table 1. Parameters of competing traffic patterns.

Type	Number streams	Target bandwidth, Mbps	Window, Kbps
UDP	4	2	–
UDP	4	1	–
TCP	4	–	60

To capture traffic and obtain statistics we used Ethereal software [9] package in conjunction with post processing Perl scripts.

3.3 End-to-end performance testing

Firstly, we had to explore performance characteristics of WLAN's access point. Several advanced UNIX-based utilities [8] have been used.

We obtained end-to-end performance parameters of the path between 'iperf-server' and 'iperf-client' using the 'iperf' utility. Performance parameters of 'iperf-server' – 'iperf-client' and 'helix-server' – 'real-client' paths are similar since 'iperf-client' and 'real-client' were on the same WLAN access point, while 'iperf-server' and 'helix-server' were on the same Ethernet LAN segment.

To obtain stable characteristics of WLAN we had to measure them on a wide time scale. To get values for each column of Table 2 we performed 120 minutes of testing. The following parameters were of particular interest: maximum throughput of WLAN, end-to-end round trip time and jitter.

Table 2. Performance parameters of unloaded WLAN under different SNR ranges.

Parameters	SNR				
	≤ 10	$10-20$	$20-30$	$30-40$	≥ 40
Max throughput, Mbps	0.22	0.35	3.84	3.86	3.93
Min RTT, ms	5.79	4.02	3.09	3.32	3.23
Avg. RTT, ms	12.41	8.20	6.31	4.84	3.76
Max RTT, ms	371.59	107.94	43.53	34.26	12.29
Jitter, ms	135.92	31.02	4.06	4.02	4.47

4 Results

In our testbed environment we evaluated the performance of live video streaming service in WLAN environment under different SNR levels and different competing traffic loads. Traffic was generated by user of 'real-client' node by posing the request on live streaming video from 'helix-server'.

The summary of statistics under different SNR ranges and unloaded network condition is presented in Table 3. Despite TCP's 3-way handshake procedure was successful and mean throughput was measured to be 0.22Mbps (Table 3), one can note that the RTSP connection has not been established when SNR was $\leq$ 10 dB. It stems from the fact that under $\leq$ 10dB condition the quality of the air interface was very unstable and, therefore, there were too much bandwidth renegotiation performed by RTSP. After the server had failed to establish connection in 30 seconds the connection request was rejected by the client.

Table 3. Summary of statistics under unloaded network condition and different SNR ranges.

Parameters			SNR		
	≤ 10	$10-20$	$20-30$	$30-40$	≥ 40
3-way handshake, ms	0,0063	0,0055	0,0034	0,0026	0,0019
Conn. est. phase, ms	–	5,109	4,790	4,605	1,356
Avg. packet size, bytes	–	1036,92	1063,34	1065,66	1069,33
Avg. packets per second	–	42,80	40,28	41,56	41,16
Avg. throughput, Mbps	–	0,355	0,343	0,354	0,352

We have to note that when SNR was in $10-20$dB range the RTSP connection was successfully established and it did not take substantially more time compared to $20-30$dB range. It should also be noticed from Table 3 that when SNR increases, the average size of IP packet gets larger. At the same time, to maintain the constant mean rate as required by our live streaming video the number of packets per second also increases. Note that the time taken by 3-way handshake procedure has often been claimed as a drawback for real-time services [10]. One can realize that this time is relatively small compared to the time of RTSP connection establishment. The latter one is at least one thousand times higher.

We found that the most critical point in live streaming service is RTSP connection establishment phase. Indeed, there were cases when the SNR was fluctuating a lot while the RTSP connection was setting up. It caused a lot of errors at the wireless link and as consequences very frequent bandwidth negotiations. Therefore, sometimes client's connection establishment timer had expired before the connection was set up.

Then, we provide the same tests given the different load conditions of the WLAN (Table 1). The results are given in Table 4 where in each cell the first value is for the case of UDP 4$\times$2Mbps competing traffic, the next one is for UDP 4$\times$1Mbps and the last one is for TCP 4$\times$60Kbps. The competing traffic adds additional (to that given by SNR) fluctuations to the bandwidth available at the path between client and server and makes the bottleneck at the WLAN.

Table 4. Summary of statistics under load conditions and different SNR ranges (UDP 4×2Mbps/UDP 4×1Mbps/TCP4×60Kbps).

Parameters	SNR	
	30 – 40	≥ 40
3-way handshake, ms	– /9,893/0,250	– /0,016/0,032
Conn. est. phase, ms	– /28,52/24,341	– /7,61/1,303
Avg. packet size, bytes	– / – / –	– /1037,78/1060,37
Avg. packets per second	– / – / –	– /39.95/40,51
Avg. throughput, Mbps	– / – / –	– /0,332/0,344

We have to note that the RTSP connections have not been established under any type of competing traffic when the SNR were in 00 – 10dB, 10 – 20dB and 20 – 30 ranges. These ranges are omitted in Table 4. The cause was in that TCP was not able to complete 3-way handshake procedure, which is necessary to establish RTSP connection.

The live video streaming service did not work when SNR was $\geq$ 40dB and the network was loaded by four UDP traffic source each of which targeted on 2Mbps. Particularly, the client permanently failed to perform TCP's 3-way handshake.

The duration of 3-way handshake procedure when SNR was in 30 – 40 range and the network was loaded by four 1Mbps UDP source was roughly ten times higher than that of the network in unloaded conditions (Table 3). At the same time duration of RTSP connection establishment phase was much greater. However, the perceived quality of live video streaming service was good and can be roughly compared to the perceived quality in the network in unloaded conditions.

Almost similar observations were made when the network was under load of four competing TCP connections each of which had 60Kbps initial window size. However, one can note that the duration of RTSP connection establishment phase was smaller compared to the previous case. It can be explained by behavior of competing TCP connections whose bandwidth changes dynamically depending on losses. Note that from the other side UDP connection tries to get as much as it needs. The quality of the picture was sometimes slightly deteriorated i.e. some frames were blurry and truncated. However, the picture quality was acceptable.

As was expected live streaming service did not operate when the networks was loaded by four UDP sources each of which tried to get 2Mbps. When the target bandwidth of each competing source has been decreased to 1Mbps Helix server and RealOne player have completed 3-way handshake and established RTSP connection. However, it took much longer time compared to unloaded network conditions. Additionally, due to high bandwidth fluctuations the server was not able to use live video streaming over UDP and actual streaming was performed over TCP. The perceived quality was unacceptably bad: there always were long pauses without picture and even without voice, most frames were corrupted. Despite of that the service operated without brake.

One can note that the durations of both 3-way handshake and connection establishment phase were smaller when the network was loaded by four TCP connections. However, even worse picture and voice quality was perceived. In this case the server was also not able to use RTSP over UDP and streaming was again performed over TCP.

The throughput graphs for each case are given in Fig. 3, Fig. 4 and Fig. 5.

5 Conclusions

Summarizing, we conclude that it is not possible to support multimedia services like live video streaming on commercial basis over current evolution of IEEE 802.11b WLAN. Despite a common belief live streaming services are not ready for wide implementation in hot-spot areas where both high traffic volume and relatively weak signal strength (less than 30 dB) may deny the service easily.

Our measurements have shown that live video streaming service cannot be successfully implemented over wireless medium. The usage of TCP at the connection establishment phase may easily deny the the service even the network in unloaded conditions. The live streaming service performs well in presence of any type of competing traffic, only when excellent channel conditions (greater than 40dB) are met. However, the perceived quality becomes unacceptable in presence of any type of competing traffic with $30 - 40$dB. High volume of competing traffic easily denies service when the signal strength is under 30dB.

The major problem is that the client cannot set up RTSP connection with the server. Frequent bandwidth fluctuations, caused by both SNR and competing traffic loads, stimulated numerous bandwidth negotiations, and therefore, connection establishment's timer often expires before RTSP connection sets up. However, these bandwidth fluctuations do not actually indicate that the streaming service fails due to scarce of the bandwidth, since the actual streaming is performed over bandwidth greedy UDP protocol. Additionally, in those cases when video streaming is performed over TCP the QoS becomes unacceptable.

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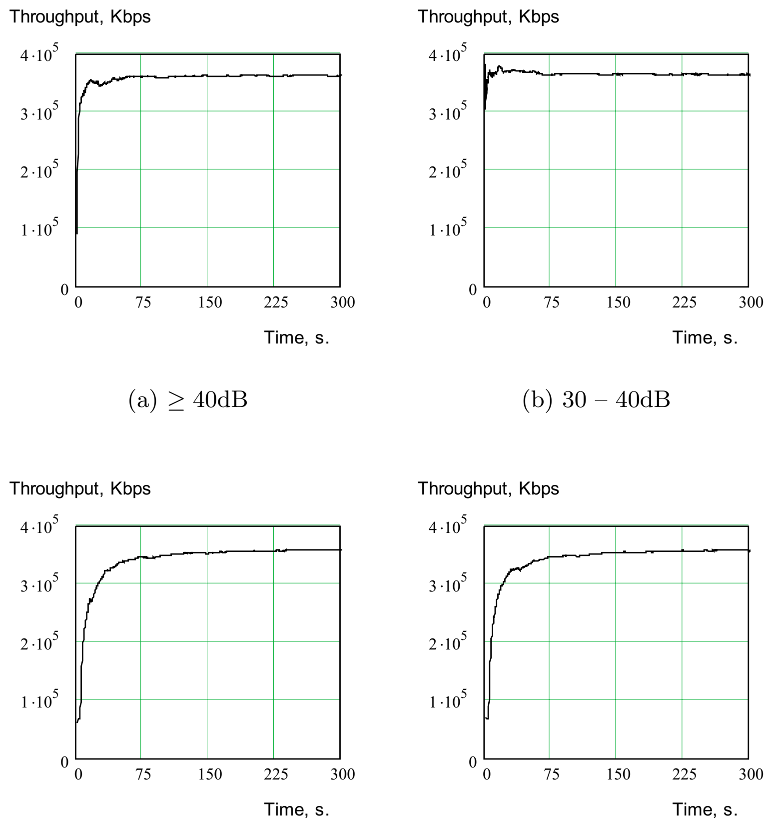
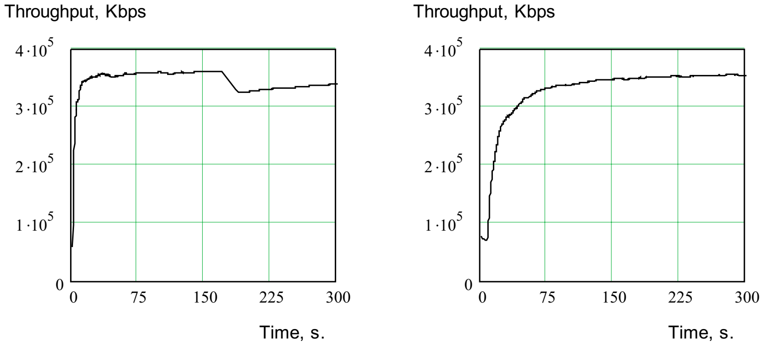


(a) $\geq$ 40dB (b) 30 – 40dB

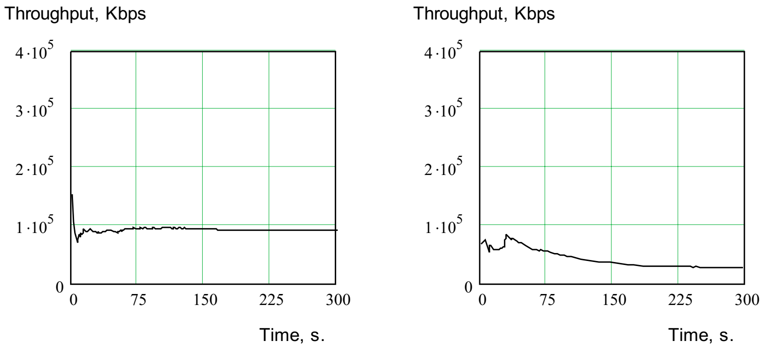
(c) 20 – 30dB (d) 10 – 20dB

Fig. 3. Throughput under different SNR rages and unloaded network condition.



(a) UDP 4×1Mbps (b) TCP 4×60Mbps

Fig. 4. Throughput under different load conditions and $\geq$ 40dB.



(a) UDP 4×1Mbps (b) TCP 4×60Mbps

Fig. 5. Throughput under different load conditions and 30 – 40dB.