OBJECTIVE ASSESSMENT OF IP VIDEO CALLS WITH ASTERISK

Lukas KAPICAK[1], *Pavel NEVLUD*[1], *Martin MIKULEC*[1], *Jaroslav ZDRALEK*[1]

[1]Department of Telecommunications, Faculty of Electrical Engineering and Computer Science, VSB-Technical University of Ostrava, 17. listopadu 15, 708 33 Ostrava-Poruba, Czech Republic

lukas.kapicak@vsb.cz, pavel.nevlud@vsb.cz, martin.mikulec@vsb.cz, jaroslav.zdralek@vsb.cz

Abstract. *The paper deals with an objective assessment of IP video calls transmission over GSM and UMTS networks. Video transmission is affected by many factors in mobile network. Among these factors belong packet loss, latency and transmission rate of the mobile network. Network properties were simulated by Simena network simulator. Our team have developed a unique technique for finding defects in video appearing in video calls. This technique is built on modified Asterisk SW PBX with enabled video recording and playback functions. Transmitted video files are compared with original video file by means of size of transmitted video file and in-video-defects. We are using MSU VQMT software for finding in video defects; more precisely we are using VQM method for comparing two video sources.*

Keywords

Asterisk, comparison video files, H.263, mobile networks, video calls.

1. Introduction

The members of our team participate on the INDECT project [1]. They are responsible for developing the IVAS (Interactive Video Audio System) system [2]. The IVAS system represents multimedia gateway between the INDECT portal and end-user equipment. End-user devices are mainly mobile devices with support for VoIP [3], [11] communication through data transmission via mobile networks. All multimedia functions are ensured by IVR (Interactive Voice Response) and IVVR (Interactive Voice Video Response) functionality [4].

The transmission medium is mostly represented by mobile networks. We tested all available mobile data transmission techniques but only networks with higher transmission rate than offer EDGE (Enhanced Data rates for GSM Evolution) have sufficient characteristics to transfer video calls over IP. Data transmission rates in mobile networks has limits in transmission speed, latency and packet loss. It is very important to find these limits for transmitted video content.

We were using Asterisk as a multimedia gateway. Asterisk provides basic functionalities such as recording and playback video files, generating IVR and IVVR menu etc. We decided to use recording and playback functionalities to assessment video files. We used two Asterisks servers for this purpose. First of them establishes video call and played pre-prepared video file and the second Asterisk records that video file.

We decided to simulate mobile networks characteristics. Transmission rate in mobile networks is very variable parameter and it is affected by many variable factors such as the number of connected users, weather, available network transmission rate etc. We used Simena network simulator [5] to simulate mobile network characteristics. The network characteristics came out from multiple measurements in tested mobile networks.

Saved video has to be analyzed with objective method. We have found project MSU VQMT (MSU Video Quality Measurement Tool) [8] specialized in comparison differences between original and encoded video file. MSU VQMT has many plugins and we have chosen plugin called VQM [9]. With this plugin, we were able to probe saved video files and compare it with original video file.

2. Testing Configuration

As was mentioned before we used Asterisk for establishing audio and video call. In the default configuration, Asterisk does not support video recording and playback so Asterisk needs to be upgraded with plugins which enable recording and playback video files. We implemented into Asterisk `app_mp4.c` plugin, which extended Asterisk for new functions – `mp4play` and `mp4save`.

These two plugins allowed us to record and play video files supported by an Asterisk and by mobile devices [6]. The `extensions.conf` needed modification to record video files. It was necessary to prepare two Asterisk servers. First Asterisk established video call and second Asterisk received the video call and then saved the video call file into predefined directory. First Asterisk server had to have appropriate configurations in `sip.conf` configuration file to forward video call to the second Asterisk server.

There was a network emulator Simena [5] located between Asterisk servers. Connection parameters were adjusted according to measured parameters in mobile networks. We measured CSD (Circuit Switched Data), HSCSD (High-Speed Circuit-Switched Data), GPRS (General Packet Radio Service), EDGE and UMTS (Universal Mobile Telecommunications System) parameters [6]. We did not measure HSPA (High Speed Packet Access) or HSPA+ network parameters.

Table one represents measured transmission speeds. We have chosen transmission rates about 214 kbps and 1990 kbps. Mobile network technologies such as CSD, HSCSD, GPRS and EDGE does not provide sufficient transmission rate for real time video transmissions. Encoded video file was pre-prepared with these parameters:

- video resolution 176x144 pixels,
- 25 frames per second,
- video codec H.263 with 304 kbps data rate.

Tab.1: Measured transmission speeds in different mobile networks [7].

Transmission technology				
	EDGE	**UMTS FDD**	**UMTS TDD**	**HSDPA R5**
Downlink [kbps]	126,57	123,19	214,77	1996,91
Uplink [kbps]	101,35	113,59	157,56	330,45
Latency [ms]	193	144	124	87

These video parameters depict that transmission speed 214 kbps will not fully cover needs of video transmission. We have chosen video data rate in case of comparison defects in picture between transmission speed 214 kbps and 1990 kbps. We set these transmission speeds in Simena network simulator and we were changing packet loss and latency. Testing architecture is displayed in the figure one.

Fig. 1: Testing architecture with Simena network emulator.

3. Video File Size Comparison

Our method allows us to save video call into predefined storage in Asterisk 2. Video has the same parameters as a video call so we can simply compare file size of these video files.

Table two and three depicts differences in video file size. It is noticeable that rising latency has not influenced on the video file size. On the other side, rising packet loss and transmission rate has influence on the video file size. Figure 3 and Fig. 4 depict decreasing character in dependence on ascending character of the packet loss.

Tab.2: Measured transmission speeds in different mobile networks [7].

H.263	**214 kbps**	**1990 kbps**
Latency [ms]	**Size [B]**	**Size [B]**
0	748112	991129
50	837544	991129
150	743340	991129
250	711935	991129
350	688161	991129
500	962987	991129

Tab.3: File size of saved video files – packet loss.

H.263	**214 kbps**	**1990 kbps**
Packet Loss [%]	**Size [B]**	**Size [B]**
5	793403	944177
15	672654	791849
25	757362	746104
35	656917	661522
45	560021	548463
55	379394	353694

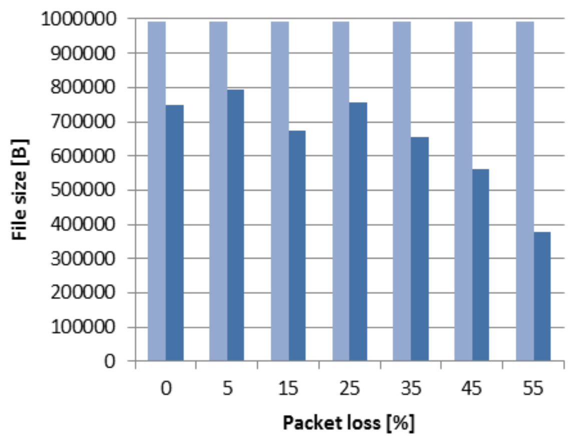

Fig. 2: Graph with packet loss with transmission rate about 214 kbps light blue – original file size, dark blue – transferred video file size).

It is noticeable to see differences between file size in figure two and three and in table two. When we analyzed packet loss between data rate 214 kbps and 1990 kbps we have found similar video file size in packet loss 25 %. Transmission rate was different in these two cases so it was necessary to analyze these two situations.

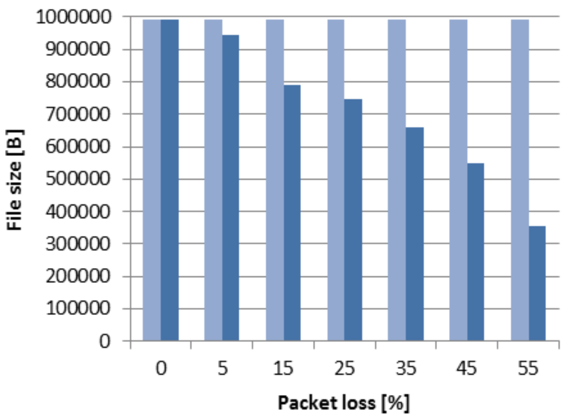

Fig. 3: Graph with packet loss with link rate 1990 kb/s (light blue – original file size, dark blue – transferred video file size).

4. Analysis of the Video Content

Saved video files should be analyzed by the objective method. We used project MSU Video Quality Measurement Tool [8]. This software allows us to compare one or more video files. It is designed for analyzing differences in encoded video files but we used this MSU VQMT for analyzing differences between original and transmitted video file. MSU VQMT contains different tools for analyzing video files.

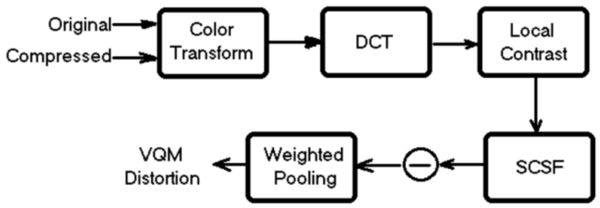

Fig. 4: Overview of Video Quality Measurement [9].

Fig. 5: Differences in the picture with different VQM – original/VQM = 15,5234.

We used DCT – based Video Quality Evaluation tool (VQM) [9]. This method is used for analyzing defects in compressed video files. DCT-based video quality metric (VQM) is based on Watson's proposal [10], which exploits the property of visual perception. This metric uses the existing DCT coefficients, so it only incurs slightly more computation overhead [9]. We had four sample groups with saved video files. Two sample groups were affected by latency in video transmission. First group was affected by low transmission rate too.

Table 4 and 5 show number of frames in saved video files. Number of frames in the original video file was 391 frames. The MSU VQMT has implemented technology (by extension) to optical comparison original video file with transmitted video file (saved video call file). With this extension, we were able to compare every single frame of the video file. Table 4 shows information about VQM values depending on packet loss. In the case of that packet loss has zero value, the VQM value is very low. We have found that the main differences in video quality are at the beginning of the video transmission and at the end of the video transmission. Average VQM value has not rising character. It is noticeable in Tab. 4.

Tab.4: VQM values depending on the packet loss.

H.263	214 kbps	214 kbps	1990 kbps	1990 kbps
Packet Loss (%)	**Frames**	**Avg. VQM**	**Frames**	**Avg. VQM**
0	255	0,02205	391	0,01354
5	297	12,87321	366	13,13988
15	266	13,45568	319	14,22181
25	297	14,83738	298	14,42438
35	260	14,28836	263	14,95820
45	225	15,11789	230	14,90110

With rising packet loss rises frame dropping too. From a statistical point of view, there are in-video defects spread out between totally dropped frames and between damaged frames.

Tab.5: VQM values depending on the transmission latency.

H.263	214 kbps	214 kbps	1990 kbps	1990 kbps
Latency (ms)	**Frames**	**Avg. VQM**	**Frames**	**Avg. VQM**
0	255	0,02205	391	0,01354
50	305	2,17959	391	0,01354
150	253	0,02214	391	0,01354
250	241	1,31047	391	0,01353
350	226	0,03398	391	0,01354
500	374	0,46453	391	0,01353

Table 5 depicts VQM values depending on the transmission latency. The VQM values are independent on the video latency. Low frame number is caused by low level transmission rate (214 kbps). In case of the situation, when transmission rate is enough to cover needs of video call, the VQM value is nearly zero. There are no differences between original video file and transmitted video file.

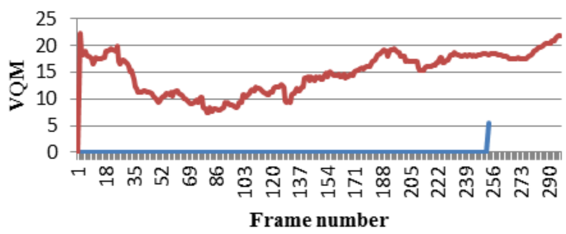

Fig. 6: Comparison graph with VQM values for 214 kbps data rate – transmission affected by latency 150 ms (blue) and 25 % packet loss (red).

Figure 6 displays differences in VQM value. The red graph shows very variable characteristics of the VQM value. The differences between original video file and transmitted are significant. On the other hand, the VQM transmission values are influenced by the latency nearby zero values. In some circumstances, the VQM value could reach a larger value but this situation can occur only at the beginning of video call transmission or at the end of the video call. Low transmission rate does not cause changing of VQM values. Low transmission changes only number of saved frames and it causes reducing of saved video file too.

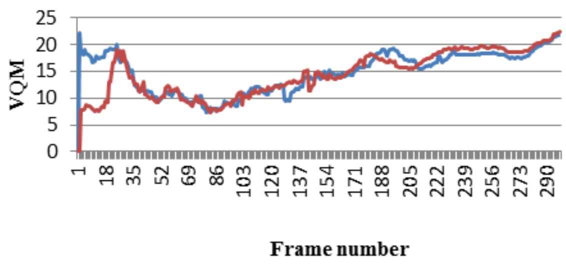

Fig. 7: Comparison graph with VQM values for 214 kbps (blue) and 1990 kbps (red) data rate – transmission affected by 25 % packet loss.

Figure 7 depicts VQM values character for transmission rate 214 kbps and 1990 kbps. The character of the curve is similar for both values (blue for 214 kbps and red for 1990 kbps). The highest VQM values are at the beginning and at the end of the video call.

5. Application

Results from these tests could be used for both video and performance benchmarking. It can describe server performance for a specific number of video calls. We can also describe transmission line performance and we can also define the number of successfully transmitted video calls at one time. Transmission errors could be divided into these categories:

- Video call file has the same size such as the original video file. Transmission line covers transmission demands of the video calls. Transmission delay caused by latency is not found.
- Video call has the smaller size compared with original video file and there are small VQM values. The transmission line does not cover demands of the video calls. Transmission delay caused by latency is not found.
- Video call has smaller size compared with original video file and there are high VQM values. All broadcasted packets do not achieve destination and the packets are lost during transmission. Video contains many in-video defects.

With these results, we can describe link properties and also define the cause of transmission errors. This approach allows to define network properties according to video calls characteristics.

6. Conclusion

Video transmission is very important part of the IVAS system. Many other services such as video call, online streaming or IVVR technology depend on video transmission. The IVAS system is designed for mobile end users, so it is necessary to count with mobile data transmission. End user devices are represented by smartphones or tablets.

We tested transmission of video calls in mobile networks. These networks have some limitations in transmission rate, latency and packet loss. Measured parameters have very variable pattern so we decided to simulate these parameters with Simena network emulator. Simena network emulator allowed us to configure virtual network parameters on the wired transmission line. Testing environment consisted of two Asterisk servers and Simena network emulator. Asterisk server was equipped by module which allowed record and play video files saved on the local storage. Video call was established through this simulated network by specific CLI commands.

The thirst tested parameter was the size of the saved video call. Video file size depends on two parameters – transmission rate and packet loss. High latency does not influence the size of the video file. We decided to investigate differences in video content. There are two ways how to analyse video content – subjective and objective analysis. An objective method has been chosen for our tests. MSU VQMT provides needed tool for video analysis. This software includes many plugins but we have chosen only VQM method. With this method, we were allowed to analyse differences and errors in saved video files. We were able to find specific errors and then we divided in-video errors into specific categories defined in this paper. According to these errors we described transmission line and found a solution to improve network ability to transfer video calls.

Testing network consisted of two Asterisk servers can be used for testing any IP transmission line with video calls. It is possible to establish benchmark and security [12] tests with appropriate configurations in Asterisk dial plan.

Results from the tests could be used for new tests. The solution is almost ready for system benchmarking. We can prepare scripts for establishing larger number of video calls in one moment. The line between two Asterisks could be wired or wireless. We are planning to test our solution in real mobile network too. We plan upgrade this solution with development of new Asterisk module which will combine technics comes from MSU VQMT. Testing video file will be then automatically tagged with error description.

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About Authors

Lukas KAPICAK received his M.S. degree in telecommunications from VSB-Technical University of Ostrava, Czech Republic in 2007. Since 2007 has been studying Ph.D. degree at the same university. His research is focused on wireless transmission and data flow analysis, simulation and optimization.

Pavel NEVLUD received his M.Sc. degree in telecommunication engineering from VSB-Technical University of Ostrava, Czech Republic in 1995. Since this year he has been holding position as an assistant professor at the Department of Telecommunications, VSB-Technical University of Ostrava. The topics of his research interests are communication technologies, networking and security.

Martin MIKULEC received the M.Sc. degree from VSB-Technical University of Ostrava, Faculty of Electrical Engineering and Computer Science in 2011. Currently, he is working toward the Ph.D. degree at the Department of Telecommunications, Faculty of Electrical Engineering and Computer Science. His research is focused on advanced services in IP telephony.

Jaroslav ZDRALEK holds position as an associate professor with Department of Telecommunications, VSB-Technical University of Ostrava, Czech Republic. He received his M.Sc. degree in Computer Science from Slovak Technical University of Bratislava, Slovakia in 1977. He received his Ph.D. degree from VSB- Technical University of Ostrava in 2002, dissertation thesis „Diagnostic system without dismantling of locomotive controller“. His research is focused on fault tolerant system, communication technologies and coding.

