

RTP analysis for the video transmission process on WhatsApp and Skype against signal strength variations in 802.11 network environments

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Abstract. Today many people communicate using video content over the internet, either one-way, two-way or even multi-way or conference-friendly communication. Skype and WhatsApp are one of them, with a huge number of users around the world. The 802.11 wireless network is still the primary choice to access on the internet, especially for users with indoor mobility. Through the mobile application platform, Skype and WhatsApp service users can move from one place to another. Unfortunately, this will create a variation in signal power reception on the device used in the scope of the 802.11 network access point. So in this study, we want to know the response of Skype and WhatsApp application when there is a change of signal strength on 802.11 wireless networks, through analysis of received RTP packets. The data show to us, WhatsApp managed to stream RTP packets of nearly 12 thousand packages, compared to Skype which only revolves around 4800 packages.

Keyword: Video Transmission, Signal Strength, Skype, WhatsApp, RTP Analysis.

1. Introduction

In the current digital era with the rapid growth of technology, led to the growing needs and use of technology in multimedia is very high, which then affects the presence of a variety of multimedia-based applications to support the needs of it all. Most of our people, especially in urban areas are very familiar with multimedia-based applications, whether image processing, image or video delivery using mobile phone applications, VoIP, video calls, video streaming, video on demand and so on. Referring to the Cisco VNI report [1], forecast that the total internet video traffic of both businesses, consumer, and the combination of both in 2021 will account for 80 percent of overall internet traffic, up from 2016 reaching 67 percent. This show that in the future era, communication with the multimedia platform will be filled by the data traffic on the internet.

Access to the Internet is central to our attention, where in some countries, especially developing countries, 802.11 wireless networks or we are familiar with Wi-Fi is still a major star, especially on the use of mobility indoors. The use of wireless networks 802.11 is different from cable networks, as



they are vulnerable to random channel conditions, such as multipath fading, interference, path loss and others especially, 802.11 networks using 2.4 GHz ISM frequencies are widely used on other devices, so it is very susceptible to interference [2]. Also, a mobility of wireless network users causes fluctuations in signal strength on the receiving device. So the use of 802.11 wireless networks on applications like Skype and WhatsApp will face problems. However, it is strongly believed that every service provider has a trick in dealing with this problem. So in this paper, we want to know how these two applications respond to changes in signal strength on 802.11 wireless networks.

Several related studies have been conducted, as in this study [3], in which the movement of the people who moves straight or random on the scope of the 802.11 wireless network can reduce the network throughput value by 20.4 percent. With some equivalent approach from above paper, in [4] the researcher try to transmitted several video transmission process using Skype running with 802.11 network on indoor building environment with human traffic inside the building. In another study [5], performance on 802.11 wireless networks was measured when delivering video and audio packets with interference from signal strength variations. Associated with the measurement of video traffic with video telephony applications like Skype [6,7], iChat, Hangouts has been done in this study [8] with fixed wired network and using network emulator to give impairment to the networks conditions and in this study [9] measurement conducted in Wi-Fi and Cellular network also using using network emulator.

So in this paper, we conduct research scene with real conditions approach, where we conduct the transmission of video content through Skype and WhatsApp application with signal strength variation, then analyzed how each application responds to the change of signal strength.

2. Methodology

Transmission process of video over the internet, more reliable use of UDP than TCP. UDP has the advantage of shipping with tight time limits, especially on interactive video, but has a deficiency in the number of packet loss that many. while TCP can provide warranty package to the destination so that the video quality can be received well but TCP is not good at maintaining the delay of data packet delivery time[10]. RTP (Real Time Protocol) and RTCP (Real Time Control Protocol) are used together to transmit video in real time. RTP / RTCP itself can not guarantee the data packet in real time, but both protocols provide an interface that can be used in applications[11] such as Skype and WhatsApp as well as other applications to implement data transport in real time. RTCP becomes a means of information that can be used on both sides, there is called a sender report and there is a known receiver report. It contains all the information that can be used to synchronize between the data has been sent separately.

In this study we want to answer a research question, connect some condition, where the application running video call seaseon, and we give some research scenario variation of signal strength the 802.11 network, and we want to known how the application service Skype and WhatsApp respond to signal strength variations, through two things: 1) How many RTP packets are successfully transmitted, in each given signal strength scenario?, 2) Percentage of maximum lenght payload RTP can be transmitted in each given signal strength scenario? Both of these research questions will be related to the quality of the video received in the context of other research opportunities that are on the topic of quality of experience (QoE) that we have not discussed in this paper.

To answer the research question above, we designed a research experiment scenario as shown in figure 1. On this research design, we will use two devices with an android operating system that have been installed the Skype and WhatsApp mobile version. Our first device uses the ASUS Padfone smartphone, next we call ANDRO, while the second device we use Android emulator named KOPlayer [12] is installed on the PC, the next we call EMUL. We use emulator of android to get fix connection to the internet by using fixed network. so that we are able to reduce the disruption that occurs when using wireless networks, making it easy to map problems when logged in at the analytical stage. For internet connection

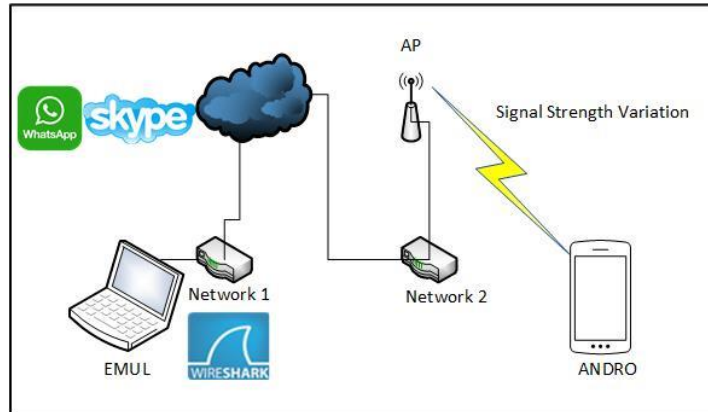


Figure 1. Research Design, while two device running one way video transmission using Skype and WhatsApp. EMUL device in fix wired network connection to the internet, while ANDRO device connect to 802.11 network on different signal strength scenario.

To achieve the goal, in this study we make the limitation that the transmission of video only one way and we not transmit the audio signal. The next limitation is, that we connect the EMUL using LAN Card on the router in network 1 and then connect to the internet using fibre optics infrastructure networks to ensure EMUL has a stable connection. While the ANDRO connects to the internet on a separate network 2 wirelessly on the access points of IEEE 802.11n, so we want to study, what will happen when the variation signal strength only on one side communications on video transmission process. Through this research design only ANDRO will be given five scenarios of variation of signal strength that is: -12 dBm, -31dBm, -54dBm, -62 dBm and last -72dBm and in each scenario, ANDRO transmits 30 seconds of video. While in EMUL we use a tool that is Wireshark[13] to capture RTP / RTCP data transmitted by ANDRO and received by EMUL

3. Result and discussion

3.1 Video transmission process using Skype

In this chapter we present the data we have obtained. In table 1, we display the IP identity and port used in the transmission process using Skype. While in table 2 is the RTP data on the video packet and audio packet. each video and audio transmission session each has a unique SSRC (Synchronous Source) identifier. In the same session the RTP is transmitted with the same SSRC ID. When the transmission session changes, a new SSRC ID is assigned.

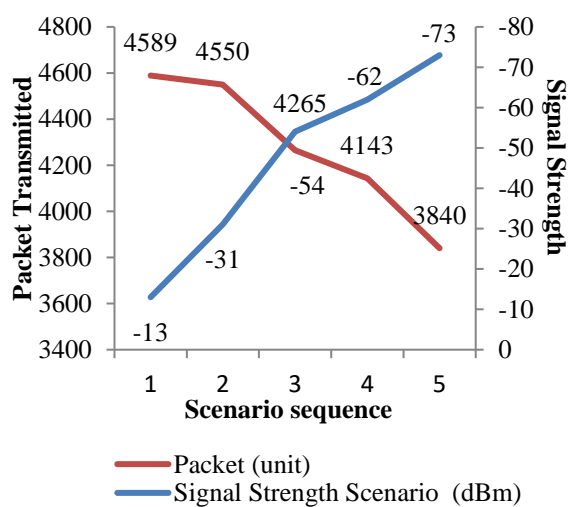
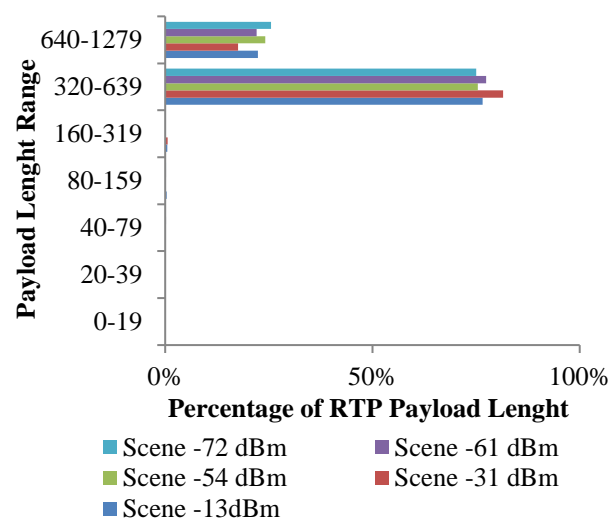
from the two tables below, it can be seen data RTP from the transmission process using Skype. Type of payload (PT) used is RTPType 122 which is in the category of dynamic Payload Type. Whereas in figure 2 it can be seen that the graph of RTP Payload that success transmitted in each scenario tends to decrease, we suspect this means that the data reduction and image quality becomes poor. From figure 3, we can see that each RTP packet that has been successfully transmitted has different payload lengths. In the graph, it can be seen that in every scenario of research has almost the same percentage in the range of successful payloads transmitted, in the range 320-629 bytes on average between 70% to 80%, while in the range above 640-1279 bytes average between 17% to 25%.

Table 1. IP identity and port for Skype transmission process in every scenario

No	Signal Strength Scenario	ANDRO		EMUL	
		IP	SRC.PORT	IP	DST.PORT
1	Data 1 Skype -13dBm	104.44.201.58	3480	192.168.100.53	59601
2	Data 2 Skype -31dBm	104.44.201.58	3480	192.168.100.53	64033
3	Data 4 Skype -54 dBm	104.44.201.60	3480	192.168.100.53	65167
4	Data 5 Skype -62 dBm	104.44.201.59	3480	192.168.100.53	55017
5	Data 6 Skype -73 dBm	104.44.201.60	3480	192.168.100.53	60764

Table 2. RTP Payload video packet for Skype

No	Signal Strength Scenario	Video Packet			
		SSRC	PT	Packet	Byte
1	Data 1 Skype -13dBm	0xC78FF0CB	RTPTYPE-122	4589	2221194
2	Data 2 Skype -31dBm	0x237B405B	RTPTYPE-122	4550	2074110
3	Data 4 Skype -54 dBm	0x68AFB2AC	RTPTYPE-122	4265	2108179
4	Data 5 Skype -62 dBm	0x40EB513B	RTPTYPE-122	4143	2066989
5	Data 6 Skype -73 dBm	0xA1B1E905	RTPTYPE-122	3840	1946290

**Figure 2.** RTP Packet Transmitted vs Signal Strength for Skype**Figure 3.** Percentage of RTP payload range that successful transmitted for Skype

3.2. Video transmission process using WhatsApp

Just like the previous sub-chapters, but the data we present in this sub-chapter is the result of the video transmission process using WhatsApp. In tables 3 and 4 there are interesting things we get, if we compare with the previous Skype data, on the ANDRO side, SRC.PORT always uses the same port 3480, and this port is a registered port on IANA. But what is interesting in WhatsApp data on the ANDRO side, SRC.PORT changes in each transmission session or on category dynamic port [14], as can be seen in table 3. Another interesting things from the video transmission process on WhatsApp is, in each video transmission session using two different SSRC IDs and two different PT types. this can be seen in table 4. This things is different when compared with Skype RTP data which only use one SSRC ID and one PT type on each of its transmission sessions.

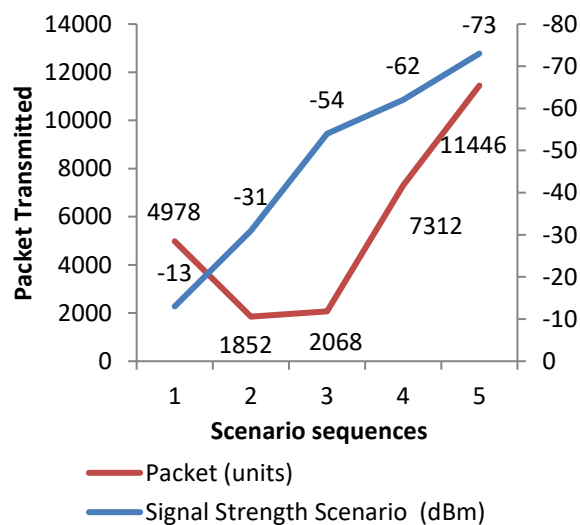
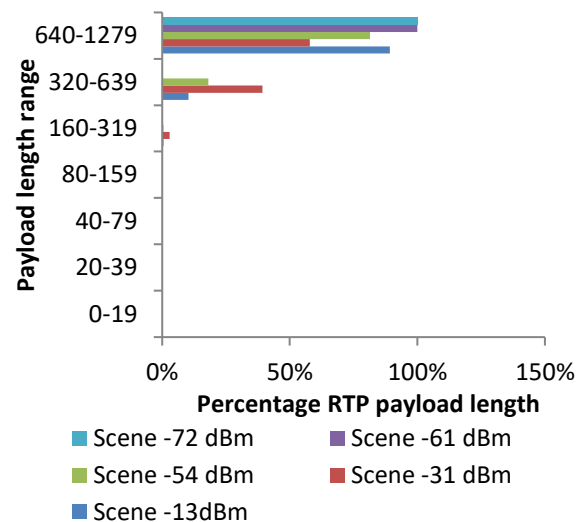
The result of number RTP packet from the video transmission process on WhatsApp is shown as in figure 4 that the number of transmitted RTP packets increases while the signal strength decreases. this is beyond our expectations, and this will becomes an opportunity for future research. Also if we look at the number of transmitted RTP packets is much greater than that of Skype, figure 5 shows to us some interesting point. Where in the transmission process using WhatsApp this can be seen, that the percentage of the length of RTP payload in the range 640 - 1279 bytes dominates between 60 - 100%. Whereas, the packet length of 320 - 639 bytes ranges between 15% - 40%. So we can conclude from this data, that WhatsApp has better video quality on the receiving side because it successfully transfers more RTP payload.

Table 3. IP identity and port for WhatsApp transmission process in every scenario

No	Signal Strength scenario	Android		Emulator	
		IP	SRC.PORT	IP	DST.PORT
1	Data 1 WA -13dBm	114.124.242.168	50215	192.168.100.53	55704
2	Data 2 WA -31dBm	114.124.242.168	33641	192.168.100.53	59331
3	Data 3 WA -54 dBm	114.124.242.168	35614	192.168.100.53	58506
4	Data 4 WA -62 dBm	114.124.242.168	59578	192.168.100.53	53489
5	Data 5 WA -73 dBm	114.124.242.168	43361	192.168.100.53	60849

Table 4. RTP Payload video packet for WhatsApp

No	Signal Strength scenario	Video Packet			
		SSRC	PT	Packet	Byte
1	Data 1 WA -13dBm	0x8C4C6D6B & 0xD571F6A0	RTPTType-102& 103	4978	4665322
2	Data 2 WA -31dBm	0x45B7AA3E & 0xAF98B0AE	RTPTType-102& 103	1852	1333930
3	Data 3 WA -54 dBm	0x563708E0 & 0xD2D6DB03	RTPTType-102& 103	2068	1691462
4	Data 4 WA -62 dBm	0x53A112C2 & 0x20DE5986	RTPTType-102&103	7312	7741138
5	Data 5 WA -73 dBm	0xD6712987 & 0xCE0EAA79	RTPTType-102& 103	11446	12207274

**Figure 4.** RTP Packet Transmitted vs Signal Strength for WhatsApp.**Figure 5.** Percentage of RTP payload range that successful transmitted for WhatsApp

In figure 6, we compare the number of packets successfully transmitted by Skype and WhatsApp in each of the research scenarios based on signal strength variations. WhatsApp can be superior to Skype in terms of the number of packets successfully transmitted under conditions or disturbance of signal strength variations in 802.11 networks. This opens up many opportunities for discussion in future research so that we can get the full picture in understanding the process of video transmission being a disturbance by signal strength in wireless network environments.

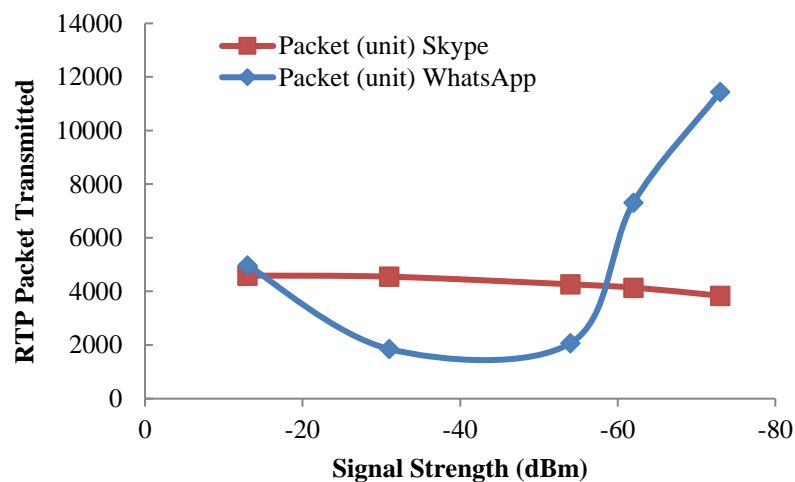


Figure 6. Skype vs WhatsApp RTP packet successful transmitted in variation signal strength

4. Conclusion

From all the discussions we have presented, we draw a conclusion in the scenario of our research. As shown in figure 3 and 5 WhatsApp obviously has the advantage of being able to stream RTP payload more when compared to Skype under the varying signal strength conditions in the wireless network being used. WhatsApp managed to stream RTP packets of nearly 12 thousand packages, compared to Skype which only revolves around 4800 packages. This condition is also indicated by the dominance of the length of data payload transmitted on WhatsApp. where, the payload range of 640-1279 bytes dominates more than 80% of which is successfully transmitted. Meanwhile, the Skype range of payload successfully transmitted is dominated by packages with a range of 320-639 bytes of 70% to 80%.

However, this raises a subsequent research opportunity to complete the answer piece of what has not been answered in this study. How the quality of video received in the context of user satisfaction within the scope of research topics Quality of Experience (QoE), when the video transmission process is disturbed by changes in signal strength occurring on the wireless network used to communicate each other. Based on the data we get, we assume or suspect that when the transmission process using WhatsApp is then disturbed by changes in signal strength, then the QoE parameter measurement results will be better than when the video transmission process using Skype, and this we must prove in the next study.

Acknowledge

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