Performance Analysis of Video Calls Using Skype

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Abstract

The widespread use of the Internet has had a significant impact on the world of communication, leading to the development of new technologies. One of these new technologies, VoIP (voice/video-over-Internet protocol), has revolutionised contemporary communication. Skype is undoubtedly the most well-known VoIP application in the current spectrum. Its features, capabilities and successes have attracted the research community and telecommunication companies, which have become interested in illustrating Skype's performance, characteristics, quality and end user experience. This paper focuses on investigating Skype's responsiveness mechanism toward bandwidth variation. Skype's sending bit rate and packet size when experiencing different levels of packet loss will be studied as well. In addition, we focus on how Skype shares its available bandwidth with other cross-traffic such as TCP traffic. Moreover, we measure end user quality of experience (QoE) under different packet loss conditions. This paper offers a number of key findings, including the fact that Skype responsiveness to bandwidth variation very dependent on the speed motion of the video calls. In addition, Skype is robust regarding minor packet loss. We also found that Skype is indeed TCP-friendly and reacts effectively to congestion. Furthermore, we have determined that the acceptable quality for the end user when Skype video calls experience different levels of packet loss is 8% of loss; beyond that, the user will not tolerate poor quality.

Keywords

Skype, video call, Quality of experience, VoIP.

1 Introduction

Communication among people has been fundamentally revolutionised by the Internet. Text messages and emails are still the traditional means of communication; however, we are now at the cutting edge, welcoming the next generation of communication: voice/video telephony. Several vendors have provided applications that support online video chat, such as Facetime, MSN messenger and Skype. The most popular example of these applications is Skype, which provides high quality voice/video transmission in a variety of network conditions. Due to the nature of video calls, which require real time communication between users, its quality can be more sensitive to network impairments, such as bandwidth variations and packet loss. In addition, as video calls demand more bandwidth than do voice calls, their traffic and quality can be also affected by other cross traffic. It is therefore of central importance for researchers, application developers and users to evaluate Skype's video call behaviour and quality under a variety of network conditions. However, to date, there has been very little research conducted in this area.

In this paper, we present our recent study of Skype's performance under adverse network conditions such as bandwidth variation and packet loss. In addition, we will investigate how Skype will compete for bandwidth with other traffic. Finally, the end user quality of experience (QoE) will be measured.

2 Related work

Evaluating Skype's performance can be categorised into two areas: studying protocol network characteristics and investigating VoIP aspects. Baset and Schulzrinne (2006) first examined Skype's traffic, peer-to-peer technology and NAT crossing mechanism. Since then, several papers have been released on Skype's P2P technology, architecture and traffic (Bonfiglio et al., 2008; Guha et al., 2006). In the second area, some researchers concentrated on Skype voice quality, providing extensive investigation into its voice service only (Chen et al., 2006; Te-Yuan et al., 2009). Some papers have been published regarding Skype video calls. Boyaci et al. (Boyaci et al., 2009) studied Skype's sending rate under different network conditions and then compared its performance with other video chat applications. They found that Skype, under adverse network conditions, performed better than other VoIP applications. In a recent paper, De Cicco et al. (De Cicco et al., 2011) investigated Skype's responses to bandwidth variation; they determined that Skype's responsiveness mechanism is too slow. The most recent investigation was conducted by Xinggong et al. (Xinggong et al., 2012), who studied Skype performance under different network conditions such as pocket loss, propagation delay and bandwidth variation. They also developed several models, such as Skype video calls' rate control and FEC redundancy. They concluded that Skype adapts well to bandwidth variation and can perfectly combat mild loss. Identifying whether Skype is TCP friendly or not was a controversial issue among some researchers (Boyaci et al., 2009), (De Cicco et al., 2011) and (Xinggong et al., 2012). In our paper we will study Skype performance under adverse network conditions using different video motions. In addition, we will investigate how Skype shares bandwidth with TCP traffic. Finally, we will measure the end user experience regarding Skype video quality.

3 Measurement Testbed Setup

A controlled testbed that consists of several components was used to study Skype video call performance, as shown in Figure 1. This testbed consisted of two clients on which the Skype program was installed. These clients were connected by NAT wireless routers where each host has a private IP address and connects to the Internet via these routers. In addition, a software-based network emulator was used to emulate the different network conditions (NEWT) (Microsoft.Resarch.Asia, 2010). To emulate video calls, we used three standard video sequences with different motions—Akiyo, Foreman and Stefan. This can comprehensively determine Skype's video adaption mechanism. Video sequences are injected into Skype using ManyCam (Manycam, 2012), which provides a constant and repeatable transmission of the video content. The measurement data are captured using Wireshark and then analysed using Matlab.

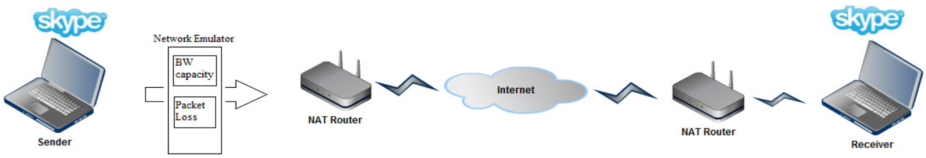

Figure 1: Testbed architecture

4 Measurement Results

In this section, we will illustrate Skype performance under adverse network conditions. This includes reporting on Skype's response to bandwidth variation when various video sequences with different speed motions are used; discussing how Skype's sending rate and packet size behave under different packet loss conditions; identifying whether or not Skype is TCP-friendly; and examining the acceptable quality for the end user when the application experiences different packet loss conditions.

4.1 Impact of available bandwidth

To examine Skype's responsiveness mechanism toward changeable network conditions such as plunge/soar available bandwidth, the capacity of bandwidth was varied according to a square wave form. The network emulator was used to change bandwidth to rates ranging from 1,000 Kbps to 160 Kbps. The measured results of Skype sending rates are demonstrated in Figure 2, which shows that Skype effectively reduced its sending rate as the available bandwidth plunged, and increased its sending rate when the bandwidth rose. This increase varied as per the speed motion of the video call.

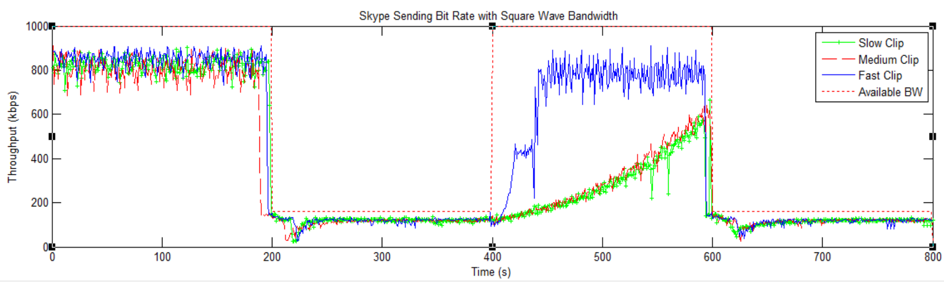

Figure 2: Skype sending bit rate per second with the square wave bandwidth

In the time interval [400,600], where the available bandwidth was increased suddenly to 1000, it was noticed that Skype adapted to the sudden increase by increasing its sending bit rate for fast, medium and slow calls. However, Skype's responsiveness mechanism for the fast video call was more effective and quicker than for the other video calls, which were medium and slow. Skype sending bit rate for the fast video call took approximately 50s to adapt to the sudden bandwidth increase, and then it increased its sending bit rate again to up to 800 Kbps. This can be explained in that due to the fact that the fast video call contains the most complex scene and the highest motion, thus; it need more bandwidth to deliver the call in high quality. In the medium and slow video calls, Skype's adaptive mechanism was too slow with respect to the sudden increase in bandwidth and this conclusion was also achieved by (De Cicco et al., 2011) . Skype's sending bit rate for medium and slow calls increased gradually over the period, reaching roughly 600 Kbps at the end of the period.

In conclusion, Skype can adapt its sending bit rate with respect to sudden decreases/increases in bandwidth. Skype video calls' adaptive mechanism for fast motion was more effective and quicker than those of the other video calls.

4.2 Impact of packet loss

We investigated how Skype behaves in term of sending bit rate and packet size when packet loss is introduced at different levels. This is particularly crucial in assessing Skype reactiveness toward different levels of packet loss. Using a network emulator, we varied the packet loss rate from 0% to 14%.

The measurement results of Skype's sending rate are illustrated in Figure 3. We found that Skype adapts its sending bit rate toward packet loss in three manners. First, when the rate of packet loss is ≤ 6%, Skype's sending bit rate remains almost constant. Second, in the range of the packet loss 6%-10%, Skype's sending rate decreases gradually. Third, when the level of packet loss is larger than 10%, Skype's sending rate is flatted at a lower value. These behaviours were also observed for the different video calls that were used in this experiment (fast, medium and slow).

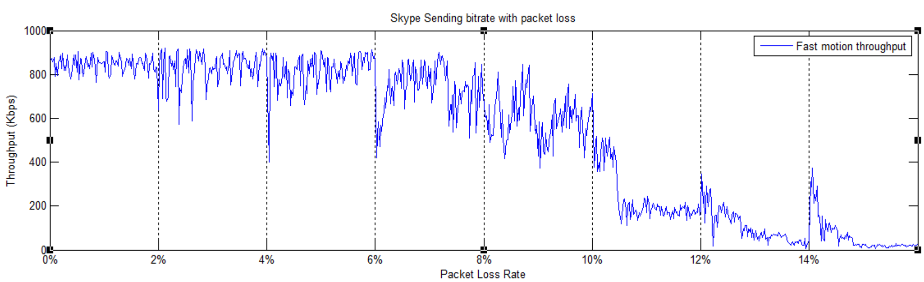

Figure 3: Skype Average sending bit rate with different rate of packet loss

To summarise, the adaptation point for Skype, when it combats different packet loss level, is 6%. Skype adapts to the packet loss via three behaviours. First, Skype functions normally when the packet loss is ≤ 6% and its sending rate becomes loss-ignorant. Second, in the range of 6% to 10% loss, Skype start reacting to the different packet loss condition by reducing its sending bitrate gradually. The third behaviour is that Skype turns to conservative behaviour when the packet loss is larger than 10% and it significantly keeps its sending rate flat at a low value. Skype follows these behaviours for fast, medium and slow video calls.

The measured results of Skype's packet size show that Skype adapts to minor packet loss by increasing its packet size proportionally as the minor packet loss rate increases. However, when the level of packet loss is significantly large, Skype relatively adapts by reducing its packet size. This is illustrated at Figure 4.

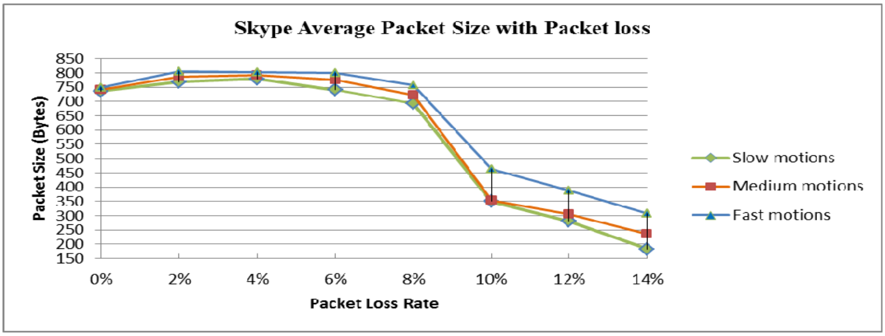

Figure 4: Skype average packet size with different packet loss levels

It is evident that Skype followed two behaviours in order to adapt its packet size toward different packet loss rates. First, when the rate of packet loss was ≤ 8%, Skype's packet sizes increase slightly as the minor rate of packet loss increased. This can be interpreted as a result of employing FEC algorithm by Skype to combat the packet loss. Second, when the level of packet loss was larger than 8%, Skype's packet size dropped dramatically. This indicates that Skype becomes more conservative when it detected that the network condition became very bad. Consequently, Skype only sends out required data at a low rate. These behaviours were observed for the different video calls with different motion speed which used in this experiment.

In summary, Skype adapts to the packet loss using two behaviours. First, when the packet loss is ≤ 8%, Skype behave normally and counter the loss by employing the FEC algorithm. Second, Skype performs in a conservative manner when the packet loss is larger than 8% and it significantly reduces its packet sizes. Skype uses these behaviours for fast, medium and slow clips.

4.3 Skype video call performance when sharing bandwidth with TCP traffic

In an actual network, Skype video calls would not propagate alone; different traffic would share the bandwidth. It is known that the majority of Internet traffic is transferred by the TCP protocol (Jiang and Dovrolis, 2005), which uses a congestion control mechanism to control the load applied on the network. However, since Skype transfers its traffic over UDP protocol (Bonfiglio et al., 2009), it lacks the sense of having a congestion control mechanism at the transport layer. Thus, it is vital that the applications that rely on UDP protocol for their transmissions should be TCP-friendly in order to maintain Internet stability (Floyd and Fall, 1999). Skype copes with this issue by placing its built-in rate control mechanism at the application layer (Xinggong et al., 2012).
Several studies have been conducted to identify whether or not Skype is TCP-friendly (Boyaci et al., 2009), (De Cicco et al., 2011) and (Xinggong et al., 2012).

However, contrary results were achieved in these studies. Boyaci et al. (Boyaci et al., 2009) stated that TCP traffic is more aggressive than Skype when they share the bandwidth, while De Cicco et al. (De Cicco et al., 2011) concluded that Skype behaved more aggressively than TCP traffic. However, Xinggong et al. (Xinggong et al., 2012) stated that Skype is TCP-friendly.
Thus, we aimed to investigate how Skype competes with TCP traffic based on HTTP protocol and how this traffic affects Skype's performance.
In this scenario, we use a network emulator to configure the bandwidth to 1,000 Kbps. In order to generate HTTP traffic, a 12 MB file was uploaded in the middle of a Skype video call to Media Fire, which is a cloud storage service website.

The measured results demonstrate that, in the absence of any other traffic, the sending rate of all video calls utilised the bulk of bandwidth. Once the HTTP traffic was introduced, Skype's sending rate adapted to the existence of HTTP traffic by reducing its transmission rate by approximately 50%, leaving the remaining bandwidth for HTTP traffic. After the HTTP traffic was stopped, Skype resumed its sending rate to utilise the majority of the available bandwidth. This was noticed for all video calls with different speed motions. The following graph shows the Skype sending rate performance of fast video calls in the presence of TCP traffic, based on HTTP protocol.

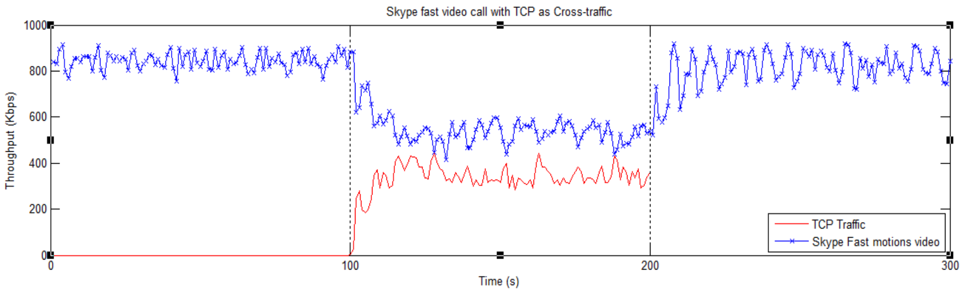

Figure 5: Skype sending rate with TCP traffic based on HTTP protocol

It was mentioned previously in the related work that there has been controversy as to whether or not Skype is TCP-friendly. We found that Skype shared the bandwidth with other TCP traffic in a fair manner, regardless of the video call speed motions. Thus, we agree with Xinggong et al. (Xinggong et al., 2012), and concur that Skype is indeed TCP-friendly.

In conclusion, Skype utilises the bulk of bandwidth in absence of other traffic. When it shares bandwidth with TCP traffic, it does so fairly; thus, it can indeed be considered TCP-friendly.

4.4 Investigating the end-user acceptable quality for Skype video calls

We also investigated the end-user acceptable quality for Skype video calls under different packet loss conditions. This is of central importance in evaluating the extent to which Skype can maintain an acceptable level of quality under different levels of packet loss. This was done by carrying out several actual Skype video calls with various people during which different packet loss rates were introduced by the network emulator.

Once the packet loss level was introduced, the end user was asked whether he/she would tolerate the video call quality or not. We considered that the end user would not tolerate the video call quality if he/she dropped the call, if the call continued with audio only or if the end user declared that he/she could not continue the call.

We found that the drop ratio among end users increased relative to the level of packet loss increase.

We observed that the majority of end users were unable to tolerate the Skype video quality when packet loss rate was 10%, and they asked to disconnect the call. This feeling was more obvious when the rate of packet loss was more than 10%. The following graph shows the cumulative distribution (CDF) of the dropping ratio versus different packet loss rates.

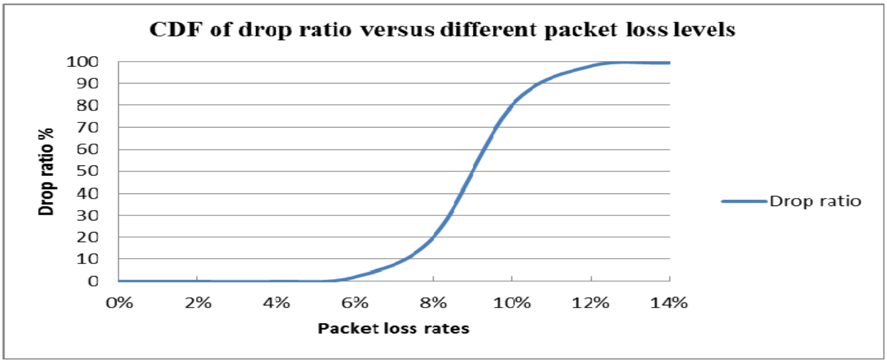

Figure 6: CDF of drop ratio versus different packet loss levels

In conclusion, we examined the end-user acceptance quality for Skype video calls under different packet loss conditions. We found that the drop ratio among end users increased proportionally as the level of packet loss increased. Therefore, it could be said that the end user would tolerate the quality of a Skype video call until the level of loss reached 8%. After this level of packet loss, Skype video call quality was determined to be unacceptable.

5 Conclusion and future work

In this paper, we look at Skype performance under different network conditions. Through extensive measurement, we have illustrated that Skype responsiveness for fast video motion is more effective and it is robust toward minor loss. When network conditions become very bad, Skype tends to reduce its sending rate significantly. In addition, our results reveal that Skype is TCP-friendly. Moreover, we found that end users will tolerate the quality of a Skype video call up to an 8% level of loss. After this level, Skype video call quality was not tolerated and was unacceptable.

For future work, we will extend our outlook to include high-speed connection and then study Skype's performance under this system. Skype performance of multi-users video conferencing is also in the plans for future research.

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