ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 3, Issue 12, December 2016

Performance Evaluation of Video Conferencing Techniques

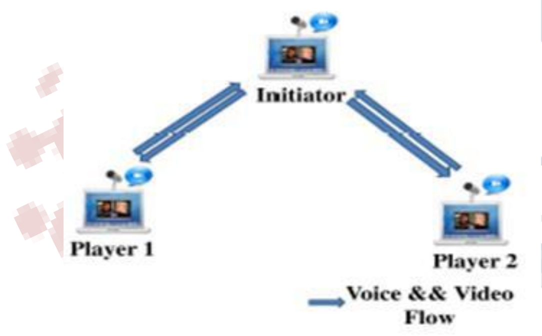
[1] Dr. Godbole Bhalchandra Balkrishna, [2] Mr. Lokhande Shashank Shivaji
[1] [2] Karmaveer Bhaurao Patil College of Engineering, Satara

Abstract:-- **Video Conferencing requires high bandwidth, low delay, and video transmission between geographically distributed nodes (users). It is challenging as well as interesting to study high quality video conference and to deliver data through the best internetwork. Presently Google+, iChat and Skype as well as SDN based unicast & multicast video conferencing techniques are in use. In present paper, we propose performance evaluation for present and developing video conferencing techniques. Here Google+, iChat, Skype and SDN based systems are compared amongst metrics like bandwidth, latency, throughput and speed. Comparison between different video conferencing techniques is analyzed technically. It is observed that google+, iChat & Skype are used for intra and internet. We also review the new techniques such as SDN based systems.**

Index Terms:— **SDN, iChat, Skype, nodes, latency, metrics, throughput..**

I. INTRODUCTION

Video conferencing is an online meeting that takes place between two or more participants where each one can see an image of other and both are also able to speak and listen to each-other in real time.



(a)

(b)

(c)

Fig.1 (a) iChat, (b) Google+ and (c) Skype techniques [1].

It has been around some time, is now gaining popularity. It does improve communication, cut down on travel time & cost and also encompasses wide range of technologies used in various situations as shown in Fig.1.

iChat: The architecture used is P2P. In case of iChat star topology is used to connect the users. The central hub is the conference initiator, i.e. the user who heads the conference. The initiator only has permission to add new user or close the conference. A normal user uploads his voice and voice data through one UDP flow and downloads others voice and voice data through one UDP flow. Normal user only connect to the initiator, two normal users cannot communicate directly. iChat cannot work if UDP flow is blocked [1].

Google+: Server centric technology is used here. Each user sends his voice and video to a dedicated proxy server, and also receives other user's voice and video from that server. Different users choose different proxy servers. [1].

ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 3, Issue 12, December 2016

Skype: For two-party video calls, Skype uses direct P2P transmission for voice and video if the two users can establish a direct connection. When three or more users are involved, the network topology is shown in Fig. 2. Voice is still transmitted using P2P. Similar to iChat, the conference initiator acts as the central hub. A normal user uploads his voice to the initiator and downloads other users' voice from the initiator. Different users normally choose different relay servers. Similar to Google+, Skype mostly uses UDP to transmit voice and video, and only switches to TCP if UDP is blocked. The other codecs with respective frame size and bit rate are as shown in (i). These small flows indicate signaling [2].

(i). Nominal Characteristics of Skype Codecs

Codecs	Frame size (ms)	Bit rate
ISAC	30,60	10/32 kbps
ILBC	20,30	13.3,15.2
G.729	10	8
iPCM-wb	10,20,30,40	80 (mean)
EG.711 A/U	10,20,30,40	48,56,64
PCM A/U	10,20,30,40	64
SVOPC	20/60	20/50
True Motion VP7	Un known	>20

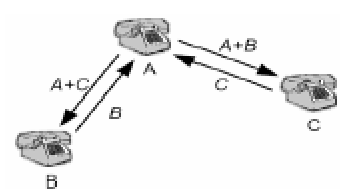


Fig.2 Skype Three-Party Audio Conference

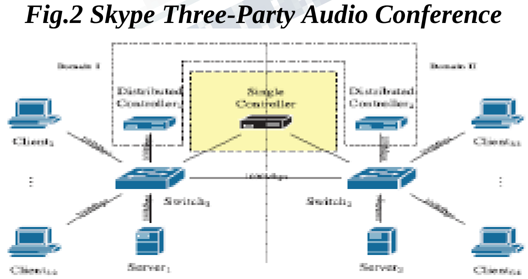


Fig.3 The network topology

SDN: It is an umbrella term encompassing several kinds of network technology aimed at making the network as agile and flexible as the virtualized server and storage infrastructure of the modern data center. A single controller, referred as logically centralized but physically distributed controllers. With distributed controller as each controller collects state information about the network, they need to exchange their views in order to build global network view as shown in Fig.

3. The consistency is employed by a specific model employed in due course. Levin and et al. [3] uses a flow simulator to study the design of distributed SDN applications. SDN based video conferencing is design methodology as well as implementation guideline. Presently it is conceptually been used to improve the performance of video conferencing with respect to latency and throughput [4]. Generally, the number of users or parties controls the design and implementation of VC system. A review is carried out by comparing active and passive parameters of different video conferencing techniques namely viz. voice and video delay, latency, throughput, incurred in capturing, encoding, decoding and rendering i.e. end to end delays perceived by users; bandwidth, video generation, protection, adaptation and distribution. It is challenging and ambitious to come up with proper inference, to address these challenges a comparative study of various parameters is done by analyzing the above mentioned techniques [5]. Video traffic is studied and analyzed by various researchers and developers [7]—[10]. Details of comparison are presented in graphical and tabular format in this paper. Rest of the paper is organized as follows. Comparative study of the old and recent video conferencing techniques with respect to different parameters is discussed in section II. The practical and theoretical aspects along with graphical analysis are presented in section III. Discussion is in section IV followed by conclusion stating the summary of the paper.

II. COMPARATIVE STUDY

A comparative study for the old and recent video conferencing techniques is carried out with respect to the parameters and number of users. Each user in conference generates voice and video packets constantly at significant rates. In our study and analysis, the duration and throughput are recorded as voice and video flows from user to user to detect right topology and differentiate between techniques, Here Google+ video calls for both cases i.e. two party and

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 3, Issue 12, December 2016

multiparty as illustrated in Fig. 1(a). iChat uses heterogeneous receivers that always receive same video versions and the receiver determines the video quality. Normally the lowest video quality for different receivers is ensured where neither video trans-coding nor layered coding is employed. The Skype is employed where the large variability of receivers is present, as shown in (ii), one way voice and delay performance between users is compared for the Google+, iChat and Skype video conferencing techniques. Because of signal processing on both ends this delay may increase. Skype is two and multiparty setup for voice and video. The multiparty Skype employs different topologies for voice and video transmissions. Here the voice mixing and recording adds delay into communication delay.

(ii). One way delay performance

System	Party	Voice (ms)	video (ms)
Google+	Two-Party	180	100
iChat	Two-Party	220	220
	initiator to normal	220	220
Skype	Two-Party	156	110
	initiator to normal	230	130
	normal to normal	230	190

Since video has to be first send to the Skype server and then relate to the receivers, Skype video delay is always greater than voice delay. Both video and voice transmission and reception is unsynchronized. The gap between voice and video delay is of the order of 100 ms. Here the iChat transmits video and voice for two party calls in synchronized way. For multiparty conferences the delay performance is same as two party cases. Due to synchronized data communication network delays are negligible for initiator to combine packets generated by different sources.

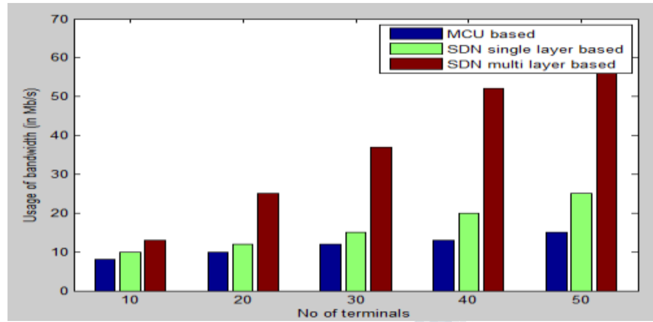


Fig.4 Bandwidth usage in SDN (Mb/s)

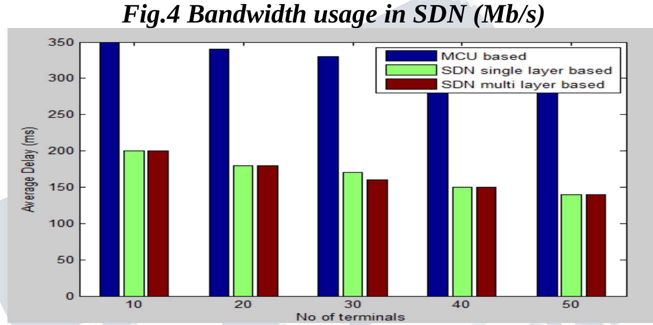


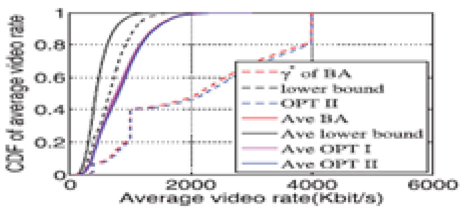
Fig.5 Average delay in SDN (ms)

Video conferencing systems are deployed mainly by means of a MCU (Multipoint control Unit) and SDN. MCU is a bridge that interconnects calls from several resources. Performance evaluation of Bandwidth usage and average end to end delay incurred during the video conferencing in case of SDN is as per shown in Fig. 4 and Fig.5. In case of bandwidth usage as the number of users increases SDN enables multicast architecture performs better than the other methods i.e. MCU & SDN single layered as far as bandwidth usage is concern. The performance of the SDN-enabled layered video multicast solution is almost the same as that of the SDN-enabled single-layer multicast conferencing system [4]. To investigate how Skype responds to bandwidth available in the network, the bandwidth capacity in network emulator is varied from 50 kbps to 1000 kbps while fixing the PLR and propagation delay.

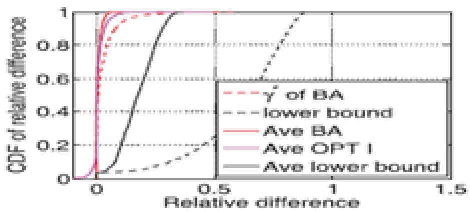
III. GRAPHICAL ANALYSIS

In Fig. 6(a), the optimized performance of heterogeneous multiparty video conferencing is shown for achieved video rates for CDF of average video rate against video speed for OPT-I, OPT-II with lower bound. Here the optimized performance of average lower bound MPVC is better average OPT-I and OPT-II.

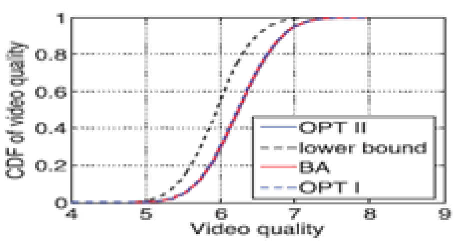
International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 3, Issue 12, December 2016



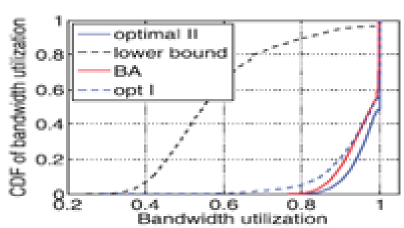
a) achieved video rate



b) relative performance



c) average video quality



d) bandwidth utilization

Fig. 6 Performance of heterogeneous MPVC with 1600 random viewing scenarios

The bandwidth allocation algorithms further improve the performance to earlier methods, as shown in Fig.6 (a). The relative performance of bandwidth allocation algorithm is superior to average OPT I and lower bound techniques as shown in Fig.6(b). Fig.6(c) depicts video quality for different methods and it is nearly same for OPT I, OPT II, Lower bound and Bandwidth allocation algorithm. The bandwidth utilization of OPT I and BA algorithm is impulsive at 80% to 100% utilization whereas bandwidth utilization for lower bound algorithm is increasing steadily as bandwidth utilization increases from 40% to 80%. Above 80% bandwidth utilization of lower bound algorithm, the performance saturates and the gradient is low as shown in Fig.6(d).

Maximizing Aggregate Video Quality: The first design objective is to maximize the total video quality received by all peers. We adopt a PSNR-type of video quality model, which quantifies the quality of a video stream at rate ri as log (ri). The optimal peer bandwidth allocation is to maximize the total video quality of the conference given in (1)

$$OPTI : \max_{U,R,B} \sum_{l\in S} |Gi| \, log(ri) \qquad (1)$$

Due to the log video utility function, the optimal solution of OPT I achieves the weighted proportional fairness among all video sources, with the weight for a sub conference be the number of viewers as shown in Fig. 6.

Achieving Max-Min Fairness: Another widely used fairness metric is the Max-Min fairness. Intuitively, we prefer all sources to achieve the same rate as long as it is allowed by the individual source's upload capacity and the available bandwidth resource in the whole MPVC system. To achieve this, we want to find a video ratesuch that if a video source *i*'s upload capacity u_i is less than , it should be able to stream its video at rate $r_i = u_i$, for any other source with $u_i \geq$ it should stream its video at the common rate $r_i =$. Under this setting, the capacity of the system is defined as the maximal supportable , which can be calculated by using (2)

$$OPTII : \max_{U,R,B} \gamma \qquad (2)$$

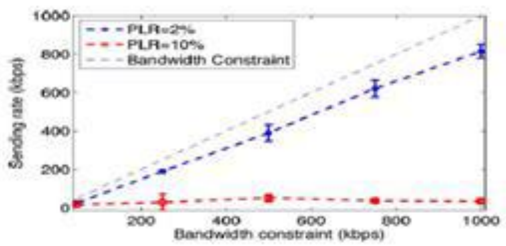
Lower Bound of Max-Min Capacity: While the max-min capacity *for each one-view MPVC scenario can be iteratively solved for the corresponding optimization problem OPT II, similar to the homogeneous case, it is important to obtain lower bounds of *for heterogeneous systems that is independent of specific watching relations, and even better, independent of conference sizes..

Bandwidth Allocation Algorithm: Y. Zhao and et al [5] presented Bandwidth allocation algorithm based on three guidelines. Two-level hierarchy is adopted for bandwidth management. At the top level, a centralized tracker manages the helper pool shared by all sub-conferences. It keeps track of the bandwidth contributed by peers in sub-conferences with surplus bandwidth, and allocates helper bandwidth to

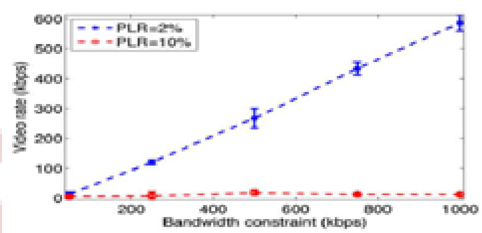
International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 3, Issue 12, December 2016

sub-conferences with bandwidth deficit. At the bottom level, the bandwidth allocation among peers in each sub-conference is coordinated by the video source. In addition to OPT II, BA and the lower bound, OPT I defined in (1), the bandwidth allocation optimized directly for video quality. The average curves of OPT I, OPT II, and BA algorithm are clustered together, and the gap between them and the average rate curve of the lower bound is smaller than the max-min capacity gap. Fig.6 (b) which shows the relative performance difference of OPT I, BA algorithm, and lower bound compared with OPT II.

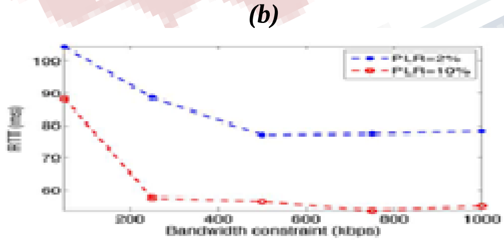
In Fig.7 (a) when, Skype is in NORM state and increases its sending rate proportionally as the bandwidth capacity increases.



(a)



(b)



(c)

Fig. 7 Impact of available network bandwidth. The two way propagation delay is fixed at 50 ms. (a) Sending rate, (b) Video Rate and (c) Round-trip time.

On the other hand, when, Skype is in CONS state and its sending rate almost remains unchanged even when the bandwidth capacity increases. When it detects PLR is larger than 10%, it will switch to the CONS state and sends data at the lowest rates. Similar trend can be detected in video rate as illustrated in Fig.7(b) video rate in NORM state changes linearly with bandwidth capacity, and remains unchanged in CONS state. It is also noticed that the video rate increases proportionally with the sending rate. The changes of RTT with available bandwidth are shown in Fig.7 (c). It is observed that RTT decreases in general as bandwidth capacity increases From these results, [6] have presented the mean performance of Skype under different propagation delays for sending rate, video rate and RTT at Skype can closely keep track of the available network bandwidth and adjust its sending rate and video rate to efficiently utilize available bandwidth without causing excessive congestion.

IV. DISCUSSION

Video conferencing is a popular industrial application in distance learning, Telemedicine, judicial and governmental activities. Here we surveyed the video conferencing techniques such as SDN; this is not VC techniques but an independent Platforms to carry out VC in improved fashion to perform better as compared to the previous one [7], [8]. Fig.1, 2 and 3 provide details of video conferencing techniques and their features. The data in (i) provides some details of characteristics of Skype Codecs. Most codecs provide audio and video signal with error detection and correction facility. Here full pan/Tilt/Zoom capability of camera with required resolution and microphones placed tabletop/sealing/wireless for equalized audio stereophonic or multi track signal. Also table depicts the frame sizes with respect to bit rates of the codec. The emphasis on ease of use and flexibility is intended to drive broad usage across many different remote setup configurations, allowing users at home to telecommute and users in remote corporate locations to reduce long distance travel. The solution therefore serves as an enabler for tele-commuting, as it gives telecommuters the opportunity to be virtually present during any meeting [9].

Studies show that, given the right equipment, there is often only a small difference between a face-to-face meeting and a virtual meeting when non-verbal causes are visualized. The solution's portability and low cost makes it a candidate for many other situations where one wants to virtually bring people together and communicate effectively.

Though it is new methodology SDN is directly applicable to video conferencing techniques and it is popular, therefore it is adopted in different applications. Fig. 4 & 5 show the

ISSN (Online) 2394-6849

International Journal of Engineering Research in Electronics and Communication Engineering (IJERECE)
Vol 3, Issue 12, December 2016

bandwidth usage and average delay for SDN based video conferencing in case of SDN enable multicast solutions. We presented graphical analysis w.r.t video rates, video quality, relative performance and bandwidth utilization. This analysis is for Skype specifically as shown in Fig. 6 and 7. Along with this the performance is presented in terms of optimization statistics. OPT I & OPT II are given mathematically with respect to min-max capacity and bandwidth allocation Algorithm.The Skype mean performance in terms of propagation delay is shown in (iii).

(iii). Mean performance of Skype under various propagation delays [10]

Sending rate	Video rate	RTT
354.4	343.6	64.9
358.7	342.3	220.5
352.7	353.9	416.1
355.8	338.9	1017.7
368.8	350.9	2019

V. CONCLUSION

Here, the study of different popular video conferencing systems is presented for video conferencing systems. The comparative information regarding design, operation and performance is given. iChat, Google+ and Skype are techniques which are better improved and used for P2P applications. Various voice/video processing delays, incurred in capturing, encoding, decoding and rendering, account for a significant portion of the end-to-end delays perceived by users. Compared with multi-version video coding, layered video coding can more efficiently address user access heterogeneity. With layered video coding, prioritized selective retransmissions can further enhance the robustness of conferencing quality against various network impairments. In this paper, we present study on three different popular video conferencing systems along with SDN.

REFERENCES

[1] Yang Xu, Chenguang Yu, Jingjiang Li and Yong Liu , "Video Telephony for End-consumers: Measurement Study of Google+, iChat, and Skype," IEEE/ACM Transactions on Networking, Volume 22, 2014, pp. 826-839.

[2] Dario Bonfiglio, Marco Mellia, Michela Meo, and Dario Rossi ," Detailed Analysis of Skype Traffic, " IEEE transactions on multimedia, vol. 11, no. 1, Jan. 2009.

[3] D. Levin et al., "Logically centralized: State distribution trade-offs in software defined networks," in Proc. 1st Workshop Hot Topics SDN, 2012, pp. 1–6.

[4] Mohamed Aslan and et. Al., "On the Impact of Network State Collection on the Performance of SDN Applications 'Utility Maximization in Peer-to-Peer Systems'," IEEEE communications letters, vol. 20, no. 1, Jan. 2016 5Proc. ACM SIGMETRICS Int'l Conf. of Measurement and Modelling of Computer Systems, 2008.

[5] Y Zhao & et al., "Enabling P2P One-View Multiparty Video Conferencing, Optimal Bandwidth Sharing in Multi-Swarm Multi-Party P2P Video Conferencing Systems," IEEE transactions on parallel and distributed systems, vol. 25, no. 1, January 2014 IEEE/ACM Trans. Networking, vol. 19, no. 6, pp. 1704- 1716, Dec. 2011.

[6] Xinggong Zhang, & et al. "Modelling and Analysis of Skype Video Calls: Rate Control and Video Quality, " IEEE transactions on multimedia, vol. 15, no. 6, Oct. 2013

[7] C. Fuchs, "The Implications of New Information and Communication Technologies for Sustainability," Environ., Dev. And Sustainability, 10:3 (2008), 291–309

[8] Y Zhao, Yong Liu, C. Chen, and Jianyin Zhang "Enabling P2P One-View Multiparty Video Conferencing," IEEE transactions on parallel and distributed systems, vol. 25, no. 1, Jan. 2014, pp. 73-82.

[9] V.P. Kompella, J.C. Pasquale and G.C. Polyzos, "Multicast Routing for Multimedia Communication," IEEE/ACM Trans. Networking, Jun. 1993, pp. 286-292.

[10] Xing Zhang, & et. Al., "Modeling and Analysis of Skype Video Calls: Rate Control and Video". IEEE transactions on multimedia, vol. 15, no.6, Oct. 2013, pp. 1446 – 1457