JOURNAL OF ADVANCEMENT IN
ENGINEERING AND TECHNOLOGY

Journal homepage: http://scienceq.org/Journals/JAET.php

Research Article Open Access

Development of Campus Video-Conference System Based on Peer-To-Peer Architecture

Ogirima, Sanni Abubakar Omuya, Afolabi, Adeolu .O, Adigun, Abimbola Adebisi, Akande, Noah Oluwatobi
Department of Computer Science & Engineering, Ladoke Akintola University of Technology, Ogbomoso, Nigeria

*Corresponding author: Ogirima, Sanni Abubakar Omuya
Department of Computer Science and Engineering,
Ladoke Akintola University of Technology,
Ogbomoso, Oyo State, Nigeria.
E-mail: ogirima_saow@yahoo.com, soogirima@lautech.edu.ng
Received: March 15, 2014, Accepted: March 31, 2014, Published: April 2, 2014.

ABSTRACT

Peer to Peer (P2P) systems inherently have high scalability, robustness and fault tolerance because there is no centralized server and the network self-organizes itself. This is achieved at the cost of higher latency for locating the resources of interest in the P2P overlay network. This paper describes the design and implementation of campus video conference system based on P2P architecture that was tested within premises of Ladoke Akintola University of Technology, Ogbomoso, Nigeria. The proposed Campus video conference system is made up of five modules which are the media stream engine, the conferencing control protocol, transmission module, TCP/UDP module and the user interface module. The media stream engine is responsible for audio/video capture and playback, the conferencing control protocol defines a set of conventions governing the structure and behavior of communication messages, the transmission module consists of a peer and a distribution network constituting of the peers also the delivery and exchange of streaming data while the audio manager and video manager use TCP/UDP to broadcast to other peer. The proposed system will offer smooth video conferencing with low delay and seldom and short freezes. It is believed that this videoconference system will bring video telephony to a new level of quality and will lead to a new trend in everyday communications in the university community.

Keyword: Campus, Peer-to-Peer Systems, Video Conference, Multimedia, Multipoint Control Unit

INTRODUCTION

Recent technological improvements in the domain of networks, multimedia hardware and compression algorithms have brought videoconferencing and other distributed multimedia applications to the desktop. An example of a multimedia application whose popularity is continuously increasing is videoconferencing (Dimitris and Constantin, 2007). Videoconferencing is a full-motion, two-way, video/audio system that permits two or more people in different locations to communicate with each other. Two-way videoconferencing is often used for large groups and by colleges and universities that offer video courses (Dennis, 1998). Videoconferencing is also used in distance learning and Web-based courses. Colleges and Universities have been providing live broadcasts of lectures and seminars to some of their students who are unable to travel to class sites. The justification of using videoconferencing in education may also come from concepts and principles related to social learning and social constructivist theory (Beverly, 2000). In business settings, videoconferencing is used for employee training, group work or to introduce a new product, service or procedure. Understanding what are required for videoconferencing and what software programs are available becomes more and more important to people working in education and business, and also for those who are enthusiastic about personal communication in this way.

Generally, there are two types of systems used to implement video conferencing applications. The first type is the centralized video conferencing system. This type is further classified into two systems: the Multipoint Control Unit (MCU) device system. MCU is a central device with a larger bandwidth for Internet connection than a regular participant (Ramadass, 2010). MCU has the capability to serve an N number of participants in a multipoint conference. The number of participants relies on the type of device and the Server based system; these systems use a centralized server to distribute the video signal among the participants. Each participant requires software such as WebEx 2003 to be able to log in to the conference and communicate directly with other participants. Generally, one of the biggest drawbacks of the centralized system is that it is not scalable. In addition, it requires a higher bandwidth to disseminate a single video signal among participants. The second type is the overlay network video conferencing system. As what happens in ALM, the concept of the overlay network video conferencing system is the possibility of implementing a multicast functionality at the application layer (Suman et al, 2002). The Application Layer

Multicast (ALM) approach has the ability to disseminate video signals faster, but the main disadvantage of this system is that it copes badly with a heterogeneous network. The slow node cannot bear the burden of the flow; thus, the video stream loses packets.

Video conferencing applications have scalability issue limitation. Therefore, the overlay network is used to overcome all other limitations through the utilization of peer resources (Alhamza et al, 2012). An overlay network is a computer network built on top of another network. Nodes in the overlay are considered connected by virtual or logical links each corresponding to a path, perhaps through many physical links in the underlying network (Anderson et al, 2001). Three kinds of overlay networks transfer the content parts: the Application Layer Multicast (ALM) network similar as in Pendarakis et al, 2001, the Peer-to-Peer (P2P) network such as Napster and the Content Distribution Network (CDN) (Saroiu et al, 2003). P2P is an extremely popular method in which nodes in the network, called peers, offer resources such as bandwidth, processor and storing capacity to other nodes. Consequently, as the number of users increases, the global resources of the network also grow. Peers that serve another peer can also be chosen using proximity network criteria to avoid bottlenecks.

The main advantage of P2P approach is scalability; each peer offers resources such as bandwidth, processor and memory to other nodes. Unfortunately, the bandwidth demand of video communications prevents these applications to form a multipoint (MP) video conferencing environment and limits them to only point-to-point video communications (Alhamza et al, 2012). This is mostly because of the fact that low bandwidth connections (e.g., ordinary modem over a phone line or wireless GPRS) that are barely enough for point-to-point video communications make more than one video connection infeasible. Moreover, users tend to consume as much of the available bandwidth as possible to increase their video quality and, hence, a MP video system that increases the demand cannot be possible.

P2P systems became extremely popular in a short time as a file sharing mechanism. The distributed architecture where each peer could act as a server and a client at the same time, created a difference from the traditional client/server models. This has triggered P2P systems to find diverse applications. One of these applications is media streaming. Most of the recent techniques for P2P media streaming on the Internet utilize multicast model at the application layer. The main benefit of implementing application layer multicast is overcoming the lack of large-scale IP multicast deployment at the network layer. Due to the benefit inherent in P2P systems, this paper presents a videoconferencing systems based on P2P Architecture.

Review of Related Works

Providing multipoint video conferencing service is challenging because of its high bandwidth demand and strict streaming quality requirement. Compared with traditional server-based solutions in Ramadass (2010), one of the most critical limitations facing a server-based video conferencing system is a scalability bottleneck, whereby the outgoing bandwidth of the server is shared among all concurrent clients (Alhamza et al, 2012). Specifically, the more clients there are, the lesser the bandwidth each client can have. Hence, the performance of this approach deteriorates rapidly as the number of simultaneous clients' increases. In addition, this system is considered to be expensive compared with the P2P network.

P2P systems have been employed in a wide variety of distributed applications such as File Sharing and Voice over Internet Protocol (VOIP). The ubiquity, resilience and scalability provided by P2P systems make them ideal for large scale distributed applications. A P2P based multipoint video conferencing using layered video coding with two layers at the end hosts was presented by Akkus et al, (2012). The system targets end points with low bandwidth network connections (single video in and out) and enables them to create a multipoint conference without any additional networking and computing resources than what is needed for a point-to-point conference. In contrast to prior work, it allows each conference participant to see any other participant at any given time under all multipoint configurations of any number of users, with a caveat that some participants may have to receive only the base layer video. To ensure secure transmission of video, a scheme providing authentication, integrity and non-repudiation was introduced and its effectiveness was shown through simulations.

The P2P overlay network is used in different applications, for example, file sharing (Cohen, 2003), video streaming (Xin et. al, 2005) and video conferencing (Akkus et al, 2011). On the other hand, most P2P video conferencing systems use fluid encoding to distribute the video stream among participants as in Vanets (Hossain et al, 2009). Vanets is a P2P video conferencing system that distinguishes between active and passive participants (active participants are producers of video stream, whereas passive participants represent viewers only). Vanets takes advantage of transcoding to allocate streaming rates optimally for all participating peers in the conference (Vetro et al, 2003). In other words, transcoding can change the bit rate to meet the requirements of peers (Xin et. al, 2005). In transcoding, the video signal is changed by the relaying peer to meet a lower encoding rate through either re-encoding or changing key parameters such as the quantization values of (Suman et al, 2002).

Alhamza et al, (2012) proposes a Peer-to-Peer Video Conferencing Using Hybrid Content Distribution Model, a server-based infrastructure is modified into a peer-to-peer video conferencing system while preserving the same functionality and features of the existing multimedia conferencing system. The modification was achieved using a hybrid content distribution model, which is a combination of fluid and chunk content distribution models to distribute parts of the video stream fairly among participants. The hybrid content distribution model offers a better way of handling heterogeneous networks because it can distinguish between a fast peer and a slow peer, dealing with each one according to its capabilities. In the proposed system, the function server was not used for video distribution; instead, it was only used for monitoring and controlling the peers to reduce the burden on the servers. In a way, the proposed model was able to overcome the problem of scalability and a bandwidth bottleneck on the main server and achieve good way to distribute video chunks.

The multi-rate and multiply P2P video conferencing system in Ponec et al (2009) proposes that different receivers in the same group can receive a video at different video rates. In this system, an optimal set of tree structures is determined for routing multi-rate content using scalable video coding (Schwarz et al, 2009). This system divides peers into many groups. Each group can be represented by a tree and each peer in the tree can receive different video rates rather than a single rate (Chen et al, 2008).

Developing videoconferencing systems over Internet or Local Network is a challenging task, since audio and video communications require high bandwidth and low latency. Therefore, it is particularly difficult to develop scalable systems that support high number of users with various capabilities. Current videoconferencing systems such as IP-Multicast and H.323 cannot fully address the problem of scalability and universal accessibility (Ogirima,2011). Therefore, this paper is to design and implement a video-conferencing system based on client-server and peer-to-peer architecture to meet our campus need. To attain the higher scalability, two strategies are incorporated i.e., the hierarchical network structure and the promotion of active and/or rich peers.

MATERIALS AND METHOD

The detailed steps taken to design and implement the proposed campus video conference system based on P2P architecture are outlined in this section.

Architecture of the Proposed System

The proposed Campus video conference system composed some of the modules below. The core modules as showed in the figure below are the media stream engine and the conferencing protocol. The media stream engine is responsible for audio/video capture and playback. The conferencing control defines a set of conventions governing the structure and behavior of communication messages.

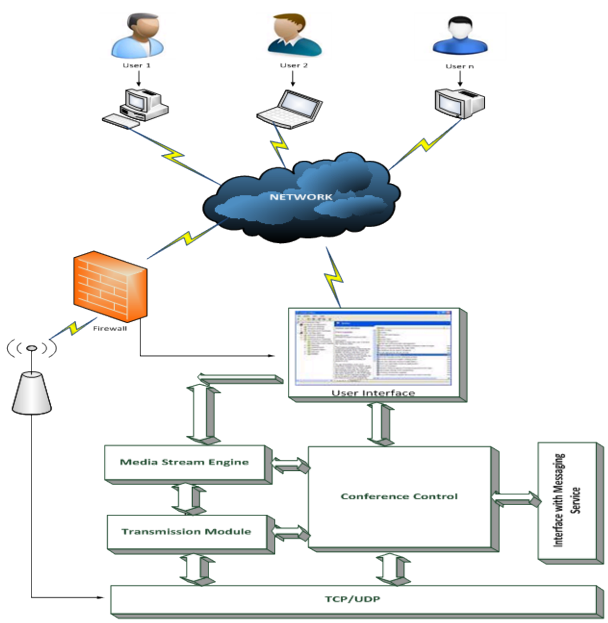

Figure 1: Architectural framework of Campus Video conference system

Media Stream Engine

The audio-video manager is the central repository and container class for all activities related to the peer-to-peer conferencing support. It contains the implementations of the conferencing session control protocol, the network socket thread procedure which receives UDP network requests, the send command methods which provide support for sending the UDP request packets to remote peers. It contains instances of associated classes which provide the audio and video streaming and the codec implementations. It provides the conference management implementation.

Some unicast audio endpoints have to rely upon an extra audio mixer to create a mixed audio stream from all the audio streams in the session. Since the audio mixer reduces the number of the streams that audio endpoints need to receive, audio mixing can relieve the processing overhead in the endpoints and reduce some audio traffic across the network. Video mixing is particularly important for users who cannot receive more than one video stream. By video mixing, these users get the pictures of multiple participants in a meeting through one video stream. Furthermore a video mixer can also decrease the number of video streams so that the bandwidth consumed by video traffic could be lowered. This optimization seems to be very necessary to support large scale videoconferencing. Image grabbers capture video streams and save them as static JPEG files. These snapshot pictures can be embedded into the control panel of each endpoint, which visualize the videoconference system set in the session.

a. Audio Mixer

Audio Mixer provides audio mixing services. An Audio Mixer can have any number of audio mixers as long as the host machine can handle. An audio mixer receives the streams from the broker network and publishes the mixed streams back on the broker network. Clients receive the mixed streams by subscribing to the mixed stream.

While some audio codecs are computing intensive, some others are not. Therefore the computing resources needed for audio mixing change accordingly. Audio mixing units need to have prompt access to CPU when they need to process received packages. Otherwise, some audio packages can be dropped and result in the breaks in audio communications. Therefore, the load on audio mixing machines should be kept at as low as possible.

b. Video Mixer

There are a number of ways to mix multiple video streams into one video stream. One option is to implement a picture-in-picture mechanism. One stream is dedicated as the main stream and it is placed in the background of the full picture. Other streams are imposed over this stream in relatively small sizes. Another option is to place the main stream in a relatively larger area than other streams. For example, if the picture area is divided into nine equal regions, main one can take four consecutive regions and remaining regions can be filled with other streams.

In our case, we choose a simpler mechanism. We divide the picture area into four equal regions and place a video stream into each region. This lets a low end client to display four different video streams by receiving only one stream. *Video Mixer* can start any number of *Mixers*. Each video mixer can mix up to 4 video streams.

Transmission Module

This module consists of a peer, and a distribution network constituting of the peers. Delivery and exchange of streaming data, i.e., video and audio are done through the distribution network. For low bandwidth requirement and management cost, we adopt shared-tree architecture to the distribution network. The distribution network consists of the core network and the distribution trees whose root is connected to the core network as shown in Figure 2.

As far as both of delay in each of distribution trees and that among them is small enough, the number of participants can be easily increased by connecting more distribution trees by the core network. In this paper, we call a peer which belongs to the core network leader peer, and all other peers' general peers. A leader peer manages the IP addresses of neighboring leader peers in the core network and its entire direct child peers. A general peer keeps the IP addresses of its parent and children. In addition, it manages the list of the IP addresses, which it knows, in its ancestor list. Peers have a limitation on the number of acceptable children called fan-out, denoted as f, in accordance to their available bandwidth.

The overview of the system behavior is as follows. First, a newly participating peer sends requests to other peer. At this time, the participating peer is notified whether it should become a leader peer or general peer. Next, it connects to either the core network or a distribution tree to join the conference. Then, the participant is involved in the conference as a speaker or an audience in accordance with the agenda. Streaming data from a speaker is once transmitted to the root of the tree to which it belongs, and then broadcasted to the other peers in the tree and to peers in the other trees via the core network.

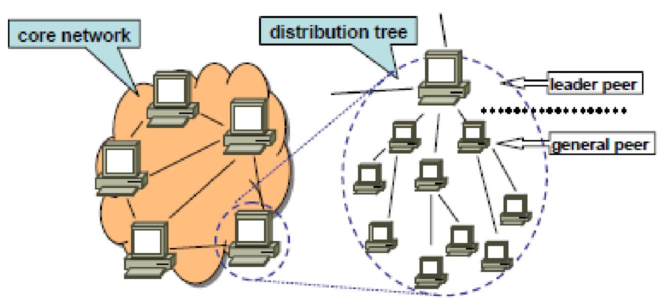

Figure 2: A hierarchical distribution network within the Campus of LAUTECH, Ogbomoso

Campus Conference Control Protocol

Our protocol is designed along with the application scenarios and it should provide the quality of service needed by a multiparty video conferencing system. As we all know, video conferencing is highly bandwidth demanding and has very stringent requirements on transmission delay. No matter which media transmission structure we use, Unicast or application-level multicast, more members in a conference leads to a lower level of quality of service.

Therefore, in our protocol, we provide a mechanism to control the number of conference members, so that we can provide better service to those members who joined the meeting earlier than others. This protocol is so concise that it uses only four communication messages:

- ***JOIN***: This message is sent from a new user (e.g. *N*) to an existing conference member (e.g. *M*). It contains the member information of the new user, such as the display name and the new user IP address.
- ***ACCEPT***: This message is in reply to the *JOIN* message, if the conference member *M* wants to accept the newcomer *N*. It consists of *M*'s member information, as well as the member list in *M*'s view.
- ***REJECT***: This message is also in reply to the *JOIN* message, if for any possible reason, member *M* rejects *N*'s join request.
- ***LEAVE***: This message is sent from a leaving member to all the other conferencing members to politely inform them of his leaving.

A typical scenario is when a new user *N* is invited to an existing conference. *N* is required to build communication channels with all the other conference members in order to keep the full mesh complete. Figure 3.3 illustrates the process of making connections between *N* and *M* (an existing member). Supposing there are three attendees (*A*, *B* and *C*) in this conference before *N* joins, this process will be executed three times.

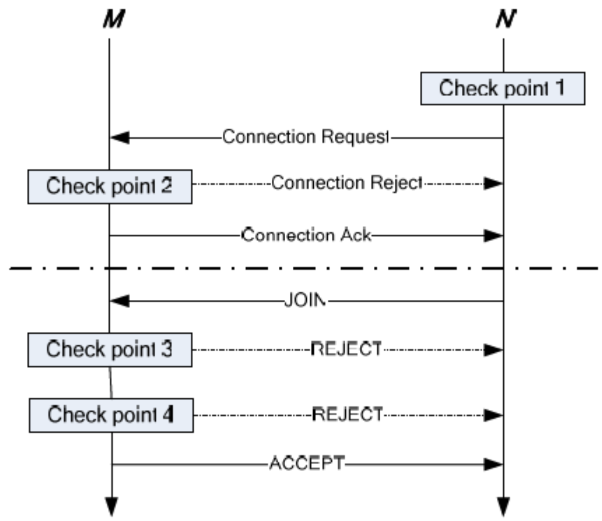

Figure 3: The process of building communication channel

In figure 3, the two vertical lines are the time line. *M* is an existing conference member and *N* is a new joiner. The actions above the bold dotted line make a TCP connection, while the actions below build a communication dialog. When *N* accepts an invitation, it sends a Connection Request to its inviter. If the request is acknowledged, *N* will send a *JOIN* message to *M*. *A* respond with an *ACCEPT* message that lists the IP address of all the conference members (i.e. *A*, *B* and *C*) in the conference. *N* receives the *ACCEPT* message and connects with members who are not in its connection list yet. In this way, a full connection mesh of four members is built. Please note that there are four check points listed in this figure. They are designed to deal with concurrency problems. If any of these check points fails, the process will terminate and the new member *N* will not be able to join the conference.

TCP/UDP and Messaging Services

Audio manager and video manager use UDP to broadcast to other peer. Port 5013 and 5014 is reserve respectively for this process. Message Services (MS) manager makes use of TCP to perform its own process. MS makes use of Port 5015 to perform message services such as call initiative

Firewall: Firewall is software that checks information coming from the internet or a network, and then either blocks it or allows it to pass through to the attempted system depending on the firewall settings. Allowing information through the firewall, sometimes called unblocking, is when an exception is created to enable a particular program to send information back and forth through the firewall.

User interface: User input data (diagnosis request) through the user interface, which consequently calls the knowledge base, feeding the user input data, the knowledge base is being consulted then the inference engine comes to a final diagnosis, which is displayed by the user interface to the user.

Interface with Messaging Services: It is an interface for business applications to invoke message handler functionality for sending and receiving messages. Similar to ODBC, JDBC, and other abstract service interfaces, it exposes the message handler functionality as a defined set of APIs for business application developers.

Hardware Requirement

One of our major goals was to keep a simple, low-cost setup that is affordable for everyday use. We wanted to avoid solutions using expensive systems, high-bandwidth networks or costly proprietary conferencing software. Our desktop-based augmented reality setup for each client consists of a 1.5GHz PC, 512Mb RAM, Webcam of any configuration, speaker, and external or internal microphone.

Study area and sample size

The population of the study conducted within Ladoke Akintola University of Technology, Ogbomoso, Nigeria, it constitutes the Undergraduate students, Postgraduates and Lectures. Purposive sample is drawn to aid the ease of data collection. The respondents were allowed to test the proposed system to give their view which will serve the research purpose.

A total of one hundred (100) copies of questionnaire were distributed among the respondents (Undergraduate students, Postgraduates students and Lectures). The respondents give their view based on usability, speed voice & picture clarity and network bandwidth of the system.

Among the respondents are:

i. Undergraduate students = 50
ii. Postgraduate students = 30
iii. Lecturers = 30

Method and Tools for Data Analysis

Microsoft Excel was used to capture and analyze the data obtained from the duly-filled copies of questionnaire while frequency and percentage distributions were the descriptive techniques used. The descriptive survey was adopted to obtain the opinion of a representative sample of the target population.

RESULTS AND DISCUSSION

Figure 4: Running environment of the Proposed system.

Figure 4 shows the running environment of the propsed application. User of the system must enter their peer IP address in order to join the videoconference system as shown in Figure 5. Clicking on “start button” trigger the call initiative process. It will send “ JOIN” message to other peer system. Figure 6 shows the message dialog form that will be display on other peer system.

Figure 5: User vedio conferencing Interface .

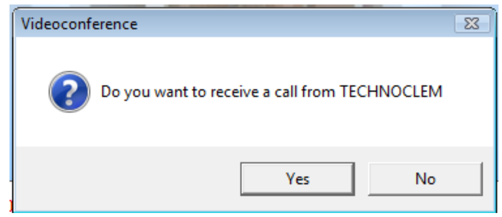

Figure 6: Message dialog Box

If the user of other peer system click on “Yes” button, then the one issued the “JOIN” message will be allow to join the videoconference system, otherwise “REJECTED” message as shown in Figure 7 will be display.

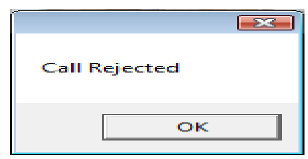

Figure 7: Call rejected Box

Campus Video Conferencing

As shown in figure 8 below, once a user has logged in successfully, the video conferencing arena is displayed which enabled the user to view other users online and start discussion. Once a user starts a discussion, a request will be sent to the intended user and on acceptance of the request, the discussion will be initiated but if the request is rejected the discussion will not be initiated.

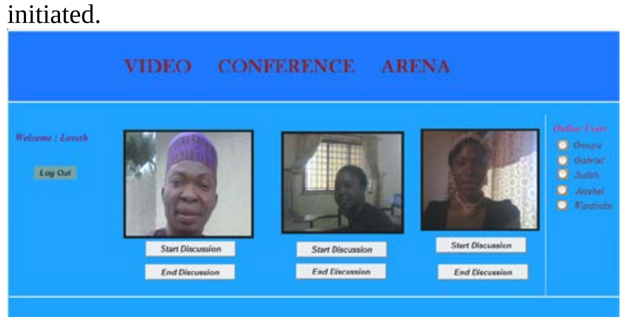

Figure 8: Proposed Campus Video Conference Interface

The assessment carried out in this work was based on users' preference of a chosen platform to access the proposed system in terms of usability, speed, and voice & picture clarity and network bandwidth of the system.

Figure 2 shows the distribution of the occupation of the respondents in the study area. 50 % of the respondents were students, 30% were postgraduate students while 20% were lecturers.

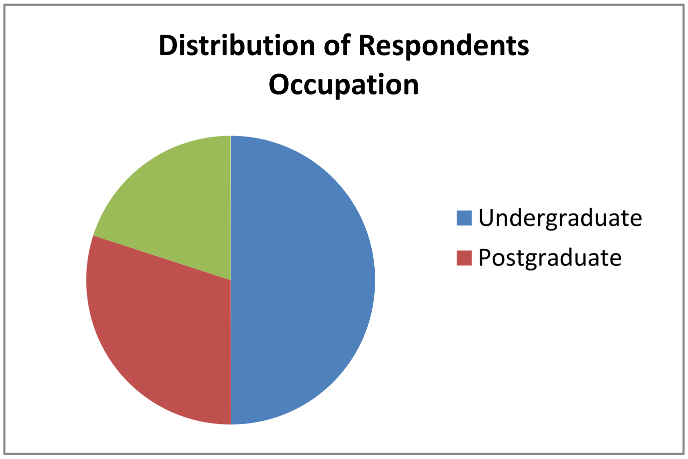

Figure 2: Distribution of Respondents Occupation

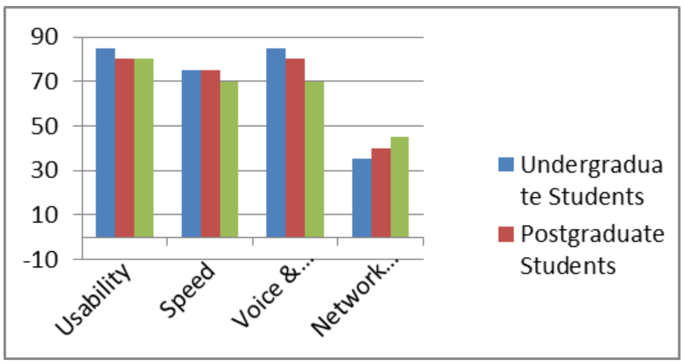

Figure 3: Results of Respondents assessment based on system Usage

The evaluated result obtained from respondents' data analyzed using Microsoft Excel's descriptive techniques including frequency and percentages show the following observations: all the respondents were able to use the developed system very well, 85% of the undergraduate students rated it high, including those that have once used a teleconferencing application and those that have not. Also, 80% of both postgraduate students and lecturers were able to use the developed system effectively rated it 35%.

The respondents were allowed to interact with the system for 30 minutes each and within the period of interaction, the system recorded a high speed with no voice and picture overlap, this shows a strong improvement over other teleconferencing systems and this can be credited to the network construction method which uses a scalable P2P conferencing system.

The respondents put the bandwidth consumption at 40% and this is considerably better when compared to other teleconferencing application and this was attributed to the shared-tree architecture used in the distribution network which is made up of the core network and the distribution trees whose root is connected to the core network.

CONCLUSION

Videoconferencing can enable individuals in distant locations to participate in meetings on short notice, with time and money savings. Technology such as *VoIP* can be used in conjunction with videoconferencing to enable low-cost face-to-face business meetings without leaving the desk, especially for businesses with widespread offices within the campus of LAUTECH, Ogbomoso. Nigeria.

The developed video conference system is based on P2P architecture, a network construction method for a scalable P2P conferencing system was used and it was showed that this mechanism can offer smooth video conferencing with low delay and seldom and short freezes. It is believed that this videoconference system will bring video telephony to a new level of quality and will lead to a new trend in everyday communications.

REFERENCES

1. Akkus, I.E., M.R. Civanlar and O. Ozkasap (2011), "Peer-to-peer multipoint video conferencing using layered video", Journal of Network and Computer Applications, Vol 34 (2011) 137–150
2. Alhamza Munther, Salah Noori, Azlan Osman, Ayad Hussain, Imad Jasim , Ali Shanoon (2012). "Peer-to-Peer Video Conferencing Using Hybrid Content Distribution Model", Journal of Computer Science Vol. 8, No 7: 1134-1142, 2012, ISSN 1549-3636.
3. 3.Andersen, D., H. Balakrishnan, F. Kaashoek and R. Morris (2001), "Resilient overlay networks", SIGOPS Proceedings of the 18th ACM Symposium on Operating Systems Principles (SOSP), (SOSP' 01), Banff, Canada, pp: 131 145
4. Beverly Abbey (2000), "Instructional and Cognitive Impacts of Web-Based Education", Pennsylvania: Idea Group Publishing.
5. Bram Cohen (2003), "Incentives Build Robustness in BitTorrent", the Pennsylvania State University
6. Chen, M., M. Ponec, S. Sengupta, J. Li and P.A. Chou (2008), " Utility Maximization in Peer-to- Peer Systems", Proceedings of the International Conference on Measurement and Modeling of Computer Systems, Jun. 2-6, ACM Press, Annapolis, MD, USA., pp: 169- 180.
7. Dimitris Thanos, Constantin Arapis (2007), "Teleconferencing Applications: A Survey", Communications of the ACM, Vol. 40, No. 4, April 2007, pp. 79-89.
8. Dennis O Gehris (1998) Using Multimedia Tools and Applications on the Internet. California: Integrated Media Group.
9. Hossain, T., Y. Cui and Y. Xue (2009), "Vanets: Case study of a peer-to-peer video conferencing system",
10. Ogirima, Sanni Abubakar Omuya (2011) "Design And Implementation Of A Video-Conferencing Based On Client-Server And Peer-To-Peer Architecture" Unpublished B.Sc(Hons.) dissertation, Department of Computer Science , Faculty of Communication & Information, University Of Ilorin, Nigeria
11. Perlman, R. (2004), "Models for IP multicast", Proceddigns of the 12th IEEE International Conference on Networks, Nov. 16-19, IEEE Xplore Press, pp: 678- 682.
12. Pendarakis, D., S. Shi, D. Verma and M. Waldvogel, (2001) ." ALMI: An application level multicast infrastructure", Proceedings of the 3rd USENIX

Symposium on Internet Technologies and Systems, Mar. 26-28, San Francisco, California, USA.,pp: 49-60.
13. Ponec, M., S. Sengupta, M. Chen, L. Jin and P.A. Chou (2009), “Multi-rate peer-to-peer video conferencing: A distributed approach using scalable coding”, Proceedings of the IEEE International Conference on Multimedia and Expo, Ju. 28-Jul. 3, IEEE Xplore Press, New York, pp:1406-1413.
14. Ramadass, S. (2010). “Multimedia Conferencing System”, 1st Edn., VDM Verlag, ISBN- 10: 3639238524, pp: 188.
15. Saroiu, S., K.P. Gummadi and S.D. Gribble, (2003). “Measuring and analyzing the characteristics of napster and Gnutella hosts”, Multimedia Syst., 9: 170-184
16. Schwarz, H., D. Marpe and T. Wiegand (2007), “Overview of the scalable video coding extension of the H.264/AVC standard”, IEEE Trans. Circ. Syst. Video Technol., 17: 1103-1120.
17. Suman, B., Bobby B and Christopher K. (2002). “ Scalable application layer multicast”, Proceedings of the 2002 Conference on Applications, Technologies, Architectures and Protocols for Computer Communications, Aug. 19-23, ACM Press, Pittsburgh, PA, USA., pp: 205-217.
18. Vetro, A., C. Christopoulos and H. Sun (2003), “Video transcoding architectures and techniques: An overview”, IEEE Signal Process. Mag., 20: 18-29. DOI: 10.1109/MSP.2003.1184336
19. Xin, J., C.W. Lin and M.T. Sun (2005), “Digital Video Transcoding”, Proc. IEEE, 93: 84-97.

Citation: Ogirima, Sanni Abubakar Omuya, et al (2014) Development of Campus Video-Conference System Based on Peer-To-Peer Architecture. J. of Advancement in Engineering and Technology. V1I2.