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# Implementing a LAN Video Calling Application (VinceS-Tool) which Minimizes Bandwidth and Network Resources Better than Skype

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***Abstract :***
*Video calling allows for real time face to face communication. Skype is currently one of the leading video calling applications, it uses hybrid peer to peer network architecture which requires super nodes, central servers, internet connection as part of resources. Previous researches have proven that the network architecture used by Skype demands high bandwidth and expensive resources ,the researchers proposed alternative approaches for implementing video calling applications and network architectures that could help in minimizing bandwidth usage and resource requirements in comparison to Skype.The research project implements a LAN video calling application (VinceS-tool) using the proposed alternative approaches, to minimize bandwidth consumption and resource requirements during video calling.*

***Keywords:*** *Codec, Media, MOS, SIP, VoIP*

## 1. Problem Identification

### *1.1. Introduction*

Video calling connects individuals in real time through audio and video communication over broadband networks enabling visual meetings and collaboration on digital documents and shared presentations (VCAEditors, 2014). Technology such as VoIP can be used together with desktop video calling to facilitate low cost face to face communication without having to leave the desk, especially for businesses with satellite offices-video calling is also used by businesses to establish profitable relationships quickly and efficiently without leaving their place of work.

### 1.1.2. Background

Face to face communication on the internet has become a modern culture, connecting distanced peers with the common example being Skype. Video calling has become increasingly available as the technology required to support it for example smart phones, computers and the internet are now globally available.

### *1.2. Problem Statement*

Skype cannot work on a LAN without internet connection and its central server as inclusive resources. Skype Video call automatically uses the frame rate, the packet size and the video resolution in order match most available bandwidth so it does not allow use of a lower bandwidth than the available bandwidth (Cicco et al., 2011). Since Skype was built with the assumption of fast, usable and stable connections in a high bandwidth environment, complications like network bandwidth starvation, jerky video and unreliable audio arise when they are used under inconsistent bandwidth especially in developing countries like Zimbabwe. (Paul, 2006) mentions that Skype client acts as a "super node" and makes itself available to relay calls made by other users. Having numerous super nodes on a school network increases bandwidth consumption which has led to a ban on Skype on some USA Local area networks (Paul, 2006). Each super node handles several hundred clients, based on their memory and bandwidth availability (Frank Bulk, 2004).If Skype strains network bandwidth in developed countries then it brings more strain to a developing country’s network bandwidth.

*1.3. Aim*

This project aims to implement a LAN Video calling application (VinceS-tool) which minimizes bandwidth and network resources better than Skype

*1.4. Objectives*

The research objectives are to

- Provide quality of service when video calling under a low bandwidth with lesser resources (that is internet and central servers)
- Design and implement a video call application
- Measure the performance of the video call application at different video resolutions on a LAN.

*1.5. Questions to Be Answered by the Research*

- Which value for video resolution minimizes bandwidth consumption?
- How can a video call approach be achieved without the super node aspect, central server and the internet resources?
- Does VinceS-tool minimize bandwidth consumption as assumed better than Skype?

*1.6. Hypothesis*

VinceS -tool is expected to utilize less bandwidth than Skype.

$H_0$ there is no significance difference between Skype's bandwidth utilisation and VinceS-tool's bandwidth utilisation.

$H_1$ there is significance difference between Skype's bandwidth utilisation and VinceS-tool's bandwidth utilisation.

*1.7. Justification*

The success of this product will greatly benefit individuals in developing countries where unstable and low bandwidth connections prohibit them from participating in effective video calling. In addition, various results from experimentations run during the project could be helpful for future research on development of Internet based video calling solutions for low bandwidth and unstable Internet conditions.

*1.8. Definition of Terms*

Opinion Score: a scale of 1-5 in which 5 is best.

- SIP

Session Initiation Protocol. A signaling protocol developed to set up, modify and tear down multimedia sessions over the Internet.

- VoIP

Voice over Internet Protocol. The technologies used to transmit voice conversations over a data network using the Internet Protocol.

**2. Literature Review**

*2.1. Introduction*

This paper is a review of the literature on improving the video calling user experience under low bandwidth and unstable Internet connection. A good understanding of how specific degradation of Internet Quality of Service (QoS) can alter the Quality of Experience (QoE) helps to develop mechanisms to deliver acceptable content on top of poor networking conditions.

2.1.2. <u>Online Video Calling</u>

Video call applications are tools offering a virtual environment for face to face communication among geographically dispersed participants. Mobile network service providers are concentrating more on provisioning of multimedia services such as online video chatting, mobile classroom educational system, mobile entertainment and mobile Facebook in addition to its conventional voice services. However, providing such multimedia services to the end users with the expected QoE and QoS within the available bandwidth is a challenging task (Mariappan & Pandian, 2011)**.** Video call applications can be used by anyone as long as the participant has any mobile device (laptops, cell phones, or personal computers) which has a webcam and is internet enabled, to talk face to face in real time but a user has to install some software or use any of the web video call applications that have been made available by different companies (VCAEditors, 2014)

*2.2. Applications of Video Call*

The list below presents some business application of video calling with examples(Weinstein, 2012) These include In-branch/in-store remote advisors ,Adding resources to meetings in client offices ,Expanding the point of sale footprint, Enhancing the contact center experience ,Empowering remote workers ,Creating the "no-office" company , Super-charging distance education programs , Creating the "no-building" school , Expediting the hiring process and Leveraging distributed staff management.

*2.3. Factors Affecting Video Calling*

Several factors come into play when we consider quality of a video call. Below are some of the factors.

2.3.1. Bandwidth

Video chat requires real-time communication. If the application over utilizes the link, it causes unfairness to other traffic and if it under-utilizes the link, it may cause low quality of video chat (Jan, et al., 2013). The technical quality of the videoconference depends critically on bandwidth availability (Baecker, 2006) .

2.3.2. Packet Loss Ratio

Packet Loss Ratio is the fraction expressed in percentage of packets lost by the network. The maximum packet loss ratio that is acceptable depends upon the exact codec and network deployed. The following maximum packet loss ratios are recommended for MPEG-2 and H.264 codecs without error concealment and/or correction.

- Maximum of 0.1% for H.264 video codec.
- Maximum of 0.5% for MPEG-2.

When operating above these packet loss ratios, the quality and thus acceptability of video sequences drops rapidly as packet loss increases (Pinson, 2006).

| | |
|---|---|
| **MPEG-1** | The first lossy compression scheme developed by the MPEG committee, is still in use today for CD-ROM video compression and as part of early Windows Media players |
| **MPEG-2** | Standard evolved to meet the needs of compressing higher-quality video. MPEG-2 is used in today's video DVDs and digital broadcasts via satellite and cable. |
| **MPEG-4** | Has emerged as much more than a video and audio compression and decompression standard. The MPEG committee designed MPEG-4 to be a single standard covering the entire digital media workflow from capture, authoring and editing to encoding, distribution, playback and archiving H.264/MPEG-4 AVC |
| **H.264/ MPEG-4AVC** | To provide significantly enhanced compression performance and provision of a "network-friendly" packet-based video representation addressing "conversational" (video telephony) and "non-conversational" (storage, broadcast or streaming) applications |

*Table 1: Moving Picture Experts Group codecs*

2.3.3. (Compression-Decompression) Codec

In the electronic meeting context, the video stream consumes most of the bandwidth (around 10 times more than the audio) (Chen, 2002). The compression aims then to reduce the bandwidth needed when keeping an acceptable quality for the user experience. The CODEC has two components: the coder and the decoder. The coder takes the local sound and vision signals and converts them into a form that can be transmitted over a digital network. The decoder performs the reverse function i.e. it takes the remote site's digital signals from the network and converts or decodes them into a form that enables the picture monitor to display images and the loudspeaker to radiate sound from the remote site. A CODEC is thus required at each end of the link (Down, 2009).

Table 1 below shows information on Moving Picture Experts Group (MPEG) compression standards (RTC, 2013).

2.3.4. Coder Type and Properties

Coder bit rate is the amount of information (in bits per second) output by the video coder to the network, excluding all network overhead (e.g., transport & protocol overhead).Different coding algorithms will have different minimum recommended coder bit rates. For example a Minimum of 1.5 Mbps for MPEG-2, Minimum of 768 kbps for H.264, also known as MPEG-4(Pinson, 2006).

2.3.5. Frame Rate

Frame rate is the rate at which a video system can produce unique consecutive images, called frames. Frame rate is measured in frames per second (fps). A minimum of 10 fps is recommended (Pinson, 2006). Videos having frame rates of 10 -15 give poor quality of perception. Frame rate above 15 gives an acceptable level of quality. A proposal was made for a low frame rate face video transmission strategy, which can, in real time compress and transmit face video at very low bit-rates (Dias & Fernando, 2009).Scholl et al., (2005) state that when video is used only to provide a sense of presence, for example to identify basic emotions in a video chat, one frame every five seconds may be acceptable; however if complex emotions are to be portrayed, then 0.2fps will not suffice.

An example is the Portholes project which demonstrated that a frame rate of one update every five minutes could provide alertness in a work setting but may not be adequate for remote classrooms (Chen, 2002). His experiments showed that lowering the frame rate from 25 to 15 and 5 fps did not decrease a person's understanding of the content of the video and suggested that 5 fps may be the minimum required frame rate. However experiments have shown that video could still be useful at 1 fps.

2.3.6. Spatial Resolution (image size/picture ratio)

Spatial resolution refers to the number of pixels in each frame, or the dimensions of a frame in terms of height and width. The more bandwidth in a signal the more potential visual resolution the converse is also true. (Baidu, 2001)

2.3.7. Transmission rate adaptation

Research in this domain aims to optimally adapt the multimedia stream rate to user bandwidth capacity in order to provide the best QoS possible (McCanne, 1996). The main challenge is to accommodate heterogeneous environments were several users have different bandwidths (Gill, 2008).

2.3.8. Latency

Video calling connects small numbers of individuals or groups or rooms together to speak and view one another in almost real-time, response time delays may be up to a quarter second (Baecker, 2006). Audio is presented ahead of video in some video calling systems since audio requires less time to process (Chen, 2002), thus reducing latency as well as bandwidth. The conventional approach to synchronizing audio and video is to delay the audio so that the audio and video latencies are matched, however the time required to process video can exceed the maximum perceived audio latency that is acceptable in a conversation.

*2.4. Audio Quality*

Low bandwidth also affects the production of audio, resulting in a variety of audio distortions. Parts of the encoded audio may be deleted at low bitrates while coding high frequency content under low bandwidth situations (Pyykkö, 2006).

2.4.1. Audio Video Quality (bitrate ratios)

Pyykkö (2006) observed that an audio-video bitrate of 24/76 kbps (6fps) was more pleasant than 16/84 (12.5fps). He noticed that when the overall quality reduced, viewers relied more on the audio information. His study also showed that the complexity of a scene also affected the relation of audio to video. The more complex a scene was, the more bits were needed for audio in low bitrates(Pyykkö, 2006). Thus the scene complexity and the supposed clarity of the video affected the weight that was given to the audio channel. However finding the audio-video threshold ratio for different content might be vital for producing acceptable audio-video content with limited bandwidth.

*2.5. Effects of poor internet/network transmission on the quality of user experience*

Table 2 below summarizes how some important QoS degradations can affect the user experience quality (Scholl, 2005)(Chen, 2002), (Watson, 1996) and (Hargreaves, 2008).

| QoS degradation effects on user experience QoS degradation | Consequence on user experience |
|---|---|
| Long and random delays and packet loss | Directly affect fluency of multimedia stream, leading to poor user concentration |
| Frequent interruptions | Badly alter the sense of “presence”, can be very irritating for the user |
| Low bandwidth | Delivery of poor quality content which cuts off the user enjoyment and immersive experience |

*Table 2: Effects of poor network*

*2.6. Peer to Peer Network.*

Peer-to-peer (P2P) is an alternative network model to that provided by traditional client-server architecture (HKSAR, 2008) citing Intel.com. P2P networks use a decentralized model in which each machine, referred to as a peer, functions as a client with its own layer of server functionality. A peer plays the role of a client and a server at the same time. The peer can initiate requests to other peers, and at the same time respond to incoming requests from other peers on the network. It differs from the traditional client-server model where a client can only send requests to a server and then wait for the server’s response..Peer to peer networks can be roughly classified into two types “pure P2P networks” and “hybrid P2P networks”. In a pure P2P network, all participating peers are equal, and each peer plays both the role of client and of server. The system does not rely on a central server to help control, coordinate, or manage the exchanges among the peers (HKSAR, 2008)**.**

In a hybrid P2P network, a central server exists to perform certain “administrative” functions to facilitate P2P services. For example, in Napster, a server helps peers to “search for particular files and initiate a direct transfer between the clients”. Only a catalogue of available files is kept on the server, while the actual files are scattered across the peers on the network. The central distinction between the two types of P2P network is that hybrid P2P networks have a central entity to perform certain administrative functions while there is no such server in pure P2P networks. Compared to the hybrid P2P architecture, the pure P2P architecture is simpler and has a higher level of fault tolerance (HKSAR, 2008)

*2.7. Review of Skype video call applications*

According to the company’s CEO, Niklas Zennström. , Skype is both a product and the name of the company. As a product, it is the software that “gives people new power to affordably stay in touch with their friends and family by taking advantage of their technology and connectivity investments (skype.com/company, 2004). Table 3 below shows the bandwidth requirements for specific Skype calls (Skype.com, 2014).

| Call type | Minimum download / upload speed | Recommended download / upload speed |
|---|---|---|
| Calling | 30kbps / 30kbps | 100kbps / 100kbps |
| Video calling / Screen sharing | 128kbps / 128kbps | 300kbps / 300kbps |
| Video calling (high-quality) | 400kbps / 400kbps | 500kbps / 500kbps |
| Video calling (HD) | 1.2Mbps / 1.2Mbps | 1.5Mbps / 1.5Mbps |
| Group video (3 people) | 512kbps / 128kbps | 2Mbps / 512kbps |
| Group video (5 people) | 2Mbps / 128kbps | 4Mbps / 512kbps |
| Group video (7+ people) | 4Mbps / 128kbps | 8Mbps / 512kbps |

*Table 3: Skype bandwidth*

2.7.1. <u>Skype Network structure</u>

Skype uses an overlay peer-to-peer network. There are two types of nodes in this overlay network, ordinary hosts and super nodes (SN). An ordinary host is a Skype application that can be used to place voice calls and send text messages. A super node is an ordinary host's end-point on the Skype network. Any node with a public IP address having sufficient CPU, memory, and network bandwidth is a candidate to become a super node. An ordinary host must connect to a super node and must authenticate itself with the Skype login server (Schulzrinne, 2006). Skype is therefore a Hybrid peer to peer system as shown by figure 1 below, since it is a peer to peer system which depends partially on central servers or allocates selected functions to a subset of dedicated peers (super nodes) Central servers act as central directories where either connected users or indexed content can be mapped to the current location Dedicated peers direct control information among other peers (Moltchanov, 2013).

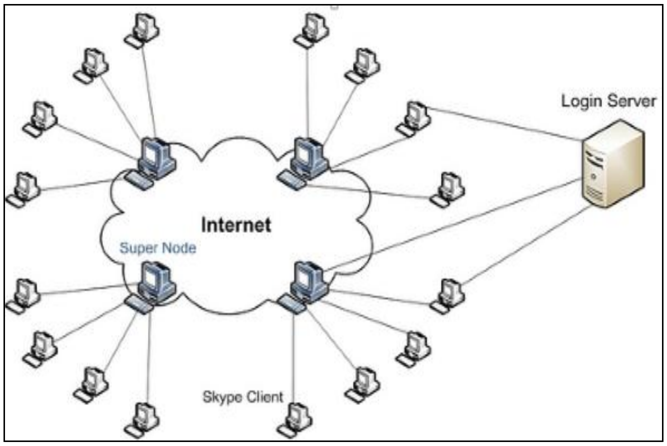


*Figure 1: Hybrid peer to peer network architecture.*

2.7.2. <u>Skype's Strengths</u>

Skype achieved its functions classified into startup, login, user search, call establishment, media transfer, and presence messages. Compared to Yahoo, MSN and Google Talk clients, Skype changes its priority to High priority, when a call is established. It has the least difference between the time the words are spoken on one voice client, and the time they are heard at the other voice client given the two voice clients are already in a voice session bridging(Schulzrinne, 2006). Skype automatically sets video resolution in accordance to the available bandwidth (community.skype.com, 2012).Skype Video call uses the frame rate, the packet size and the Video resolution in order to throttle its sending rate to match the network available bandwidth
(Cicco, et al., 2008).

2.7.3. Skype's Weaknesses

Skype's rate adaptation mechanism somehow addresses the subtle relationship between sending rate and user satisfaction; there are yet discrepancies towards consistent user satisfaction (Te-Yuan Huang, 2011). Paul (2006) also mentions that Skype client acts as a "super node" and makes itself available to relay calls made by other users. Having numerous super nodes on a school network increases bandwidth consumption which has led to a ban on Skype on some USA Local area networks (arstechnica.com, 2006).One cannot make Skype calls without internet connection on your network. It is a peer-to-peer Internet telephony application (support.skype.com, 2014).

2.7.4. Possible improvements on Skype

- Application of a pure peer to peer architecture

A pure peer-to-peer network does not have the notion of clients or servers, but only equal peer nodes that simultaneously function as both "clients" and "servers" to the other nodes on the network. This model of network arrangement differs from the client-server model where communication is usually to and from a central server. An important goal in peer-to-peer networks is that the bandwidth of all clients can be used, so the total bandwidth - and usually the available download bandwidth for the average user - grows with the number of nodes, instead of all clients having to share the bandwidth of one server, where adding more clients could mean slower data transfer for all users.

- Why pure peer to peer architecture?
  - *No overhead processing:*

  All end systems have equivalent capabilities and responsibilities and either party can initiate a communication session The participants share a part of their own hardware resources for example storage capacity, link capacity, CPU power, unlike in hybrid peer to peer were super nodes do overhead processing(Schulzrinne, 2006).

  - *Reduce costs:*

  There is no need to buy more special machines to be servers. Every computer can be a server and a client at the same time. This removes super nodes which consume more bandwidth on networks when they do video call services for other ordinary nodes (Schulzrinne, 2006).

  - *Improve Robustness and Reliability:*

  Session Initiation Protocol is a signaling protocol that is used to establish, tear down and control multimedia sessions (Schulzrinne, 2006).

In Skype SIP telephony is treated as a hybrid peer to peer system with static set of super-nodes (SIP servers). However, using a pure peer to peer architecture instead of static set of SIP servers improves the reliability and allows the system to dynamically adapt to node failures (Schulzrinne, 2006).

*2.8. Conclusion*

A good understanding of how specific degradation of Internet QoS can alter the QoE helps to develop mechanisms to deliver acceptable content on top of poor networking conditions.

It has been shown that various tools and techniques can be employed to improve video calling effectiveness, and video calling systems. In addition video call applications are sensitive to bandwidth fluctuations and unstable connections. Thus specific factors such as codec, bitrates, frame rates, audio quality, video quality and audio-video quality have to be carefully chosen to suit the user's preferences and ensure pleasant use of the software tool.

**3. Research Methodology**

*3.1. Introduction*

This project aims to develop an internet independent video calling application that has a lesser bandwidth demand and better quality of service as compared to Skype This section discusses design choices, network architecture and user interface and their corresponding implementation to meet the above project objective. The final experiment prototype should deliver:

- an audio stream conveying perceptible speech using the smallest bandwidth possible
- a video stream that provides a good sense of presence and improves the user experience

*3.2. Development Environment*

This project is implemented with Delphi Technologies. These products include the Delphi programming language which is an object oriented language and Delphi developing application software. For VinceS-tool the SIP Delphi component library for Delphi allowed to build the VoIP application, The VinceS-tool windows were are also implemented using Delphi language.

3.2.1. System Requirements

- Network: This will allow for the setup of a pure peer to peer network architecture aided by the Session Initiation Protocol to establish connection between peers.
- Computer: The computer will be responsible for running the VinceS-tool executable file; this will provide an interphase to the user to be able to control the application.
- User: Responsible for feeding a destination IP address or host name to call and carry out the video conversation.
- Webcam: Responsible for recording the video.
- Microphone and Head phone: Responsible for recording sound and playback of sound

*3.2. Design and Implementation of the Audio Sub-System*

The objective is to deliver an audio stream clear enough conveying an intelligible speech, when using the smallest bandwidth possible. The audio communication will go through six main steps: recording, compression, transmission, decompression, buffering and playback.

3.2.2. Audio Recording, Compression and Streaming

Sound recorded through the microphone needs to be digitized from its analog form. This process involves two steps

Steps:

- Sampling: consists of dividing time axis into a number of discrete blocks called samples
- Quantization: consists of dividing the signal strength into several discrete levels.

Vince S-tool was developed to use ZIP format, every sound packet is compressed using the ZIP format at 28kbps enough for the context of limited bandwidth. This format can compress without data loss and it provides a resulting clearer sound. Audio compression and streaming process is shown in Figure 2

Different algorithms and formats have been proposed for audio compression, including: A-Law, M-Law, MP3 and Groupe Speciale Mobile (GSM) (Foo et al.1999). During VinceS-tool prototyping, these formats were tested:

- A-Law and M-Law required recording the sound at 16 Khz. The uncompressed sound stream needs 128 kbps. After compression, the stream is delivered at 64 kbps which is not enough for the context of limited bandwidth.
- MP3 offers different level of compression. But with 32 kbps, the sound quality degrades substantially and the speech is hardly intelligible.
- GSM offered a constant compressed stream of 13 kbps; but as for MP3, the speech is hardly intelligible.

Figure 2 illustrates the recording and compression.

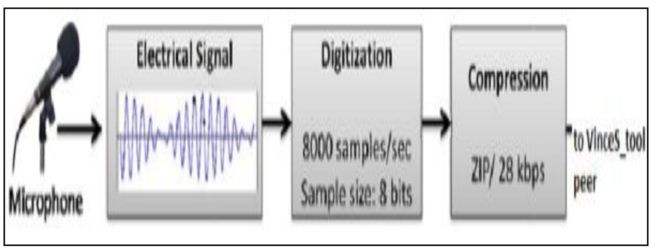


*Figure 2: Audio compression and streaming process*

3.2.3. Stream reception, decompression, buffering and playback.

VinceS-tool was developed to use ZIP format decompression to modify the audio packets received at the destination which are decompressed and buffered into a queue collection, which is an implementation of a first in first out stack. The queue size is dynamic and grows or decreases when needed, performing audio playback. Figure 3 below shows the Stream reception, decompression, buffering and playback.

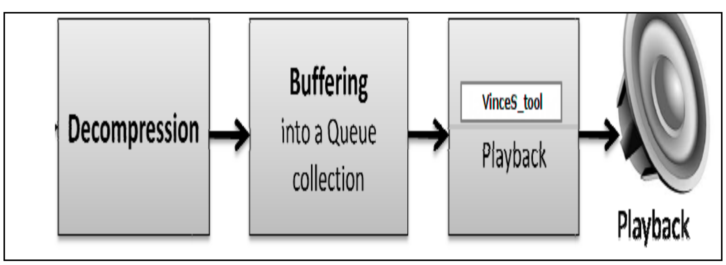


*Figure 3: Stream reception, decompression, buffering and playback.*

*3.3. Design and implementation of the video system*

3.3.1. Video Recording
The video was recorded using a webcam or any digital video recorder.

3.3.2. Video Compression
The raw video stream from the webcam is not compressed and natively has a high bit rate (around 18Mbps for a 320X240 video at 15 frames per sec). Such a bit rate can obviously not be afforded in the video calling context. VinceS-tool applies YUY2 for video compression and decompression.YUY2 is a format used by PC-based display cards, allows for variation in features like picture size and aspect ratio. This assisted varying manual selection of VinceS-tool video resolution and byte size.

3.3.3. Resolution
For experimental purposes the VinceS-tool was implemented to use an option form three different video resolutions for connection as noted in the literature review the higher the resolution the more the bandwidth consumed also the lower the resolution the lower the bandwidth required. So it used a high resolution of 640*480, average case 320*240 and best case 160*120.

*3.4. Vince S-tool (Network Architecture)*
The VinceS-tool does not have any central server but will only rely on its other VN_ tool clients under pure peer to peer network architecture .It will also eliminate the super node approach which is used by Skype's hybrid peer to peer. Since VinceS-tool will be sever less it will support instant video calling without having to login to any central server and this also reduces total bandwidth usage. Figure 4 below shows the proposed VN_tool architecture.

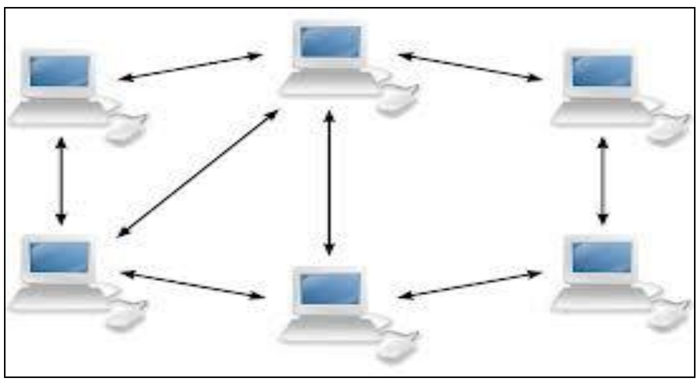

*Figure 4: VinceS-tool pure Peer to Peer network architecture.*

The application (VinceS-tool) is designed using VoIP (voice over IP) protocol allowing a pure peer to peer architecture as shown above .Peer-to-Peer (P2P) Architecture is an architecture in which nodes (peers) cooperate together to perform tasks. Each node has essentially equal importance and performs the same tasks within the network. The call establishment and call management is handled by the Session initiation protocol (SIP). The SIP server and SIP client features are embedded within the VinceS-tool (application).As a result the VinceS-tool applications on both ends can independently communicate directly without the need of another SIP agent or facility.

*3.5. User interface*
3.5.1. VinceS windows
The main window shown in Figure 5 below consists of

- Top frame which on the left side shows the identity of the host machine, in its middle is the name of the application and on the right side shows identity of the caller. On the far right are minimize and close window options.
- The middle frame is split into two, the left side is the video panel for the host machine and the right side is the video panel for calling/guest machine.
- The bottom frame consist of :
    - ➢ Settings icon on its far left, this will lead to settings window
    - ➢ Start VinceS call button, this button will establish connection when a call is requested on host machine and will send a call request to a guest machine with the specified name or IP address.
    - ➢ An IP/host name input box, this allows for specification of target machine Ip address or name. This serves as input to the VOIP connection establishment.
    - ➢ On the right is the end VinceS call button this will terminate the video call (VOIP connection).


*Figure 5: Main window*

3.5.2. <u>Settings Button in Depth</u>

The application aims to minimize bandwidth usage and at the same time, still provide good quality of service by allowing the user to specify a better performing set of video factors.

The control aspect is resolution.

- Settings window A
  - ➢ Allows for call customization, which is selecting ringtone, selecting max time for call ringing and host caller ID specification
- Window B leads to webcam options and video format options
- Window C leads to the video format factors, this option is not provided by Skype
  - ➢ Resolution
  - ➢ Codec
  - ➢ Byte size(bitrate)

Figure 6 below shows screenshots of windows A, B and C linked from the settings button on VinceS-tool

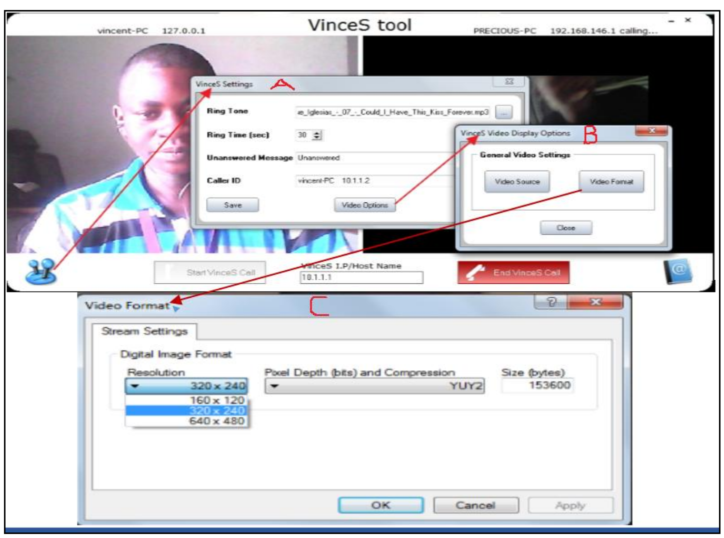


*Figure 6: VinceS-tool windows A, B and C*

3.5.3. <u>How to make a call using VinceS-Tool</u>

The following steps are a guide on how to use the VinceS-tool .

Run the VinceS-Tool .exe on host machines

- Enter the IP address or computer name of the host you need to call in the VinceS IP/Host Name box. The following Figure 3.6.3.b shows the VinceS IP/Host Name box.
- Click the start VinceS Call green button to send call connection request to host machine

- If host specification is invalid it returns an (VinceS-tool host address error)
- If host specification is valid call request is sent, it shows the calling icon and the green button turns white. In case you want to cancel the request you click on the End VinceS Call button (RED).

- On the receiving side call alert comes in the form of a ringing tone, to accept the call press the Start VinceS Call button(Green) or to reject it click on the End VinceS Call button (RED).Case 1 when call is accepted conversation begins as shown in figure 8 .

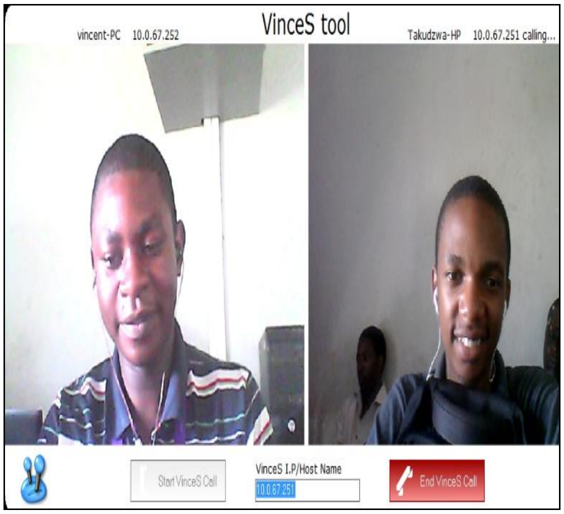


*Figure 7: Video call on VinceS-tool*

Figure 8 is a use case diagram showing what happens when making a call on VinceS-tool
Case 2
When call connection is rejected or not answered a message appears informing the caller of an unanswered call.

- To end call click on the End VinceS Call button (RED).
-

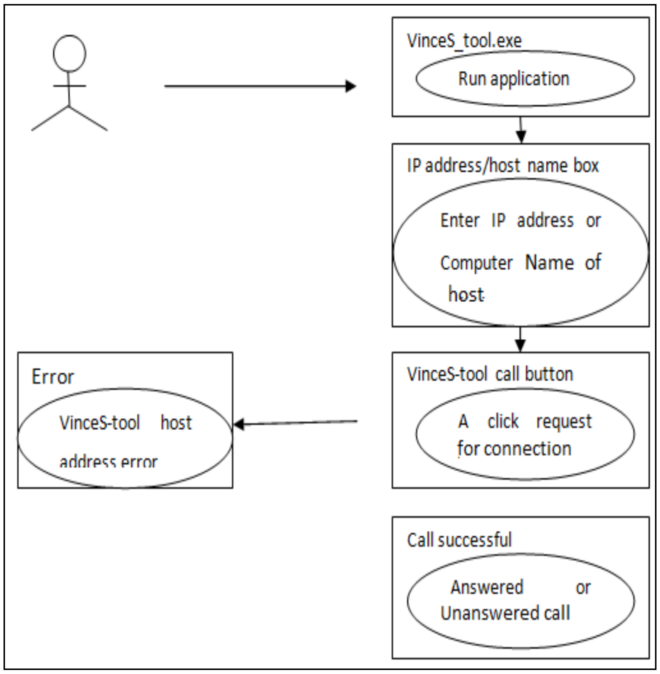


*Figure 8: Call use case diagram*

*3.6. Data Collection*

The projects main goal was to emerge with the VinceS-tool as a less bandwidth demanding application and with provision of good quality of service better than Skype.. With Net Balancer you can:

- Set for any process a download and/or upload network priority or limit
- Manage priorities and limits for each network adapter separately
- Define detailed network traffic rules
- Group local network computers and balance their traffic synchronized
- Set global traffic limits
- Show network traffic in system tray

This project used Net balancer because it has the ability to show bandwidth usage per application in this case bandwidth usage by Skype only and bandwidth usage by VinceS-tool only on a host computer. Other bandwidth monitoring tools exist but some show bandwidth consumption as a total of all running application on the computer and some are not specifically for 32 bit Windows 7 operating systems on which the VinceS-tool runs on.

3.6.1. <u>Quantitative evaluation approach (Experiment setup and procedure)</u>

The objective of the experiment was to determine which application between Skype and VinceS-tool utilizes more bandwidth when carrying out a video call conversation. Tests were carried out at Bindura University of Science Education's (Astra) LAN and the experiment was set up in computer science Lab 1.It applied two Windows 7 based laptops, each equipped with a webcam, headphone and headset.

Time for each experimental call was 10minutes. In order to minimize the experimental error and increase the accurateness of experiments, each daily experiment for a single resolution was repeated seven times and their average bandwidth was calculated. This was done for two days.

- Control factor -Resolution

Each experiment included running the VinceS-tool at 3 different video resolution all in comparison to Skype's recorded bandwidth in that experiment. The higher the resolution the higher the bandwidth consumed. VinceS-tool 9 experiments were ran in a single day as follows:

- Best case lower resolution (160*120) for lower bandwidth, 3 runs find average.
- Average case average resolution (320*240) for average bandwidth consumption, 3 runs find average.
- Worst case high resolution (640*480) for higher bandwidth consumption, 3 runs find average

Since this was done for two days in total 18 experiments were performed.

To maintain a constant number of users on the LAN all experiments were ran from 7pm till they were complete, during this time most students had gone home meaning lesser number of users on the LAN and a greater bandwidth available to experiment with. To monitor the bandwidth usage Net balancer 8.0.2 was used to indicate the maximum bandwidth used by the laptops during each experiment, after each experiment the maximum bandwidth was recorded. For this experiment net balancer was installed on a single laptop. Since the connection uses full duplex bandwidth, the maximum upstream and downstream bandwidth were added to give the maximum full duplex bandwidth.

- Experiment procedure
  - Launch net balancer
  - Launch Skype video call connection between the two laptops
  - Document the maximum bandwidth occupied by Skype application as shown by Net balancer during the 5 minute call.
  - Close the Skype application and launch the VinceS-tool video call on both laptops.
  - Document the maximum bandwidth occupied by VinceS-tool application as shown by Net balancer during the 5 minute call.
  - Change VinceS-tool resolution
  - Repeat steps 4 to 6 for average resolution then low resolution

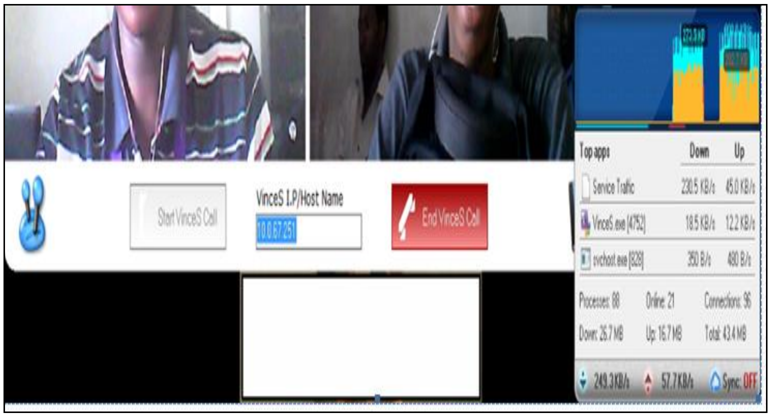


*Figure 9: Vince S-tool*

Figure 9 shows a screen shot from an experiment with VinceS-tool on its worst case with high resolution (640*480,) for higher bandwidth consumption, in this test Net balancer logged a maximum upstream bandwidth of 12.2kb/s and maximum downstream bandwidth of 18.5kb/s to give a full duplex (total) bandwidth of 30.7kb/s.


*Figure 10: Skype*

Figure 10 shows a screen shot from an experiment with Skype whose resolution is adjusted by its central servers. In this test Net balancer logged a maximum upstream bandwidth of 288.1kb/s and maximum downstream bandwidth of 447.9kb/s to give a full duplex (total) bandwidth of 736kb/s.

3.6.2 Data Analysis

Quantitative data on bandwidth utilisation by the VinceS-tool at high, average and low resolution, and the data on bandwidth utilisation by Skype were analysed using independent samples T- Test on SPSS version 16which is software that provides various functionalities for analyzing statistical data. Results were displayed using tabular and bar graph form.

3.6.3. Summary

The proposed system is based on a pure peer to peer network architecture unlike Skype's hybrid peer to peer network architecture, this eliminates central servers, super nodes and the internet.

Each peer can either record and send audio-video stream or receive and play it back using video and audio compression and decompression favorable to limited bandwidth.

Net balancer bandwidth monitoring tool was be used to evaluate if the system consumes less bandwidth than Skype.

## 4. Data Presentation Analysis and Interpretation

### *4.1. Introduction*

This section assesses VinceS-tool and Skype in terms of bandwidth utilisation on Astra's LAN. Tests were conducted in order to determine the bandwidth requirements by the VinceS-tool and by Skype. The researcher collected the quantitative data which are presented in the second section of this chapter using Net balancer.

### *4.2. Presentation of results*

When making a video call upstream and downstream bandwidth usage are considered the addition of these two gives us the total bandwidth consumed which is termed full duplex bandwidth. The tables below use this principle in representing figures for upstream, downstream and full duplex bandwidth consumption. The tables also group the experiments by the controlling factor which is the resolution (high (640*480), average (320*240) and low (160*120)). Table 5 below shows Vince S-tool and Skype experimental results from the experiments ran on day one.

| **Vince S-tool** | **Skype** |
|---|---|
| **Test 1 High resolution bandwidth (640*480)** | |
| (upstream)+(downstream)=(full duplex) 12.2kb/s + 18.5kb/s = 30.7kb/s | (upstream)+(downstream)=(full duplex) 288.1kb/s + 447.9kb/s = 736.0kb/s |
| 12.6kb/s + 17.3kb/s = 29.9kb/s | 231.8kb/s + 98.7kb/s = 329.7kb/s |
| 12.7kb/s + 21.2kb/s = 33.9kb/s | 182.3kb/s + 66.6kb/s = 248.8kb/s |
| 11.8kb/s + 20.0kb/s = 31.8kb/s | 223.8kb/s + 111.8kb/s = 335.6kb/s |
| 13.1kb/s + 20.1kb/s = 33.2kb/s | 139.1kb/s + 241.7kb/s = 380.8kb/s |
| 18.8kb/s + 22.9kb/s = 41.7kb/s | 155.4kb/s + 256.9kb/s = 412.3kb/s |
| 22.4kb/s + 11.5kb/s = 33.9kb/s | 242.2kb/s + 128.0kb/s = 370.2kb/s |
| **Test 2 Average resolution bandwidth (320*240)** | |
| (upstream)+(downstream)=(full duplex) 10.7kb/s + 11.4kb/s =22.1kb/s | (upstream)+(downstream)=(full duplex) 332.8kb/s + 122.6kb/s = 445.4kb/s |
| 10.4kb/s + 11.9kb/s =22.3kb/s | 223.5kb/s + 134.8kb/s = 358.3kb/s |
| 10.2kb/s + 12.4kb/s =22.6kb/s | 302.7kb/s + 217.5kb/s = 520.2kb/s |
| 12.9kb/s + 10.8kb/s =23.7kb/s | 170.6kb/s + 245.1kb/s = 415.7kb/s |
| 13.5kb/s + 7.8kb/s =21.3kb/s | 108.7kb/s + 212.7kb/s = 321.4kb/s |
| 10.7kb/s + 9.4kb/s =20.1kb/s | 221.7kb/s + 155.1kb/s = 376.8kb/s |
| 11.2kb/s + 11.4kb/s =22.6kb/s | 234.5kb/s + 174.7kb/s = 409.2kb/s |
| **Test 3 Low resolution bandwidth (160*120)** | |
| u(uptream)+(downstream)=(full dupleduplex) 8.8kb/s + 9.7kb/s =18.5kb/s | (upstre(upstream)+(downstream)=(full duplex) 389.2kb/s + 235.1kb/s = 624.3kb/s |
| 8.2kb/s + 10.2kb/s =18.4kb/s | 248.9kb/s + 132.7kb/s = 381.6kb/s |
| 8.5kb/s + 10.5kb/s =19.0kb/s | 265.2kb/s + 206.6kb/s = 471.8kb/s |
| 11.6kb/s + 7.1kb/s =18.7kb/s | 109.7kb/s + 264.9kb/s = 374.6kb/s |
| 10.8kb/s + 8.6kb/s =19.4kb/s | 232.5kb/s + 173.2kb/s = 405.7kb/s |
| 8.3kb/s + 9.5kb/s =17.8kb/s | 126.9kb/s + 257.7kb/s = 384.6kb/s |
| 8.5kb/s + 10.5kb/s =19.0kb/s | 157.5kb/s + 254.2kb/s = 411.7kb/s |

*Table 4 : Day 1 Experiment*

Table 6 shows VinceS-tool and Skype experimental results from the experiments ran on day two.

| **VinceS-tool** | **Skype** |
|---|---|
| **Test 1 High resolution bandwidth (640*480)** | |
| (upstream)+(downstream)=(full duplex) 14.6kb/s + 19.3kb/s = 33.9kb/s | (upstream)+(downstream)=(full duplex) 127.3kb/s + 250.8kb/s = 378.1kb/s |
| 15.3kb/s + 18.9kb/s = 34.2kb/s | 239.5kb/s + 121.2kb/s = 360.7kb/s |
| 22.9kb/s + 20.3kb/s = 43.2kb/s | 332.4kb/s + 173.2kb/s = 505.6kb/s |
| 11.6kb/s + 18,6kb/s = 30.2kb/s | 113.8kb/s + 287.9kb/s = 401.7kb/s |
| 12.3kb/s + 20.1kb/s = 32.4kb/s | 215.2kb/s + 167.4kb/s = 382.6kb/s |
| 15.8kb/s + 24.7kb/s = 40.5kb/s | 243.1kb/s + 150.3kb/s = 393.4kb/s |
| 14.9kb/s + 16.9kb/s = 31.8kb/s | 108.0kb/s + 261.8kb/s = 369.8kb/s |
| **Test 2 Average resolution bandwidth (320*240)** | |
| (upstream)+(downstream)=(full duplex) 11.3kb/s + 12.7kb/s =24.0kb/s | (upstream)+(downstream)=(full duplex) 214.9kb/s + 124.7kb/s = 339.6kb/s |

| | |
|---|---|
| 11.6kb/s + 10.4kb/s =22.0kb/s | 145.7kb/s + 152.8kb/s =298.5 kb/s |
| 10.1kb/s + 12.1kb/s =22.2kb/s | 243.1kb/s + 152.7kb/s = 395.8kb/s |
| 10.8kb/s + 13.7kb/s =24.5kb/s | 155.8kb/s + 223.6kb/s = 379.4kb/s |
| 12.8kb/s + 9.6kb/s =22.4kb/s | 298.3kb/s + 103.0kb/s = 401.3kb/s |
| 9.4kb/s + 11.5kb/s =20.9kb/s | 173.1kb/s + 207.5kb/s = 380.6kb/s |
| 9.6kb/s + 10.8kb/s =20.4kb/s | 221.0kb/s + 176.8kb/s = 397.8kb/s |
| **Test 3 Low resolution bandwidth (160*120)** | |
| (upstream)+(downstream)=(full duplex)<br>10.3kb/s + 8.2kb/s =18.5kb/s | (upstream)+(downstream)=(full duplex)<br>312.4kb/s + 231.9kb/s = 544.3kb/s |
| 9.6kb/s + 7.1kb/s =16.7kb/s | 125.6kb/s + 117.3kb/s = 242.9kb/s |
| 7.2kb/s + 9.7kb/s =16.9kb/s | 235.2kb/s + 169.4kb/s = 404.6kb/s |
| 9.3kb/s + 10.3kb/s =19.6kb/s | 181.8kb/s + 205.3kb/s = 387.1kb/s |
| 9.0kb/s + 7.4kb/s =16.4kb/s | 176.9kb/s + 231.5kb/s = 408.4kb/s |
| 6.9kb/s + 9.3kb/s =16.2kb/s | 213.7kb/s + 198.6kb/s = 412.3kb/s |
| 10.7kb/s + 7.5kb/s =18.2kb/s | 197.2kb/s + 185.5kb/s = 382.7kb/s |

*Table 5: Day 2 Experiment*

### *4.3. Analysis and Summary of Research Findings*

Bandwidth utilisation for all experiments, by the VinceS-tool at (high, average and low) resolution in comparison to bandwidth utilisation by Skype are shown in the figures below (figure 11, 12 and 13) respectively.

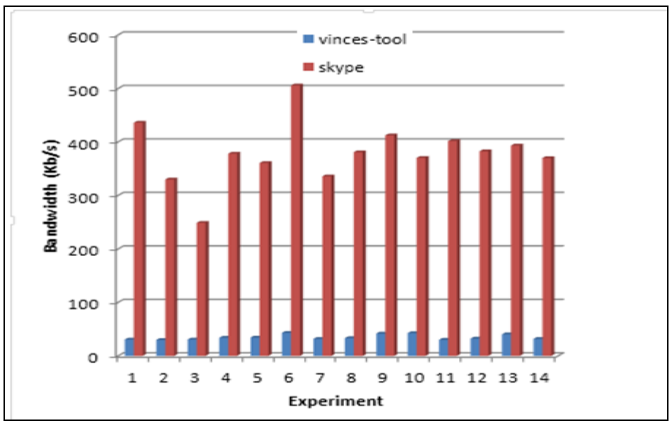


*Figure 11: Experiments with Vince S-tool at high resolution*

A comparison of bandwidth utilisation by VinceS-tool at (high resolution) to Skype showed a significant difference (P<0.05) at 95% confidence interval.

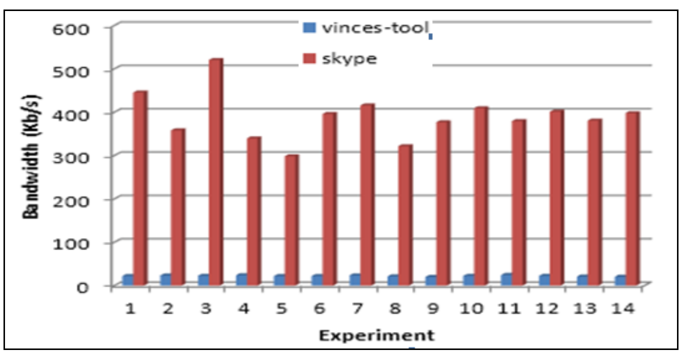


*Figure 12: Experiments with Vince S-tool at average resolution*

A comparison of bandwidth utilization by Vince S-tool at (average resolution) to Skype showed a significant difference (P<0.05) at 95% confidence interval.

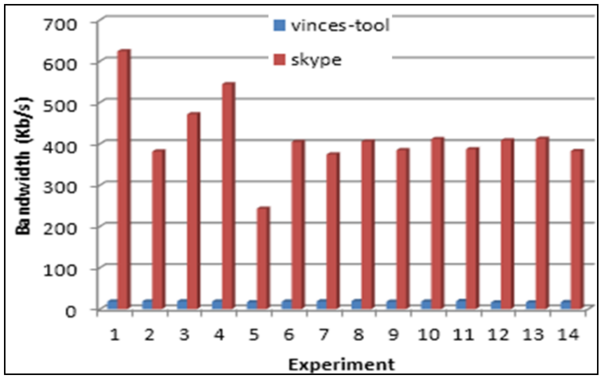


*Figure 13: experiments with Vince S-tool at low resolution*

A comparison of bandwidth utilisation by VinceS-tool (at low resolution) to Skype showed a significant difference (P<0.05) at 95% confidence interval. Furthermore a comparison of bandwidth utilisation by VinceS-tool (at all resolution data combined) to Skype showed a significant difference (P<0.05) at 95% confidence interval.

*4.4. Summary*

In all the 42 experiments VinceS-tool proved to minimize bandwidth usage, this is shown by its low figures in bandwidth consumption as compared to Skype. An analysis on the SPSS showed that there was a significance difference between Skype's bandwidth and resource utilisation and VinceS-tool's bandwidth and resource utilisation.

## 5. Conclusion and Recommendations

*5.1. Conclusions*

For this report, it was shown in the experiments that the hybrid peer to peer network architecture employed by Skype consumes more bandwidth and requires more resources which are central servers and the internet, than the Vince S-tool which applies pure peer to peer network architecture and has no need for the internet and central servers. The VinceS-tool proved to work using a pure to pure network architecture this means connection was established without the need of any central servers and the internet.

From the hypothesis mentioned in the first chapter and the experimental results the researcher concluded

$H_1$ there is significance difference between Skype's bandwidth and resource utilization and Vince S-tool's bandwidth and resource utilisation.

Therefore it is advisable to put higher priority on pure peer to peer architecture when developing LAN video calling systems since they require less bandwidth

*5.2. Future work*

The researcher focused on using pure peer to peer video calling network architecture on a single LAN, research and development should be done to implement pure peer to peer video calling network architecture on LAN to LAN connection. Video calling also exists between mobile phones as such research should also be considered in implementing a pure peer to peer video calling network architecture on these devices.

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