CHERI

January 19, 2007

Instructor Lakshman One
School of Engineering Science
Simon Fraser University
Burnaby, BC
V5A 1S6

Re: ENSC 440 Project proposal for a Voice Bandwidth Saving System

Dear Instructor One,

Please find attached the project proposal for a Voice Bandwidth Saving System (VBSS), a communication system that reduces the bandwidth usage of VOIP application. This document provides an overview of the involved technology and our proposed product. In addition, we outline our design approach, budget and sources of funding, company structure and detailed profile of our development team.

Cheri Perception is composed of four SFU undergraduate engineering students with a range of technical skills and experience in software system development: Bryan Cua, Cathy Zhang, Tilson Chung, and Hubert Pan. If you have any questions or concerns regarding our proposal, please contact me by phone at 604.619.0841 or by email at ensc440-cheri@sfu.ca.

Sincerely,

Bryan Cua

Bryan Cua
CEO of Cheri Perception

*Enclosure: Proposal for a Voice over IP Bandwidth Saving System.*

Cheri Perception, c/o SFU School of Engineering Science | Burnaby BC, Canada V5A 1S6
604.619.0841 | ensc440-cheri@sfu.ca

CHERI

# VoIP Bandwidth Saving System
# Project Proposal

# VoIP Bandwidth Saving System

## Project Proposal


| | | |
|---|---|---|
| Project Team: | Hubert Pan<br>Tilson Chung<br>Cathy Zhang<br>Bryan Cua | |
| Contact Person: | Bryan Cua<br>bcua@sfu.ca | |
| Submitted to: | Steve Whitmore<br>Lakshman One<br>Ash Parameswaran | ENSC305<br>ENSC440<br>ENSC440 |



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Revision: 1.0

# Executive Summary

Ending 2005, approximately two-thirds of Fortune 2000 companies have a certain degree of VoIP equipment installed [1]. While some companies have implemented a full VoIP phone network, others may use this new technology for long distance communication between different branches in their company. What new benefits did this technology create for these companies?

Nowadays, a majority of companies already have a network infrastructure available for connection to the internet. VoIP rides on this already pre-existing infrastructure with easily installed software compatible with most affordable, modern personal computers (PCs). These benefits allow companies to deploy VOIP systems with the required features by purchasing inexpensive hardware and maintaining a low-running cost. On the other hand, the remaining companies have to rely on specific vendors for their specialize hardware that may lack some of the required features.

**Voice over Internet Protocol (VOIP) provides the capability to break up voices into small pieces, and place them into packets that could travel through the Internet. Today, VOIP is used in many companies for their internal calls.**

So, what hinders these remaining companies from joining this trend? First, the bandwidth usage of a VoIP system depends on the number of users. However, small growing companies usually find it difficult to upgrade their network infrastructure to suit the VoIP needs. Another problem concerns the security of conversations being passed through the public (insecure) internet. Both these concerns – bandwidth and security – hinder these small companies from implementing a VoIP system.

We, at Cheri Perception, aim to produce a solution for these small enterprises. The proposed system, named Voice over IP Bandwidth Saving System (VBSS), will be able to greatly reduce the bandwidth requirement for deploying a satisfactory VOIP communication and uphold the quality and security of the transferring signals.

Cheri Perception consists of four undergraduate engineering students from SFU: Bryan Cua, Cathy Zhang, Tilson Chung, and Hubert Pan. Armed with a range of technical skills and industrial experience, we will develop our proposed system through proper software development stages and extensive testing through 13 weeks of development time. With an estimated budget of \$1740, we plan to deliver a working prototype on April 10th, 2007.

# Revision History

| Date | Version | Description |
|---|---|---|
| 01/16/2007 | 0.1 | Initial Draft |
| 01/17/2007 | 0.2 | Contribution by Tilson Chung |
| 01/18/2007 | 0.3 | Contribution by Cathy Zhang |
| 01/19/2007 | 0.4 | Contribution by Hubert Pan |
| 01/20/2007 | 0.5 | Revision by Bryan Cua |
| 01/21/2007 | 0.6 | Revision by Team |
| 01/22/2007 | 0.7 | Additional Formatting |
| 01/22/2007 | 1.0 | |

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# Introduction

As technology has improved, so has the telephone evolved greatly from its humble beginnings in Graham Bell. With the Atlantic cable of 1858, global telephony was finally possible, and hence, improved communication efficiency within large international organizations.

Enter into the new millennium. The Internet has taken its place as a replacement to data transfer between groups of people. Email, instant messaging, and, especially, online video and audio conversations allow people miles apart to communicate almost instantly, as if side by side. The use of the Internet as the channel for communication continues to grow further, fostered by its stable and developed protocol for packet transmission.

Another popular application through this protocol is the VoIP (Voice over Internet Protocol). Currently, many companies have replaced their traditional private phone system with VoIP. It allows businesses to have long distance conversations charged only on their internet connection fee. At present, developers face several challenges including quality of service and security.

A growing enterprise, in order to ensure good conversation quality, will have to continuously upgrade their existing internet connection. On the other hand, sending phone conversations through a public service poses its own security risks, mandating third-party security software.

Enter the VBSS. It acts independently of any proprietary system, and can potentially half the amount of data for a single transfer. This performance increases as the number of users grows. In addition, it ensures quality of service without sacrificing security. We, at Cheri Perception, will aim our product at small businesses, and as the VoIP market matures, we perceive larger companies, or even telephone service providers, becoming our potential customers.

This document provides a system overview, a proposed design solution, project schedule, and financial information. In addition, our team organization and company profile are included.

# System Overview

The VBSS was incepted as a filter for voice data going between two networks. Hence, it requires two sections – a transmitter and a receiver as shown in Figure 1. Imagine two distant entities, a call center and a city, both having their own internal phone network. At the moment, they have established multiple phone conversations. Assume two call center attendants, John and Jill, are currently conversing with Bob and Bill. The transmitter will merge the voices of John and Jill, and send it through the internet as a "super-packet". On the other side, the receiver will split this "super-packet" back into John and Jill's voices, and transmit them to Bob and Bill, respectively.

In more technical terms, the transmitter adds the magnitude of the two voices together. On the other hand, we implement our receiver using the Harmonic Enhancement and Suppression (HES) system [2]. The HES system assumes two speakers, one of which is stronger in characteristics, for example, in volume. It will then enhance the stronger speaker's characteristics to completely extract his/her voice. Finally, it will suppress that voice to obtain the weaker speaker's data.

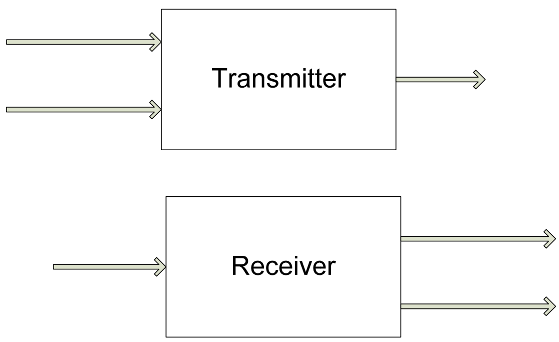



**Figure 1: System input and output**

In a very simple VoIP system, we will find a SIP Server, a router, and VoIP phones over an Ethernet network. Figure 2 shows a general VoIP system with our VBSS product. The SIP Server controls, for example, the ringing of the phones, and instructs the phones on the other party – where they should connect. The Ethernet network is their phone lines. As they transmit messages, our system will intercept it, and repackage it in a more efficient manner before sending through to their destination.

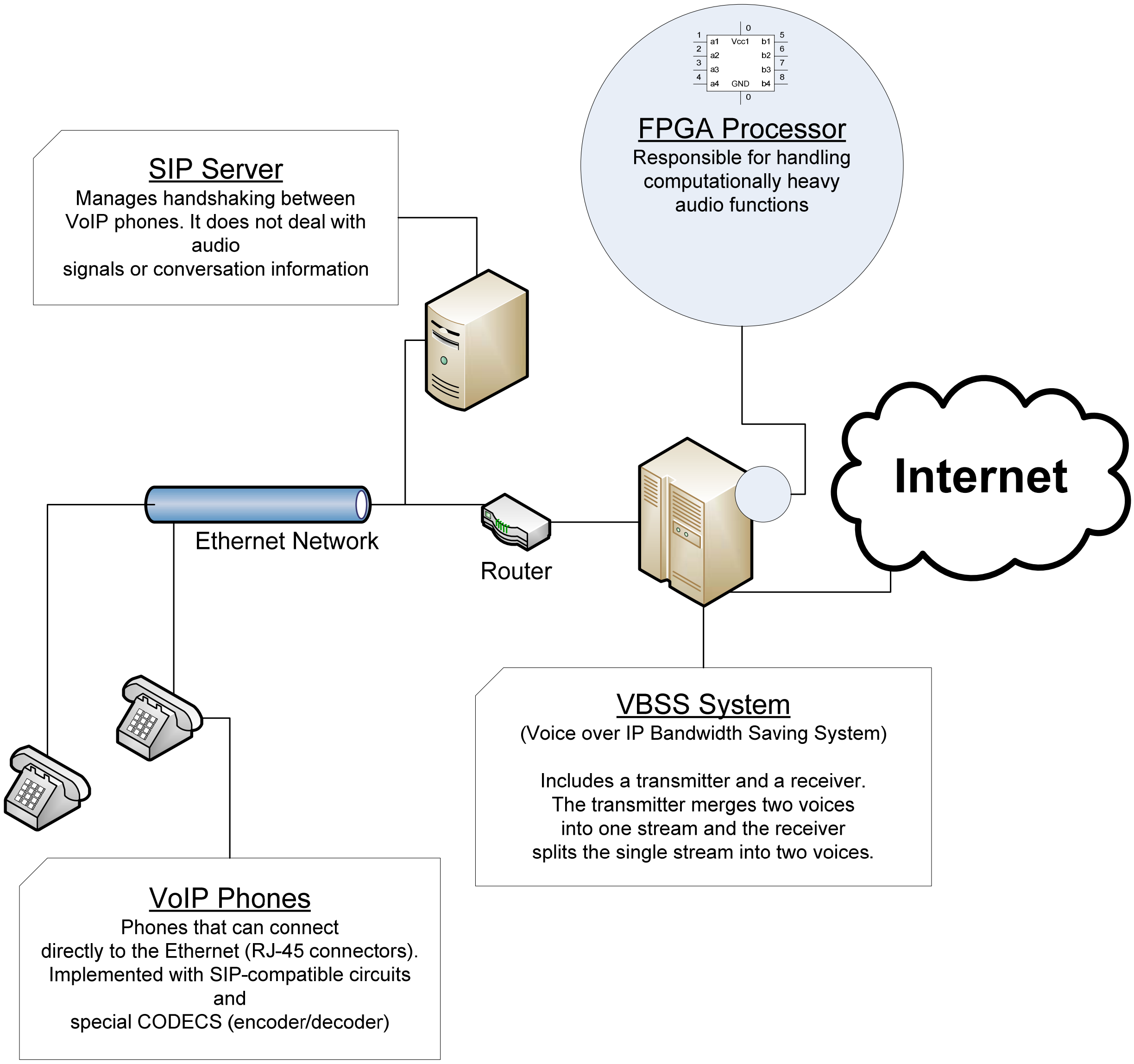



**Figure 2: System Overview**

# Possible Design Solutions

A large problem in VoIP is its use of bandwidth. Two areas in which bandwidth can be saved are the header, and the voice data. Since header compression has already been implemented by many, such as Cisco Systems®, and is orthogonal to voice compression, we will focus on the latter.

## 1. Transparent Connector

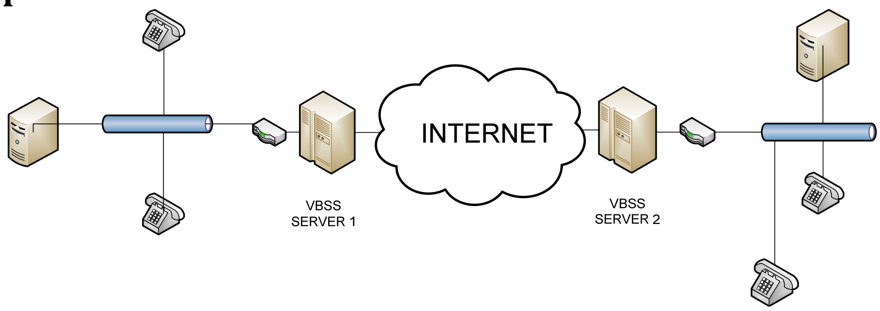



**Figure 3: Transparent Connection**

A first possibility is as Figure 3, where the VBSS server is placed outside of the phone network, thereby making the whole operation transparent to the existing system, save some additional delay time.

The transmitter will filter out packets sent by the router for voice information, and merge two like packets together. Then, it will send the packets to the internet. Likewise, the receiver will filter out packets coming into the router for voice information, and split the packet into its two parts.

## 2. Ethernet Node

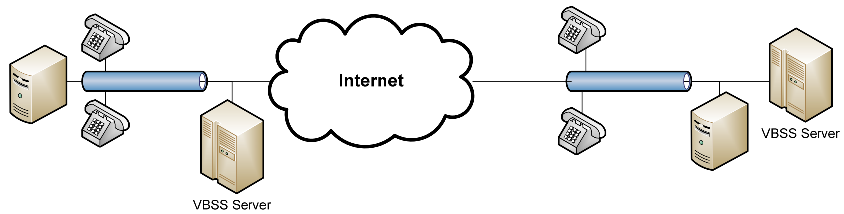



**Figure 4: Ethernet Node**

The approach outlined in Figure 4 shows a VBSS server that can be placed at any node in the network, making the system a plug-in to an existing SIP network. The transceiver will no longer need to filter voice packets from others. As well, it does not need to open the packet wrapping and rewrap. However, modifications to the SIP server will be required. Specifically, voice information must all be routed to the VBSS server before entering into the Internet.

# Proposed Design Solution

We propose to build a system transparent to existing systems in order to limit complications in the original network. Components of the system include transmitter; receiver; controller; and buffer and Ethernet interface. The Ethernet interface will be implemented on a processor while the other components on a FPGA.

## 1. Transmitter

The transmitter receives two voice segments from the controller, and adds them together. However, it may not be this simple. Additional information strategically added into the voice sample may help the receiver in its job. As well, time synchronization might need to be implemented, and so with dynamic merging.

## 2. Receiver

The receiver encompasses the most complex part of the whole system; it splits merged voice signals into their respective halves and sends them back to the controller. Another mechanism to be present in the receiver deals with missing packets. Since we are using RTP/UDP/IP, if a packet (a portion of the voice) is missing, we will have to regenerate it through the previous and next packet.

## 3. Controller

The controller's tasks include differentiating between voice and non-voice packets, and sending them to the transmitter/receiver or the buffer, respectively; determining the number of concurrent phone conversations and their destinations; and repackaging voice packets.

The condition for which a packet is to be stopped is first, it is a voice packet, and second, there is more than one conversation at a common destination. If both conditions are valid, the voice packet will be placed on a buffer for a specified amount of time, whence it will wait for the second packet so as to merge.

## 4. Ethernet Communicator

Another component of this project is the routing of packets in and out of our system, and protocol management. This component will communicate with external interfaces.

# Sources of Information

In researching and developing our project, we gather information for analysis and problem solving from a variety of sources. These sources include textbooks, publications, online information, University Faculty, and specification sheets from related products.

## 1. The Internet

The internet is a very informative source of information. Sites such as Wikipedia can give us a broader and more easily understood overview of the problem we wish to face, whereas *IEEE Xplore* [3] can give us technical papers which outline in more detail possible algorithms or methodologies we may use, and their limitations.

## 2. Publications

A book we are currently consulting is *Voice over IP Fundamentals* [4], which goes into great detail about current telephony systems and standards used in the industry. This book is beneficial because of its up-to-date information, as it is published in 2007.

Two books that have also been beneficial are *Linear Prediction of Speech* [5] and *Voice and Speech Processing* (4), as they have given us basic understanding of our topic.

# Budget and Funding

## 1. Budget

Our general approach in building the VBSS will require items as reflected in Table 1. We have discussed some of our choices with our course adviser, Lakshman One, and agree that an expensive FPGA embedded platform is necessary for our application to work under a satisfactory latency.

On the other hand, we seek the lowest-cost for the remaining items. This includes our main PC (personal computer), routers and the VOIP-phones. To cut down the budget furthermore, some members of Cheri Perception have donated their own old PC's to our project.

**Table 1: Budget required for hardware**

| Item | Description | quantity | estimated total cost |
|---|---|---|---|
| **FPGA embedded platform** | Virtex-4 ML403 (HW-V4-ML403-USA) | 1 | 773 |
| **Low-end PC** | 600MHz + 512MB RAM | 3 | 0 |
| **PC** | 1.8Ghz + 1GB RAM | 1 | 400 |
| **Routers** | Linksys routers | 2 | 70 |
| **VoIP phone, SIP compatible** | Grandstream | 4 | 180 |
| **Software License Fees** | TIMIT corpus | 1 | 117 |
| **Miscellaneous** | cables, shipment fees | n/a | 200 |
| | | **Total** | **1740** |

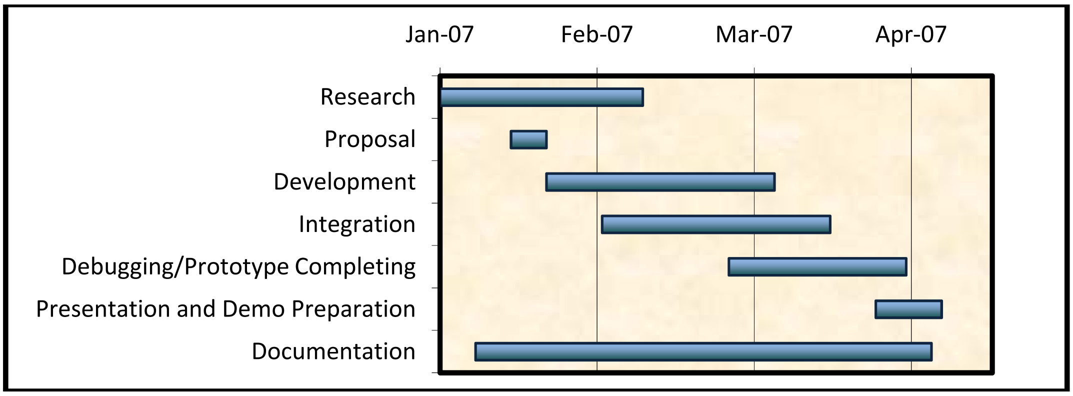
## 2. Funding

Cheri Perception is eager to find funding from multiple sources in order to subsidize for expensive parts and components. With a team of four, we are entitled to a fund of $200 provided from the School of Engineering Science. In addition, we are in the process of applying for the Engineering Science Student Endowment Fund and John Wighton Fund, which we estimated to gather $400 according to the amount obtained from previous projects.

Today, there are already 2000 patents related to VOIP being registered at US Patent and Trademark Office. The number is still growing, and some of the companies involved include Cisco, Broadcom, VTech, Ascalade and Nokia. We believe our project idea can become one of the major intellectual properties in the VOIP area, so we plan to approach some local telecommunication companies such as Broadcom and Nokia for possible source of funding.

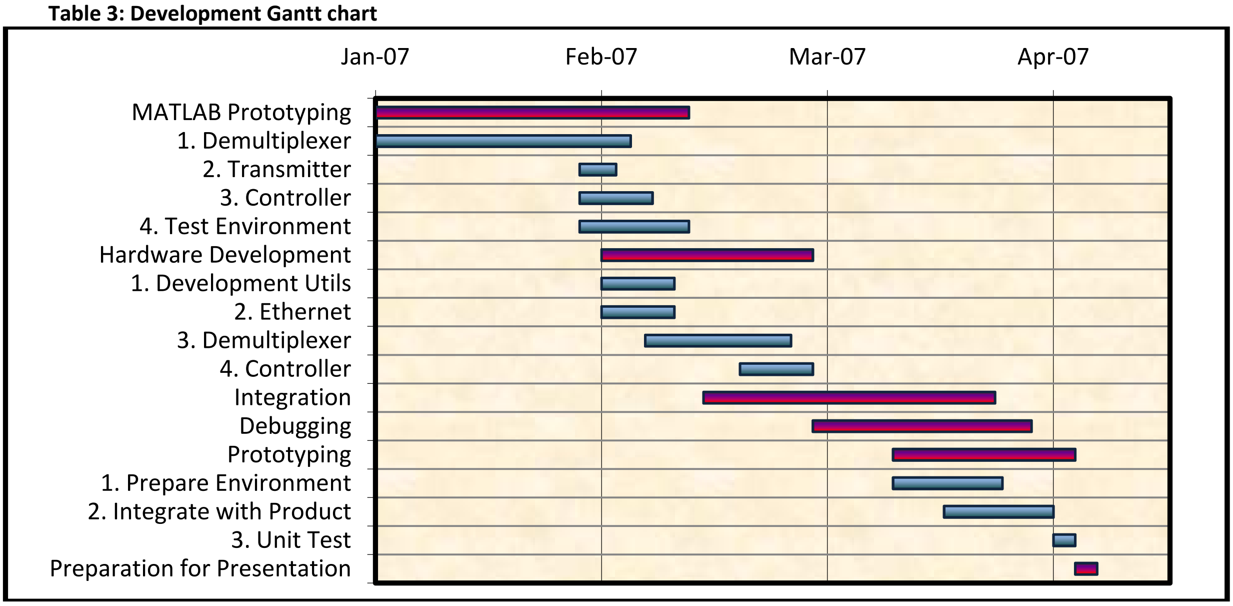
# Schedule

**Table 2: Overall Gantt chart**




Above, in Table 2, is our overall Gantt chart, describing the overlap of activities over time including research, development, integration, debugging, prototyping, presentation, and documentation. Also, the proposal has been included as it signifies the approved beginning of the project. Below, in Table 3, a Gantt chart specifically created for developmental activities, but also includes integration, debugging, prototyping, and demonstration preparation. Lines in red indicate large milestones while blue are subsections.

**Table 3: Development Gantt chart**



The milestone chart below (Figure 5) displays our milestones and date at which we wish to finish them. Special note has been taken to include *Hardware (Implementing Demultiplexer)* because the demultiplexer, or receiver, is a very large portion of our system.

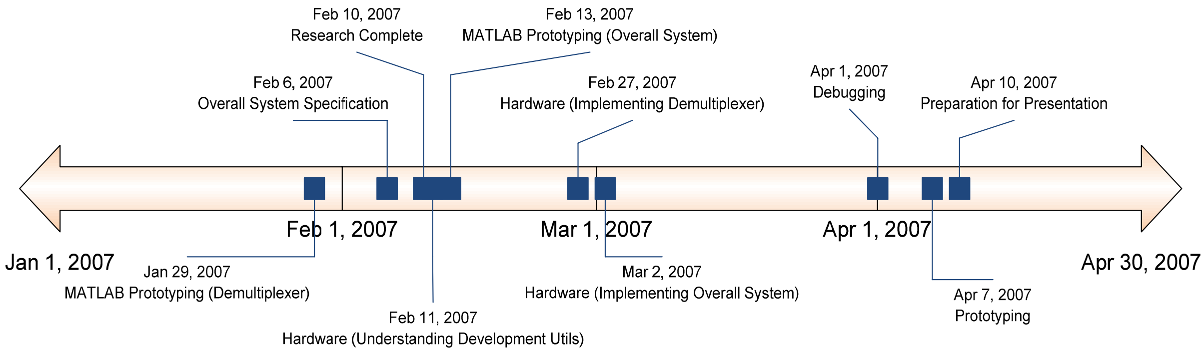



**Figure 5: Milestone chart**

# Team Organization

Cheri Perception consists of four hard working and talented undergraduate engineering students from Simon Fraser University, Bryan Cua, Tilson Chung, Hubert Pan, and Cathy Zhang. We have knowledge and experience in the areas of electronics, computer programming, and systems design. All members have different program specialization interests, and each of the team members will contribute his/her expertise to make this project a success.

Our structure is loosely organized in which each team member is responsible for a specific area of operation according to his/her expertise and related experience. **Bryan Cua** is the Chief Executive Officer (CEO) and is responsible to oversee the entire project and to maintain team organization. **Tilson Chung** is the Chief Technical Officer (CTO), responsible for project specifications and development of the project. **Hubert Pan** is the Chief Operating Officer (COO), in charge of the entire operation in regards to the project. **Cathy Zhang** is the Chief Financial Officer (CFO) and is responsible for all financial concerns, such as managing the budget and resolving financial issues.

All our team members meet regularly every week to ensure proper group dynamics and to discuss project progress. Our team depends much on the SFU Caucus System and Windows Live Messenger, where these allow each of us to post the information we find useful to share and to discuss ideas to increase efficiency and productivity whenever we cannot meet in person.

In order to ensure that all the team members are working on the project in the proposed direction, we keep meeting minutes and send it to all group members. During the meeting, all team members share their thoughts, ideas, information and solutions to the projects, and together we discuss the issues or problems we faced, then we finally determine the next step we should progress to. All decisions made in this project are agreed upon by all team members, and are all fully discussed to ensure the best outcome.

# Company Profile

**Bryan Cua – Chief Executive Officer (CEO)**

Bryan Cua, a systems engineering student at Simon Fraser University, is currently working as an intern in HSBC doing software development for web applications using J2EE and Portlet, and is also familiar with C, C++, and VB. As well, he is designing a website for Vango Intercultural Network Canada Inc. and works with aTIMI Media. As a student, he has taken courses in the analysis and design of linear feedback/modern control and real-time embedded systems. In his spare time, he likes to play the piano, from which comes his interest in audio.

**Tilson Chung – Chief Technical Officer (CTO)**

Tilson Chung is a fifth year electronics engineering student at Simon Fraser University. He has worked as a embedded system programmer for Dr. Scratchley at SFU, where his responsibility was to analyzes real-time multi-thread activities. During his time in SFU doing research with Dr. Payandeh, he developed image processing software for MIROHOT (Micro Hockey Tournament). In addition, he has worked at VTech Telecommunication Company to assist in the development of Automated Testing Environment to develop mute-tone detection circuits, Bluetooth interface PCB, and DSP function programming.

**Hubert Pan – Chief Operating Officer (COO)**

Hubert Pan is a fifth year computer engineering student at Simon Fraser University. He has worked as a java programmer for a Rough Mill Project with the National Research Council, where he implemented a program that simulates a rough mill process with agent architecture. He is also experienced in other programming languages such as C, C++ and visual basic. During his time in Mustang, he was responsible in designing a cryptography system for securing company documents.

**Cathy Zhang – Chief Financial Officer (CFO)**

Cathy Zhang is a fourth year electronics engineering student at Simon Fraser University, and has finished work terms doing automated testing environments in Fortinet, from which she learned about protocol security, and VTech, in which she tested various typ including the WIFI phone.

# Conclusion

Internet communication technology is still developing; with large room for improvement, there are many new technologies waiting to be discovered. It is no surprise that VBSS, in its future endeavors, be an even more revolutionary technology in its capability to save bandwidth and ultimately lower running costs of, hopefully, any VoIP network.

Our group will accomplish the first VBSS implementation for SIP based networks as per the schedule included in this document, with a prototype available by April 10th, 2007. We will focus on satisfying the main specifications for this system within this 3 month period. Our funding will mainly be from the Engineering Science Student Endowment Fund and telecommunication companies.

At the end of this period, we expect to see a newborn product, working in line with existing technologies to better serve the community.

# Sources and References


[1] Deloitte Touche Tohmatsu. [Online] 2004. www.deloitte.com/dtt/research/0,1015,cid=64018&pv=Y,00.html.

[2] *Cochannel Speaker Separation by Harmonic Enhancement and Suppression.* **Morgan, P. David, et al.** 5, s.l. : IEEE Transactions on Speech and Audio Processing, 1997, Vol. 5.

[3] *IEEE Xplore.* [Online] IEEE, 2006. http://ieeexplore.ieee.org/.

[4] **Davidson, Jonathon, et al.** *Voice over IP Fundamentals.* Indianapolis : Cisco Press, 2007.

[5] **Markel, J.** *Linear Prediction of Speech.* New York : Springer-Verlag Inc, 1976.

[6] **Parsons, Thomas.** *Voice and Speech Processing.* New York : McGraw-Hill Inc, 1987.

[7] G.7xx Audio (Voice) Compression Protocols (CODEC) Overview. *Javvin network management and security.* [Online] Javvin Technologies Inc., 2005. http://www.javvin.com/protocolG7xx.html.

[8] G.711. *Wikipedia: The Free Encyclopedia.* [Online] Wikimedia Foundation, Inc., 2007. http://en.wikipedia.org/wiki/G.711.