CHERI

February 19, 2007

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

Re: ENSC 440 Functional specification for a Voice Bandwidth Saving System

Dear Instructor One,

Please find the attached functional specification for the Voice Bandwidth Saving System (VBSS), a communication system that reduces the bandwidth usage of VOIP applications. This document outlines functional requirements for this system. Specifications for two stages of our system – conceptual and production-ready – will be presented. These specifications are the basis for our design.

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: Functional Specifications for a Voice over IP Bandwidth Saving System.

VBSS Functional Specification

February 19, 2007

CHERI

VBSS

Functional Specification

VoIP Bandwidth Saving System (VBSS)

Functional Specification

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Submitted to:

Steve Whitmore	ENSC305
Lakshman One	ENSC440
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Revision: 1.0

Executive Summary

By the end of 2005, approximately two-thirds of Fortune 2000 companies have installed a certain degree of VoIP equipment [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.

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.

For most, the only solution is a costly upgrade to their internet connection. Cheri Perception provides a cheaper solution for these companies.

However, the large bandwidth usage of the VoIP system hinders remaining companies from joining this trend. For most, the only solution is a costly upgrade to their internet connection. Cheri Perception provides a cheaper solution for these companies.

Development of our system will occur in three phases. The first stage is the prototype stage in which we will attempt to prove our concept and attain the following:

1. Develop within a budget of $1500
2. Simulate an internet environment
3. Default application specific for G711-compliant phones running in *SipX* v3.4
4. Simulate running four phones
5. Save bandwidth by at least 45%

The second stage of development concerns production for which our goals are:

1. Increase compliance to other standards G726 G729 and ITU-T standards
2. Increase compatibility to other VoIP servers
3. Dual internet ports as per design
4. Mass production: single product price within $50
5. Include smart arbitration for a mass network

The aims of our third stage are:

1. Partner with current networking companies
2. Realign current design to integrate with current routers and switches from partnered companies

Revision History

Date	Version	Description
01/29/2007	0.1	Initial Draft
02/12/2007	0.2	Added terms to Glossary
02/15/2007	0.3	Initial Test Plans
02/16/2007	0.4	Reordering
02/17/2007	0.5	More test plans
02/18/2007	0.6	Added modular specifications
02/19/2007	1.0	Final Edit

Table of Contents

Glossary

Term	Definition
Bandwidth	A measure for communication capacity of a channel. Channel, here, primarily refers to the internet connection. A larger bandwidth means more information can be carried through that channel.
Channel	Medium for transfer of information. We will mostly refer to the “internet channel” as the “channel”, for which we mean the lines or wires that carry information.
CODEC	**Co**der-**Dec**oder A method of encoding and decoding information to save space while transferring. Analogous to using abbreviations. The writer will encode a word into an abbreviation while the reader decodes it to the full word.
cRTP	**C**ompressed **RTP** Protocols use some bandwidth to store header information. cRTP compresses the header information to save bandwidth.
G.711	An audio CODEC standard used readily in VoIP phones. It utilizes PCM to compress information.
G.7xx	A group of audio CODEC standards that are used in VoIP applications. Each CODEC uses a different way to compress information.
Header Information	Information added onto the data in a protocol, such as the RTP protocol, which directs the data to its destination.
IP	**I**nternet **P**rotocol A protocol for moving data from one location to another. There are two types of information – header and data information.
Layer 2	Data-Link Layer The second layer of networking is the data-link layer, which provides a means to transfer data between elements in a network.
Layer 1	Physical Layer The first layer of networking is the physical layer, which only provides a means to transfer bits of data rather than packets
Peripheral Network	The Ethernet Network that is connected to our VBSS System
Packet	Pieces of information passing through the Ethernet Network are commonly known as packets

Term	Definition
Receiving System	System receiving information (data and header)
RTP	**R**eal **T**ime **P**rotocol A type of internet protocol that deals with data that cannot be delayed.
Transmitting System	System transmitting information (data and header)
VoIP	**V**oice **o**ver **I**nternet **P**rotocol The transfer of voice data over the internet is commonly referred to as VoIP
VoIP Phones	Phones that send information through the internet
μ-law algorithm	A compression algorithm used by G.711. G.711 standard has two algorithms. One is μ-law, and the other is A-law. While μ-law is used in North America and Japan, A-law is used in Europe.

Introduction

VBSS is designed to be a transparent bandwidth saving utility that can be easily installed on a pre-existing VoIP system. Being readily installable, however, also poses requirements on the systems that interface with VBSS. Hence, VBSS's functional requirements not only define the VBSS system, but also restrict its deployment environment.

There are three stages in our development plan – prototype (concept), production, and integration. The first stage of development (conceptual) is planned to be finished on April 10th, 2007, while the second and third stages are due on a future date.

1. Scope

This document describes the functional requirements of VBSS and its peripheral systems. Examples of the peripheral system's components are Ethernet networks, the VoIP phones, and human staff. While our conceptual framework will be fully defined by the functional specification, requirements for production can only be defined partially. Hence, these requirements form the basis for our design and implementation, but production specifics are still unknown. More strictly, our design document should mirror this document.

2. Intended Audience

Members of *Cheri Perception* are to read this functional specification to guide their design and implementation of VBSS, which ensures that VBSS will meet these functional criteria. During development, this document will serve as a progress indicator, and during quality assessment, this document will be our examination material.

3. Convention

Each requirement will be indexed as in Figure 1

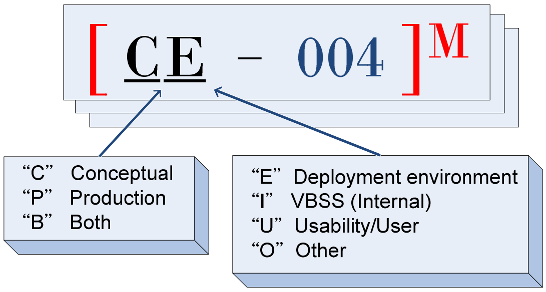

Figure 1: Indexing Format

Our indexing can be split into several parts.

1. The first letter, exemplified by **C**, can take on three different values, **C**, **P**, and **B**. It refers to the stage of development relevant to the item. Some requirements are only meaningful when undergoing a proof of concept, while others may only pertain to production. Still others may relate to both of the stages.
2. The second letter, denoted by the **E**, refers the requirement's field of influence, for which we mean the party that is restricted by the requirement. While some of the requirements are meant to specify our product, others give required qualifications for the operation of our ideal system. More information will be presented in the *Requirements* section of this document.
3. The third part of our index, **004**, is a three digit number uniquely defining each specification. Any specification can be identified by this three digit number alone. Hence, all other qualifiers are meant as indicators for characteristics.
4. The last part of our index (**M**) may or may not appear. **M** stands for Module and will only appear if the requirement is for a specific module.

System Overview

Our conceptual system can be broken down into five different components:

- Channel Simulator
- Packet Management
- Information Retrieval Service
- Voice Processor
- Master controller

as shown in Figure 1 (the Ethernet interface is a physical interface). Those components which only apply to the prototype are prefixed by the word "simulated", while the remainder applies to both prototype and production. Each component will only communicate with its neighbors.

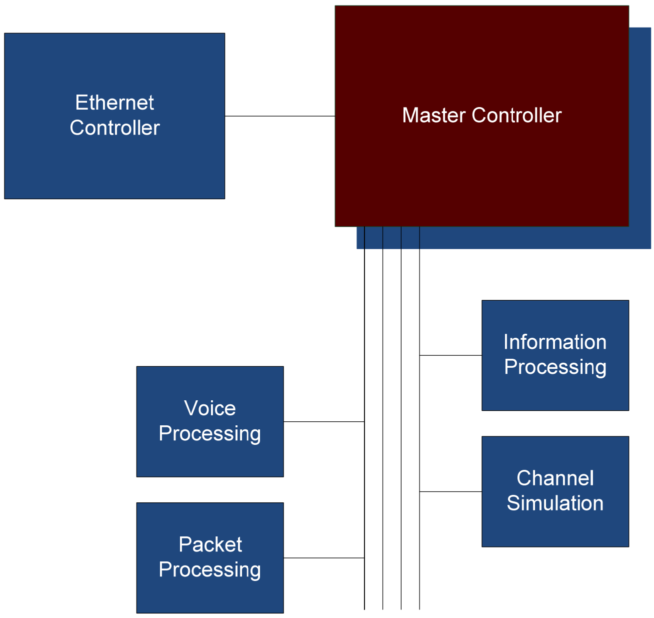

Figure 2: System components (from left to right)

The channel simulator's function is to simulate an internet channel, and equally, the Ethernet interface simulator's function is to simulate an Ethernet interface. The packet processing layer inspects the packet for voice information, while the information processing layer is responsible for extracting certain attributes from the packet, such as destination. Finally, the voice processing layer will process the packet voice data.

Requirements

This section will describe different types of requirements required for our system, and will provide listings of all functional specifications. Each different requirement category refers to Figure 3.

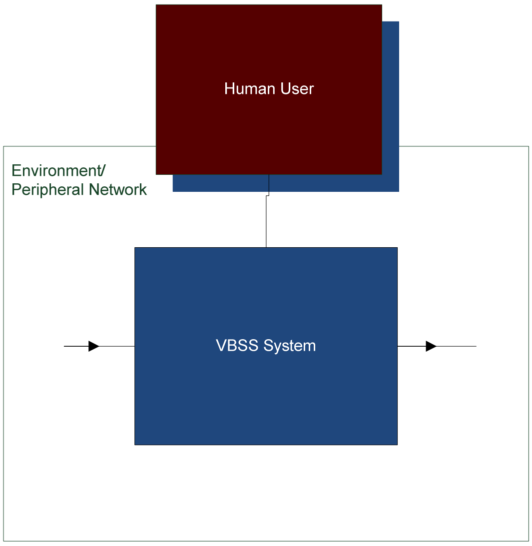

Figure 3: Black Box Diagram with Environment

1. Categories

Our specifications are split into four categories:

- VBSS Internals (**I**): refers to the internal characteristics of the VBSS system. This category may include static properties, such as power usage, and overall system delay. It may also entail other characteristics such as startup, processing, shutdown, and outgoing packets. In other words, requirements describing the relationship between input and output of a single VBSS device fall into this category.

- Deployment Environment (**E**): refers to the necessary attributes that the peripheral environment needs to have. This category includes VoIP phones, the Ethernet network, and incoming packets. Necessary characteristics of the "Environment" white box in Figure 3 are described in this category.

- Usability (**U**): refers to maintenance and use of our product. Examples of requirements in this category include minimum number of phones required for testing, how it should be connected, and how others can maintain its integrity. This category corresponds with the red box above.

- Other (**O**): refers to any specifications that do not fit in the above categories. Price, weight and dimensions may be included in this category. This category is not represented above.

2. Listings

Listing 1: Common VBSS Specifications

Index	Description
[BI- 001]	System must not change existing peripheral network characteristics or endanger them by introducing security holes.
[BI- 002]	System must route packets correctly through the network
[BI- 003]	System must not distort signal past recognition (of speaker and speech)
[BI- 004]	System must not have more than 50ms delay
[BI- 005]	System must acquire power either from an Ethernet cable or a CSA approved adapter
[BI- 006]	System must be modular in design

Listing 2: VBSS Specifications (prototype specific)

Index	Description
[CI- 007]	System must support G.711 audio CODEC (μ-law algorithm)
[CI- 008]	System must support *Sip X Server v3.4* type Network
[CI- 009]	System must transfer all packets through channel characteristics simulation
[CI- 0010]	System must simulate channel noise and interference
[CI- 0011]	Packet controllers implemented as Layer 2

Listing 3: VBSS Specifications (production specific)

Index	Description
[PI- 012]	System must support G.7xx and ITU-T audio CODECs
[PI- 013]	System must not add more than 10ms latency for filtering packets
[PI- 014]	System must support *Sip X* Server, Asterisk® Server
[PI- 015]	System must support most commercial VoIP phones, including Cisco, Broadcom, VTech
[PI- 016]	System must be able to handle noisy environment
[PI- 017]	Transmitting System should resolve if receiving party has a VBSS system
[PI- 018]	System must not hinder packets when starting up or shutting down
[PI- 019]	Packet controller implemented as Layer 1

Listing 4: Common Deployment Environment Specifications

Index	Description
[BE- 020]	Environment must be an Ethernet network
[BE- 021]	Ethernet network routers should support 10/100 Mbps
[BE- 022]	Environment must be free of large electromagnetic noise
[BE- 023]	Connected Phones must be VoIP enabled phones
[BE- 024]	System must be able to operate between -5°C and 40°C.
[BE- 025]	System must be able to withstand domestic pressure and humidity

Listing 5: Deployment Environment Specifications (prototype specific)

Index	Description
[CE- 026]	Phones must use G.711 (μ-law) audio CODEC
[CE- 027]	Environment must be *Sip X Server v3.4* running on a Fedora Core
[CE- 028]	Environment must be installed on a computer with minimum of 500 MHz Pentium-II with 768 MB ram.
[CE- 029]	Packet header information should not be coded in cRTP
[CE- 030]	Node IP addresses must be static
[CE- 031]	Nodes are connected through a switch
[CE- 032]	Environment has a maximum of four phones

Listing 6: Deployment Environment Specifications (production specific)

Index	Description
[PE- 033]	Environment must be connected through a network
[PE- 034]	System must be powered by a CE certified power supply
[PE- 035]	Network must have compliant phones and a compliant server

Listing 7: Common Usability Specifications

Index	Description
[BU- 036]	System must automatically start once installed
[BU- 037]	System must not require constant human supervision
[BU- 038]	System must only require at most one maintainer
[BU- 039]	System must be able to return to default configuration
[BU- 040]	System must include monitoring services available through any internet browser

Listing 8: Usability Specifications (production-specific)

Index	Description
[PU- 041]	System must install default configuration automatically upon plug-in
[PU- 042]	System must be easily configurable via any internet browser
[PU- 043]	System maintainer must understand network structure and security
[PU- 044]	System maintainer must not have to understand cryptography or signal processing to operate VBSS
[PU- 045]	System maintainer should understand VoIP fundamentals and basic communication rules
[PU- 046]	System must be marketed on the Internet
[PU- 047]	System must have a quick start guide available upon purchase and downloadable online
[PU- 048]	System must have status indicators for on/off, activity, connectivity
[PU- 049]	System configuration must be secure and password protected
[PU- 050]	System should provide error reporting
[PU- 051]	System must be plug-and-play

Listing 9: Miscellaneous Specifications (prototype-specific)

Index	Description
[CO- 052]	System must cost less than $1500 CAD
[CO- 053]	System must be tested under influence of background noise
[CO- 054]	System must be tested with presence of irrelevant packets
[CO- 055]	System must be tested in conditions of high network traffic

Listing 10: Miscellaneous Specifications (production-specific)

Index	Description
[PO- 056]	System must cost less than $150 per unit
[PO- 057]	System must be capable of mass production
[PO- 058]	System must be able to integrate into existing router systems – as part of, for example, a Cisco router.
[PO- 059]	System must not contain small detachable objects
[PO- 060]	Individual System must be less than or equal to 1U thickness
[PO- 061]	Individual System must be able to be placed on a rack
[PO- 062]	Individual System weight must be less than 500g
[PO- 063]	System must be able to handle 4G of force
[PO- 064]	Device casing should not be made of corrosive material
[PO- 065]	Device should not require reprogramming at each startup
[PO- 066]	System must be tested under high/low temperature conditions similar to when transported through air
[PO- 067]	System must test for compliance with peripheral system.
[PO- 068]	When incompatible with peripheral system, system must report error conditions and disable all packet processing

3. Modular Listings

Listing 11: Functional Specification for the Master Controller

Index	Description
[BI-069]^{M}	Master controller must be able to communicate with subcomponents
[BI-070]^{M}	Master controller must redirect data to the correct subcomponent
[BI-071]^{M}	Master controller must keep a record of current phone connections

Listing 12: Functional Specification for the Ethernet Controller

Index	Description
[BI-072]^{M}	Ethernet Controller must redirect any packets that do not pertain to it.
[BI-073]^{M}	Ethernet Controller must read data from a website for configuration
[BI-074]^{M}	Ethernet Controller must output received error messages into a website

Listing 13: Functional Specification for the Voice Processing Section

Index	Description
[BI-075]^{M}	Voice processing subcomponents (transmitter and receiver) must run in parallel, one not affecting the other
[BI-076]^{M}	Voice processing block must encode output in µ-law (G.711)
[BI-077]^{M}	Voice processing block must remember output of recent frames

Listing 14: Functional Specification for the Information Processing Section

Index	Description
[BI-078]^{M}	Information processing block must occur in parallel with voice processing
[BI-079]^{M}	Information processing block must not return headers as output
[BI-080]^{M}	Information processing block must return all required information to create a header as output

Listing 15: Functional Specification for the Packet Processing Section

Index	Description
[BI-081]^{M}	Packet processing subcomponents (constructor and processor) must work in parallel
[BI-082]^{M}	Packet constructor must return valid headers as output
[BI-083]^{M}	Packet processor must return two pieces of data as output – voice and network data
[BI-084]^{M}	Network data must be transmitted to the information processing block
[BI-085]^{M}	Voice data must be transmitted to the voice processing block

Test Plan

Our team is currently divided into two sections.

1. The first section sets up runtime environments
2. The second develops code for different sections of the project

Hence, two different test categories must be executed, one pertaining to each group above. Unit tests of each category will be executed concurrently with development. Additionally, these tests only correspond to the prototype system.

1. Runtime Environment

The runtime environment includes *Fedora Core, SipX* Server, *Sip* compliant phones, and the packet routing process.

Runtime Test Case #1

Requirement:

Sip-X server must run smoothly on the Linux system, and it shall allow external phones to register with the server.

Procedure:

1. Inspect the status of each service under *Sip-X* server.
2. Connect four Grandstream phones and the *sip-X* server box to the 8-port switch.
3. Ping other devices on the network from the sip-X server box.
4. Using the Sip-X management interface, add four users and assign each of them a unique phone number.
5. Modify the phone profiles with correct IP of related servers.
6. Restart the Grandstream phones manually.
7. Wait for the phones to restart, and check whether they have downloaded new profile information from *Sip-X* server.
8. Test the phone by checking if it could accept incoming calls and also could make outgoing calls.

Expected Result:

After *Sip-X* server is started up, executing "/sbin/service sipxpbx" should display 'Ok' for each sub-services under *Sip-X*. User should be able to ping all external devices from *Sip-X* server through the 8-port switch. Grandstream phones should be able to download new profiles from *Sip-X* server, and are able to register with the *Sip-X* server. Each Grandstream phone should be equipped with a unique phone number, and calls can be made by all phones to any remaining phones.

Runtime Test Case #2

Requirement:

Arp-Poisoning server must be able to perform Man-In-The-Middle (MITM) attack between any two of the Grandstream phones. The Arp-Poisoning server is necessary to allow us to implement our protype system on a one-Ethernet-port FPGA board.

Procedure:

1. Connect four Grandstream phones and a *Sip-X* server to the 8-port switch.
2. Let the phones register with the *Sip-X* server
3. Check whether the Grandstream phones can receive and make calls.
4. Now connect the *Arp-Poisoning* server to the 8-port switch.
5. Arp-poison one pair of phones.
6. For the pair selected at step (5), make a call from one of the pair to another of the same pair.
7. From the *Arp-poisoning* monitor, check the traffic status and see if it is able to capture the voice packet.
8. Un-poison the phones.
9. Repeat step 6 to see if phones can function normally.
10. Go back to step 5 with another new pair.

Expected Result:

The *Arp-poisoning* server should be able to perform MITM attack and intercept the voice data between any two phones. From the monitor screen, user should find a complete session of recorded voice conversation. The phones should also function normally after the *Arp-poisoning* server release the poison from the phones.

2. VBSS System

The VBSS system has many components. Each component is described in the **System Overview**.

System Test Case #1

Requirement:

FPGA must be able to connect to the network as an HTTP server. FPGA must be able to send and receive packets.

Procedure:

1. Add the FPGA into the DNS list and create static IP
2. Ping the FPGA, awaiting a return packet
3. Program the board with example code from the development CD
4. Run the program on the board
5. Enter IP address or domain name of the board into all web browsers
6. Navigate through all pages

Expected Result:

Pages should appear on the web browser as created.

System Test Case #2

Requirement:

Components must be able to communicate with on-chip microprocessors

Procedure:

1. Add VHDL code for G711 Codec
2. Modify HTTP Server to communicate with the peripheral
3. Modify HTTP Server to send textbox information from the browser page into the peripheral
4. Modify HTTP Server to show the output of the peripheral in the browser page
5. Run the modified HTTP Server
6. Input different test variables

Expected Result:

A dynamic output appears on the web browser

System Test Case #3

Requirement:

Microprocessor should send information to the correct component

Procedure:

1. Create a shell for the peripherals
2. Create internals that set the output to equal the input
3. Modify microprocessor to communicate with each of the peripherals in an order
4. Modify HTTP Server pages to receive more than one input for different required parameters for different conditions
5. Modify HTTP Server to output the signals between the different peripherals
6. Run the microprocessor
7. Input different test variables

Expected Result:

The components are stringed together in a specific order. This order is dependent on the type of packet that enters the microprocessor. If the packet is a non-voice packet, it should be sent through immediately. If the packet is a voice packet, the microprocessor should send it to packet processing. Then, concurrently, the voice data will be passed to voice processing, and the information data will be passed to information passing. When finished, both pieces of information will be passed back to packet processing, and back to the microprocessor.

Conclusion

Cheri Perception is on its way in providing a means for companies to adopt a VoIP system. This document outlines some of the functional requirements needed to fulfill this goal. While the conceptual system may not appear rigorous, this rigor is not required at that stage of development, because its purpose is to demonstrate the feasibility of the project. However, the final product specified in this document is definitely robust.

April 2007 is the deadline for conceptual development, and by that time, a prototype will be available. All functional requirements prefixed by **B** or **C** will be manifested in this prototype.

Sources and References

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