Radio Over IP Voice and Signalling Characterization Through
System-of-Systems Radio Over IP Solution Deployment

by

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Abstract

The focus of the thesis is on the characterizing Radio over IP voice and signalling delay, availability and voice quality by deploying Radio over IP (RoIP) System-of-Systems solution over large distances. System-of-Systems RoIP solution interfaces various mobile radio systems over multiple third party Wide Area Networks (WANs) based on Multiprotocol Label Switching Internet Protocol (MPLS IP) links to enable different departments such as Police, Military, Fire, Ambulance, Oil, Correctional Services, Forestry, Mining and etc. to communicate using their existing mobile radios. This thesis focuses on 4 major parts. The first part identifies various mobile radio systems and interfaces to extract and convert radio signaling and voice into the IP packets. The second, researches various interoperability management systems (servers/applications) to interconnect various mobile radio clients (agencies and departments) through the voice patches/conferences. The third focuses on the RoIP System-of-Systems solution deployment, testing under real-world condition using both unicast and multicast enabled networks, developing baseline characteristics such as Codecs, Delay, Jitter, Packet Loss, for secure and reliable communication over the third party WAN (MPLS IP backbone). The fourth focuses on RoIP System-of-Systems architectures and their performances when deployed over large distances and multiple third party WAN networks. The fourth primarily focuses on the systems latency characterisation, availability and voice quality. This thesis is the first one to examine radio over IP interoperability into a great level of details and provide a good foundation in understanding where interoperability would work and where not by characterizing major delay contributors in the overall System-of-Systems RoIP solution, from technology perspective. In this thesis RoIP communication over large distances has achieved 99.99%-99.999% availability and audio quality was found to be loud and clear 5/5 (signal strength 5/clarity 5).

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List of Acronyms

APCO 25:	Association of Public Safety Communication Officials
CAI:	Common Air Interface
CSI:	Console Subsystem Interface
CTCSS:	Continuous Tone Coded Squelch
COR	Carrier Operated Switch
DAQ:	Delivered Audio Quality (DAQ)
DFSI:	Digital Fixed Station Interface
DCS:	Digital Coded Squelch
DTMF:	Dual Tone Multy Frequency
ERTT:	Emergency to Talk
E&M:	Earth and Magneto/Ear and Mouth
FDMA:	Frequency Division Multiple Access
FM:	Frequency Modulation
FSI:	Fixed Station Interface
GRE:	Generic Routing Encapsulation
ISSI:	Inter Subsystem Interface
IMBE:	Improved Multi-Band Excitation
IPSec:	Internet Protocol Security
IPICS:	Interoperability and Collaboration System
LMR:	Land Mobile Radio
LAN:	Local Area Network
LTE:	Long Term Evolution

MPLS IP:	Multiprotocol Label Switching Internet Protocol
NAC:	Network Access Code
P25:	Project 25 (standard based digital waveform and radio features)
PTT:	Push to Talk
QoS:	Quality of service
RoIP:	Radio over IP
RTT:	Request to Talk
RTP:	Real Time Transportation Protocol (RTP)
SIP:	Session Initiation Protocol
TAG:	Trunked Analog Gateway
UDP:	User Datagram Protocol UDP Real Time Transportation Protocol (RTP)
VLAN:	Virtual Local Area Network
VoIP:	Voice over IP
WAN:	Wide Area Network

Chapter 1: Introduction

Various agencies and departments such as Police, Military, Fire, Ambulance, Oil, Correctional Services, Forestry, Mining and etc. have needs to communicate during their day-to-day operations. This is especially important in mission critical situations that involve loss of infrastructure, property and in severe cases loss of life. Due to the lack of interoperability, agencies and departments are not able to communicate and share critical information. Some interoperability initiatives to develop standard based interfaces such as P25 Open Interface, Common Air Interface (CAI), Fixed Station Interface (FSI), Digital Fixed Station Interface (DFSI), Console Subsystem Interface (CSSI) and Inter Subsystem Interface (ISSI) are not likely to solve immediate interoperability needs. Some of the reasons for the lack of the standard based interfaces are due to development cost, lack of cooperation among radio systems vendors (tend to keep market share with proprietary based solutions), long funding cycles for radio systems, legislative problems and long deployment timeframes for radio systems.

In this thesis, System-of-Systems Radio over IP (RoIP) solution was deployed to enable various non-interoperable agencies and departments to communicate using their existing mobile radio infrastructure in order to characterize RoIP system delay, system availability and audio quality.

In this chapter, Section 1.1 introduces System-of-systems RoIP interoperability solution, Section 1.2 explains the motivation behind this research, Section 1.3 summarizes approach undertaken to complete this study, Section 1.4 lists the contribution of the current study to the mobile radio interoperability and Section 1.5 outlines the scope of the thesis.

1.1 System-of-Systems Radio over IP (RoIP) interoperability solution

System-of-Systems RoIP interoperability solution that was deployed for this research is a solution that enables existing mobile radios such as analog legacy radios, digital radios, P25 digital in both conventional and trunking radio architectures to communicate over the MPLS IP dedicated or shared links. The solution enables non-interoperable mobile radios to communicate regardless of the frequency band such as VHF, UHF, HF and 700MHz-800MHz and vendor type such as Motorola, Codan, Cassidian/EADS and Harris.

System-of-Systems RoIP solution uses an enterprise approach by deploying off-the-shelf products such as interoperability servers and Land Mobile Radio (LMR) gateways to enable radio systems to communicate over the enterprise Multiprotocol Label Switching Internet Protocol (MPLS IP) based Wide Area Networks (WANs) by hosting radio conferences. The solution involves establishing voice over IP (VoIP) network which can be divided into two major parts: the core (MPLS IP based WAN) where the centralized interoperability servers are housed and access network consisting of LMR gateways to interface radio systems located at each agency or department. The servers are used to host multiple simultaneous voice conferences, and it can be a server-class computer [1], rack server, or blade type server. System-of-Systems RoIP solution takes advantage of IP standards such as Session Initiation Protocol (SIP) to establish calls, Protocol Independent Multicast (PIM) protocol to direct multicast traffic, User Datagram Protocol (UDP)/Real Time Transportation Protocol (RTP) for voice/data transport and it is multicast based when possible to provide flexibility and save bandwidth. Bandwidth usage between multicast and unicast enabled networks is outlined in Figure 1.1.

Figure 1.1: Bandwidth usage: Multicast vs. Unicast IP [31]

1.2 Motivation

There is widely evident need for inter-organization and intra-organization mobile radio interoperability. In recent years modern society has witnessed an increase in loss of property, infrastructure and life due to natural disasters such as the Oklahoma City bombing (1995), Columbine High School shooting (1999), Hurricane Katrina (2005), terrorist attacks such as the 9/11 (2001), Boston Marathon (2013), Alberta flood (2013) etc. In many cases this could be prevented or minimized if various departments and agencies, both nationally and internationally, could communicate to exchange valuable information and coordinate their actions [1][2][4].

Over the years various agencies and departments such as police, military, fire, ambulance, oil, correctional services, forestry, mining and other public and non-public safety organizations have deployed mobile radio systems that best suits their needs and comply with their budget constraints. Those radio systems are provided by various vendors such as Motorola, Harris, Cassidian/EADS, Codan, and etc and in most cases they are proprietary solution (not based on open standards). The deployed radio systems operate over 10 different frequency bends. Radio spectrum allocation for both Canada and US are outlined in Figure 1.2 and Figure 1.3, respectively.

Figure 1.2: Radio spectrum allocation in Canada [47]

UNITED STATES FREQUENCY ALLOCATIONS

THE RADIO SPECTRUM

Figure 1.3: Radio spectrum allocation in United States [48]

Once joint operations among agencies and departments became ever more significant starting in 1980s, it became evident that communication silos have made agencies and departments non-interoperable [3]. It has been realized that there is a significant need to overcome communication problems and that agencies and departments should focus on deploying an open standard based radio system. Unfortunately this process will takes years since all parties need to share the same vision such as users, vendors as well as a political support needs to be obtained. It happens that mobile radio over IP would be a promising, immediate solution to bridge existing communication silos.

Traditionally the lack of interoperability among mobile radios was remedied by an approach “one radio for all”, replacing existing radio system with new (such as standard based P25 digital mobile radios) and/or developing standard based interfaces DFSI, ISSI, CSSI and etc. The reason why this does not work, or does not provide an immediate solution, is that replacing radio systems infrastructure at once would be extremely expensive; vendors do not play well together in complying with standards; vendors want to keep their market share by providing proprietary solutions; long funding cycles for radio systems, legislative problems and long deployment timeframes for radio systems.

Voice over IP Interoperability solution provided by various vendors such as Twisted Pair, Cisco and Motorola show promising results in solving interoperability problems. Twisted Pair and Cisco are the pioneers in providing standard based interoperability applications (management server and media server) to interconnect various radio channels over the IP backbone. On the other hand Motorola Motobridge [55] interoperability solution enables primarily interoperability among existing Motorola radios and contains proprietary based components. All three vendors claim that they tested their solutions on a small scale such as lab environment, or the private networks with limited number of different radio systems and radio systems architectures. Unfortunately, the vendors did not publish the testing results such as radio systems tested, baselines characteristics bandwidth, delay, jitter and packet loss over the WAN technologies and how change in baselines characteristics affect mission critical communication.

The motivation of this thesis is to contribute in determining the baselines characteristic for reliable mission critical communication and to characterize an overall RoIP delay by identifying major system

delay contributors and how delay can be reduced. Another motivation of this thesis is to determine if an overall RoIP delay can be mathematically approximated to determine in advance if interoperability would make sense among various radio systems over the WAN. The Systems-of-Systems RoIP interoperability solution that was deployed for the purpose of this thesis examines voice interoperability of various radio systems over the MPLS IP based WANs, determines the baselines characteristics, identifies issues with proposed deployment solutions such as Cisco IP Interoperability and Collaboration System (IPICS) and Motorola Motobridge and provide recommendation for future interoperability deployments. System-of-Systems solution has been deployed and tested over the private dedicated network and public WANs infrastructures provided by Telus, BELL and AT&T.

This thesis is conducted using an enterprise approach by deploying and testing off-the-shelf standard based products (routers and switches), selecting and engineering radio interfaces to interface LMR gateways and using Wave by Twisted Pair and Cisco IPICS management and media servers for radio channel mixing. Wave and Cisco IPICS management and media servers are the most promising standard based servers for radio channel management and mixing/conferencing [6][7].

1.3 Approach

This section will briefly explain the approach taken to characterize RoIP delay, availability and voice quality by deploying System-of-Systems Radio over IP solution for this thesis. System-of-Systems RoIP solution is combination off-the-shelf products to enable interoperability between legacy radios, P25 standard based radio systems (both conventional and trunking), IP based P25 radio system over the MPLS IP based WANs using both unicast and multicast.

Firstly it was researched what type of interface would be the most suitable to interface various radio systems to LMR gateways and what protocol to use to establish voice calls (SIP) or to direct voice streams (PIM) over IP. Next, various available radios in VHF, UHF, 700-800Hz frequency bands were interconnected using point-to-point trunks in direct connection LMR-to-LMR gateways to establish voice calls and subjectively measure voice quality as per TSB 88 Delivered Audio Quality (DAQ) [42]. According to the TSB 88 DAQ is subjective measurement. Our target was DAQ 3.4: speech understandable with repetition only rarely required and some noise/distortion is acceptable. The voice signal was rated “loud and clear” in terms of the signal strength/ signal clarity. The signal strength and the signal clarity were rated on scale 1-5. For example, if voice signal was heard loud and clear with no truncated audio it would be rated 5/5.

Next, LMR gateways were connected to a switch and a router was used to simulate WAN. Network encryption was introduced to secure IP traffic using generic routing encapsulation (GRE) tunnels with internet protocol security (IPsec) with AES 256 encryption. Once steady voice calls were established a Wide Area Network (WANem) emulator was added to the network to emulate real network conditions.

WANem is a general-purpose tool which is used to emulate impairments in IP network [22]. The WANem emulation tool was used to emulate changes in bandwidth, delay, jitter and packet loss to establish minimum baseline characteristics requirements for reliable communications. In addition various voice codecs were tested and voice clarity was subjectively evaluated [42].

Once baseline conditions were properly determined and set, interoperability application servers Wave and IPICS were deployed and tested for radio channel management and channel mixing (radio channel conferencing/radio channel patching) simulating two agencies. Multicast was enabled in each server location and connection between simulated agencies was established using unicast connection.

After the testing in the simulated environment was completed it was decided to conduct testing over the MPLS IP WAN network provided by BELL/Telus and AT&T. The Systems-of-Systems RoIP interoperability solution was deployed and tested in two different architectures simulating agencies and departments. For each System-of-Systems RoIP architecture delay, availability and voice quality were examined. Once delay metrics was obtained it was analysed and characterised to understand what major contributors were and how delay can be reduced. We examined if an overall end-to-end systems delay can be approximated by examining delay of each systems component independently.

Testing results were analysed and recommendations were provided.

1.4 Contributions

The following are the contributions of the thesis:

- A proposed approach to deploy RoIP System-of-Systems solution over large distances and multiple networks with focus on the voice and signalling delays characterisation, systems availability (packet loss) and voice quality. This includes determining and understanding how much of the end-to-end system's delay is contributed by each system's component such as inherent delays of radio systems, LMR gateways, patching/conferencing, IP networks, connection trunks etc.

- A proposed approach to show that an overall end-to-end system delay can be approximated mathematically by adding predetermined delays of the system's components. Therefore we propose that by examining RoIP System-of-Systems components we can get a good sense if interoperability between certain radio systems over large distances would make sense. In addition we propose multicast service reflection to avoid multicast-over unicast-multicast trunks to connect separate agencies in order to reduce delay.

1.5 Outline of this Thesis

This thesis consists of five chapters. Chapter Two provides an overview of the work relevant to the present study. Chapter Three covers stages taken to deploy System-of-Systems Radio over IP (RoIP) interoperability solution including determining a radio over IP baseline characteristics such as bandwidth, delay, jitter, and packet loss. Chapter Four provides the various architectures used to deploy System-of-Systems Radio over IP (RoIP) interoperability solution over the dedicated/private and public WANs. This chapter also covers work completed to determine overall System-of-Systems RoIP delay as well as how much that delay is contributed by each system component, availability and voice quality . Chapter Four provides testing results and compares the proposed solution to other available solutions as well as discuss the lack of available test data from vendors. Chapter Five provides overview of the thesis contributions, provides recommendations and guidance for future research.

Chapter 2: Background

In this chapter, background information is provided on topics that play central role to deploy System-of-Systems RoIP solution such as: analog and digital (conventional and trunking) mobile radio systems, interfaces between radio systems and IP backbone, connection trunk connections (unicast/multicast), multicast over unicast to multicast (MUM) trunks, generic routing encapsulation (GRE) tunnels with internet protocol security (IPsec) encryption, baseline characteristics (bandwidth, delay, jitter and packet loss) and codecs, quality of service (QoS) and interoperability application servers.

2.1 Analog and Digital (Conventional and Trunking) Mobile Radio Systems

Various agencies and departments such as military, police, fire, ambulance, oil, correctional services, forestry and mining have been deploying variety of mobile radio systems over the years. Those mobile radios are both analog and digital and they have been deployed in conventional and trunking system architectures. The radios are two-way which means that they can both transmit and receive signals (transceivers). A two-way radio enables user to communicate with another user using the same frequency channel.

Radio systems operate over various frequency bands which can be summarized as follows:

- VHF-Low Band: 25 MHz to 50 Mhz
- VHF-High:138 Mhz to 174 Mhz
- UHF:408 Mhz to 512MHz
- 700MHz (fairly new)
- 800 Mhz
- 4.9 GHz (new: microwave links)

2.1.1 Analog Mobile Radio Systems

Most agencies and departments have been equipped with the traditional two-way analog radio communication systems. Analog radio systems operates on radio channels which can include one or more frequencies and various tones which contain important messages such as Continuous Tone Coded Squelch System (CTCSS), Digital Coded Squelch (DCS) and carrier squelch tone or noise squelch, as it is shown in Figure 2.1. CTCSS tones are sub-audible and continue tones sent with audio to activate receivers for transmitted audio. A receiver that has CTCSS decode capability locks out all signals except ones that were encoded with the correct tone and it will activate its speakers. DCS is similar to CTCSS but uses a digital code instead of an audio tone on analog radio systems. Carrier squelch or noise squelch receiver receives any signal.

Most of the agencies use FM radios where FM stands for frequency modulation. Frequency modulation means changing the frequency of the transmitter in proportion of the audio being picked up by the microphone. On the receiver side the change in transmitter frequency is detected and used to reproduce the audio signal at the speaker. Frequency modulation can be classified as wide-band if the change in the carrier frequency is much higher than the signal frequency or narrow-band if the change in the carrier frequency is about the same as the signal frequency.

One of the advantages of analog radios comparing to digital radios is the way they handle background noise.

Figure 2.1: Analog mode overview

High level overview of the analog radio communication is shown in Figure 2.2.

Figure 2.2: High level overview of analog radio voice communication

2.1.2 Digital Mobile Radio Systems

Digital mobile radio systems take analog signals and convert it into digital through a process called sampling. Analog signal (human voice) is converted through the microphone into voltage and from voltage into binary numbers 0 or 1. Binary information (0s and 1s) is transmitted over the air comparing to analog (continuous voltage) information. The idea behind using digital signals instead of analog is to increase number of radio channels in a given frequency spectrum (reduces necessary bandwidth per channel), improve audio quality, eliminate fading problems in mobile environment, provide additional data transfer capabilities and to maintain radio coverage with less power transmitted [11]. To accomplish better spectrum efficiency (reduce bandwidth) vocoding algorithms have been used. Voice coder (vocoder) listens a sample of audio and transmits only certain characteristics that represent sound. Once signal is received, receiver uses these basic characteristics to synthesise an equivalent of the input sound. Vocoders are designed to primarily rely on known human speech characteristics to encode and decode data. The negative side to this is that in cases that noise (outside of the human voice spectrum) is presented during the transmission, vocoders can produce issues such as garbled sound and metallic sound, loss of communication and etc. Digital radio system block diagram is shown in Figure 2.3.

Figure 2.3: High level overview of digital radio voice communication

2.1.3 P25 Digital Radio system

In order to improve interoperability among various radio systems, provided by different vendors, public safety and federal government users in North America had formed Association of Public Safety Communication Officials -Project P25 (APCO-P25). APCO-P25 is an open standard for digital radios. P25 has been developed in two phases P25 phase 1 and P25 phase 2. Most of the vendors currently comply with most of the P25 phase 1 standard specification. It is developed to establish digital standard for common-air interface (between mobile, portable and base station radios), encryption, data transmission and trunking protocols [8][9][13]. The P25 standard is intended to provide the basis for voice communication interoperability. Common-air-interface standard eventually should enable radio systems from various vendors to interoperate. In plain words this means that if two P25 compliant radios are obtained from two different vendors they should be able to communicate over the common-air-interface.

P25 compliant radios operate in digital and analog mode; support Frequency Division Multiple Access (FDMA); can be deployed in conventional and trunking architecture; works in VHF/UHF/800MHz bands and use 12.5 KHz wide channels (P25 phase 1) and will eventually use 6.25 KHz (P25 Phase 2). In comparison analog FM radios require 25 KHz wide channels. Since P25 compliant radios can work in analog mode they are backward compatible with existing analog radio systems [13].

When used in analog mode, the P25 compliant radio behaves the same as the conventional analog radios. This means that the P25 radio supports standard analog features such as CTCSS, DCS and wideband or narrowband operations.

In P25 digital mode the P25 transmitter converts analog audio into digital using Improved Multi-Band Excitation (IMBE) vocoder. Once converted, the signal is sent over the channel and de-vocoded back to analog in the receiver.

The IMBE vocoder listens to a sample of audio input and only transmits certain characteristics that represent the sound (88 bits of encoded speech every 20ms). IMBE vocoder is heavily optimized for human speech and does not work well in reproducing other sounds. Analog CTCSS and DCS codes are replaced with Network Access Code (NAC). NAC codes represent a 12 bit codes that prefixes every packet of data such as voice packets [13]. P25 Radio system architecture is shown in Figure 2.4.

Figure 2.4: P25 Radio system architecture

Having stated above, it can be concluded that P25 multiband compliant radios are good alternative to solve interoperability issues. Unfortunately radio manufacturers are protecting their proprietary rights and do not cooperate in developing the P25 compliant radios. Also P25 compliant multiband radios are very expensive; to replace existing radio systems with P25 radio system at once would be cost prohibitive.

In this thesis we have deployed radio over IP (RoIP) interoperability solution as an alternative to solve immediate interoperability issues of the existing radio systems, including P25 radio systems, and cost a fraction compared to replacing entire existing radio systems with the P25.

2.1.4 Conventional and Trunking Mobile Radio System

2.1.4.1 Conventional Mobile Radio Systems

The conventional radio systems include base stations, mobiles, portable and filtering devices. These devices can have separate frequencies assigned on a functional basis in frequency bands such as VHF and UHF [8]. Conventional means that radios operate on fixed radio frequency (RF) channels and that each user or a user group is assigned a fixed frequency or a set of frequencies. Conventional radio systems allow the user to talk on one channel at a time. The user manually selects the proper channel on a radio device if the radio is programmed for multiple channels. When multiple users have the same frequency assigned only one user can talk on that channel and all other users on that channel need to listen. If multiple users attempt to talk at the same time on the same channel they can cause interference. This is also known as the “party” line.

The conventional radio systems are simple to design, operate, with dedicated frequency (no channel set-up time) and much cheaper comparing to the trunked radio systems. On the other hand the conventional radio systems are not RF spectrum efficient due to the fixed frequency assignment and easy to intercept by analog devices [8].

2.1.4.2 Trunking Mobile Radio System

Trunking mobile radio systems combine multiple radio repeaters at the same site to share frequencies among users. Comparing to the conventional radio systems, the trunking takes advantage of probability that not all the users will need channel access at the same time. Therefore trunked radio systems allow more users to share a limited number of radio channels dynamically [12].

User radios on the trunked radio system are assigned to the talk groups which are actually virtual channels and it is possible to have many more talk groups than available frequencies. Each user radio, when not transmitting, monitors a dedicated “control channel”. The control channel is a communication channel between a user mobile radio and the central controller (computer driven) [12].

When a user presses the Push-To-Talk (PTT) button on a mobile radio, the radio sends a digital message to the system controller over control channel requesting a channel to be assigned. In case that a channel is available, the system controller sends a message back to the mobile radio with info of the assigned channel. Once the “handshake” is completed (channel granted) a user's radio provides “talk beep sound” and user can proceed with a voice conversation on the designated channel. In addition, the system controller broadcast a similar message over the control channel to other mobile radios that belong to the same talk group to tune in to the designated channel. The switch from monitoring the control channel to the designated channel happens automatically. Once conversation is over, all radios return to monitoring the control channel. Since frequencies are dynamically assigned, a user may be on a different frequency every time it transmits. Note that entire procedure occurs so fast that user does not notice any difference compared to a conventional radio system. But, in case that all voice channels are in use, a user that re-

quest a channel grant will be placed in a queue (priority based) by the system controller [12]. Trunked radio system can be both analog and digital, even though the control channel is used to send digital data.

Trunked radio systems have advantages. In case that a channel fail in a trunked radio system, radios that have been affected can be automatically moved to another channel with a minimum degradation of service. Trunked radio system provides inherent security features due to the way they are designed and operate. For example, trunked radios needs to register before they can be used and therefore in case that radio is lost or stolen, the radio can be disabled individually based on its registration credentials.

On the other hand trunked radio systems do have downsides. Trunked radios are controlled by the central computer based controller which makes the channel assignment. In case that the channel controller fails (i.e. the power outage) the trunked radio system would be seriously affected. Trunked radio systems cover large geographical areas and provide services to many users. Small problems on the radio system can have a major impact on the users. Digital messages that are sent over the controller channel can be hacked by an experienced person who can put the radio systems at a high security risk.

The most important thing for this research is that trunked radio systems have a significant delay from the time a radio PTT button is pressed until the user can start transmitting due to the channel assignment process. If all the channels are used, the user will be placed in a queue. This can cause significant delay. Therefore for this interoperability research it is important to ensure that IP backbone and all the processing that take place in the IP core (channel conferencing/mixing) does not introduce any

significant delay. Otherwise communication would provide unsatisfactory experience for the users especially if the radio system is used to carry a mission critical communication.

Trunking radio systems are usually based on proprietary software provided by manufacturers such as Codan, Motorola, Harris, Cassidian/EADS and etc. These companies, in desire to protect their market share, do not co-cooperate significantly when it comes to information sharing. Therefore interoperability among trunked radio systems provided by these manufacturers is complex. That is why in this thesis we focus on deploying a strong IP backbone core to provide interoperability among radio systems and to push proprietary radio systems to the access side of the System-of-Systems RoIP interoperability solution.

2.2 Interface between Radio Systems and IP backbone

To enable mobile radios to communicate over the IP backbone it is important to determine what interface would be suitable to extract voice and signalling, easy to build and cost effective. Standard based interfaces such as P25 open interface, CAI, FSI, DFSI, CSSI and ISSI are in the process of development, standardization and testing. It will take a long time until those interfaces are proven and extensively tested among various vendors who, as it was stated many time in this thesis, tend to resist developing open standard based radio interfaces and architectures. Currently, standard based interfaces cost between 100K and million. Even though some of the radio system vendors claim that they are compliant with standards, or that they are heading in that direction, they still tend to involve their proprietary components in the overall solution.

This research focuses on finding a cost-effective interfaces to extract audio and radio signalling such as PTT and Carrier Operated Switch (COR) from various radio systems such as Motorola Quantar/DIU (Astro 25 digital), Cassidian IP based (P25 Digital) , Codan MT3/MT4 (analog/digital), Harris CS7000 (digital proprietary) and etc. convert it into IP packets and vice versa. Therefore this research also focuses on determining interface between radio and IP backbone.

Approach used was to:

1. Identify what interfaces radio systems support to extract audio, radio signalling such as PTT/COR and special tones (i.e Dual-Tone Multy-Frequency (DTMF)).
2. Identify cost effective interfaces on the IP backbone that would convert audio, signalling and tones received from radio system into standard IP packets and vice versa.

This research identified that various radio systems use the following interfaces T1, Ear and Mouth or Earth and Magneto E&M (4-wire or 2-wire) and P25 digital IP (not standard IP header). In terms of the IP backbone, the System-of-System RoIP solution IP core is built mainly of the Cisco switches and routers. Therefore this research was focused on Cisco interface cards that would support interfaces outlined above. Cisco voice cards VIC3-2E&M and VIC3-2MFT T1/E1 were chosen as compatible and cost-effective solution to support E&M (4-wire or 2-wire) and T1, respectively. By deploying VIC3-2E&M and VIC3-2MFT T1/E1 into a Cisco router (i.e 2800, 2900) and properly configuring voice ports, router is turned into the LMR gateway. For IP based P25 digital radio systems a trunked analog gateway (TAG) was used to convert IP based P25 digital signals into 4-wire analog to interface VIC3-2E&M.

2.2.1 T1 Interface

This research showed that many older/legacy radio systems can interface traditional T1 leased lines (1.54Mbps) using their own T1 interface or through a device such as channel bank. To leverage on the existing radio to T1 interface capabilities, this research showed that Cisco voice card VWI3-2MFT T1/E1 used in Cisco routers (i.e 2800/2900 grade routers), with the digital signal processors (DSP) to encode/decode the audio streams, can be used to interconnect those radios to the System-of-Systems radio over IP core (IP backbone) [14]. This was proved in our lab facilities but the testing was not characterized into the great level of details. One of the reasons is that T1 leased lines are very expensive (\$1000/month or more per link) and that those lines have to be dedicated for voice traffic. This research showed that T1 lines can be replaced using point-to-point trunks (multicast or unicast) sharing MPLS circuits with other type of traffic. To interface old legacy radios over the MPLS IP using point to point trunks E&M interface was used instead of T1 interface. Please note that System-of-Systems core is MPLS IP based (shared infrastructure). MPLS IP based core provides many advantages such as traffic engineering, quality of service, network redundancy, easy and cost effective expansion, protocol independent forwarding, bandwidth allocation, service level agreement, and etc. For this research VWI3-2MFT T1/E1 found its use in multicast-over unicast-multicast trunks. This is explained in greater details in the section 2.3.2 below.

2.2.2 Ear and Mouth /Earth and Magneto (E&M) analog interface

Ear and Mouth or Earth and Magneto (E&M) trunking interface is a type of control line signalling that uses DC signals on separate leads called “E” and “M” to communicate status between trunk circuit and signalling unit. Originally E&M analog interface was developed by Bell labs to trunking circuit and

the network. The trunking circuit can be PBX and signalling unit can be telco, central office (CO), channel-bank, or for this research most important Cisco voice enabled platform such as Cisco router with VIC3-2E&M interface card and digital signal processing (DSP) module to enable voice processing [14]. Origin of E&M interface name comes from earth - electrical ground and magneto - the electromagnet that generates tone. The trunking circuit and signalling unit exchange their status over the E and M leads using combination of battery and ground (earth). The battery signal used is nominally -48VDC. E&M interface applications require that positive terminal (+) of the battery is connected to a shared ground (Earth).

This research showed that E&M analog interface can be effectively used to connect radio systems to the IP network. To accomplish this it was important to understand E&M analog interface parameters, which E&M parameters are supported by each radio system and to match radio E&M parameters with cisco VIC3-2E&M interface card. In addition it was important to perform impedance matching between radio interface and VIC3-2E&M interface (i.e 0dBm at the input - 0dBm at the output).

Analog E&M interface contains four main parameters which combination determines E&M deployments [15].

- E&M interface types and wiring arrangement (type I, II, III, IV and V)
- Audio implementation: 2-wire or 4-wire operation
- Start dial Supervision signalling (immediate, wink and delay)
- Address signalling (i.e. pulse, DTMF)

Great level of details about E&M interface can be found in [15][16]. This research examined radio systems such as Motorola, Codan, Harris and Cassidian P25 IP (through the trunk analog gateway (TAG) determined an appropriate E&M type V, II, V and III, respectively. The radios tested support 2-wire and 4-wire operation, immediate start dial supervision signalling and DTMF address signalling. Cisco VIC3-2E&M voice card supports all E&M types. Cisco VIC3-2E&M pin layout is shown in Figure 2.5 [16]. This research was focused on understanding how to extract E&M voice and signalling (PTT/COR) out of the radios to properly interface with VIC3-2E&M. This included signal extraction through modified connectors (I,e DB25/RJ11 to RJ45) and opto-cuplers to match VIC3-2E&M voice interface card pin layouts. Also current limiting and impedance matching was performed between radio interface and the voice card. The voice card was configured to match E&M radio types, audio implementation (4-wire operation), signalling etc. A VIC3-2E&M voice interface card pin layout is shown in Figure 2.5.

Figure 2.5: E&M VIC3-2E&M pinouts [16]

Cisco Interface E/M mapping is outlined in the following table:

Table 2.1: Cisco E/M Interface 2-wire and 4 wire operation - Type I, II, III, IV and V mapping

CISCO E/M INTERFACE										
Pin	Signal	Description	Two-Wire Type 1	Two-Wire Type 2	Two-Wire Type 3	Two-Wire Type 5	Four-Wire Type 1	Four-Wire Type 2	Four-Wire Type 3	Four-Wire Type 5
1	SB	-48V signalling battery		SB	SB			SB	SB	
2	M-lead	Signaling input	M	M	M	M	M	M	M	M
3	R	Ring, audio input					R	R	R	R
4	R or R1	Ring, audio input/output or output	R	R	R	R	R1	R1	R1	R1
5	T or T1	Tip, audio input/output or output	T	T	T	T	T1	T1	T1	T1
6	T	Tip, audio input					T	T	T	T
7	E-lead	Signaling output	E	E	E	E	E	E	E	E
8	SG	Signaling ground return		SG	SG			SG	SG	

2.2.3 Trunked Analog Gateway (TAG)

One of the radios that was part of this research was Cassidian COR^{25} P25 digital, IP based radio system. COR^{25} is software-centric native IP architecture. Our research showed that COR^{25} IP packets contain proprietary content and that it cannot be interfaced directly to the System-of-Systems Radio over IP. As many times as we mentioned in this thesis, vendors try to keep their proprietary solutions to any extent possible. Also COR^{25} is P25 digital, it uses IMBE vocoder to encode traffic and it cannot interface directly the System-of-System RoIP since the System-of-System RoIP architecture uses E&M analog interface (supported by VIC32-E&M interface card).

To interface COR^{25} radio system to the Systems-of-Systems RoIP solution it was necessary to convert P25 digital signal into the E&M analog. This research showed that COR^{25} radio system use trunked analog gateway (TAG) to interface older analog consoles. The TAG converts P25 digital (decrypt and decode) signal into 4-wire analog E&M. In addition it has capability to encode signal using G.711 com-

pression codec. Therefore the TAG was a good candidate to interface *COR*²⁵ P25 digital, IP based system to the System-of-Systems radio solution. One of the key points of this research was to determine and match E&M interface pinouts (4 -wire audio and signalling) of the TAG to interface Cisco VIC3-2 E&M and to determine TAG processing delay.

2.3 Connection trunk connections (unicast/multicast)

To establish voice over IP communication between end points (voice gateways) this research focuses on understanding multicast and unicast trunks. A trunk or a tie line can be defined as permanent communication link between the two voice points [17]. This research showed that multicast trunks provide better flexibility as well as they reduce an overall system delay. But, the System-of-Systems RoIP solution provides interoperability among various departments and agencies which networks support or do not support multicast. Therefore it is important to understand both multicast and unicast trunks. Since Systems-of-Systems core (IP backbone) is mainly developed using Cisco switches and routers it is important to understand point-to-point trunks from the Cisco Internetwork Operating System (IOS) software perspective.

2.3.1 Unicast point-to-point connection trunks

The unicast connection trunks create a permanent VoIP call between the two VoIP gateways (point-to-point trunk). Point-to-point trunks simulate a trunk connection by creating virtual trunk tie lines between two telephony points. When a trunking circuit (in this case a radio system) is connected to the VoIP gateway it appears as if the radio system is directly connected to the T1 trunk [17]. For the purpose of this thesis, point-to-point unicast trunks were used for T1 (leased lines) replacement.

The key features of unicast connection trunks are:

- Permanent connection: always up
- Transport signalling: end-to-end
- Can be used only with other Cisco IOS based devices that support unicast trunks

Unicast connection trunks establish connection (channel) among voice end points (VoIP gateways) using standard based H.323 or SIP connection setup protocols. Once connection is established between the two end points it enables real time protocol (RTP)/RTP control protocol [19] datastream (audio/video/data) flow between the end points. On the other hand multicast connection trunks do not use the connection setup protocols and direct real time protocol (RTP)/RTP control protocol datastream to an multicast IP address so it can be received by a number of end points (anyone who registers and wants to receive multicast traffic). Multicast trunks use multicast routing protocol instead of the connection mechanisms such as H.323/SIP. For the System-of-System RoIP solution, protocol independent multicast - sparse mode (PIM-SM) [20] protocol was used to enable multicast routing and to direct RTP/RTCP datastream flow among the voice end points. It is important to explain that from the voice-end point perspective both the unicast and the multicast connection trunks present the same permanent tie-line style connection.

For this thesis, originally a H.323 connection mechanism was used to establish the unicast trunks, but since SIP protocol provides more advantages and it is standard based [18] and it was decided that for the System-of-Systems RoIP deployment, SIP is to be used setup unicast trunks. Based on the testing completed, which is outlined in the later sections, it was concluded that the multicast trunks should be used when possible.

2.3.2 Multicast over Unicast to Multicast (M1:U12:M2) configuration

The System-of-Systems RoIP solution, deployed for the purpose of this research, takes advantages of the multicast IP by enabling the multicast IP in the core. Multicast addressing enables multiple voice end points to receive a traffic generated by one source (traffic is sent one-to-many) unlike unicast IP where traffic is sent one-to-one. Therefore multicast provides a great scalability.

Agencies and departments that were simulated for the purpose of this research have multicast IP enabled on the local networks but to have multicast enabled end-to-end was not an option. One of the reasons was that multicast IPs would need to be shared and that would create security issues. Another reason was that firewall used for this testing did not support multicast traffic. Multicast provides great flexibility but in the same time it can create network management headaches if needs to be jointly managed by multiple parties. To enable communication between the multicast “islands” this research showed that multicast-over unicast-to-multicast (M1:U12:M2) alternative can be used [21]. M1:U12:M2 connection trunks are developed based on the connection trunk functionalities explained in the previous section 2.3.1.

The key features are:

- Permanent connection – always up
- Transport signalling end-to-end
- Connects multicast islands using the unicast point to point trunks

The unicast portion of M1:U12:M2 employs a SIP connection setup mechanism to establish a channel to another voice gateway and directs the real-time transport protocol (RTP)/RTP control protocol (RTP/RTCP) datastream to that single endpoint. Therefore the multicast traffic between the multicast

islands is piped over the unicast connection trunks. The M1:U12:M2 connection trunk maps the multicast traffic from a multicast channel M1 (multicast island 1) to a unicast address U1 to transport this traffic across the unicast VoIP network. Once unicast is received on the other side it is mapped to the multicast address M2 (multicast island 2). Connection trunks M1:U12:M2 are shown in Figure 2.6.

Figure 2.6: M1:U12:M2 Connection trunks

To accomplish M1:U12:M2 connection trunks VIC3-2MFT T1/E1 voice card is used to send multicast traffic over the unicast connection. This is accomplished by using a physical loopback cable between the ports on the T1/E1 card (connecting DS0 pairs 0/0:0 and 0/1:0) [21]. For the System-of-System RoIP solution M1:U12:M2 connection trunks are used to interconnect the core to various agencies and departments that do not have multicast enabled end-to-end.

2.4 GRE Tunnels with IPsec encryption

2.4.1 Generic Routing Encapsulation (GRE) Tunnels

Tunnelling provides the mechanism that transports packets of one protocol within another protocol. The protocol that has been transported is called the passenger protocol and the protocol that is used to

transport the passenger protocol is called the transport protocol. In addition tunnels have a carrier protocol which is used to encapsulate the passenger protocol.

GRE tunnels use IP protocol as the transport protocol and they can be used to transport many other passenger protocols. The tunnels are virtual point-to-point links that have the tunnel source and tunnel destination addresses at each endpoint [23]. From the System-of-Systems perspective GRE tunnels with IPsec encryption are used to enable and secure multicast traffic over the unicast WAN network which is provided by a third party provider. Bell, Telus and AT&T WAN network providers; used for this research as the WAN providers, currently do not support multicast traffic over the MPLS IP circuits and therefore GRE tunnels are used to encapsulate our multicast voice traffic into unicast for the transport between end points [24]. Therefore the multicast traffic is the passenger protocol, unicast is used as the carrier protocol to encapsulate the multicast traffic and IP is used as the transport protocol. The Systems-of-Systems RoIP solution deploys GRE tunnels in a hub spoke architecture where the hub is the System-of-Systems core and spoke sites are agencies or departments that are interfaced to the core.

2.4.2 IPsec

Internet Protocol Security (IPsec) is a suite of protocols that provide network layer crypto security to internet communications at the IP layer. IPsec is used to provide secure communication gateway-to-gateway (between two locations) or remote host-to-gateway (enterprise) security but also it can provide end-to-end or host-to-host security [28] [29]. IPsec framework is standardized and therefore it is not tied to any specific vendor product. It can be found on routers, firewalls as well as on the client desktops such as Windows and Mac. IPsec provides a great scalability since it was designed to accommodate large enterprise requirements and it has built-in crypto key management [29]. In terms of the System-of

System RoIP solution deployed IPsec is used to secure multicast traffic over GRE tunnels that connect Systems of Systems end points over third party WAN.

2.5 Baseline characteristics

To ensure that the System-of-Systems RoIP core provides IP backbone to successfully interoperate among various radio systems, each radio systems was put under the test to determine what network characteristics are required for the radios to successfully send or receive VoIP traffic. The network characteristics to be determined are:

- Network bandwidth/one channel (Kbps)
- Transport protocol (UDP, TCP, RTP, RTCP)
- Protocol port number
- Delay
- Jitter
- Packet Loss
- Compression Codecs

To emulate real network conditions and to vary the network variable under test (bandwidth, delay, jitter and packet loss), a WANem emulator was used. The WANem emulator is a network emulating tool for emulating impairments in IP networks [22]. For the System-of-Systems RoIP solution the WANem was used to emulate the worst possible condition on the network so that minimum bandwidth, acceptable, delay, jitter and packet loss could be determined for reliable VoIP communication for each radio system.

To enable radio systems to interoperate over the System-of-System RoIP IP backbone core it is important to find common supported compression codec. Otherwise if we have mismatch in compression voice codec, the voice communication would fail. Cisco VIC3-2E&M support the following standard-based compression codecs that allow communication over the IP: g711 ulaw, g711 alaw, g729r8, g729br8, g723ar53, g723ar63, g723r53, g723r63, gsmfr, g725r16, g.728, iLBC, g.722. For the System-of-Systems RoIP solution this research and testing showed that G.711 codec is the most suitable providing the best voice quality. One of the most important concerns for mission critical communication is the overall RoIP system delay. For example G.711 codec in comparison to G.729 creates 25ms less delay per sample, no voice quality degradation due to transcoding and packet loss has less impact on the voice quality.

G.711 codec or pulse code modulated (PCM) codec is the most popular voice coding schemes for digital coding of waveforms and it is usually used to encode voice signals that have non stationary nature [22]. G.711 codec is standard based and meets International Telecommunication Union - Telecommunication (ITU-T) recommendations. G.711 codec uses pulse code modulation scheme operating at 8KHz sample rate with 8bits/sample and therefore encode audio at 64Kbps channel. This is important to know to be able to plan bandwidth allocation for voice channels. In Figure 2.3, it can be seen three major parts to convert analog signal to digital: sample and hold, quantization and encoding. The PCM algorithm follows the same procedure. The quantization can be divided into uniform and non-uniform quantization. The non-uniform quantization provides great algorithm for non-stationary voice coding (i.e speech) [22]. G.711 codec comes in two different variety G.711 μ -Law and G.711 A-Law. μ -Law and A-Law compressions are defined by Equation (2.1) and Equation (2.2), respectively:

- μ -Law:

$$y(n) = F[x(n)] = X_{max} \frac{\ln(1+\mu \frac{x(n)}{X_{max}})}{\ln(1+\mu)} \text{sgn}[x(n)] \quad (2.1)$$

$x(n)$ represents sample of the input signal, $y(n)$ represents output signal after compressing and transforming, X_{max} is the maximum amplitude of input voice signal, μ is constant value which represents degree of compression. Therefore the higher μ means higher compression [22].

- A-Law:

$$y(n) = F[x(n)] = \begin{cases} \frac{A|x(n)|}{1+\ln A} \text{sgn}[x(n)], & 0 \leq \frac{|x(n)|}{X_{max}} \\ X_{max} = \frac{1+\ln[\frac{x(n)}{X_{max}}]}{1+\ln A} \text{sgn}[x(n)] \end{cases} \quad (2.2)$$

A- represents the compression parameter and $|x|$ is the normalized integer to be compressed.

The main difference between G.711 μ -law and G.711 A-law is in the number of bits that codec uses to encode PCM audio samples. G.711 μ -law encodes 14-bit linear PCM audio samples and G.711 A-law encodes 13-bit linear PCM audio samples into 8-bit logarithmic samples. G.711 A-law requires less computer processing power.

G.711 μ -law is mainly used in North America and Japan and G.711 A-law is used in most other countries outside North America. The System-of-Systems RoIP uses G.711 μ -law codec.

2.6 Quality of Service (QoS)

This research showed that deploying RoIP over WANs, owned by a third party, requires QoS to be enabled over the network. One of the reasons is that voice is sometimes used in mission critical operations and, as it was explained before, seconds can prevent loss of life, infrastructure and property. Therefore it is important to make sure that voice has guaranteed the highest priority over the network over other type of traffic such as video, data, file transfer and etc.

The internet protocol version 4 (IPv4) originally was defined to have four key mechanisms to provide service: type of service, time to live, options and header checksum [34]. IPv4 header is outlined in Figure 2.7.

Figure 2.7: IPv4 Header [35]

The type of service mechanism is used to provide capability of selecting the quality of the service desired. Type of service field is used by gateways to select desired transmission parameters for particular network when routing an internet datagram. IPv4 TOS byte is shown in Figure 2.8.

Figure 2.8: IPv4 TOS Byte [35]

To enable end-to-end QoS on the IP-network, the Internet Engineering Task Force (IETF) has defined two models: integrated services (IntServ) and differentiated services (DiffServ).

IntServ provides signaled-QoS model using end-to-end signalling, state maintenance for each flow and admission control at each network element. In IntServ QoS-model end-hosts signal the network about what QoS they need/expect. DiffServ provides a much simpler QoS mechanism compared to IntServ to categorize traffic into different classes since it eliminates the need for per-flow state and signaling at every hop. DiffServ provides provisioned QoS model where network elements (i.e gateways) are configured to service multiple classes of traffic which has different QoS requirements [35]. These classes are also known as class of service (CoS) which QoS is applied. To accomplish this DiffServ model separates packets into different CoS classes by marking the type of service (ToS) byte in the IP header. The

first 6 bits of the ToS byte are called differentiated service code point (DSCP) in the IPv4. Service-compliant network elements include a classifier that selects packets based on the DSCP field as well as provide buffer management and packet scheduling mechanism for the specific forwarding treatment which is indicated by the DSCP field [36]. A 6-bit DSCP in the IPV4 ToS octet is shown in Figure 2.9.

Figure 2.9: Differentiated Service Codepoint (DSCP) [35]

The Systems-of-System RoIP has been deployed and tested over Local Area Network (LAN) and WAN networks where DiffServ QoS-models have been deployed.

Therefore, there are two main aspects of QoS:

1. QoS within the LAN - owned and maintained by each agency or department
2. QoS over the WAN – owned by third party.

2.6.1 QoS within the Local Area Network (LAN)

Each agency and department who maintains its own LAN needs to make sure that they properly mark multicast/unicast traffic within their network in order to make sure that voice has the highest priority. This is especially important for the networks that transport mission critical voice so that, in case of congestion on the network, voice has priority over email, video and etc. For the System-of-Systems RoIP QoS has been deployed over the LAN using DiffServ QoS-Model and VLANs to separate voice traffic from other type of traffic present on the network.

2.6.2 QoS over the Wide Area Network (WAN)

Wide Area Network (WAN), which connect various networks (LANs) owned by agencies and departments, is usually provided by a third party.

The QoS over the third party WAN is usually requested as part of the Service Level Agreement (SLA) between an agency/department and a third party WAN provider. In the SLA the agency/department request a QoS over the network to be enabled in such fashion so that traffic over the WAN gets prioritized as per agencies/department needs. For reliable radio over IP communication over the third party WAN (such as BELL/Telus/AT&T), this thesis suggests that the QoS network class-maps are implemented as following (in order of importance):

- Class: VOICE, Priority: Highest: example: Radio over IP traffic.
- Class: VIDEO, Priority: High: example video conferencing and streaming video.
- Class: NETWORK-CONTROL, Priority: Medium: example: Control traffic such as access to routers by maintenance staff.

- Class: Call Signalling, Priority: Medium: example SIP phones.
- Critical: Important Applications.
- Class: PRIORITY-DATA, Priority: Low-Medium: example: Agency/Department database query.
- Class: DATA, Priority: Low: example: Email, Internet.
- Class: Class-Default, Priority: Lowest: example: FTP file transfer.

It is important to point that third party WAN providers, for the purpose of this research Telus/Bell/AT&T, provide 40% guaranteed QoS. This means that out of 100% ordered circuit bandwidth the provider guarantees 40% availability at any point in time. This is important to know when planning bandwidth allocation for radio over IP channels.

From the System-of-Systems RoIP perspective to calculate bandwidth per voice channel the following requirements need to be understood: codec (G.711), voice payload, headers, number of packets/second sent, layer 2 headers, encryption [21].

- G.711 (64K PCM) – 8KHz sampling * 8bits/sample = 64Kbps
- Voice payload - By default in Cisco IOS voice payload size can be set to 10ms, 20ms or 30ms. It is recommended for optimal communication to use 20ms payload (default value). 20ms sample (G.711 codec) = 160 bytes payload/ packet
- IP/UDP/RTP headers – 40 bytes/packet
- Number of packet/second Cisco IOS by default (all VoIP traffic- media RTP) – 50 packets/second

- Layer 2 – generic routing encapsulation (GRE) tunnel – 4 bytes
- Encryption – internet protocol security (IPSec) – 16 bytes

Therefore the amount of bandwidth consumed by each VoIP call can be calculated:

$$BW = (160 \text{ bytes} + 40 \text{ bytes} + 4 \text{ bytes} + 16 \text{ bytes}) * 50 \text{ packets/second} = 88\text{Kbps}$$

Our testing showed that for reliable voice communication under real world network conditions (i.e change in delay, jitter and packet loss) actual bandwidth should be 128Kbps.

Since the third party WAN providers guarantee 40% QoS it is recommended to allocate 60% more bandwidth to ensure that 128Kbps bandwidth is allocated per voice channel for voice communication at any given point in time. Bandwidth will be discussed in more detail further in the next chapter.

2.7 Application Servers

For the System-of-Systems RoIP deployment we have tested two off-the-shelf RoIP application servers for channel management and conferencing/mixing: IP interoperability and collaboration system (IPICS) from Cisco Systems and Wave from Twisted Pair.

2.7.1 IP Interoperability and Collaboration System (IPICS)

Cisco IPICS interoperability platform enables various radio end points to communicate over IP network. Cisco IPICS hosts virtual radio conference calls by creating virtual talk groups (VTGs) to facilitate push-to-talk (PTT) communications between users of multiple technologies and types. Each radio channel can be assigned to desired talk group. Talk groups can be predefined or created on-the-fly [27][28].

The Cisco IPICS server is a linux based platform with enhanced security installed on the multiservice platform server (hardware based). Other Cisco IPICS components include the Cisco IPICS dispatch console (IDC), Cisco mobile client, Cisco IPICS IP phone client, Cisco IPICS operational views (Ops Views), Cisco land mobile radio (LMR) gateways, unified media server, and session initiation protocol (SIP) telephony gateways [28]. Cisco IPICS server provides configuration, administration and security services. Cisco IPICS platform is outlined in Figure 2.10.

Figure 2.10: Figure2.10: Cisco IPICS interoperability platform [28]

For the purpose of System-of-Systems RoIP to test voice interoperability the following components of the IPICS platform were used: IPICS server, IPICS dispatch console (IDC), land mobile radio (router with VIC3-2E&M voice card) gateways and unified media server (UMS). IPICS server and unified media server were centralized in the System-of-System core and land mobile radio (LMR) gateways/

IPICS dispatch console (IDC) were located within various agencies/departments to manage radio channels/talk groups.

2.7.2 Wave Interoperability platform

Another interoperability platform that was tested to host radio channel conferencing and to provide real-time radio channel management capability was Wave interoperability platform provided by Twisted Pair. Wave is a software based platform and suite of applications that hosts and manages radio channels and radio channel conferences using voice over IP technologies [29].

Wave interoperability platform has core servers called Wave management server, Wave domain server and Wave media server that provide control mechanism and ability to manipulate audio (receive, process and transmit) from different communication technologies including two-way radio.

Wave management server is a web based application that provides configuration services for a wave system, administration and security services for wave domain and it can be hosted on windows server 2003 SP2 or windows server 2008 R2 system. Wave domain server is wave management server that controls a collection of wave components. Wave media server enables non IP based devices such as two-way radio to communicate over wave domain. Wave media server converts audio streams (i.e voice) into IP packets and vice versa. Wave media server creates patches between audio streams from different radio systems (radio channel conference). Patches can be predefined and created on-the-fly [29].

Wave system also has number of wave clients such as wave dispatch console. Wave dispatch console enables users manipulate radio channels such as radio channel patching (conferencing). Wave interoperability platform is outlined in Figure 2.11.

Figure 2.11: Wave Interoperability platform [31]

Chapter 3: System-of-Systems Radio over IP Network Characterization

To deploy Systems-of-Systems Radio over IP over the WAN technologies the basic approach is to first characterize network characteristics through emulation and live circuits (MPLS). The purpose is to determine baseline requirements such as bandwidth, delay, jitter, packet loss for reliable communication for various radio systems under test such as Motorola, Harris, Codan and Cassidian. Also it is important to determine: transport protocol (i.e UDP, TCP), protocol port number and compression codec. To achieve the above proposed we test:

1. Radio over IP SIP solution (using point to point trunks and/or M1U12M2)
2. Conventional radios such as Codan and Motorola
3. Trunked radios such as Cassidian, Harris and Motorola.

Also from the radio feature perspective our approach is to determine the impact of network impairments (change in network characteristics: bandwidth, delay, jitter and packet loss) on signals generated by features such as:

- a) Push-To-Talk (PTT)
- b) Request-To-Talk (RTT)
- c) Emergency Request-To-Talk (ERTT)
- d) DTMF tones

The goal of this chapter is to provide: i) connection setup protocol, transport protocol, protocol port number and compression codec, ii) test setup to characterize the network, iii) baseline requirements such

as bandwidth, latency, jitter, and packet loss for radio over IP deployment, iv) radio features and network change impact on the radio features.

3.1 Connection setup, transport protocol, port number and compression codec

In the previous chapter it was explained that our research showed that choice of connection setup protocol and transport protocol depends on what type of connection trunks between radio over IP end points is used. In case of unicast point-to-point trunks, SIP protocol is used to establish call between end points and UDP-real time protocol (RTP)/RTP control protocol [19] is used as transport protocol to enable datastream (audio/voice) flow between the end points. In case of multicast connection trunks, multicast connection trunks do not use connection setup protocol. Multicast connection trunks use protocol independent multicast (PIM) to enable multicast routing and to direct RTP/RTCP datastream (audio/voice) packets among end points. Systems-of-Systems radio over IP solution deploys both unicast and multicast connection trunks.

Protocol port number for SIP is 5060 (non-encrypted signalling) and 5061 (encrypted signalling), UDP/RTP and UDP/RTCP, 5004 and 5005 respectively. Application servers use 21000.

In the previous chapter it was explained that this research showed the G.711 codec is the most suitable for 4-wire and 2-wire analog audio. During our testing various codecs, supported by VIC3-2E&M were tested and voice quality was subjectively measured. It was concluded that G.711 codec provides the best voice quality with no noticeable voice quality degradation due to transcoding and packet loss.

3.2 Overview of the test setup to characterize the network

The test setup to determine network requirements for reliable radio over IP communication was developed in phases.

In phase one an approach was to develop a simple lab test setup to prove that radio-to-radio can communicate over IP using unicast point-to-point trunks establishing VoIP calls between end points (LMR gateways) using SIP Protocol. Once voice call between radios over unicast point-to-point trunks was successfully established and tested, GRE tunnels with IPsec were configured and voice flow was tested again. Phase I lab test setup is shown in Figure 3.1 and Figure 3.2.

Figure 3.1: Phase I: Lab test setup point-to-point trunks

Figure 3.2: Phase I: Lab test setup point-to-point trunks over GRE/IPSEC Tunnel

In phase two the approach was to deploy WANem emulator tool between end points to simulate real WAN conditions (impairments in the network) to determine baseline requirements (bandwidth, delay,

jitter and packet loss) and codec for reliable radio over IP communication. Phase II test setup is shown in Figure 3.3.

Figure 3.3: Phase II: Lab test setup point-to-point trunks over simulated Wide Area Network (WAN)

In phase three the approach was to deploy unicast point-to-point trunks with GRE/IPSEC over third party WAN (MPLS IP circuits without QoS enabled) during the field test trial to test voice and radio features. In addition 24hr continuous audio recording tests were performed to determine radio over IP availability (audio (packet loss) over 24hr). 24hr test were performed over 10 consecutive days .Once testing was completed, QoS was enabled as per section 2.6.2 and tests were repeated. To record and analyze audio over the RoIP network a laptop with Audacity software was used. Audacity is a free open source digital audio editor and recording software application available for Windows, Mac OS X, Linux and other operating systems [25][26]. Audacity has capabilities to detect silence gaps present in audio recording. For this testing audio file of 24 hr (mp3) was recoded and analyzed by Audacity silence finder tool to ensure that no silence gaps was present in the recording. Audio was then played by laptop over the RoIP network for 24hrs and recoded on another end using other laptop with Audacity software. Recoded audio was tested with Audacity silence finder to ensure that no audio was missing. If audio was

found missing, network analysis was performed to understand what happened over the network at the time when silence was recorded. Phase III test setup is shown in Figure 3.4.

Figure 3.4: Phase III: Field test setup point-to-point trunks over third party (BELL/Telus) WANs

Phase four's test setup was developed in the lab environment using M1:U12:M2 trunks. Multicast traffic was enabled in multicast islands (simulating agencies or departments) which were connected over simulated WAN using M1:U12:M2 trunks. In this phase, Wave and IPICS servers (management/media) and dispatch consoles were deployed to provide management and radio channel mixing capabilities (to host radio channel conferences). This test setup showed not only Radio over IP interoperability but also that agencies/departments can deploy either of these two RoIP platforms (IPICS/Wave) to enable RoIP communication since they are standard based solution (IP based, supports multicast/unicast deployment, standard codecs i.e G.711). Phase four's test setup is shown in Figure 3.5.

Figure 3.5: Phase IV: Lab test setup - M1:U12:M2 trunks over simulated WAN

Also in this phase our contribution was to determine how much of an audio delay was contributed by M1:U12:M2 connection trunks multicast-to-unicast conversions. In our conversation with Cisco, this information was not available as delay due to M1:U12:M2 conversions were not examined. To achieve this, our approach was to determine end-to-end delay between simulated agencies with communication path established by enabling end-to-end multicast and then with communication path established using M1:U12:M2 connection trunks. Delay due to M1:U12:M2 conversions was to be determined as the difference between the two delays.

Firstly end-to-end multicast between two simulated agencies (A and B) was enabled. In simulated agency A, a Codan radio connected to the LMR gateway was used to generate PTT/COR signalling and audio. On the server two multicast channels Ch1 and Ch2 were configured. On a VIC3-2E&M voice card in the LMR gateway ports were configured with the multicast IPs (Ch1, Ch2). In agency B, the same multicast channels were provisioned on the server. Also, in agency B Ch1 and Ch2 were patched/conferenced to create a loopback. In this way an audio in agency A was sent from a radio to a port on the VIC3-2E&M voice card (Ch1) to agency B, looped back using patch on Ch2 and received on another port of the same VIC3-2E&M voice card (associated with Ch2) in agency A. An oscilloscope was used to measure audio delay difference between Ch1 and Ch2. Round trip audio delay difference between Ch1 and Ch2 on average was found to be 117ms.

Secondly, in the very same test setup connection between two agencies was switched to M1:U12:M2 connection trunks. M1:U12:M2 connection trunks associated with Ch1 and Ch2 were established between the gateways and audio delay was measured again. In this way audio in agency A was sent from a radio to a port on the VIC3-2E&M voice card (Ch1), over M1:U12:M2 trunks to agency B, looped back using patch in agency B on Ch2 and received on another port of the same VIC3-2E&M voice card (associated with Ch2) in agency A. Round trip audio delay difference between Ch1 and Ch2 was found to be 467ms.

M1:U12:M2 delay contribution was determined to be $= 467\text{ms} - 117\text{ms} = 350\text{ms}$ (round trip - two conversions). Therefore delay of the one M1:U12:M2 conversion was determined to be $350\text{ms} / 2 = 175\text{ms}$.

Our conclusion from this test was that if M1:U12:M2 connection trunks are to be used to connect agencies, 175ms has to be accounted for every M1:U12:M2 conversion.

Phase five was field deployment of two centralized RoIP platforms (IPICS) in two different System-of-Systems RoIP networks connected by M1:U12:M2 trunks over third party WAN (MPLS IP). Each IPICS RoIP platform connected multiple departments/agencies using GRE/IPsec tunnels over third party WAN (MPLS IP).

In Phase six, QoS was enabled over the networks, overall system delay was measured, analysed and characterised in the great level of details. Systems-of-Systems stability was examined with 24 hrs continuous audio test for multiple days. Phase six will be assessed in greater details in the next chapter. Phase six architectures represent the final System-of-Systems radio over IP to provide interoperability among various agencies and departments to characterize RoIP delay, availability and audio quality over the large distances and multiple third party WAN, as the main focus of this thesis.

3.3 Overview of baseline requirements bandwidth, latency, jitter, and packet loss for Radio over IP deployment

Baseline requirement, bandwidth, latency, jitter and packet loss requirements for reliable radio over IP communication were determined using the test setups developed in the Phase II (lab test setup) and Phase III (field verification test setup including third party WAN) explained in 3.1 section above. To determine network requirements for various radio systems over the MPLS IP circuit the following approach was used: analog radio-to-analog radio, P25 digital radio - to P25 digital radio and P25 trunking radios. Baseline requirements for reliable voice communication are outlined in the following tables:

Table 3.1: Baseline characteristics - Analog radio-to-Analog radio

Network requirements for reliable voice communication - Analog radio-to-Analog radio	
Bandwidth	128Kbps per voice channel
Jitter	System dependant ($\leq 12\%$ of latency)
Latency	≤ 200 ms
Packet Loss	$\leq 1\%$

Table 3.2: Baseline characteristics - P25 Digital radio-to-P25 Digital Radio

Network requirements for reliable voice communication – P25 Digital radio-to-P25 Digital radio	
Bandwidth	128Kbps per voice channel
Jitter	System dependant
Latency	≤ 200 ms
Packet Loss	$\leq 1\%$

Table 3.3: Baseline characteristics - P25 Trunking

Network requirements for reliable voice communication – P25 trunking	
Bandwidth	128Kbps per voice channel
Jitter	< 25% of Latency
Latency	< 200 ms
Packet Loss	≤1%

Bandwidth: This testing proved that bandwidth allocation per voice channel is directly proportional to requirements of a selected codec. To prove this various codecs (i.e G.711, G.729) were programed on the VIC3-2E&M voice card and bandwidth allocation over the network was limited using WANem emulator tool. While bandwidth allocation testing was performed a voice quality of each codec was subjectively measured as well. It was concluded for radio over IP (analog, digital and P25 digital - both trunking and conventional) that 128Kbps of bandwidth per radio channel should be allocated for reliable RoIP communication under real world conditions (i.e. with network impairments present).

Latency: This testing showed that latency did not affect the RoIP technology but it affected user experience when it was over 250-300ms. It is important to understand that human ear can tolerate around 200ms of latency without noticing [57]. This means that overall delay of the RoIP IP backbone (digitalization of the signal, packetization of the signal and encryption) should be ≤ 200 ms. If delay is above 500ms, the delay becomes noticeable and creates problem for user experience [32]. In our testing of var-

ious radio systems over IP network latency was measured to be $< 200\text{ms}$. It is important to understand that each radio system (analog, digital, P25 - conventional and trunking) has different inherent delay. This means that an overall end-to-end delay will exceeds 200ms . Our goal is not to make existing radios better but to ensure that delay over the IP backbone is properly characterized and propose an approach to keep the delay $\leq 200\text{ms}$. This will be addressed in more detail during next chapter.

Jitter: Jitter is variation in the amount of latency in consecutive packets and it can severely impact voice services [32]. This testing showed that jitter can have significant impact on the RoIP network and that QoS over the network must be enabled. Our testing showed that jitter affected the various radio architectures in varying degrees of severity (≤ 12 or $\leq 25\%$ of latency, system dependant). As it was explained before analog voice gets digitized into chunks of data that is transmitted in packets and then at the other communication end (receiver) it gets converted back to analog stream. The digitization produces a steady stream of packets with constant time between each packet. It is important that RoIP network devices have built-in buffers and it is crucial to understand their capability to handle network jitter. If jitter is not managed properly (i.e. de-jitter buffer is not set as needed to reduce the impact of jitter) and latency of consecutive packets differs too much then the conversion back to analog signal can fail because the required packets are not available in the buffer at the time when they are required to be converted into meaningful analogue signal [32].

Packet Loss and QoS: As it was explained through this thesis, G.711 was selected as optimum codec solution for System-of-Systems RoIP deployment (64Kbps bandwidth per voice channel). In the section 2.6.2 it was explained that GRE tunnels with IPsec (to secure voice traffic over the network) increase

bandwidth requirements for an additional 20Kbps. It was shown mathematically and proven in our lab that for one voice channel it is needed to allocate 88Kbps of bandwidth. This was the case if no network impairments were present. On the other hand the testing showed that network characteristics (i.e. jitter, packet loss) can vary and therefore it is necessary to ensure that two levels of QoS are enabled as per section 2.6.2. Our testing showed that bandwidth of 128Kbps is sufficient enough to manage jitter, latency and packet loss for reliable voice communication. Having said that, third party WAN providers only guarantee 40% QoS of the ordered circuit bandwidth and therefore it is recommended for reliable communication to allocate 60% more bandwidth per voice channel to ensure that desired channel bandwidth is available at all times. For example, to make sure that required 128Kbps of bandwidth per voice channel is available at all times over the third party WAN (assuming that third party network does not experience some major service disruptions) is necessary to allocate 320Kbps. Please note that circuit bandwidth can be ordered in increments of 128Kbps. Therefore circuit of 384Kbps needs to be ordered to ensure that 128Kbps of bandwidth per voice channel is available at all times. Our testing with QoS and without QoS was conducted over 24hr long tests (24 hr continuous audio) over 10 consecutive days. Number of packet lost without QoS and with QoS over 24 hr continuous audio tests for 10 consecutive days is outlined in Figure 3.6.

Figure 3.6: Voice traffic packet loss over WAN (MPLS IP circuits) without and with QoS enabled

It is necessary to distinguish acceptable packet loss in terms of the audio (voice) packet loss and packet loss of the data generated by the radio features (for radio features and radio features packet loss please refer to the next section 3.4). Our testing showed that audio packet loss with QoS enabled was found to be $< 1\%$.

It is necessary to understand that RoIP (voice) traffic coexists with other types of traffic, such as video and data, over the third party WAN. Therefore it is important to have QoS enabled to make sure that voice has the highest priority over other type of traffic. The reason for this is that voice can be sent over the network in mission critical situations (i.e potential loss of life, property and infrastructure) and it has to have higher priority over other types of traffic such as email, video download etc.

3.4 Overview of radio features and network change impact on the radio features.

Radio features tested over IP network are:

- Push-To-Talk (PTT)
- Request-To-Talk (RTT)
- Emergency Request-To-Talk (ERTT)
- DTMF tones

Push-to-Talk (PTT), in two-way radio is also known as press-to-transmit. This feature enables conversations or talking on half-duplex communication lines using a button to switch from voice reception mode to transmit mode [33].

Request-To-Talk (RTT) is a radio feature which enables radios to send data packet to request a channel to transmit signals (i.e voice). In case that there is no channels available, RTT gets placed into the queue until channel is available (used in trunking radio systems).

Emergency Request-To-Talk (ERTT) is a radio feature which enables radios to send an emergency message which contains the radio ID. This feature can be of high importance in mission critical situations. ERTT is usually sent over a specific (dedicated) radio channel rather than over the channel selected by the operator using the channel selector [36].

DTMF tones – DTMF stands for Dual Tone-Multi frequency. When buttons on a radio are pressed a connection is made that generates two tones at the same time. Each button presents combination of the two frequencies [37]. DTMF tones are shown in Figure 3.7.

1	2	3	697 Hz
4	5	6	770 Hz
7	8	9	852 Hz
*	0	#	941 Hz
1209 Hz	1336 Hz	1477 Hz

Figure 37: DTMF -Tones

Each feature explained above plays a critical role for mission critical situations and loss of a packet is not accepted. For mission critical radio system maximum tolerable packet loss when it comes to radio features is $\leq 1\%$. For example, out of 100 PTTs or ERTTs only one can fail. For this testing each feature was tested 100 times and packet loss was found to be $\leq 1\%$

Chapter 4: System-of-Systems RoIP deployment examples and Experimental results

In this chapter, firstly architectures of the System-of-Systems RoIP solution that were deployed and tested over the LAN and WAN technologies are discussed. Secondly test results, with focus on the system delay characterization, system availability (packet loss) and voice quality are presented, compared and discussed for various radio systems. Following this, recommendations are provided to to reduce overall IP backbone delay. In the last part of this chapter, the System-of-Systems RoIP deployed solution is compared to other proposed RoIP interoperability solutions and it is discussed lack of testing results availability from other researchers and vendors.

4.1 System-of-System RoIP deployment examples

The System-of-Systems RoIP interoperability solution was deployed and tested in two architectures. The first architecture has one centralized IPICS RoIP platform (IPICS server and UMS server) and simulates RoIP network of one agency with multiple departments connected in the hub-spoke architecture. This architecture has multicast enabled in LANs as per Figure 4.1 below and connections among sites are established using GRE tunnels with IPsec (AES 256) encryption. The second architecture simulates RoIP networks where two agencies with multiple departments were connected over the M1:U12:M2 trunks. Please note that for this testing two centralized IPISC RoIP platforms (IPICS Server and UMS Server) were deployed (one per simulated agency) as per Figure 4.12.

In the Phase IV, which is outlined in the section 3.2, it was tested that both IPICS and Wave platforms can be deployed simultaneously since they are standard based technologies and therefore interoperable (both platforms support: unicast/multicast, codecs (g.711, g.729 etc), RTP/RTCP protocols etc). The

reason for continuing testing of the System-of-Systems RoIP solution with IPICS RoIP platforms only, was that the thesis follows an enterprise approach by deploying (cost vs. performance) solution using off-the-shelf products. Wave RoIP platform was equally good RoIP platform performance wise, but more expensive. Supporting our decision is that our test network was built mainly from the Cisco standard based components (i.e routers and switches) so it made sense to propose IPICS as RoIP interoperability platform for radio channel management and channel mixing/conferencing.

4.1.1 System-of-Systems RoIP deployment architecture I - (one agency-multiple departments)

The first architecture System-of-System RoIP that was deployed over the third party WAN (BELL) is outlined in Figure 4.1.

Figure 4.1: System-of-Systems RoIP architecture I (simulating agency X with multiple departments)

As it is shown in Figure 4.1, the System-of-System RoIP architecture has one centralized IPICS server and one unified media service (UMS) server. The system was deployed in the hub-spoke architecture where hub site was the site where the centralized IPICS server and UMS servers were housed. Spoke sites represented various departments of one agency where radio systems and LMR gateways were located. Connections between spoke sites and the hub site were established using GRE tunnels with IPsec encryption over third party WAN. Multicast was enabled in both, the hub site and in the spoke sites LANs using the PIM-SM. The core's gateway router was declared as a multicast rendezvous point. A rendezvous point is a router in a multicast network domain that is configured to be a shared root for a multicast shared tree [58]. In this architecture all the LMR gateways and the dispatch consoles had to register with the rendezvous point first and send/receive desired multicast to/from the rendezvous point thereafter. Routing was deployed by enabling open shortest path first (OSPF) routing protocol [38][39]. LMR gateways were located in close proximity to the third party WAN access point. Bandwidth of 128Kbps was allocated per radio channel, codec was selected to be G.711, and QoS which was marking voice at the highest priority was deployed in both LAN and WAN networks. Once communication paths were established, voice traffic was tested among radio sites using portable radios, base station radios and dispatch consoles. Firstly we tested one channel at the time. Secondly, radio channels were conferenced/mixed and communication was tested again with success among radio sites. Voice quality was subjectively measured as per TSB 88 DAQ 3.4 [42]. The traffic was recorded and analysed using Audacity software to test any impairments in audio (packet loss).

4.1.1.1 Test Strategy

Test strategy was developed to examine overall delay of the proposed System-of-Systems RoIP architecture as well as to examine how much of that overall system delay is contributed by each system's component. This testing determines packet loss over the network with QoS enabled as well subjectively measures voice quality. The testing also determines inherent processing delay of each radio system under the test which includes both trunked and conventional radio systems such as: Codan (VHF analog/VHF-UHF-800 P25 digital), Harris (UHF digital, proprietary), Cassidian (UHF P25 digital - IP based), and Motorola (VHF, ASTRO 25 digital).

To accomplish what was stated above the following steps were taken:

- Determine delay of the IP core at the hub site.
- Determine delay between spoke sites and the hub site including inherent delay of each radio system under test.
- Determine system availability (packet loss) and voice quality.

4.1.1.2 Test Results

4.1.1.2.1 Delay of the IP core at the hub site

Delay of the IP core at the hub site was examined as per the test setup in the Figure 4.2 below in the following steps:

- Determine processing delay of the VIC3-2E&M port (analog signal digitization and packetization (IP) processing delay).

- Determine UMS processing delay - patch/conference of the radio channels.
- Determine network delay of the IP backbone at the hub site.

Figure 4.2: Delay measurement test setup at the HUB site

VIC3-2E&M voice port processing delay was determined using the following approach. Two ports (0/0 and 0/1) on one VIC3-2E&M card in the LMR gateway were programmed with the same multicast IP. Test radio (Codan) was connected to one port on the VIC card to generate PTT/COR signalling and audio. An oscilloscope was used to capture audio wave forms: probe 1 (audio in - port 0/0) was connected to audio output at the back plane of the radio and probe 2 was connected to the output of the second VIC3-2E&M port (audio in - port 0/1). The delay difference between the two captured audio

wave forms was measured to be on average 72ms. Therefore processing delay of one VIC port was calculated $\sim 72\text{ms}/2 = 36\text{ms}$.

UMS processing delay was determined using the following approach. Two ports on one VIC3-2E&M card in the LMR gateway were configured with two different multicasts IPs. Two channels with same multicast IPs were configured on the IPICS server as well and assigned to a user profile (i.e tester). Using Dispatch console, the user created patch/conference for those two channels. Digital P25 VHF Codan radio was used to generate PTT/COR signaling and audio. An oscilloscope was used to capture audio wave form on the probe 1 (audio in - port 0 on the VIC3-2E&M) and probe 2 was used to capture audio waveform at the output (port 1 on the VIC3-2E&M). Taking into account that processing delay of the VIC3-2E&M port was measured to be $\sim 36\text{-}40\text{ms}$, processing delay of the UMS was found to be $\sim 40\text{ms}$.

Network delay of the IP backbone was determined using “ping” test. From the LMR gateway continuous ping test (with payload of 1000 bytes) was issued towards the UMS from dispatch console and LMR gateway and it was determined that on average network delay was found to be: 1ms (maximum 5ms).

Therefore IP backbone at the hub site adds maximum processing delay of $\sim 40\text{ ms}$ for the conference and 1ms for the network delay.

4.1.1.2.2 Delay between a spoke site and hub site

To determine delay from a spoke site to the hub site the following steps were taken:

- Determine inherent latency of the radio system - processing delay of the radio system from the moment PTT was issued to the moment that audio was present on the VIC3-2E&M port and vice versa. Determine inherent audio latency thereafter.
- Determine network delay from spoke sites (LMR gateways) to the hub site (UMS).
- Determine round trip delay from a spoke site (LMR gateway) to the hub site and vice versa.
- Determine end-to-end delay (user-to-user, a real life scenario)

Inherent latency of the radio systems was determined at one radio system at the time. The test setup that was used to determine inherent delay of various radio systems is outlined in Figure 4.3.

Figure 4.3: Radio inherent delay - test setup

Inherent latencies of the radio systems are outlined below:

1. Department A - Codan (VHF/UHF/800MHz, P25 digital and analog) conventional radios.

The testing showed that processing delay of the radio system is not the same when signal was sent from the radio to the LMR gateway and when the signal was sent to the radio from the LMR gateway. Inherent latency of the Codan analog radio system with CTCSS tone activation was found to be: ~378ms radio to LMR Voice card (~212 for CTCSS tone processing and ~ 166 for voice processing) and ~262ms (~107 for CTCSS tone processing and ~ 155 for voice processing). On the other hand inherent latency of the Codan P25 digital radio system with NAC codes activation was found to be ~ 229ms radio to LMR Voice card (~63ms for the NAC code processing and ~166 for voice processing) and ~257 (~63ms for the NAC code processing and ~193 for voice processing) (Appendix A1). Please note that for Codan analog radio system with CTCSS tone activation, CTSS tone is present for the entire period. Therefore and overall processing delay is 378ms and 262ms, respectively.

2. Department B - Harris (UHF, digital proprietary) trunking radios

Inherent delay of the Harris radio system needs to be discussed in more details. Since this is a trunking radio system, there is a channel controller involved to allocate a channel prior to audio transmission. Therefore delay from the moment PTT was issued and the moment at which voice actually could be transmitted needed to be determined as well as audio processing delay thereafter. The testing showed that processing delay of the radio system is not the same when signal was sent from the radio to the LMR gateway and when the signal was sent to the radio from the LMR gateway. This is due to the way that radio system architecture was designed.

Inherent latency of the Harris (P25 digital - proprietary) trunking radio system was found to be 1081ms-1174ms for channel allocation and 551-553ms for audio processing (Appendix a1).

3. Department C - Cassidian (UHF, P25 digital - IP based) trunking

Similarly as in the previous case, Cassidian radio system is trunking based radio system and therefore delay from the moment PTT was issued and the moment at which voice actually could be transmitted and audio processing delay thereafter needed to be determined. Also it was noted that processing delay of the radio system is not the same when signal is sent from the radio to the LMR gateway and when the signal is sent to the radio from the LMR gateway. This is due to the way that radio system was designed. Inherent latency of the Cassidian (P25 digital, IP based, trunking) radio system was found to be from 757ms-1s for the channel grant plus audio delay of ~ 250-262 ms. Also we have examined trunked analog gateway (TAG) audio processing delay which was determined to be ~24ms (Appendix a1).

4. Department D - Motorola (VHF, Astro 25) conventional

Since this is a conventional radio system there is no channel controller and therefore audio can be sent immediately after the PTT was issued. The testing showed that the processing delay of the radio system is not the same when a signal is sent from the radio to the LMR gateway and when the signal is sent to the radio from the LMR gateway. Inherent latency of the Motorola (VHF Astro 25) conventional radio system was found to be from 482ms - 489ms for PTT processing and 266ms - 425ms for audio processing (Appendix a1).

Based on the data obtained and the worst case scenario (longest system delay), inherent latency of radio systems is graphically presented in Figure 4.4.

Figure 4.4: Inherent radio system latency: PTT to RX and audio latency

Analysing obtained results we observed that processing delay of the radio systems was found to vary based on the technology that was used such as analog, digital, P25 digital and the way the radio systems were deployed in the conventional or in the trunking architecture. In general inherent latency of the trunking radio systems, due to channel request/channel grant processing time, it was found to be longer. In our testing, it was found that Harris trunking architecture (digital-proprietary) had the longest inherent radio system delay.

Network delay of the IP backbone from a spoke site to the hub was determined by “ping” test. From the LMR gateway at each spoke site, continuous ping command (with payload of 1000 bytes) was issued towards the IPICS Server and UMS to determine network delay. Based on the spoke site distance from the hub site network delay was measured to be: Hub-Spoke site B ~ 60ms on average (maximum 63ms), Hub-Spoke site C ~ 11ms on average (maximum 14ms), Hub-Spoke site D ~ 6ms on average (maximum 7ms).

Spoke sites (LMR gateway) to the hub (core) site delays and vice versa

1) Spoke site (LMR gateway) to the hub (core) site delays

Delays from spoke sites to the hub (core) site were determined using both soft patches/conferences (created by UMS) and hard patch using a physical loopback cable. In both cases LMR gateways (i.e VIC3-2E&M ports 0/0 and 0/1) in spoke sites B, C and D were programmed with appropriate dedicated multicast IPs associated with spoke sites radio Ch1 and Ch2. In the IPICS server channels with the same multicast IPs were configured accordingly for each radio site.

a) Soft patch (UMS)

IPICS dispatch console was used to create a loopback by patching/conferencing Ch1 and Ch2 for every spoke site. Once patches were created voice traffic was injected into the voice port 0/0 directly (Ch1) using IFR (i.e 1KHz test tone generator). The voice traffic was sent to the hub (core) site and through the patch was looped back to the spoke site on Ch2. Oscilloscopes in the spoke sites were used to capture both Ch1 and Ch2 audio waveforms directly of the voice ports

and to determine audio delay. The test setup that was used for the testing is outlined in Figure 4.5.

Figure 4.0.5: Spoke site to hub site delay (traffic looped with soft patch)

Delays between spoke sites A, B, C and D (LMR gateways) and the hub (core) site using soft patch were determined to be on average 59ms, 97ms, 64ms and 62ms, respectively (Appendix A2).

- b) Hard patch (loopback cable) – Physical loopback cable along with an additional LMR gateway were used in the hub/core to loopback traffic to the spoke sites as per the test setup in Figure 4.6 below. In the core, LMR gateway VIC3-2E&M voice ports were configured with the same multicast IPs (Ch1 and Ch2) as in the spoke sites. Delay was measured between the spoke sites and the core one at the time. To connect back-to-back two VIC3-2E&M voice ports the following needed to be done:

- Configure spoke sites VIC3-2E&M ports associated with radio channels Ch1 and Ch2 as E&M type V. Type V enables two routers (two signalling units) to be connected back-to-back.
- Engineer RJ-45 loopback cable as per the table below to connect two VIC3-2E&M ports.
- Place loopback cable into the ports.

Table 4.1: Physical loopback cable pin layout – router back to back connection

RJ-45 Pin	Router A Function	Router B Function	RJ-45 Pin
1	Sig. Battery	Sig. Ground	8
2	M-Lead	E-Lead	7
3	Ring	Tip	6
4	Ring-1	Tip-1	5
5	Tip-1	Ring-1	4
6	Tip	Ring	3
7	E-Lead	M-Lead	2
8	Sig. Ground	Sig. Battery	1

Figure 4.6: Spoke site to hub site delay (traffic looped with loopback cable)

It should be noted that measuring delay using the hard patch involved two additional traffic conversions (IP-digital-analog-digital-IP) due to the additional 2xVIC3-2E&M comparing to loopback done by UMS patch. This needs to be taken into the account when analysing the test results.

1) Loopback cable placed in the hub (core site)

Delays between spoke sites B, C and D (LMR gateways) and the hub (core) site with loopback placed in the Hub site were determined to be on average 116ms, 90ms and 87ms, respectively (Appendix A2).

Note: Since spoke site A and the hub site are collocated, only soft patch test was conducted.

2) Loopback cable placed in the spoke sites

Latency from the hub site to the spoke sites was measured using hard patches only. The reason was that the System-of-Systems RoIP solution had the centralized UMS server to host channel conferences and therefore we could not take advantage of the soft patches to loopback traffic to the hub site. To loop

traffic back from the spoke sites to the hub site a physical loopback cable was used at the spoke sites LMR gateways. The rest of the procedure was the same as per the hard patch procedure above. Delays between the hub site and spoke sites B, C and D with loopback placed in the spoke sites were determined to be on average 114ms, 91ms and 88ms, respectively (Appendix A2).

End-to-End delay (user-to-user, a real life scenario)

To simulate a real life scenario and to determine portable-to-portable (end-to-end) round trip delay, two portable radios were programmed on two different RF channels, in each department, to generate PTT/COR and audio signals (voice). The radios (base stations) were connected to two different voice ports on the LMR gateway (i.e 0/0 and 0/1). The voice ports were configured to transmit on two different multicast IP addresses. Two radio channels with the same multicast IP, as the one on the voice ports, were programmed on the IPICS server and using the dispatch console (in the spoke site or in the core) they were patched (placed into conference or virtual talk group). Oscilloscope probe 1 (Ch1) was connected to the mic of the radio 1 and probe 1 (Ch2) was connected to the speaker (audio out) of the receiving radio 2. Round trip (end-to-end) latency of the radio-to-radio (portable-to-portable) over IP was measured as the difference between Ch1 and Ch2 audio waveforms. In addition, spoke site to the hub site latency was calculated as round trip latency/2. Test setup is outlined in Figure 4.7.

Figure 4.7: End-to-End audio latency test setup

For each radio system tested, spoke site to the hub site and vice versa, end-to-end latency test results are outlined in the tables below:

- 1) Department A - Codan (VHF analog and P25 digital VHF/UHF/800MHz) end-to-end latency.

End-to-end latency between Codan VHF analog radios was found to be 432ms for PTT to Rx audio and 318ms for audio processing thereafter. End-to-end latency between Codan digital UHF/800MHz radios was found to be 245ms for PTT to Rx audio and 356ms for audio processing thereafter (Appendix A3).

- 2) Department B - Harris (UHF, digital proprietary) end-to-end latency.

End-to-end latency between Harris (UHF, digital proprietary) radios was found to be 2422ms for PTT to Rx audio and 1275ms for audio processing thereafter (Appendix A3).

- 3) Department C - Cassidian (UHF, P25 digital) end-to-end latency

End-to-end latency between Harris (UHF, digital proprietary) radios was found to be 1885ms for PTT to Rx audio and 759ms for audio processing thereafter (Appendix A3).

4) Department D - Motorola (VHF, Astro 25) end-to-end latency.

End-to-end latency between Motorola (VHF, Astro 25) radios was found to be 1095ms for PTT to Rx audio and 820ms for audio processing thereafter (Appendix A3).

4.1.1.2.3 Packet loss and voice quality

Packet loss was determined with QoS enabled as well as voice quality was subjectively measured as per TSB 88 DAQ [42]. To achieve this following approach was used:

Mp3 audio (song) in length of 24hr was created using Audacity software [25] [26]. After the mp3 audio file was created, it was analysed using Audacity silence finder feature to make sure that no silence gaps were found. Audacity and silence finder features are shown in Figure 4.8.

Figure 4.8: Audacity and silence finder features

The test setup to conduct 24hr continuous audio test is shown in Figure 4.9.

Figure 4.9: 24 hr audio recording and analysis test setup

The test setup used Mp3 player and radio test set IFR 2975 [41] to inject audio into a radio system on Ch 1. Audio was sent to the hub (core) on Ch1, mixed in the UMS and looped back on Ch2. Both Ch1 (outbound audio) and Ch2 (inbound audio) were recorded using laptop with Audacity software. Ch1 was recorded to make sure that audio processed through the radio system did not have any packet loss and Ch2 was recorded to determine if any packet loss took place over that System-of-System IP backbone. The 24hr test was repeated for ten consecutive days. Testing was conducted for each radio system, results were obtained and analysed and the packet loss over a 24 hr period was found to be < 0.01%.

Please note that as per service level agreement (SLA) with third party WAN provider (BELL) system availability for MPLS-HP circuits was 99.90%. This testing showed that system availability for the System-of-Systems RoIP setup was found to be > 99.99% as per Figure 4.10.

Figure 4.10: System availability – 24hrs continuous audio test for 10 consecutive days

Audio quality was subjectively measured as per TSB 88 DAQ to reflect the user experience. Our target was DAQ 3.4 which stands for understandable with repetition only rarely required and some noise/distortion acceptable [42]. In addition we have used received signal strength Indicator (RSSI) to measures signal strength in order to ensure that audio signal (i.e 1KHz test tone, audio level -10dBm) that was send through the system-of-system RoIP backbone was received on the receiver side with no distortion (i. 1KHz, audio level -10dBm). From the user perspective our obtained results showed high voice quality which for a user created experience that was comparable to the VoIP phone.

4.1.1.3 System-of-Systems RoIP architecture I - Analysis

Based on our testing results interoperability call flow and delay metric can be outlined as per Figure 4.11.

Interoperability Call Flow and Delay Metrics

Figure 4.11: Interoperability call flow - one agency/multiple departments

Our testing showed that inherent latency of various radio systems under the test was determined to be 235ms - 1.174s and that network delay, based on the distance, between spoke sites and hub site was found to be from 1ms-30ms. Also our testing examined the processing time of the VIC3-2E&M port as well as the processing time of the UMS, 36ms-40ms respectively. Our testing showed that LMR gateway to LMR gateway (VIC3-2E&M-to-VIC3-2E&M port) delay, based on the distance, was found

to be 117ms-194ms. With our largest tested distance of 8674Km we have proved that LMR gateway-to-LMR gateway delay was found to be 194ms < 200ms. As we pointed before, the human ear cannot detect voice delays less than or equal to 200ms. Our loopback delay test showed that delay in both direction between departments and the hub site was approximately the same 231ms, 177ms, 173ms and 228ms, 181ms, 175ms, respectively. Our testing proved that end-to-end RoIP System-of-Systems delay can be mathematically approximated by adding delays of each systems component in the communication path. In our continuous audio testing we have achieved availability greater than 99.99% which is much greater than the 99.90% guaranteed by the third party provider. Our System-of-Systems architecture has achieved high voice quality which for a user created experience that was comparable to the VoIP phone. All transmission were heard loud and clear 5/5 (signal strength 5/clarity 5). Our testing showed that trunking radio systems have larger inherent latency comparing to conventional and that that can be limited factor in communication. For example our testing showed that if end-to-end (portable-to-portable) delay is larger than 2.5s it is getting confusing when multiple parties trying to communicate. One party can start talking thinking that other party has finished. This can create confusion and parts of very important communication (mission critical) can be lost. Having said that, it is important to point again that System-of-Systems interoperability solution is not designed with goal to improve existing radio systems but to enable interoperability among the systems. The contribution of this thesis is to characterize major delay contributors such as various radio systems, LMR gateways and IP network backbone as the key design elements for interoperability. Failure to examine key design elements can results in delays, inefficiencies and ineffective decisions and solutions. Our goal is to provide benchmarks for IP backbone delay to limit LMR gateway-to-LMR gateway delay to 200ms and overall user radio to user radio (portable) less than 2.5s. Since we have determined VIC3-2E&M port and UMS

processing delays to be 36ms and 40ms, respectively we recommend that IP network delays should be limited to 80-90ms. In our second architecture example, outlined later in this chapter, we have examined IP network delay between furthest sites (11,677Km) and found to be ~107ms.

Our testing showed that by understanding the delay of each identified major systems component in the communication path, we could approximate an overall end-to-end system delay as is shown in Table 4.2. In support, we have compared end-to-end system delays obtained during our end-to-end system tests and calculated delay mathematically by summing the determined delays of the major systems-of-systems components such as: inherent latency of radio systems, LMR gateway latency, network latency and UMS channels patching /conferencing processing latency. This is based on the worst case scenario (longest delay).

Table 4.2 End-to-End System-of-Systems delay: tested vs calculated delay values based on the longest delay

Systems-of-Systems delay - tested	Systems-of-Systems delay - calculated
Department A - Codan (VHF Analog) ~ 757 ms	~ 753 ms
Department A- Codan(P25 Digital VHF/UHF /800MHz ~ 601ms	~ 599 ms
Department B - Harris (UHF, Digital proprietary) ~ 2422 ms	~ 2427 ms
Department C - Cassidian (UHF, P25 digital) ~ 1885 ms	~ 1900 ms
Department D - Motorola (VHF, Astro 25) ~ 1095 ms	~ 1089 ms

This means that by having the delay benchmarks available for each of the System-of-Systems components will help to make a decision if interoperability makes sense between certain radio systems

and their locations. This is significant in a sense that we can approximate an overall end-to-end system delay in the early stage of the project such as project planning phase.

Based on the determined delays of each identified major System-of-Systems components and our conclusion that end-to-end system delay can be approximated mathematically, the following interoperability delay matrix can be developed to reflect delays among examined sites. The delay matrix is based on the longest determined inherent latency of radios. The delay matrix is shown in Table 4.3.

Table 4.3: Interoperability delay matrix

Interoperability Delay Matrix	Site A (Hub) Codan, Analog, VHF, Conventional	Site A (Hub) Codan, P25 Digital, VHF,UHF/800 MHz,	Site B Harris, Digital proprietary, UHF, Trunking	Site C Cassidian, P25 Digital, UHF, Trunking	Site D Motorola, Astro 25, VHF, Conventional
Site A (Hub) Analog Codan VHF, Conventional	753 ms	747 ms	1694 ms	1254 ms	926 ms
Site A (Hub) Codan, P25 Digital, VHF,UHF/800MHz, Conventional	603 ms	599 ms	1545 ms	1105 ms	777 ms
Site B Harris, Digital proprietary, UHF, Trunking	1485 ms	1480 ms	2427 ms	1987 ms	1659 ms
Site C Cassidian, P25 Digital, UHF, Trunking	1381 ms	1376 ms	2323 ms	1900 ms	1555 ms
Site D Motorola, Astro 25, VHF, Conventional	764 ms	759 ms	1803 ms	1362 ms	1089ms

As per the interoperability matrix above, it can be concluded that interoperability between trunking radio systems can cause some significant delays due to the large inherent latency of the radio systems. In our case, interoperability among radio systems under the test reflected that end-to-end delay in the worst case scenario was found to be less than or equal to 2.422s. This was acceptable from the user perspective. In case that the worst case scenario exceeds 2.5s, operability will have to take place in the sense that end users will have to be trained to ensure that one conversation is completed prior to starting a new one. Otherwise parties involved in the conversation would talk over each other and parts of conversation would be lost.

Our research observed a significant find during our testing which was related to the patching mechanism provided by UMS. The testing showed that COR/PTT signalling was “lost” when channels associated with the VIC3-2E&M ports were patched using UMS. Having said that, the testing did confirm that VIC3-2E&M voice card did accept 4-wire audio along with COR signalling as per design. Unfortunately when channels were patched and COR was issued, the other end was not activated if no voice was present (PTT was not detected). This was discovered when COR signalling was simulated on the card itself by shorting “M” lead to signal battery (pins 1 and 2) with no voice present and balanced input impedance to 600Ohm.

To confirm that signalling is lost due to the patch created by UMS, we have conducted another test with physical loopback. We have bypassed UMS and patched channels using LMR gateway on another end with physical loopback cable. It was confirmed that PTT gets and stays active as long as the COR was applied on other end of the communication path. Our research showed that Cisco is using RTP packets

to map PTT/COR to M and E leads, respectively, on an E&M router interface (VIC-32E&M) [50]. But this is not the case if channels are patched through UMS. Based on the Cisco's documentation if no RTP (in our case voice) packets are present in both send and receive, the call is considered inactive.

This behaviour is like a voice-operated transmits (VOX)/ voice-active detection (VAD) which is another type of voice activated communication which is usually used in the older radio systems. VAD/VOX like behaviour does not raise issues in the conventional radio systems but it can create significant issues in the trunked based radio systems. The reason for this is that in the trunking radio systems a channel resource is allocated by the channel controller for a conversation that takes place and PTT/COR signalling ensure that gates are open and that channels does not gets dropped in case that voice communication is muted for any reason. If signalling PTT/COR is interrupted/lost the channel controller would drop the channel. To continue voice communication a new channel would have to be allocated by the channel controller. Therefore we would have to go through the channel setup process again. This can create some significant delays and therefore provide ineffective interoperability solution.

This behaviour was not observed when radio equipment was connected to the VIC3-2E&M card. Our research showed that the radio equipment under the test was creating enough noise to keep RTP packet flow and therefore to activate and to keep PTT signalling steady. This was the case as long as COR was issued on other end (port). We conducted tests to determine what would be the lowest noise level needed to ensure that PTT signalling was activated and present as it was desired. Our testing found that with no voice present, noise levels at pins 3 and 6 (audio in) was -63dB. This was not enough to activate PTT. Our testing showed that the lowest noise level that would activate PTT signalling and keep it

steady would be -60dB. We examined each radio system and tested that Codan, Motorola, Harris and Cassidian emit -54dB, -60dB, -60dB and -58dB of noise, respectively. This noise level was good enough to activate and keep PTT signalling. This would be workaround for the current setup but we do not consider this as elegant solution. We have made Cisco aware of our finding and we work with them to resolve this issue.

4.1.2 System-of-Systems RoIP deployment architecture II (two agencies - multiple departments)

The second large scale of the System-of-Systems RoIP architecture which was deployed over the third party WANs is outlined in Figure 4.12.

Figure 4.12: System-of-Systems RoIP architecture 2 (simulating interoperability between agency X and agency Y)

Each agency had multicast enabled within their network with the core's gateway router declared as the multicast rendezvous point. Connection between agency X and agency Y was established using M1:U12:M2 connection trunks which were created between the inter-agency gateways as per Figure 4.13. M1:U12:M2 connection trunks created inter agency permanent connections to which departmental voice traffic was mapped using patches/conferences, hence provided interoperability among agency X and agency Y departments. Having patching capability in both agencies enabled both parties to have control over the conversation. This means that both parties must agree for the conversation to occur. Traffic was encapsulated using GRE tunnels with IPsec encryption. Each agency was shielded with firewalls. Each agency hub/core test setup was identical as per the System-of-Systems RoIP large scale architecture 1. Agency Y departments use Motorola (VHF, Digital Astro 25) trunking and conventional radio systems.

Figure 4.13: M1:U12:M2 Connection trunks and radio channel mapping using patch/conference

The test strategy was developed to:

- Determine delay between the two agency hub sites over the IP backbone including M1:U12:M2 conversion delay and IP backbone (network) delay.
- Determine inter-agency delay between agency X departments (LMR gateways) and agency Y hub site.
- Determine network delay between agency Y departments and agency Y hub site.
- Determine inter-agency delays between departments (LMR gateway-to-LMR gateway).
- Determine inherent latency of radio systems in agency Y.
- End-to-end delay (user-to-user).
- Determine voice quality.
- Determine overall System-of-Systems RoIP availability (packet loss).

4.1.2.2 Test Results

4.1.2.2.1 Inter-agency hub-to-hub delay

To determine inter agency (hub-to-hub) delay and to understand how much of that delay each major component was contributing, the following approach was used. Overall delay between the hub sites was determined using the test setup as shown in Figure 4.14.

Figure 4.14: Test setup to determine inter-agency (hub-to-hub) delay

In simulated agency X (hub site), a Codan (VHF, conventional) radio was connected to the LMR gateway to generate PTT/COR signalling and audio (voice). M1:U12:M2 trunks were established between the inter-agency gateways. On a VIC3-2E&M voice card, in the LMR gateway, ports were configured to match a M1 multicast pair (Ch1 and Ch2 as per Figure 4.13). In agency Y (hub site), VIC3-2E&M ports in the LMR gateway were configured to match M2 multicast IPs. A patch between M2 multicast channels was created. In this way, an audio from agency X hub was sent from the radio to the VIC3-2E&M voice card (Ch1- port 0), and then over M1:U12:M2 trunks to agency Y. Once audio was received in agency Y it was looped back to agency X on Ch2 using a patch created by UMS. The looped-back audio was received in agency X on the VIC3-2E&M voice card (Ch2 - port 1). Oscilloscope probes were connected to audio in and audio out of the VIC2-2EM voice card to determine delay in outbound and inbound audio. Hub-to-hub round trip audio delay on average was measured to be 523ms.

Secondly IP backbone (network) delays between the two inter-agency gateways, used to establish M1:U12:M2 trunks, was determined using “ping” command. This was accomplished by logging into one of the gateways and issuing “ping” towards the loopback address of another gateway and sending 1000 bytes packets:

“ping loopback_address -t -l 1000”. Network delay (roundtrip) on average was determined to be 57ms.

Thirdly M1:U12:M2 delay was determined. Since we have examined audio delay contribution of every component it was easy to calculate M1:U12:M2 delay:

M1:U12:M2 delay~ (hub-to-hub delay -2x VIC3-2E&M port delay - IP backbone (network delay) – UMS patch processing delay) / 2 = (523ms -2x36ms - 57ms - 40ms) / 2 = 354ms / 2 = 177ms. Obtained result confirmed our expectation based on our M1:U12:M2 delay findings in the phase four lab testing (175ms).

4.1.2.2.2 Inter agency delay between agency X LMR gateways and agency Y hub (core) site

To determine inter agency delay between agency X departments (LMR gateways) and agency Y hub site the following approach was used. Multicast radio channels were configured at the agency X - LMR gateways (VIC3-2E&M voice ports), in each department, as well as on the IPICS server. Radio channels were patched/conferenced to appropriate M1:U12:M2 connection trunks using dispatch console. In agency Y, patches/conferences were created, between matching radio channels and M1:U12:M2 connection trunks to loopback voice traffic back to agency X departments.

In agency X departments, voice traffic was generated using radios (i.e connected to VIC3-2E&M port 0/0). Voice traffic was sent from the LMR gateways to agency X hub and through the patch/conference the voice was sent over the M1:U12:M2 connection trunks to agency Y hub site. Once voice traffic reached agency Y it was looped back using patches/conferences. In agency X an oscilloscopes were used to measure delay between outbound and inbound audio (i.e VIC3-2E&M ports 0/0 and 0/1).

Inter-agency delays between agency X departments A, B,C, D (LMR gateways) and agency Y hub site was found to be 302ms, 335ms, 309ms and 306ms, respectively.

4.1.2.2.3 Network delay between agency Y departments and agency Y hub

To determine network delay between agency Y remote departments and agency Y hub department ping test from LMR gateways to the UMS server was used the same way as it was explained in previous sections. Network delay was found to be:

- Agency Y department B – agency Y hub ~ 96ms
- Agency Y department C – agency Y hub ~ 50ms

4.1.2.2.4 Inter-agency delay between departments (LMR gateway-to-LMR gateway)

To determine inter agency department-to-department delays similar testing approach to the one explained in 4.1.2.5 was used but this time instead of a patch/conference a physical loop was used to loop back signalling and audio traffic. In Agency Y departments, a physical loopback cables were inserted to LMR gateways to connect VIC3-2E&M voice card ports to loopback voice traffic to agency X departments. In agency Y hub, UMS patching was still but this time used to patch channels associated with LMR gateway to appropriate M2 multicast channels of the M1:U12:M2 connection trunks. Therefore voice traffic was sent from agency X departments, through agency X and Y hubs to the agency Y departments. Once voice traffic reached agency Y departmental LMR gateways, the traffic was looped back using the physical loop. In agency X an oscilloscopes were used to measure delay between out-bound and inbound audio (i.e VIC3-2E&M ports 0/0 and 0/1).

Inter-agency delay between agencies X departments A, B, C, D and agency Y department B was determined to be: 410ms, 444ms, 418ms and 415ms, respectively. Inter-agency delay between agencies X departments A, B, C, D and agency Y department C was determined to be: 434ms, 467ms, 441ms and 438ms ms, respectively (Appendix B1).

4.1.2.2.5 Inherent latency of radio systems in agency Y

Inherent latencies of agency Y radio systems are outlined below:

1. Department B - Motorola (VHF, Astro 25 digital) conventional radio

As in the case of the agency X department B - Motorola Astro 25, conventional radio system we had obtained similar results. The testing showed that processing delay of the radio system is not the same when signal is sent from the radio to the LMR gateway and when the signal is sent to the radio from the LMR gateway. Inherent delay of the radio system was found to be: 480ms-488ms for PTT/COR signalling processing and ~233-427ms for audio processing (Appendix B2).

2. Department C - Motorola (VHF, digital proprietary) trunking

Since this is trunking radio system, similarly as in the agency X, there is a channel controller involved to allocate a channel and therefore the channel grant processing time was needed to be determined as well. The testing showed that processing delay of the radio system is not the same when signal was sent from the radio to the LMR gateway and when the signal was sent to the radio from the LMR gateway. This is due to the way that radio system architecture was designed.

Inherent delay of the radio system was found to be: 560ms-935ms for PTT/COR signalling processing and ~444-425ms for audio processing (Appendix B2).

4.1.2.2.6 End-to-End latency (user-to-user)

To simulate a real life scenario and to determine inter-agency end-to-end (i.e portable-to-portable) delay the following approach was used. Firstly we determined portable-to-portable delay between portable radios in each department of the agency X through the UMS patch that was created in the agency Y hub. Secondly we determined portable-to-portable delays between portable radios in agency X through the

physical loopback in agency Y departments (LMR gateways). This test was done between the furthest departments of agency X and agency Y. Thirdly we determined portable-to-portable delay between furthest agency X and agency Y departments, where it was possible. The reason was that this test was only possible between departments with an overlapping RF coverage so that portable radios could be collocated and oscilloscope could be used to measure the delay difference. In this testing we were interested in the worst case scenario PTT to Rx audio (longest delay). The testing results are outlined in below:

- 1) Agency X portable-to-portable delay – traffic looped back by the agency Y UMS. End-to-end delay for agency X Codan (VHF, analog, conventional), Codan (VHF/UHF/800MHz digital, conventional), Harris (UHF, digital proprietary), Cassidian (P25 digital, IP based, conventional) and Motorola (VHF, Astro 25 digital, conventional) radios was determined to be 924ms, 730ms, 2.902s, 2.390s and 1.624s, respectively (Appendix B3).
- 2) Agency X (departments B and C) portable-to-portable delays through the agency Y LMR gateways (departments B and C) - traffic looped back by the physical loopback cable. Agency X site B - Harris (digital proprietary, UHF, trunking) and site C - Cassidian (P25 digital, UHF, trunking) end-to-end delays were determine to be 3.106ms and 2.550ms, respectively (Appendix B3).
- 3) End-to- end delay between agency X (site B, Harris, digital proprietary, UHF, trunking) radio and agency Y portable radios was determined to be 2.575 for PTT to Rx audio and 1.458s for audio processing thereafter (Appendix B3).

4.1.2.2.7 Inter-agency voice quality

Voice quality was subjectively measured as per the TSB 88 DAQ [42]. In our testing no truncated or missing audio was recorded when delay was ≤ 2.5 s. All transmissions were heard loud and clear 5/5 (signal strength 5/clarity 5) except in cases where delays were > 2.5 s. If delay was > 2.5 s sometimes the first syllable would be lost or if the delay was ~ 3 s the first word sometimes would be lost.

Testing was conducted among multiple parties located in multiple locations. The testing was conducted using both portable/mobile radios and dispatch consoles.

4.1.2.2.8 System-of-Systems RoIP availability (packet loss)

The test setup in Figure 4.15 was used to generate record and analyze 24hr continuous audio to determine Systems-of-Systems RoIP inter-agency availability. In agency X, (department X), Mp3 player was used along with a radio test set IFR 2975 [41] to inject audio into a radio system on Ch 1 (VIC3-2E&M port 0/0). Audio was sent to the agency X (core) on Ch1, mixed in the UMS with M1:Ch1 and sent to Agency Y core over the M1:U12:M2 trunks. In agency Y, UMS was used to mix/patch M2:Ch1 to agency Y (department X) Ch1. In agency Y (department X) audio out of the Ch1 (VIC3-2E&M port 0/0) was looped back using physical loop to the Ch2 (VIC3-2E&M port 0/1). Ch2 was sent to the agency Y (core) where it was patched to the M2:Ch2 and over the M1:U12:M2 trunk to agency X (core). In agency X (core), M2:Ch2 was patched to the department X Ch2. In agency X, (department X) both Ch1 (outbound audio) and Ch2 (inbound audio) were recorded using laptop with Audacity software. Ch1 was recorded to make sure that audio that was processed through the radio system did not have any packet loss and Ch2 was recorded to determine if any packet loss took place over that System-of-System IP backbone.

Figure 4.15: Inter-agency availability test setup

The 24hr continuous audio test was repeated for ten consecutive days. Testing was conducted for between 4 furthest agency X and agency Y departments:

- 1) Agency X, department B – agency Y, department B (distance 11,677Km)
- 2) Agency X, department C – agency Y, department C (distance 4,761Km)

Results were obtained and analysed and the packet loss over 24 hr period was found to be <0.1%.

Please note that as per Service Level Agreement (SLA) with third party WAN provider (BELL/AT&T) system availability for MPLS-HP circuits was 99.90%. This testing showed that system availability for the System-of-Systems RoIP setup was found to be > 99.90% as is shown in Figure 4.16.

Figure 4.16: System availability – 24hrs continuous audio test for 10 consecutive days

4.1.2.3 System-of-Systems Architecture II - Analysis

Based on our testing results inter-agency interoperability call flow and delay metric can be outlined as per Figure 4.17.

Figure 4.17: Inter-agency interoperability call flow and delay metric

Our testing determined that IP network delay and round trip delay between agency X and agency Y hub LMR gateways on average were found to be 57ms (one way ~ 29ms) and 603ms (one way ~ 302ms), respectively. Therefore we have found that LMR gateway-to-LMR gateway, between the two closest sites in this architecture had a delay that exceeded the desired maximum delay of ≤ 200 ms. This was a significant find in the sense that we determined that even with a zero IP network delay the overall delay between the two sites would exceed 200ms. Since this is the centralized solution and all traffic had to go through the hub sites it could be concluded that delay between agency X and agency Y departments (LMR gateways) would be greater than 200ms. This was confirmed in our further testing to determine delay between agency X LMR gateways and agency Y hub (~ 302ms, ~ 335ms, ~ 309ms and ~ 306ms)

and between agency X LMR gateways and agency Y LMR gateways $\sim$ (401ms, 434ms, 444ms, 467ms, 418ms, 441ms, 415ms and 438ms).

In our further testing we have determined that M1:U12:M2 conversion latency was $\sim$ 177ms. This confirmed our result that we originally found in our lab testing $\sim$ 175ms. Therefore we determined that M1:U12:M2 conversion was a major source of delay from the RoIP IP backbone between inter-agency hub sites. If this delay was eliminated, an overall delay between agency X LMR gateways and agency Y hub would be reduce to $\sim$ (125ms, 158ms, 132ms and 129ms) and between inter-agency LMR gateways to $\sim$ (233ms, 257ms, 267ms, 290ms, 241ms, 264ms, 238ms and 261ms), respectively. Therefore by eliminating the delay caused by M1:U12:M2 trunks we would significantly reduce RoIP IP backbone delay towards the desired 200ms.

One of the reasons why we had decided to deploy and test inter-agency system-of-system architecture with M1:U12:M2 trunks was that firewalls that we had tested with (McAfee) between the two agencies did not support multicast at the time. Another reason was that multicast between two agencies could create some significant problems from the multicast IP management perspective (each agency has its own internal multicast configuration). In Chapter 5 of this thesis, in recommendation for future work, we propose removing M1:U12:M2 trunks and enabling multicast between agencies using multicast service reflection [51]. We have tested multicast reflection successfully in our lab environment and reduced an overall inter-agency System-of-Systems architecture delay for $\sim$ 350ms ($\sim$ 175ms one way). With multicast service reflection each agency reserve management of their private multicast IP and only public multicast IP are shared between agencies as per Figure 4.18.

Figure 4.18: Proposed multicast service reflection test setup to eliminate M1:U12:M2 delay

Besides the point that removing MUM trunks would significantly reduce IP backbone delay is that establishing MUM trunks is challenging when Network Address Translation (NAT) is used to enhance security in the firewalls. The way MUM trunks are designed is that unicast portion of the MUM is used to establish SIP calls by pointing to the loopback address of the other end (router) [21]. When NAT is used we are not pointing to the exact loopback address but to the NATed one. Firewalls would need to be smart enough to inspect IP address and ports from which the packets were received and detects the type of NAT device through the packets that were sent. Unfortunately the firewalls that we tested with were not capable of doing this operation. This issue could be potentially resolved by STUN, TURN and ICE methods proposed by IETF [56], but there are some potential security issues involved and our recommendation is to remove MUM trunks and enable multicast through the multicast service reflection.

Our testing determined that IP backbone latency in agency Y over the third party WAN (AT&T) was much greater, distance wise, comparing to the IP backbone latency obtained in agency X. For example, distance between agency Y hub site and spoke sites was 5080km and 1870km and round trip latency

was found to be 96ms and 50ms, respectively; In agency X distances between hub site and spoke sites were 4337km, 631km and 200km and round trip latencies were determined to be 60ms, 11ms and 7ms, respectively. Therefore we have determined that agency Y had larger network delay comparing to the agency X and hence it introduced an additional delay between inter-agency LMR gateways exceeding our goal to limit delay to 200ms. This finding was significant in a sense that no assumption should be made about IP backbone that connect radio sites. Each IP backbone connection in the overall system-of-system interoperability solution needs to be carefully examined to ensure that overall site to site (LMR gateway-to-LMR gateway) does not exceed 200ms.

In this System-of-Systems architecture, IP backbone latency between the two furthest departments of agency X and agency Y was found to be ~ 107 ms (one way). If we recall, in the first System-of-Systems architecture we had concluded that delay of the IP backbone between LMR gateways should be limited to 88ms ($200\text{ms} - 2 \times 36\text{ms} - 40\text{ms}$) to limit delay to 200ms. In the second System-of-Systems architecture we had an additional ~ 40 ms of delay due to the second channel-patch processing. Therefore IP backbone delay between two furthest inter-agency sites, with centralised IPICS servers for channel mixing, should be limited to ~ 48 ms (one way) to limit an overall LMR gateway-to-LMR gateway delay to 200ms. Having said that, this is valid statement if M1:U12:M2 is replaced with the multicast service reflection.

Our testing confirmed our conclusion found in the first System-of-Systems architecture that that end-to-end delay can be approximated mathematically by adding predetermined delays of individual systems components:

1. Inter-agency audio delays (round trip) between agency X departments (LMR gateways) and agency Y hub site tested (563ms, 629ms, 578, 572) vs. calculated (559, 619, 570, 566), respectively.
2. Inter-agency audio delays (round trip) between departments (LMR gateway-to-LMR gateway) tested (739ms, 787ms, 807ms, 853ms, 756ms, 802ms, 750ms, 796ms) vs. calculated (729ms, 775ms, 789ms, 835ms, 740ms, 736ms, 782ms), respectively.
3. Portable-to-portable (PTT to Rx audio) delay (round trip) between agency X radios, with traffic looped back by in agency Y patch/conference (UMS) tested (924ms, 730ms, 2902ms, 2390ms, 1590ms) vs. calculated (918ms, 724ms, 2914ms, 2367ms, 1577ms), respectively.
4. Portable-to-portable delay (PTT to RX audio) between agencies X radios with traffic looped back by physical loopback cable in agency Y (LMR gateways) tested (3106ms, 2550ms) vs. calculated (3122ms and 2529ms), respectively.
5. End-to-End delay (user-to-user) between two furthest departments in Agency Y and Agency Y (both PTT to Rx audio and audio processing) tested (2575ms, 1458ms) vs. calculated (2550ms, 1482ms), respectively.

Our testing confirmed that interoperability between trunking systems is prone to significant delay due to their inherent latency. In our end-to-end testing we have found that delay between the two furthest

trunking radio sites (user-to-user) was measured to be ~ 2.575 s for the communication path setup (PTT to RX audio) and 1.458s for audio processing thereafter. As we pointed out before, this would be on the edge of the reliable user-to-user communication since we have determined that overall System-of-Systems delay should be ≤ 2.5 s. As we explained in the previous System-of-Systems architecture, based on our testing if an overall System-of-Systems delay exceeds 2.5s what could occur is a user starts talking before the other finishes, and due to the prolonged silence period other parts of the conversation could be lost. If delay is ≥ 2.5 s, communication is still possible but users will need to be trained to ensure that one party has finished communication before other should start communicating. We recommend that this should be avoided in a mission critical application. Also observed was that if end-to-end delay was ≥ 2.5 s that sometimes the first syllable would be lost or if the delay was ~ 3 s the first word sometimes would be lost.

In our continuous audio testing we have achieved availability greater than 99.92% which is greater than 99.90% guaranteed by the third party provider. Having said that, availability was not as great as it was found in the first System-of-Systems architecture where we have achieved 99.99% - 99.999% availability.

Our inter-agency system-of-system architecture has achieved good voice quality. All transmissions were heard loud and clear 5/5 (signal strength 5/clarity 5) except in cases where delays were > 2.5 s. Transmission between the two most distant sites were heard 4/4. It was observed that at the beginning of the transmission the first syllable sometimes was lost. Therefore we recommend that the worst case end-to-end delay should be < 2.5 s.

Our testing of various radios and radio interconnects revealed that some of them have implemented an additional delay on their transmission side (200-300ms). The reason for this is to prevent audio clipping at the beginning of the transmission. This is necessary to take into account when planning for an overall System-of-Systems delay since audio will have to be buffered on the network side to accommodate radio delay in order to avoid audio to be truncated.

4.1.3 Validation and comparison

To the best of our knowledge in communication with other researchers, such as Canadian Research Center (CRC) and a vendor community (Cisco, Twisted Pair and Motorola) the proposed solution is the first one to deploy and test complex RoIP architectures over the large distances in the real world conditions. The proposed solution is compared to other RoIP solutions such as proposed RoIP solution in [1] and Motobridge solution proposed by Motorola [55]. In [1] researchers were in the early stage of their lab testing and we have not found evidence that their solution was successfully deployed over the large distances. In our communication with Cisco, Motorola and Twisted Pair researching available published documentation we tried to obtain information such as: under which conditions their RoIP solutions were tested (i.e networks, distances etc.), what radio systems and radio systems architectures were tested and to obtain benchmarks such as bandwidth, delay, jitter and packet loss accordingly, to understand overall interoperability performances of their RoIP solution. By the time this thesis was completed we have not been provided with valid information. Therefore we are not in position to validate our results in direct comparison with others.

Our testing showed that understanding an overall delay between radio systems, and how much of that delay was contributed by each system's component, was critical for interoperability to be deployed successfully over the large distances and multiple WAN networks. Our testing showed that interoperability between trunking radio systems over large distances was challenging due their large inherent delay. We have showed that overall end-to-end RoIP system's delay can be mathematically approximated by adding individual delays of major RoIP system components. We have built an interoperability matrix to show in which case interoperability would work and in which would not. This is a significant in a sense that it could help others to decide would interoperability work between their radios systems in the early stage of their interoperability projects. In [1] we did not observe that their research will focus on understanding these critical aspects of interoperability among mobile radios.

We have successfully examined various radio systems interfaces to extract PTT/COR signalling and audio to interface Cisco LMR gateways. To do this we have successfully determine E&M types and build interfaces or identified alternatives (TAG). In [1] no radio systems or radio systems architectures were identified in relation to how they would be interfaced to their RoIP core. In our communication with Motorola's and their available documentation, Motobridge interoperability solution is Motorola's RoIP solution which has number of Motorola's proprietary components and it was primarily built to interface existing Motorola radio systems. We could not find evidence that their solution has successfully interconnected various vendors' systems and no similar published results were available as in this thesis. It is important to mention that Motorola has recently purchased Twisted Pair. This leads us to assume that Motorola is looking for more competitive standard based RoIP solution as oppose to their proprietary Motobridge RoIP solution.

We have shown that M1:U12:M2 connection trunks were a major contributor of IP backbone delay between radio sites (LMR gateways) ~350ms for round trip. To eliminate this delay we have proposed and tested in our lab multicast service reflection. With multicast service reflection we have eliminated this significant delay and also showed that each agency can maintain their multicast addressing ownership with no sharing. In our communication with Cisco no evidence was provided that Cisco actually has examined M1:U12:M2 conversion as well as that they have tried to examine multicast service reflection as an alternative.

Chapter 5: Conclusions and Future Work

This chapter concludes the thesis with a summary of its findings and suggests a direction for further research.

7.1 Summary

This thesis began with research of various RoIP technologies to enable existing mobile radios such as old analog legacy radios, digital radios, P25 digital in both conventional and trunking radio architectures to communicate over the MPLS IP dedicated or shared links. We were looking for the solution to enable non-interoperable mobile radios to communicate regardless of the frequency band such as VHF, UHF, HF and 700MHz-800MHz and vendor type. Our goal was to characterize voice and signalling delay, availability and voice quality by deploying System-of-Systems RoIP solution over large distances and multiple third party WANs.

We have examined various radio systems and radio systems architectures and obtained baseline requirements for reliable radio over MPLS IP communication. We have concluded that network baseline characteristics need to be: bandwidth (128Kbps/radio channel), latency (<200ms), jitter (system dependent - <12% and <25%) and packet loss (<1%) for radio features.

We have concluded that RoIP voice and signalling delay can be characterised by examining individual systems components in the communication path. In our testing we have examined inherent latency of each system's component such as inherent latency of radio systems, LMR gateways (VIC3-2&M voice card), patching/conferencing, IP networks, M1:U12:M2.

We have concluded that end-to-end delay between certain trunking radio architectures over IP backbone can exceed 2.5s and therefore create an unacceptable user experience. This means that even though we have enabled non interoperable radio systems to communicate from the technology perspective that delay would be too significant for communication to be acceptable, especially from the mission critical communication point of view.

We have concluded that M1:U12:M2 connection trunks were major source of delay on the RoIP IP backbone. We have determined that M1:U12:M2 connection trunks introduce delay of ~350ms (round trip). Since our goal is to ensure that IP backbone does not introduce more than 200ms delay we concluded that M1:U12:M2 connection trunks should be avoided by enabling multicast end-to-end by using multicast service reflection. We have proved in our lab testing that using multicast service reflection we reduced delay for ~350ms (round trip) and that each agency would preserve their sovereignty over their private multicast IPs.

In our IP backbone delay characterization we concluded that by determining delays of LMR gateways (VIC3-2&M voice card) and patching/conferencing we can determine what IP network latency needs to be in order to obtain an overall IP backbone delay $< 200\text{ms}$. This approach can be applied to any other LMR gateway and patching mechanism.

We have concluded that that an overall end-to-end system delay can be approximated mathematically by adding predetermined delays of the system's components. Based on our findings, the worst case scenario (longest end-to-end system delay), we built an interoperability latency matrix and showed that this ap-

proach can be used to determine if interoperability makes sense between certain radio systems over large distances and multiple networks. This is significant find in a sense that interoperability can be planned in advance.

We have deployed and tested System-of Systems RoIP solution in the two different architectures over multiple LAN and WAN networks and we have achieved system availability $\geq 99.99\%$ (packet loss $\leq 0.01\%$) and $\geq 99.92\%$ (packet loss $\leq 0.08\%$), respectively. From the mission critical perspective we have concluded that system availability should be 99.99% - 99.999% for reliable and user acceptable communication.

Audio quality was subjectively measured as per TSB 88 DAQ to reflect the user experience. We have achieved our target DAQ 3.4 which stands for understandable with repetition only rarely required and some noise/distortion acceptable [42]. We have concluded that all transmission were heard loud and clear 5/5 (signal strength 5/clarity 5) when an overall system delay was $< 2.5s$

We have discovered that Cisco is using RTP packets (voice packets) to preserve radio signalling (COR/PTT). COR/PTT signalling must be preserved from radio side through the network independent of voice presence. In case that COR/PTT signaling is dependent on the RTP packets presence in the trunking radio systems the channel controller would drop the channel if a user pause for some reason. This would cause significant delays since channel acquisition process on the trunking system would have to be reinitiated. We have concluded that is not a good approach and we recommend that this needs to be avoided.

7.2 Future Directions

Future testing will consist of deploying multicast service reflection in the real world condition between two agencies to ensure that our lab results can be validated from the delay, availability, voice quality and multicast IP management perspective.

Our testing showed that even though QoS prioritize voice over the network over other types of traffic such as data, it cannot make a difference among other types of voice traffic. This is significant from the mission critical voice perspective and needs to be investigated further since we can not allow situation that radio voice does get dropped due to other voice traffic present on the network (i.e VoIP telephony).

Extensions to this work are numerous such as interoperability of data, sensor data and video in a secure manner. In many cases we are facing a lack of mobile or portable radio coverage. We are looking into possibilities to take an advantage of 3G/4G/LTE mobile coverage as part of our System-of-Systems interoperability solution.

Our goal is to motivate other researchers and vendors to conduct similar tests and compare their findings to ours, working with vendors and sharing lessons learned to direct their future research in making technology more reliable for mission critical use.

References

- [1] J. H. Mock, W. T. Miller, “*A voice over IP system to solve the problem of mobile radio interoperability*”, Proceedings IEEE Vehicular Technology Conference (VETECF2002), Fall 2002, vol.3, pp.1338-1341.
- [2] F. Yu, J. Zhang, H. Tang, H. C. B. Chan, and V. C. M. Leung, “*Enhancing Interoperability in Heterogeneous Mobile Wireless Networks for Disaster Response*”, “*Wireless Communication*”, Proceedings IEEE Military Communication Conference (MILCOM2007), Orlando, Florida, USA, October 2007, vol.8, Issue 5, pp.2424-2433.
- [3] J. Brito, “*Sending Out an S.O.S: Public Safety Communications Interoperability as a Collective Action Problem*”, <http://www.fclj.org/wp-content/uploads/2013/01/7-Brito.pdf> (accessed September 20, 2013)
- [4] R. Miller, “*Hurricane Katrina: Communication & Infrastructure impact*”
<http://www.carlisle.army.mil/DIME/documents/Hurricane%20Katrina%20Communications%20&%20Infrastructure%20Impacts.pdf> (accessed October 5, 2013).
- [5] Public Safety and Homeland Security Bureau, “*Tech Topic: 1 Interoperability*”.
<http://transition.fcc.gov/pshs/techtopics/tech-interop.html> (accessed October 5, 2013).
- [6] CISCO “*IPICS Server Software*”,
http://www.cisco.com/en/US/prod/collateral/ps6712/ps6718/product_data_sheet0900aecd8034fd94.html (accessed October 6, 2014).
- [7] Twisted Pair, “*Media and Management Servers Form the Core of a Wave system*”
<http://www.twistpair.com/index/enterprise-server> (accessed October 6, 2014).

- [8] R.E. Lavigne, “*Trunking vs. Conventional Radio Systems*”, IEEE Aerospace and Electronic Systems Magazine, Vol. 16, Issue 5, May 2001, pp.3-6.
- [9] P. Kiley, T. Benedett, “*Public Safety Interoperability With an SCA Military Radio Using the P25 Waveform*”, Proceedings IEEE Military Communication Conference (MILCOM2007), Orlando, Florida, USA, October 2007, pp.1-8.
- [10] C. Hattingh, D. Sladden and Z. Swapan, “*SIP Trunking*”, Cisco Press, February 18, 2010, ISBN-10: 1-58705-944-4.
- [11] R. M. Dansereau, “*Digital Signal Processing (SYSC5602, Course notes, Carleton University, Ottawa, Ontario, 2011, p.p 2-24.*
- [12] T. C. Farrell, Scott AFB, W.H. Shaw, “*A computer simulation analysis of conventional and trunked land mobile radio for a system with a small number of user groups*”, Proceedings IEEE Military Communications Conference, (MILCOM 1989), Boston, MA, Vol.3, pp. 923-927
- [13] Telecommunications Industry Association, “*Project 25 FDMA Common Air Interface New Technology Standards Project Digital Radio Technical Standards*”,
http://www.qsl.net/kb9mwr/projects/voip/apco25/tia-102-baaa-a-project_25-fdma-common_air_interface.pdf (accessed November 7, 2013).
- [14] Cisco Systems, “*Cisco Land Mobile Radio over IP Solution Reference Network Design*”,
http://www.cisco.com/en/US/docs/wireless/lmr/design/guide/lmrsrnd_1.html (accessed May 7, 2014)
- [15] Cisco Systems, “*Analog E&M Voice Signaling Overview*”,
http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080093f60.shtml (accessed November 9, 2013).
- [16] Cisco Systems, “*Understanding and Troubleshooting Analog E&M Interface Types and Wiring*”,

http://www.cisco.com/en/US/products/hw/gatecont/ps2250/products_tech_note09186a008009452e.shtml
1 (accessed November 9, 2013).

[17] Cisco Systems, “*Gateway to Gateway Connections: Transport*”,
<http://www.cisco.com/en/US/docs/wireless/lmr/design/guide/lmrgwcon.pdf> (accessed November 13,
2013).

[18] T. Zahardis and S. Spanops, “*The Clearest Voice [SIP vs. H.323]*”, Proceedings IET Communications Engineer Magazine, Vol.2, Issue 2, 2004, pp. 14-17.

[19] H. Schulzrinne S. Casner,, R. Frederick and V. Jacobson, “*RFC: 3550: RTP: A Transport Protocol for Real-Time Applications*”, Internet Engineering Task Force (IETF), July 2003,
<http://www.ietf.org/rfc/rfc3550.txt> (accessed December 3, 2013).

[20] M. Handley, H. Holbrook, I. Kouvelas and B. Fenner “*RFC 4601 : Protocol Independent Multicast - Sparse Mode (PIM-SM)*”, Internet Engineering Task Force (IETF), August 2006.
<http://tools.ietf.org/html/rfc4601> (accessed December 7, 2013).

[21] Cisco Systems, “*Solution Reference Network Design for Cisco IPICS, Release (4.0.(2))*”,
http://www.cisco.com/en/US/docs/interoperability_systems/c_ipics/402/design/guide/sr402mod.html#wp1219280 (accessed December 15, 2013).

[22] H. K. Kalitay and M. K. Nambiarz, “*Designing WANem: A Wide Area Network Emulation Tool*”, Proceedings IEEE Communication Systems and Networks (COMSNET2011), Bangalore, India, January 2007, pp.1-4.

[23] Cisco Systems “*Configuring IP Tunnels*”,
http://www.cisco.com/en/US/docs/switches/datacenter/sw/4_2/nxos/interfaces/configuration/guide/if_tunnel.html (accessed December 18, 2013).

- [24] Cisco Systems “*Multicast over a GRE tunnel*”,
http://www.cisco.com/en/US/tech/tk828/technologies_configuration_example09186a00801a5aa2.shtml
 (accessed December 18, 2013).
- [25] C. Schroder, “*The book of Audacity record, edit, mix, and master with the free audio editor*”, No Starch Press, San Francisco, CA, 2011 ISBN-10: 1-59327-270-7.
- [26] <http://audacity.sourceforge.net/> (accessed December 20, 2013).
- [27] S. Frankel and S. Krishnan Ericsson “*RFC 6071 - IP Security (IPsec) and Internet Key Exchange (IKE) Document Roadmap*”, Internet Engineering Task Force (IETF), February 2011.
- [28] Cisco Systems, “*An Introduction to IP Security (IPSec) Encryption*”,
http://www.cisco.com/en/US/tech/tk583/tk372/technologies_tech_note09186a0080094203.shtml (ac-
 cessed December 21, 2013)
- [29] Twisted Pair, “*Understanding Wave, A Technical Brief for Potential Users & Other Interested Parties*”,
www.twistpair.com/.../WAVE%20Technical%20Brief%206-25-2012.pdf (accessed December 21,
 2013).
- [30] Twisted Pair, “*Media and Management Servers Form the Core of a WAVE System*”
<http://www.twistpair.com/index/enterprise-server> (accessed December 21, 2013).
- [31] Twisted Pair, “*Wave Technical Brief*”
<http://www.wbc-technologies.com/PDF/wave-app-ab-8.5x11.pdf> (accessed December 21, 2013).
- [32] Juniper Networks, “*Deploying Basic QoS*”,
http://hydra.ck.polsl.pl/~helot/ipad/DayOne-Book/DO_Deploying_Basic_QoS.pdf (accessed December
 23, 2013).

- [33] Wikipedia, “*Push-to-Talk*”,
<http://en.wikipedia.org/wiki/Push-to-talk> (accessed January 2, 2014).
- [34] IETF, “*Internet Protocol*”,
<http://tools.ietf.org/html/rfc791> (accessed January 2, 2014).
- [35] K. Nicols, S. Blake, F., D. and Black, “RFC 2474 – Definition of the Differentiated Service Field (DS Field) in the IPv4 and IPv6 Headers”, December 1998.
<http://www.ietf.org/rfc/rfc2474.txt> (accessed January 5, 2014).
- [36] Wikipedia, “*MDC-1200*”
http://en.wikipedia.org/wiki/MDC-1200#Emergency_button (accessed January 2, 2014).
- [37] DTMF Format
<http://www.genave.com/dtmf.htm> (accessed January 2, 2014).
- [38] Cisco Systems “*OSPF Design Guide*”,
http://www.cisco.com/en/US/tech/tk365/technologies_white_paper09186a0080094e9e.shtml (accessed January 5, 2014).
- [39] J.Moy, “*RFC 2328, OSPF Version 2*”, April 1998
<http://www.ietf.org/rfc/rfc2328.txt> (accessed January 5, 2014).
- [40] Wikipedia, “*Ping (networking utility)*”,
[http://en.wikipedia.org/wiki/Ping_\(networking_utility\)](http://en.wikipedia.org/wiki/Ping_(networking_utility)) (accessed January 5, 2014).
- [41] Aeroflex, “*Wireless Radio Test Set IFR2975*”,
<http://www.aeroflex.com/ats/products/prodfiles/datasheets/82/2975.pdf> (accessed January 7, 2014).
- [42] Tait Communications, “*Under the Hood - Specifying voice quality for your digital network*”,

<http://blog.taitradio.com/2013/08/01/under-the-hood-specifying-voice-quality-for-your-digital-network/>
(accessed January 7, 2014).

[43] CTE Solutios and NetRunner, “*Certified Information Systems Security Professional Preparation Bootcamp (CISSP)*”, Course Notes, Canada 2006, Revision 2011.

[44] Cisco Systems, “*Configuring Access Control List*”,
http://www.cisco.com/en/US/products/sw/secursw/ps1018/products_tech_note09186a00800a5b9a.shtml
(accessed January 20, 2014).

[45] Homeland Security, “*Charter for the Project 25 Compliance Assessment Program*”,
http://www.pscr.gov/outreach/safecom/p25_cap/charter/Project25ComplianceAssessmentProgramCharter.pdf (accessed May 7, 2014).

[46] G. Baldini, “*Interoperable communications for Safety and Security*”, The European Commission Joint Research Centre Scientific and technical Reports, Report of the workshop, Ispra Italy, June, 2010

[47] Industry Canada, “Radio Spectrum Allocation Canada”,
[http://www.ic.gc.ca/eic/site/smt-gst.nsf/vwapj/spectallocation-08.pdf/\\$FILE/spectallocation-08.pdf](http://www.ic.gc.ca/eic/site/smt-gst.nsf/vwapj/spectallocation-08.pdf/$FILE/spectallocation-08.pdf) (accessed September 20, 2013).

[48] U.S Department of Commerce, National Telecommunication and Information Administration, Office of Spectrum Management, “Radio Spectrum Allocation United States”,
http://www.ntia.doc.gov/files/ntia/publications/spectrum_wall_chart_aug2011.pdf (accessed September 20, 2013)

[49] National telecommunications and Information administration US Department of Commerce and Federal Emergency Management: “*Improving Interoperable Communications Nationwide: Overview of Initial State and Territory Investments*” .

[http://www.ntia.doc.gov/legacy/psic/PSIC%20Investment%20Data%20Analysis%20\(report%20only\).pdf](http://www.ntia.doc.gov/legacy/psic/PSIC%20Investment%20Data%20Analysis%20(report%20only).pdf)
f (accessed September 20, 2013).

[50] Cisco Systems, “*Cisco Land Mobile Radio Gateway*”,

http://www.cisco.com/en/US/prod/collateral/routers/ps259/product_data_sheet0900aecd8034ef85_ps5854_Products_Data_Sheet.html (accessed May 3, 2014)

[51] Cisco Systems, “*Multicast Service Reflection*”,

http://www.cisco.com/c/en/us/td/docs/routers/7600/ios/15S/configuration/guide/7600_15_0s_book/multicast_sr.html (accessed March 6, 2014).

[52] Lumension Scan, “*Credentialed canning vulnerability assessment tool*”,

<http://www.lumension.com/vulnerability-management/vulnerability-assessment-software.aspx> (accessed, March 7, 2014).

[53] Nessus, “*Port scanning tool*”,

<http://www.tenable.com/products/nessus> (accessed March 7, 2014).

[54] McAfee Vulnerability Manager (MVM), “*Credentialed and port scanning tool*”

<http://www.mcafee.com/us/products/vulnerability-manager.aspx> (accessed March 7, 2014).

[55] Motorola, “*Motobridge Interoperable IP Solution*”,

http://www.motorolasolutions.com/US-EN/Business+Product+and+Services/Two-Way+Radios+-+Public+Safety/Infrastructure/Gateways/MOTOBIDGE_Interoperable_IP_Solution_US-EN (accessed March 7, 2014).

[56] Ingate Systems, “*Solving the Firewall/NAT Traversal Issue of SIP: Who Should Control Your Security Infrastructure?*”,

http://www.ingate.com/files/Solving_Firewall-NAT_Traversal.pdf (accessed, March 10, 2014).

[57] A. Petlund, K. Evensen, C. Griwodz and Pai Halvorsen, “*TCP Enhancements for Interactive Thin-Stream Applications*”,

<http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.208.8267&rep=rep1&type=pdf> (accessed May 7, 2014).

[58] Cisco Systems “*Rendezvous Point Engineering*”,
http://www.cisco.com/c/en/us/products/collateral/ios-nx-os-software/ip-multicast/whitepaper_c11-508498.html (accessed May 8, 2014).

Appendix A -System-of-Systems RoIP deployment architecture I - complete test results

The following represent completed test results. Please note that in the tables are representative data samples. During our testing each characteristic outlined in the tables was retested many times on multiple occasions. For example, any time we would do optimization on the network or on the radio side results would be retested.

Appendix A1 – Inherent latency of the radio systems

- a) Inherent latency of Codan (VHF/UHF/800MHz, analog/P25 digital, conventional) radios - Radio to LMR gateway

Test #	Radio to LMR Gateway (VIC3-2E&M) Analog (VHF)		Radio to LMR Gateway (VIC3-2E&M) Digital P25 (VHF/UHF/800MHz)		Missed Audio
	PTT to RX audio CTCSS tone processing time (ms)	Audio Pro- cessing (ms)	PTT to RX audio NAC# code processing (ms)	Audio pro- cessing (ms)	
1	190	160	62	165	No Missed Audio
2	210	165	62	165	No Missed Audio
3	192	160	67	170	No Missed Audio
4	260	170	67	170	No Missed Audio
5	207	165	62	165	No Missed Audio
6	208	170	57	160	No Missed Audio
7	226	170	57	160	No Missed Audio
8	214	170	62	165	No Missed

					Audio
9	188	160	67	170	No Missed Audio
10	222	170	62	165	No Missed Audio
Average delay ~	212	166	63	166	No Missed Audio
Total average processing delay ~	378		229		No Missed Audio

b) Inherent latency of Codan (VHF/UHF/800MHz, analog/P25 digital, conventional) radios - LMR gateway to Radio

	LMR Gateway (VIC3-2E&M) to Radio		LMR Gateway (VIC3-2E&M) to Radio		
Test #	Analog (VHF)		Digital P25 (VHF/UHF/800MHz)		Missed Audio
	PTT to RX audio CTCSS tone Processing time (ms)	Audio Processing (ms)	PTT to RX audio NAC# code Processing (ms)	Audio Processing (ms)	
1	132	162	62	196	No Missed Audio
2	126	152	62	194	No Missed Audio
3	110	170	67	183	No Missed Audio
4	94	164	67	181	No Missed Audio
5	112	152	62	212	No Missed Audio
6	118	150	57	197	No Missed Audio
7	100	154	57	195	No Missed Audio
8	104	146	62	190	No Missed Audio

9	100	150	67	191	No Missed Audio
10	78	152	62	192	No Missed Audio
Average delay ~	107	155	63	193	No Missed Audio
Total average processing delay ~	262		257		No Missed Audio

c) Inherent latency of Harris (digital proprietary, trunking) radio

	Radio to LMR Gateway (VIC3-2E&M)		LMR gateway (VIC3-2E&M) to Radio		Missed Audio
Test #	PTT To RX Audio (ms)	Audio Pro- cessing (ms)	PTT To RX Audio (ms)	Audio Pro- cessing (ms)	
1	1075	555	1174	562	No Missed Audio
2	1080	555	1170	570	No Missed Audio
3	1070	550	1206	550	No Missed Audio
4	1085	550	1206	540	No Missed Audio
5	1045	550	1164	560	No Missed Audio
6	1095	550	1164	540	No Missed Audio
7	1065	545	1218	540	No Missed Audio
8	1120	560	1110	565	No Missed Audio
9	1075	560	1146	535	No Missed Audio
10	1100	550	1178	550	No Missed Audio
Average ~	1081	553	1174	551	No Missed Audio

d) Inherent latency of Cassidian (UHF,P25 digital, IP based, trunking) radio

	Radio to LMR Gateway (VIC3-2E&M)		LMR gateway (VIC3-2E&M) to Radio		Missed Audio
Test #	PTT To RX Audio (ms)	Audio pro- cessing (ms)	PTT To RX Audio (ms)	Audio pro- cessing (ms)	
1	1000	250	750	260	No Missed Audio
2	1000	250	750	260	No Missed Audio
3	1000	250	760	260	No Missed Audio
4	1000	250	760	265	No Missed Audio
5	1000	250	750	270	No Missed Audio
6	1000	250	760	260	No Missed Audio
7	1000	250	760	260	No Missed Audio
8	1000	250	760	250	No Missed Audio
9	1000	250	760	270	No Missed Audio
10	1000	250	760	260	No Missed Audio
Average ~	1000	250	757	262	No Missed Audio

e) Inherent latency of Motorola (Astro 25 digital, conventional) radio

Test #	Radio to LMR Gateway (VIC3-2E&M)	Radio to LMR Gateway (VIC3-2E&M)	LMR gateway (VIC3- 2E&M) to Radio	LMR gateway (VIC3- 2E&M) to Radio	Missed Audio
	PTT To RX Audio (ms)	Audio pro- cessing (ms)	PTT To RX Audio (ms)	Audio processing (ms)	
1	484	268	496	424	No Missed Audio
2	484	264	496	420	No Missed Audio

3	484	266	496	426	No Missed Audio
4	480	260	480	422	No Missed Audio
5	484	272	490	424	No Missed Audio
6	478	268	488	434	No Missed Audio
7	482	262	466	420	No Missed Audio
8	480	268	496	426	No Missed Audio
9	482	266	498	434	No Missed Audio
10	484	262	482	422	No Missed Audio
Average ~	482	266	489	425	No Missed Audio

Appendix A2 – Delay between spoke sites (LMR gateways) and the hub (core) site

a) Soft patch (UMS) – Spoke sites to the Hub site delay

Test #	Spoke site A to Hub audio delay (ms)	Spoke site B to Hub audio delay (ms)	Spoke site C to Hub audio delay (ms)	Spoke site D to Hub audio delay (ms)	Missed Audio
1	120	202	125	122	No Missed Audio
2	120	194	130	120	No Missed Audio
3	120	204	130	126	No Missed Audio
4	110	186	120	126	No Missed Audio
5	120	188	130	126	No Missed Audio
6	130	190	130	128	No Missed Audio
7	110	192	125	122	No Missed Audio

8	110	192	130	124	No Missed Audio
9	110	196	130	120	No Missed Audio
10	120	198	120	122	No Missed Audio
Average round trip delay ~	117	194	127	124	No Missed Audio
Average one way delay ~	59	97	64	62	No Missed Audio

b) Hard patch - loopback cable paced in the Hub (core) site

Test #	Spoke site B to Hub audio delay (ms)	Spoke site C to Hub audio delay (ms)	Spoke site D to Hub audio delay (ms)	Missed Audio
1	228	180	180	No Missed Audio
2	254	170	174	No Missed Audio
3	248	180	174	No Missed Audio
4	230	170	170	No Missed Audio
5	230	170	174	No Missed Audio
6	228	180	170	No Missed Audio
7	222	170	170	No Missed Audio
8	230	180	170	No Missed Audio
9	224	190	180	No Missed Audio
10	224	180	170	No Missed Audio
Average round trip	231	177	173	No Missed Audio

delay~				
Average one way delay~	116	90	87	No Missed Audio

c) Hard patch - loopback cable paced in the spoke sites

Test #	Hub to Spoke site B audio delay (ms)	Hub to Spoke site C audio delay (ms)	Hub to Spoke site D audio delay (ms)	Missed Audio
1	230	170	170	No Missed Audio
2	240	190	180	No Missed Audio
3	230	190	180	No Missed Audio
4	230	180	180	No Missed Audio
5	220	180	170	No Missed Audio
6	220	190	180	No Missed Audio
7	210	180	170	No Missed Audio
8	240	180	170	No Missed Audio
9	230	180	170	No Missed Audio
10	230	170	180	No Missed Audio
Average round trip delay ~	228	181	175	No Missed Audio
Average one way delay ~	114	91	88	No Missed Audio

Appendix A3 - End-to-end delay (user-to-user, a real life scenario)

a) Codan (analog/P25 digital, conventional) radios end-to-end latency

Test #	Analog		P25 Digital		Missed Audio
	PTT to RX Audio (CTSS tone) (ms)	Audio pro- cessing (ms)	PTT to RX Audio (NAC# code) (ms)	Audio pro- cessing (ms)	
1	442	322	244	361	No Missed Audio
2	456	317	249	354	No Missed Audio
3	422	330	254	353	No Missed Audio
4	474	324	249	346	No Missed Audio
5	439	317	239	382	No Missed Audio
6	446	330	234	367	No Missed Audio
7	446	314	239	340	No Missed Audio
8	438	306	249	340	No Missed Audio
9	408	300	249	356	No Missed Audio
10	420	322	244	357	No Missed Audio
Average round trip de- lay ~	439	318	245	356	No Missed Audio
Total average round trip de- lay	757		601		

b) Harris (UHF, digital proprietary, trunking) radios end-to-end latency

Test #	PTT To RX Audio (ms)	Audio processing (ms)	Missed Audio (ms)
1	2279	1295	No Missed Audio
2	2324	1220	No Missed Audio
3	2279	1250	No Missed Audio
4	2504	1245	No Missed Audio
5	2364	1295	No Missed Audio
6	2394	1270	No Missed Audio
7	2409	1280	No Missed Audio
8	2379	1285	No Missed Audio
9	2549	1310	First Syllable
10	2734	1295	First Syllable
Average round trip de- lay ~	2422	1275	No Missed Audio

c) Cassidian (UHF, P25 digital, trunking) radios end-to-end latency

Test #	PTT To RX Audio (ms)	Audio Difference (ms)	Missed Audio
1	1850	750	No Missed Audio
2	1850	750	No Missed Audio
3	1850	745	No Missed Audio
4	1900	750	No Missed Audio
5	1900	800	No Missed Audio
6	1900	750	No Missed Audio

7	1900	745	No Missed Audio
8	1900	750	No Missed Audio
9	1900	745	No Missed Audio
10	1900	800	No Missed Audio
Average round trip de- lay ~	1885	759	No Missed Audio

d) Motorola (VHF, Astro 25, conventional) radios end-to-end latency

Test #	PTT To RX Audio (ms)	Audio Difference (ms)	Missed Audio (ms)
1	1102	826	No Missed Audio
2	1100	828	No Missed Audio
3	1106	844	No Missed Audio
4	1086	820	No Missed Audio
5	1100	832	No Missed Audio
6	1094	832	No Missed Audio
7	1070	828	No Missed Audio
8	1100	804	No Missed Audio
9	1100	796	No Missed Audio
10	1088	792	No Missed Audio
Average round trip de- lay ~	1095	820	No Missed Audio

Appendix B - System-of-Systems RoIP deployment architecture II - complete test results

The following represent completed test results. Please note that in the tables are representative data samples. During our testing each characteristic outlined in the tables was retested many times on multiple occasions. For example, any time we would do optimization on the network or on the radio side results would be retested.

Appendix B1 – Inter-agency hub-to-hub delay

a) Inter-agency delay between agency X LMR gateways and agency Y hub (core) site

Test #	Agency X (Department A) to Agency Y Hub – audio delay	Agency X (Department B) to Agency Y Hub –audio delay	Agency X (Department C) to Agency Y Hub –audio delay	Agency X (Department D) to Agency Y Hub –audio delay	Missed Audio
1	610	629	620	613	No Missed Audio
2	600	677	620	621	No Missed Audio
3	600	669	625	605	No Missed Audio
4	610	669	620	599	No Missed Audio
5	610	667	620	611	No Missed Audio
6	600	659	625	603	No Missed Audio
7	600	661	610	605	No Missed Audio
8	600	673	600	621	No Missed Audio
9	600	697	620	621	No Missed Audio
10	600	689	620	621	No Missed Audio
Average	603	669	618	612	No Missed

round trip delay ~					Audio
Average one way delay ~	302	335	309	306	

b) Inter-agency delay between departments (LMR gateway-to-LMR gateway)

Test #	Agency X (Dept. A) to Agency Y (Dept. B) – audio delay	Agency X (Dept. A) to Agency Y (Dept. C) – audio delay	Agency X (Dept. B) to Agency Y (Dept. B) – audio delay	Agency X (Dept. B) to Agency Y (Dept. C) – audio delay	Agency X (Dept. C) to Agency Y (Dept. B) – audio delay	Agency X (Dept. C) to Agency Y (Dept. C) – audio delay	Agency X (Dept. D) to Agency Y (Dept. B) – audio delay	Agency X (Dept. D) to Agency Y (Dept. B) – audio delay	Missed Audio
1	760	826	799	845	790	836	783	829	No Missed Audio
2	790	836	867	913	810	856	811	857	No Missed Audio
3	790	836	859	905	815	861	795	841	No Missed Audio
4	770	816	829	875	780	826	759	805	No Missed Audio
5	770	816	827	873	780	826	771	817	No Missed Audio
6	790	836	849	895	815	861	793	839	No Missed Audio
7	790	836	851	897	800	846	795	841	No Missed Audio

8	790	836	863	909	790	836	811	857	No Missed Audio
9	780	826	877	923	800	846	801	847	No Missed Audio
10	760	806	849	895	780	826	781	827	No Missed Audio
Average round trip delay ~	779	827	847	893	796	842	790	836	No Missed Audio
Average one way delay ~	410	434	444	467	418	441	415	438	No Missed Audio

Appendix B2 - Inherent latency of radio systems in agency Y

a) Inherent latencies of agency Y Motorola (VHF, Astro 25 digital) conventional radios

	Radio to LMR Gateway (VIC3-2E&M)		LMR gateway (VIC3-2E&M) to Radio		Missed Audio
Test #	PTT To RX Audio (ms)	Audio Processing (ms)	PTT To RX Audio (ms)	Audio Processing (ms)	
1	480	230	490	420	No Missed Audio
2	475	235	490	420	No Missed Audio
3	480	240	490	420	No Missed Audio
4	485	240	485	470	No Missed Audio
5	480	232	485	415	No Missed Audio
6	475	235	490	420	No Missed Audio
7	485	230	480	420	No Missed Audio

8	480	220	490	430	No Missed Audio
9	478	236	490	430	No Missed Audio
10	486	230	485	420	No Missed Audio
Average ~	480	233	488	427	No Missed Audio

b) Inherent latencies of agency Y Motorola (VHF, Astro 25 digital) trunking radios

	Radio to LMR Gateway (VIC3-2E&M)		LMR gateway (VIC3-2E&M) to Radio		Missed Audio
Test #	PTT To RX Audio (ms)	Audio Differ- ence (ms)	PTT To RX Audio (ms)	Audio Differ- ence (ms)	
1	520	425	900	400	No Missed Audio
2	580	410	965	385	No Missed Audio
3	550	460	895	480	No Missed Audio
4	590	460	965	390	No Missed Audio
5	530	460	915	415	No Missed Audio
6	575	450	955	440	No Missed Audio
7	520	450	985	435	No Missed Audio
8	540	435	915	420	No Missed Audio
9	600	450	945	430	No Missed Audio
10	590	435	905	450	No Missed Audio
Average delay ~	560	444	935	425	No Missed Audio

Appendix B3 - End-to-End latency (user-to-user)

a) Agency X portable-to-portable delay – traffic looped back by the agency Y UMS

Test #	Site A (Hub) Codan, Analog, VHF, Conventional (ms)	Site A (Hub) Codan, P25 Digital, VHF, UHF/800MHz, (ms)	Site B Harris, Digital proprietary, UHF, Trunk- ing (ms)	Site C Cassidian, P25 Digital, UHF, Trunk- ing (ms)	Site D Motorola, Astro 25, VHF, Con- ventional (ms)	Missed Audio
	PTT to RX audio	PTT to RX audio	PTT to RX audio	PTT to RX audio	PTT to RX audio	
1	932	734	2920	2380	1640	First Word/First Syllable if delay > 2.5s
2	944	737	2920	2380	1630	First Word/First Syllable if delay > 2.5s
3	902	734	2920	2380	1620	First Word/First Syllable if delay > 2.5s
4	964	739	2860	2380	1620	First Word/First Syllable if delay > 2.5s
5	929	729	2900	2680	1608	First Word/First

						Syllable if delay > 2.5s
6	926	714	2900	2380	1620	First Word/First Syllable if delay > 2.5s
7	926	719	2900	2380	1595	First Word/First Syllable if delay > 2.5s
8	923	734	2900	2380	1630	First Word/First Syllable if delay > 2.5s
9	898	739	2900	2280	1602	First Word/First Syllable if delay > 2.5s
10	900	724	2900	2280	1671	First Word/First Syllable if delay > 2.5s
Avera ge round trip delay ~	924	730	2902	2390	1624	

b) Agency X portable-to-portable delay- through agency Y LMR gateways

Test #	Site B Harris, Digital proprietary, UHF, Trunking, through agency Y site B (ms)	Site C Cassidian, P25 Digital, UHF, Trunking (ms)	Missed Audio
	PTT to RX audio	PTT to RX audio	
1	3017	2600	First Word/First Syllable, respectively
2	3022	2600	First Word/First Syllable, respectively
3	2978	2500	First Word/First Syllable, respectively
4	3208	2600	First Word/First Syllable, respectively
5	3064	2500	First Word/First Syllable, respectively
6	3087	2500	First Word/First Syllable, respectively
7	3112	2600	First Word/First Syllable, respectively
8	3078	2500	First Word/First Syllable, respectively

			tively
9	3244	2600	First Word/First Syllable, respectively
10	3250	2500	First Word/First Syllable, respectively
Average roundtrip delay ~	3106	2550	First Word/First Syllable, respectively

c) End-to-end delay between agency X and agency Y portable radios

Test #	Agency X- Site B Harris, digital proprietary, UHF, trunking to Agency Y - Site C Motorola, Astro 25 digital, VHF, trunking (ms)		Missed Audio
	PTT to RX audio	Audio Delay	
1	2405	1445	No Missed Audio
2	2475	1395	No Missed Audio
3	2420	1415	No Missed Audio
4	2655	1420	First Syllable
5	2510	1465	No Missed Audio
6	2580	1480	First Syllable
7	2590	1485	No Missed Audio
8	2705	1490	First Syllable

9	2705	1490	First Syllable
10	2705	1490	First Syllable
Average delay ~	2575	1458	