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THESIS

VOIPNET: A SOFTWARE BASED COMMUNICATIONS TOOL FOR LOW-BANDWIDTH NETWORKS

by

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June 2007

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Maneuver element communications can be divided into Single-Channel Voice, Data Networks, and Telephony. Classified computer networks, such as SIPRNET are pushed to Infantry and Artillery Battalions via the EPLRS radio system. However, telephone services may or may not be supported due to limited availability of Multi-Channel Digital assets. Single-Channel Radio is utilized to communicate with higher, adjacent and subordinate organizations. While this is a sufficient means of communications, it is half-duplex, cumbersome, unreliable, and subject to availability due to net traffic. Voice over IP may be the solution to deploy full duplex telephone communications services to bandwidth deprived organizations via an existing wireless network infrastructure. The development and testing of a software based "VoIPNET" prototype would prove feasibility and provide critical primary telephone services to highly mobile maneuver elements. A detailed requirements analysis and design specification is provided for the VoIPNET application. In addition, the VoIPNET Prototype was successfully tested on a variety of EPLRS Networks and the results are summarized.

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VOIPNET: A SOFTWARE BASED COMMUNICATIONS TOOL FOR LOW-BANDWIDTH NETWORKS

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ABSTRACT

Maneuver element communications can be divided into Single-Channel Voice, Data Networks, and Telephony. Classified computer networks, such as SIPRNET are pushed to Infantry and Artillery Battalions via the EPLRS radio system. However, telephone services may or may not be supported due to limited availability of Multi-Channel Digital assets. Single-Channel Radio is utilized to communicate with higher, adjacent and subordinate organizations. While this is a sufficient means of communications, it is half-duplex, cumbersome, unreliable, and subject to availability due to net traffic. Voice over IP may be the solution to deploy full duplex telephone communications services to bandwidth deprived organizations, via an existing wireless network infrastructure. The development and testing of a software based “VoIPNET” prototype proved the EPLRS Network’s ability to provide critical primary telephone services, via VoIP, to highly mobile maneuver elements. Detailed requirements analysis and design specifications were developed for future development of the VoIPNET application. In addition, the results of VoipNET Prototype tests on an EPLRS network are compiled into deployment recommendations for units attempting to establish VoIP on an EPLRS network.

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ACKNOWLEDGMENTS

I chose this thesis topic for several reasons. First, I wanted to explore the feasibility of tactical communications via VoIP. This topic allowed me to produce a concept that is at least interesting and perhaps even useful. I hope this aids in aligning the state of infantry communications with those of industry. On this regard I would like to thank the staff at MCTSSA, Camp Pendleton for their patience and expertise. In particular, Capt Jeffery Wrobel for his excellent coordination skills and Master Gunnery Sergeant(Ret) Pedro “Pete” Zenquis for his remarkable professionalism. They fielded all of my ridiculous questions and took great pains to provide a quality test environment, excellent personnel and a quality education.

Secondly, I chose to develop a small application, so as to gain experience in as many aspects of the development process as possible. I would like to thank Professor Mantak Shing for driving home the necessity for quality Requirements and sound Design. Professor Auguston for showing me that formal methods are for people much smarter than I. And finally Professor Geoff Xie for his advice, guidance and most of all, his money.

Finally, I would like to thank my wife, Regina and daughter, Jordan, for their patience and understanding. For knowing when to leave me undisturbed in my office, and when to drag me out kicking and screaming. Thank you and I love you both.

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I. BACKGROUND AND VISION

A. INTRODUCTION

Maneuver element communications can be divided into Single-Channel Voice, Data Networks, and Telephony. Infantry Regiments currently possess the capability to draw sufficient bandwidth to support robust Data and Telephone networks, providing access to classified and unclassified data/telephone systems. Classified computer networks, such as SIPRNET are pushed to Infantry and Artillery Battalions via the EPLRS radio system. However, telephone services may or may not be supported due to limited availability of Multi-Channel Digital assets. As a result, the Maneuver Element's primary means of voice communications media is Single-Channel Radio.

Single-Channel Radio is utilized to communicate with higher, adjacent and subordinate organizations. While this is a sufficient means of communications, it is half-duplex, cumbersome, unreliable, and subject to availability due to net congestion. Therefore, the preferred media for Commanders and Staff is full duplex telephone communications. With the advent of EPLRS and the increasing availability of bandwidth at the lower echelons of command, it is advantageous to consider the utilization of existing data networks to provide telephone connectivity to units that do not have access to switched telephone services. Voice over IP may be the solution to deploy full duplex telephone communications services to bandwidth deprived organizations, via an existing wireless network infrastructure (EPLRS). The development of a software based

“VoIPNet” would provide critical primary telephone services to highly mobile maneuver elements and redundant telephone networks for units with switched telephone services.

This Thesis will identify, analyze, and document the requirements for an open source VoIP application capable of supporting both Voice and Video services over the EPLRS network.

Chapter I: Background information and the Vision of the thesis and application.

Chapter II: System features and performance requirements will be developed and analyzed.

Chapter III: Supplemental Requirements Specifications will increase the granularity of the Requirement topics from Chapter II.

Chapter IV: The Risk Management Plan will identify, analyze, and plan for potential pitfalls during the development of the Application.

Chapter V: The System Design Specification will provide the static and dynamic characteristics of the application.

Chapter VI: The Testing Plan will detail the procedures followed and features tested for the prototype. The initial testing will focus on prototype usability and suitability, while the operational testing will attempt to prove the EPLRS networks supportability of VoIP services.

Chapter VII: The Test Report will document the results of the Initial and Operational Prototype Testing.

Chapter VII: The Conclusion will summarize the findings of the test report and make deployment recommendations.

A prototype will be developed or an open source application chosen, that is capable of proving the concept of VoIP during the operation testing phase. Peer-to-peer, Video Teleconference and the EPLRS network's ability to support multiple simultaneous voice and video calls will be tested.

1. Purpose of the Vision Document

The purpose of this document is to collect, analyze, and define high-level user needs and application features for VoIPNET.

2. Overview of the VoIPNET Application

The VoIPNET application is intended to increase the quality of communications at the lower echelons of command. It will provide full-duplex voice communication, peer-to-peer video teleconferencing, file transfer, and "chat" functionality.

3. References

[1] "Managing Software Requirements." Leffingwell & Widrig.

B. USER DESCRIPTION

The VoIPNET application users are a collection of infantry, artillery, aviation and support organizations within the United States Military. Common to all users is the utilization of low-bandwidth, mobile radio communications, capable of data transmission at low rates.

1. User Demographics

The U.S. Military is progressing towards “net-centric” warfare. As bandwidth to lower echelons of command increase, so does the opportunity to develop new software tools to increase the quality of communications. Current projects underway include: Command and Control On-the-move Network Digital Over-the-horizon Relay (CONDOR) and the Joint Tactical Radio System (JTRS). Both programs are focused on pushing existing Command and Control Networks to the lowest echelons of Tactical Command.

2. User Profiles

a. Infantry Regiments/Brigade

The brigade or regiment is made up of two to five battalions under the command of a Colonel with a Sergeant Major as the senior non-commissioned officer. Armored cavalry, Ranger units and USMC infantry units of similar size to a brigade are called Regiments. Special-forces units are known as groups. Each Regiment/Brigade has a Command Staff composed of Administrative, Intelligence, Operations, Logistics, and Communications representatives.

b. Infantry Battalion/Squadron

The primary combat maneuver element, the battalion or squadron is composed of four to six companies and is commanded by a Lieutenant Colonel with a Sergeant Major as the senior non-commissioned adviser. The Executive Officer is a Major and second in command. The battalion is tactically and administratively self-sufficient and can conduct independent operations of a limited scope. Battalion sized armored and air cavalry units (U.S. Army) and all aviation units (USMC) are referred to as squadrons. Each Battalion/Squadron has a Command Staff composed of Administrative, Intelligence, Operations, Logistics, and Communications representatives.

c. Infantry Company

Company (in the infantry), battery (in the artillery) or troop (in the cavalry): The company, battery or troop is made up of three to five platoons and is typically commanded by a Captain. It usually has a First Lieutenant as the second in command and a First Sergeant as the senior non-commissioned officer.

3. Users Environment

The U.S. Military continues to be engaged in areas of operation that prove difficult for half-duplex communications. Urban terrain is an ideal environment to deploy a low-bandwidth data network for VoIPNET services. The U.S. Military currently uses the EPLRS (Enhanced Position Location Radio System) network for maneuver element access to the tactical network.

EPLRS Radio Sets (RSs) are primarily used as jam-resistant, secure data radios that transmit and receive tactical data that typically includes Operations orders, Fire support plans, Logistics reports, Situation Awareness (SA) data, Cryptographic keys for RSs, Configuration files for RSs, and E-mail.

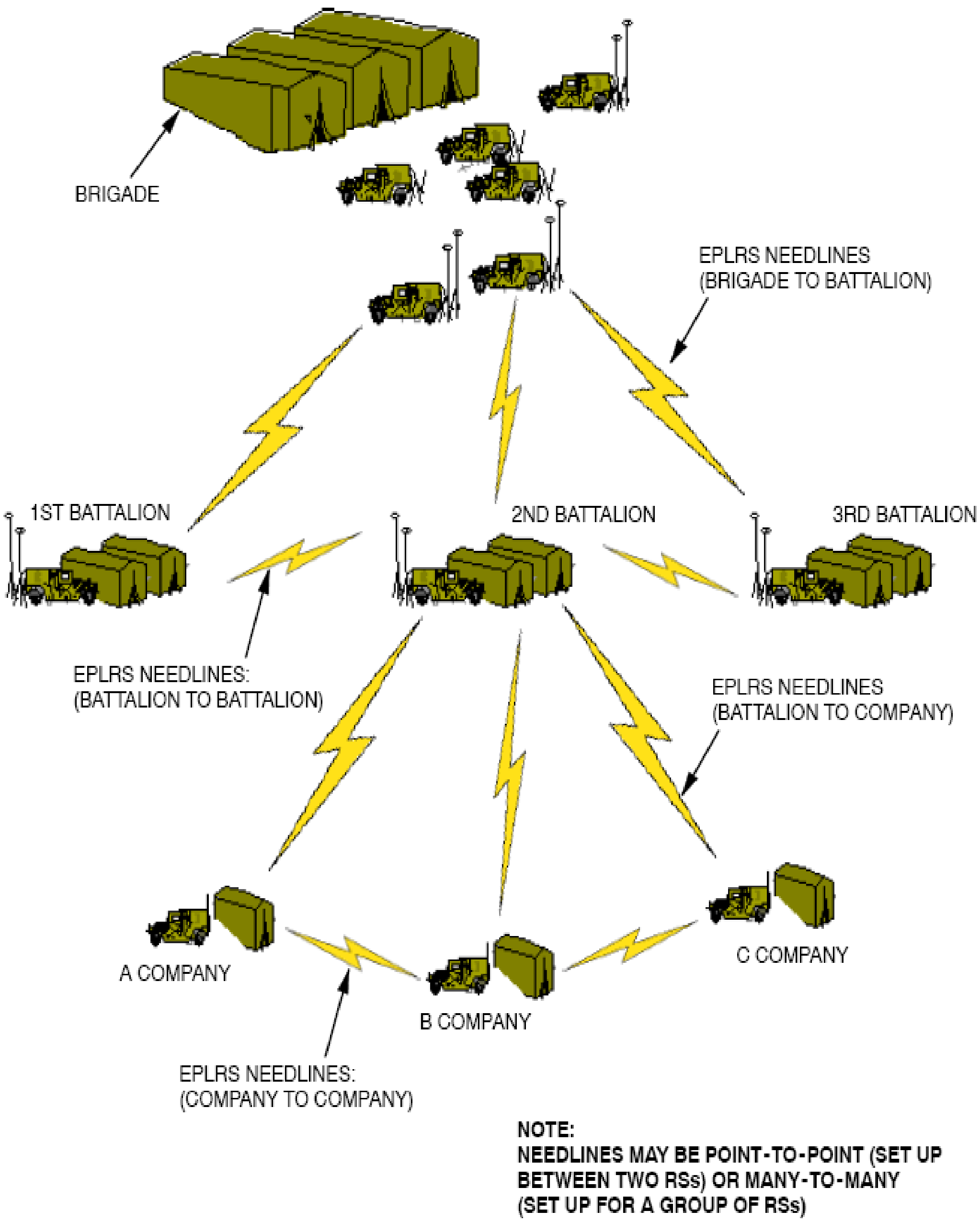


Figure 1. Sample EPLRS Network (U.S. Army, 2004).
Figure 2.

EPLRS utilizes the Time Division Multiple Access (TDMA) architecture. Each radio is assigned one or more time slots called Logical Time Slots (LTS) in which it will transmit its data onto the needline while the remaining radios listen.

Needlines

EPLRS RSs automatically route and deliver tactical data using multiple concurrent communication paths called *needlines*. A needline is a common set of time and frequency resources shared among two or more RSs to exchange data. A needline is the fundamental line of communication set up between individuals or groups of EPLRS RSs.

The needlines can be Many- to-Many, Few- to-Many or One- to-One. A single RS can support up to 32 needlines at the same time, but the maximum number is usually 28. Host computers can send and receive information too and receive from, many other host computers on the EPLRS network because the RSs can support many concurrent needlines via time division and frequency division multiplexing. RSs support needlines by sourcing data, receiving data or relaying data for other RSs. An RS can be a relay on some needlines and be a source or destination for data on other needlines virtually at the same time.

Each EPLRS network has a needline and corresponding Communication Circuit Assignments (CCA) to describe the characteristics of that needline. There are two basic types of needlines: point-to-point and broadcast. The two needline types are further sub-divided into several types:

CSMA- Carrier Sense Multiple Access provides a single cloud and shares bandwidth with all users. Each user has the ability to transmit and receive data at all times. This provides increased bandwidth flexibility but is also prone to resource “hogging” by high bandwidth applications or users. It is possible for a single user to consume all available bandwidth. Hop limits are programmed during EPLRS network planning. CSMA hops come at a price. Each planned hop on a CSMA network cuts the network's available bandwidth in half. A CSMA needline must use the same resources to support the source and destination RSs and the additional relays. For example, a CSMA network with 0 hops may provide 400kbps available bandwidth. However, the same network configuration with a single planned hop provides only 200kbps of bandwidth.

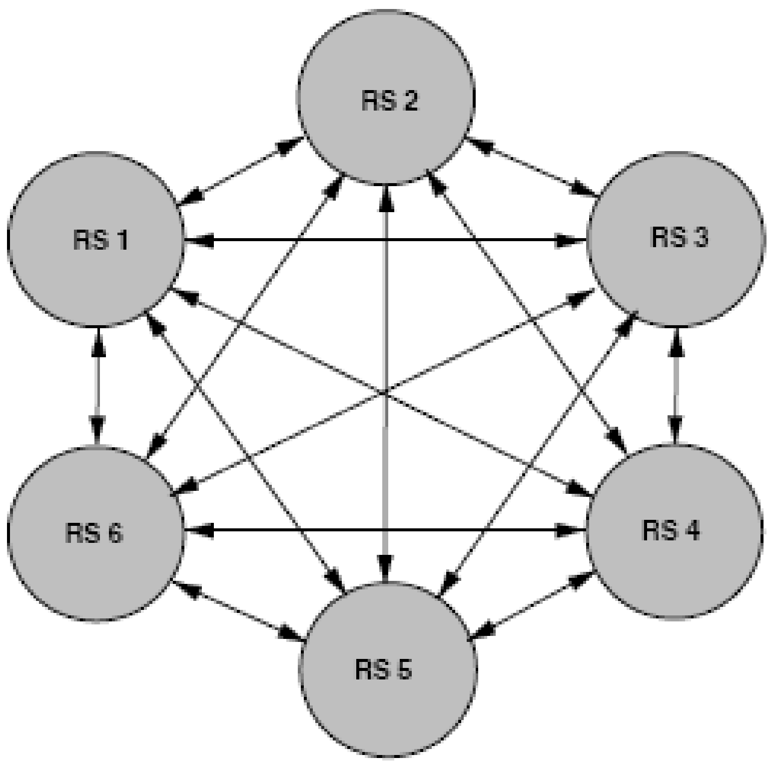


Figure 3. CSMA Many-to-Many Architecture (U.S. Army, 2004).

MSG- Multiple Source Group is a restricted CSMA. It separates the users into "core" and "fringe" members. "Core" members of the MSG have the ability to transmit and receive data at a predetermined data rate. "Fringe" members typically can receive only. Membership is determined by the allocation of shares. The network planner can allocate up to sixteen total shares. The more shares a user is allocated, the more bandwidth available. Advanced settings allow the redistribution of unused shares to other users. Converse to a CSMA needline, you can add relays to a Multi-Source Group (MSG) needline and see no reduction in throughput. The MSG needline uses an additional channel resource to support the relay. This has the added benefit of restricting the available bandwidth to each user and preventing resource "hogging."

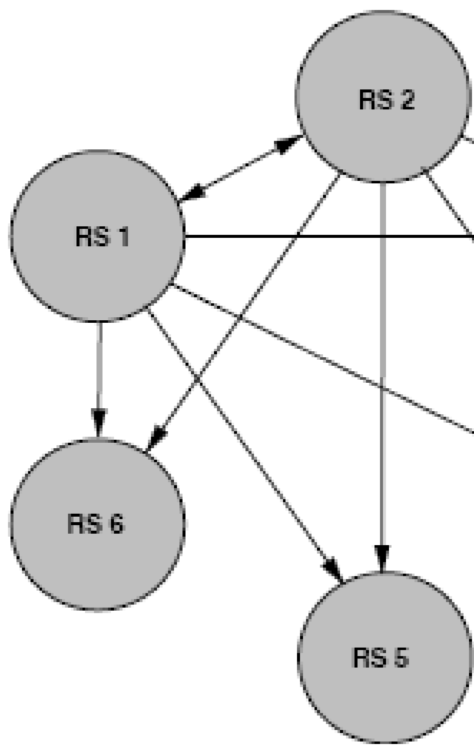


Figure 4. MSG Few-to-Many Architecture (U.S. Army, 2004).

DAP- Dynamic Permanent Virtual Circuit utilizes a single LTS to find and coordinate a virtual circuit between two radios. When the route is discovered it is added to the routing table and the DAP LTS is released. The data is then passed over a separate LTS.

HDR- High Data Rate Duplex circuits are static routes between radios, typically used to connect independent EPLRS networks. They are reliable networks with receipt delivery messages and packet retransmission. This aspect makes them unsuitable for VoIP applications.

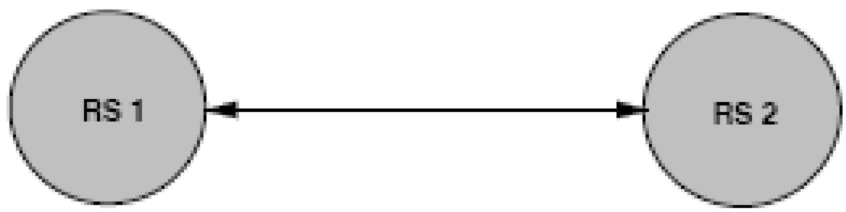


Figure 5. Point-to-Point Architecture (U.S. Army, 2004).

LDR- Low Data Rate Duplex circuits are similar to HDRs except for a lower data rate.

Needline Attribute	CSMA	MSG	HDR Duplex	LDR Duplex
Overall Characteristics	Broadcast circuit, one-to-one and many-to-many, automatic contention and collision reduction; no RS acknowledgement of data.	Broadcast needline, few-to-many (max of 4, 8, or 16 sources at any one time), no contention or collision, guaranteed or dynamic bandwidth allocation available.	One-to-one balanced, one-to-one, RS acknowledge, no contention or collision.	One-to-one balanced, one-to-one, RS acknowledge, no contention or collision.
Relay Characteristics	Up to 5 relays, selectable; automatic relay negotiation by needline participants.	Pipeline, up to 5 relays, selectable; automatic relay negotiation by needline participants.	Pipeline, if needline is one LTS or less, automatic negotiate 4 relays (5 hops). If needline is 2 or 4 LTSs, automatic negotiate 3 relays (4 hops). If more relays needed planner can configure up to 5 static relays in ENP.	Automatic relay negotiation by relays of opportunity, negotiated on LTS 2; up to 4 relays.
Bandwidth	1/4, 1/2, 1, 2, 4, or 8 LTS (bandwidth reduction for more relays).	1/4, 1/2, 1, 2, 4, or 8 LTS; source allocation assignment (no bandwidth reduction for more relays).	1/4, 1/2, 1, 2, or 4, LTS (bandwidth reduced for 5 relay option not 3 relay option).	Odd LTSs (3, 5, and 7) only.
Reliability	No RS acknowledge. High reliability reduces bandwidth by 25%.	No RS acknowledge.	Very high reliability (RSs acknowledge each transmission).	Very high reliability (RSs acknowledge each transmission).
Planning Considerations	Easy to plan, easy to enter into ENP, easy to deploy, relaying performed by RSs on needline.	Requires more planning, source and destination must be selected, bandwidth allocation decisions must be made among sources; 2 frequencies required for more relays.	Each endpoint must be selected. Relays can be automatically negotiated or pre-planned. Must be sufficient relay RSs available. 2 frequencies required for more relays.	Each endpoint must be selected, but relays automatically negotiated. Must be sufficient relay RSs available.
Advantages	Different relay schemes provide flexibility, minimum planning, supports one-to-one and many-to-many traffic.	5 relays cover wide area, guaranteed bandwidth (guaranteed speed of service for up to 16 sources per NL) and/or on-demand bandwidth, retains minimum bandwidth, supports one-to-one and few-to-many traffic, full bandwidth at 5 relays.	Increased link reliability.	Increased link reliability.

Figure 6. Needline Comparison Chart (U.S. Army, 2004).

Needline Attribute	CSMA	MSG	HDR Duplex	LDR Duplex
Disadvantages	Resources not reserved so no immediate transmit guarantee. Increased reliability reduces bandwidth; no link reliability.	Planning bandwidth can be complex, on-demand bandwidth claiming option can be slow; no link reliability.	Limited to 2 end points, equal bandwidth independent of endpoint data requirement.	Limited to 2 end points, equal bandwidth independent of endpoint data requirement.
When to Use	Requirement for many radios to have net access, transmit unicast and/or multicast IP messages.	Require guaranteed bandwidth for limited number of sources, need extended range without bandwidth penalty, guaranteed speed of service.	Exchange large size messages, require high reliability data link, require guaranteed bandwidth (guaranteed speed of service).	Require high reliability data link, require guaranteed bandwidth (guaranteed speed of service).
When Not to Use	Frequent requirement to exchange large size (>1MB) messages, require guaranteed bandwidth (guaranteed speed of service).	Many sources (radios) are required to access the network frequently and consistently. If application is not tolerant of slow bandwidth acquisition must use immediate share claim option.	Exchange messages between many-to-many sources.	Exchange messages between many-to-many sources or high bandwidth.
Typical Application	SA network for CSMA short and C2 network for CSMA normal.	Sensor netting (e.g., air defense).	TOC-to-TOC large file transfer.	Battle management data (e.g., air defense).

Figure 7. Needline Comparison Chart (Cont) (U.S. Army, 2004).

The CCA contains a waveform mode property. The Waveform Mode is either 2 or 4 Ms Modes. The waveform mode determines how much data will be transmitted per two or four ms “burst.” There are 11 different types of *waveform modes* available for EPLRS networks. The waveform mode specifies the timing format used for over-the-air message transmissions. Each waveform mode offers a unique tradeoff of operational range, jam resistance, and data rate. As a general rule the higher the waveform mode the greater the bandwidth available for the EPLRS network. The lower the mode the more resistant to jamming and more robust the needline. A robust needline translates into extended ranges for the wireless network.

Waveform Group (Timeslot)	Waveform Mode	Data Rate (KBPS)	User Data Bits per Trans-mission	User Data Bytes per Trans-mission	General Anti-Jam Performance	90% Burst Throughput (dBm)	RS-to-RS Propagation Range (No Relays) (NMI)	RS-to-RS Propagation Range (No Relays) (Km)
Tactical Internet (2ms)	0	38	80	10	Better	-100	89	165
	1	38	80	10	Best	-102	68	126
	2	77	160	20	Better	-100	63	117
	3	115	240	30	Good	-98	62	115
	4	311	648	81	OK	-94	54	100
	14	430	896	112	OK	-94	15	28
Expanded Data (4ms)	5	65	272	34	Best	-102	91	169
	6	127	528	66	Better	-100	94	174
	7	184	768	96	Good	-98	85	157
	8	238	992	124	Good	-98	85	157
	9	486	2024	253	OK	-94	58	107

Figure 8. Waveform Chart (U.S. Army, 2004).

The data is carried over a wireless UHF frequency hopping RF network. The radio hops on up to 8 channels covering a frequency range from 420-450 MHz. The fewer channels utilized the less secure and greater the bandwidth. Each radio utilizes static routing tables to route traffic throughout the network. EPLRS networks are capable of supporting from 0 to 6 hops and 5 relays. A hop occurs when a packet is routed from one radio to another. If the receiving radio must forward the packet to another radio, then a relay has occurred.

Each EPLRS network has 8 LTSs available. An LTS is a Logical Time Slot. The LTS maps the traffic on the EPLRS radio for a specific logical network. It is possible to assign a different network type to each LTS. Or assign several LTSs to one network. This affords the flexibility to split the available bandwidth between a variety of networks. Each network is assigned its own needline. The needline is a logically separate network with its own network IP address. The needline IP address serves as the default gateway for each Radio in the network.

4. Key User Needs

a. Poor Half-duplex Voice Communications

Current single-channel radio communications are cumbersome and require extensive human-to-human protocols to communicate effectively. A Software-based full-duplex “telephone” communication application will move the communication protocol implementation from the user’s domain to the applications domain.

b. Poor Information Exchange Quality

Voice media communications alone often do not provide sufficient information to convey intent and exchange ideas clearly. Often commanders are required to relocate in order to conduct face-to-face meetings to convey critical information or intent. A video teleconferencing capability would increase the quality of communication and provide face-to-face communication without relocation.

c. Inaccurate Telephone Directory Services

Current Telephone Directory procedures are decentralized and updated frequently. As Telephone numbers change they make distributed directories incorrect. A real-time Directory Services Manager would ensure a single, efficient, and accurate directory for users.

d. Retransmission in Compartmentalized Terrain

Current voice systems are line-of-site radio systems and require frequent retransmission to communicate in urban and jungle environments. A networked based communications tool, with automatic routing capability would alleviate the retransmission requirement in all but the most austere of networks.

5. VoIPNET Alternatives

a. Existing Open-Source SoftPhone

Open-Source SoftPhone are free and currently at a state of functional development that would meet and exceed the needs of the user. However, they all provide many addition complex features that are not desired by the user. Products include: SipXezPhone, Gizmo Project, and IaxComm.

b. Commercial Off-the-Shelf (COTS)

COTS provide a comparable solution to open-source, but at additional cost. Products include: Avaya IP SoftPhone, Cisco IPSoftPhone, ExpressTalk, Pingtel SIP SoftPhone, and Vonage SoftPhone.

C. PROPOSED SOLUTION

Section C provides a high-level view of application capabilities, application interfaces, and system configurations.

1. Application Perspective

The VoIPNET application is a stand-alone VoIP application developed to increase the quality of communications for lower echelons of command. It is a low-cost simple alternative to expensive COTS products. In addition, it eliminates excessive and unnecessary features found in both open-source and COTS products.

2. Application Position Statement

U.S. Military Maneuver Elements need portable software based networked communication tools. VoIPNET will increase the quality of communication at lower echelons of command. Unlike COTS products and current Open-source options VoIPNET will provide a standard set of reliable communications tools at minimum cost without unneeded features.

3. Assumptions and Dependencies

VoIPNET will be developed for Windows platforms. The development target network is the Enhanced Position Location Radio System (EPLRS).

4. Application Capabilities Summary

User Benefits	**Supporting Features**
Full Duplex Voice Communication	Button-Dial, Keyboard-Dial, Redial, Mute, Speed Dial, Phone Configuration
Face-to-Face Communication	VTC
Accurate Directory Services	Directory Distribution, Directory Update
Teleconference Capability	Initiate Teleconference, Disconnect Teleconference, Accept Teleconference, Decline Telconference
Message Center Capablity	Message Center Configuration, Save Message, Delete Message, Check Message, Leave Message
Ringer Selection Capability	Ringer-High, Ringer-Medium, Ringer-Low, Ringer-Mute, Ringer-Flash
File Transfer	File-Tx

Figure 9. VoIPNET Capabilities Summary.

D. FEATURE ATTRIBUTES

Section D will describe those feature attributes that will be utilized to track, prioritize, and manage the items proposed for implementation and development.

1. Priority

This field is established by the user. It establishes a feature hierarchy that drives development.

Critical: Essential feature. Failure to implement means the application will fail to meet the user needs. All critical features must be implemented in the scheduled release or a schedule slip will occur.

Important: A feature that is important to the effectiveness and/or efficiency of the application. Lack of inclusion will affect customer satisfaction, but will not delay release.

Useful: A feature that will be used infrequently. The lack of inclusion in a release will not affect user satisfaction, and therefore will not delay release.

2. Effort

Effort is an estimate of work required to implement a particular feature. It is utilized to gauge complexity of feature implementation.

Hard: McCabe complexity > 7.

Firm: 3 **<** McCabe complexity < 7.

Soft: McCabe complexity < 3.

3. Risk

Each feature is evaluated on its potential impact on cost, schedule delays or cancellation. Risk is measured by its probability of occurrence and its impact on the afore mentioned resources and deliverables.

HIGH/HIGH: Likely occurrence of unforeseen event and high impact if the event occurs. Occurrence will affect schedule, and or cost.

MEDIUM/MEDIUM: Moderate likelihood of occurrence of an unforeseen event and a moderate impact if the event occurs. Occurrence may affect schedule and or cost.

LOW/LOW: It is not likely an unforeseen event will occur during the implementation of a particular feature and the impact of such an occurrence is minor. Occurrence will not adversely affect schedule or cost.

4. Target Release

The target release records the intended application version that the feature will first appear.

5. Reason

The reason field is used to track the user need that initiates any particular feature. It may include a brief description of a reference to a description.

E. APPLICATION FEATURES

Section E provides a high-level description of the application features. This description provides the application capabilities that are necessary to deliver benefits to the user, the basis for definition, scope management, and project management.

1. Peer-2-Peer Voice (P2P)

P2P utilizes VoIP technology to provide the user with full-duplex voice communications over an existing data network.

2. Video teleconferencing (VTC)

VTC provides the user with face-to-face communications over an existing data network.

3. Phonebook

The phonebook provides a centrally managed, automatically distributed, accurate and reliable directory of organizational names and telephone numbers over an existing data network.

4. Message Center Services (VoiceMail)

VoiceMail provides the user with a password protected audio message repository over an existing data network.

5. Conference Call (TeleCon)

Telecon utilizes the P2P feature to provide users with the ability to conference call with up to 2 other users.

6. Ringer Selection

Ringer selection will allow the user to adjust the ringer settings for incoming calls.

7. Button Dialing

Button Dialing will allow for GUI dialing via a mouse or stylus input device.

8. Keyboard Dialing

Keyboard Dialing will allow for GUI dialing via keyboard input device.

9. File Transfer (FileTx)

File transfer will allow the user to transfer files over an existing data network.

10. Speed Dial

Speed Dial provides a user customizable one button dialing capability.

11. Re-Dial

Redial is a one button feature that calls the last number dialed.

12. Mute

Mute prevents the distant user from hearing the voice of the initiating user.

F. NON-FUNCTIONAL REQUIREMENTS

Non-Functional Requirements address applicable Standards, Usability, Dependability, and Performance.

1. Applicable Standards

The development of the VoIPNET application must conform to the following applicable standards:

Signaling	
H.323	H.323
Megaco H.248	Gateway Control Protocol
MGCP	Media Gateway Control Protocol
RVP over IP	Remote Voice Protocol Over IP Specification
SAPv2	Session Announcement Protocol
SGCP	Simple Gateway Control Protocol
SIP	Session Initiation Protocol
Skinny	Skinny Client Control Protocol (Cisco)
Media	
DVB	Digital Video Broadcasting
H.261	Video stream for transport using the real-time transport
H.263	Bitstream in the Real-time Transport Protocol
RTCP	RTP Control protocol
RTP	Real-Time Transport

H.323 Protocols Suite	
H.225	Covers narrow-band visual telephone services
H.225 Annex G	
H.225E	
H.235	Security and authentication
H.323SET	
H.245	Negotiates channel usage and capabilities
H.450.1	Series defines Supplementary Services for H.323
H.450.2	Call Transfer supplementary service for H.323
H.450.3	Call diversion supplementary service for H.323
H.450.4	Call Hold supplementary service
H.450.5	Call Park supplementary service
H.450.6	Call Waiting supplementary service
H.450.7	Message Waiting Indication supplementary service
H.450.8	Calling Party Name Presentation supplementary service
H.450.9	Completion of Calls to Busy Subscribers supplementary service
H.450.10	Call Offer supplementary service
H.450.11	Call Intrusion supplementary service
H.450.12	ANF-CMN supplementary service
RAS	Manages registration, admission, status
T.38	IP-based fax service maps
T.125	Multipoint Communication Service Protocol (MCS).
SIP Protocols	
MIME	
SDP	Session Description Protocol
SIP	Session Initiation Protocol
Graphical User Interface	
Apple Computer's Macintosh Human User Interface Guidelines	

Figure 10. Development Standards.

2. System Requirements

Windows, PC Compatible Microphone, PC Compatible Headset, Network Connection > 100 kbps.

3. Security Requirements

Password Protection for login only. Traffic encryption provided via Tactical Networks.

G. DOCUMENTATION REQUIREMENTS

Section G identifies the deliverable documentation for the VoIPNET application.

1. Prototype Users Manual

The Users Manual will detail the necessary steps to utilize all of the features available for each release.

2. Installation Guide

The Installation Guide will provide simple and easy installation and configuration instructions for both the application and data network.

3. Version Specific Read-Me File

The Read-Me file will highlight changes to this release of the application. In addition it should discuss any known compatibility issues with previous releases.

H. GLOSSARY

Effort - the amount of work required to implement a specific feature.

COTS - Commercial Off the Shelf.

JTRS - Joint Tactical Radio System.

Low-bandwidth Network - a data networks whose throughput is less than 300 Kbps.

P2P - a refernece to a Peer to Peer architecture.

SINCGARS: Single Channel Ground Air Radio Set. A frquency hopping encrypted radio system capable of transferring data at low rates. Primary readio system for U.S. military forces.

SoftPhone - Software based telphone.

Supplier [1] - The person or persons who produce a product for a customer.

System - All of the hardware and software to include VoIPNET application, EPLRS Radio Network, VoIP headset, Terminal computer, and intermediate routing devices.

System User - see User.

UA- User Agent

User [1] - The person or persons who operate or interact directly with the product. The User(s) and the customer(s) are not the same person(s).

User interface - physical manner in which the user will interact with a given application.

VoIP - Voice over Internet Protocol(IP).

VoIPNET: Voice over Internet Protocol network. The name given the application under development.

VTC - Video teleconferencing.

II. SUPPLEMENTARY REQUIREMENTS SPECIFICATION

A. INTRODUCTION

1. Purpose

This document details the functional and non-functional requirements for the VoIPNet application. Although this document is intended as a set of Requirements, not a design, some technical information has been included with the requirements description. The primary audience for this document is the software development team members, system engineers, and the customer.

2. Scope

The VoIPNET System is comprised of a single software and multiple hardware subsystems. The Software sub-system contains the Graphic User interface, Phonebook Database, and network Communications tools. The Hardware subsystem contains the EPLRS radio system, Intelligence Operations Workstation, and a PC headset. This document identifies only the software portions of the system. Although there are functional references to hardware requirements and configurations, they will only be addressed in the high-level system architecture. The software will provide a suite of quality communications tools for maneuver element commanders and staff.

3. Objectives and Success Criteria

The objective is to develop, test and demonstrate a working prototype of the VoIPNET application.

1. *Qualitative Evaluation:* An iterative and exhaustive analysis of the requirements will be conducted to ensure that the delivered product features correspond to system and sub-system requirements. In addition, a detailed testing and evaluation phase will preclude subsequent iterations of requirements analysis to ensure that discovered requirements are included in subsequent design and development.

2. *Simulation:* VoIPNET will be developed utilizing the NETBEANS 5.0 Integrated Development Environment (IDE). At each phase a workable prototype will be tested utilizing a simulated low-bandwidth network and an established EPLRS Network.

3. *Usability:*

a. A Limited User Evaluation (LUE) will be conducted with users to refine requirements.

4. Definitions, Acronyms and Abbreviations

A complete glossary of terms, definitions, acronyms, and abbreviations has been provided in Section I of this document.

5. References

[1] IEEE Std 830-1998 “Recommended Practice for Software Requirements Specifications.”

[2] IEEE Std 1233-1998 “Guide for Developing System Requirements Specifications.”

[3] IEEE Std 1016-1998 “Recommended Practice for Software Design Descriptions.”

[4] “Managing Software Requirements.” Leffingwell & Widrig.

B. CURRENT SYSTEM

1. Overview

The current primary means of maneuver element communication are single-channel voice circuits. The voice circuits provide encrypted, half-duplex voice communications. Excessive protocols are required to manage traffic and

C. PROPOSED SYSTEM

1. Overview

The VoIPNET application is intended to increase the quality of communications at the lower echelons of command. It will provide full-duplex voice communication, peer-to-peer video teleconferencing, file transfer, and “chat” functionality.

2. Functional Requirements

a. Performance Requirements

See Vision Document Chapter I, section E

b. Design Constraints

See Vision Document Chapter I, section F, par 2

3. Nonfunctional Requirements

a. Usability:

Training: No additional training is required. Users will be able to use all features within 10 minutes of reading the Users Manual.

b. Reliability:

The VoIPNET application will have an availability rate of **99%.** The following classifications are used to organize the types of failures that are prevalent in VoIP systems.

Type I - Failure: Severe operational incidents that would definitely result in the removal and reinstallation of the application. These constitute "hard-core" failures that would require the services of a trained administrator to recover. Mean Time Between (MTBF)- Type I < 2 failures per year, Mean Time To Repair (MTTR) – Type I < 10 minutes (weibull.com 2007).

Type II - Intervention: Any unplanned occurrence or failure of application mission that requires the user to manually adjust or otherwise intervene with the application or its output. These are "nuisance failures" that can be recovered by the customer, or with the aid of phone support. Depending on the nature of the failure mode, groups of the Type II failures could be upgraded to Type I if they exceed a predefined frequency of occurrence. Examples include, dropped calls, poor Voice quality, Message Center Errors, and incorrect connections, not deemed network related. Mean Time Between (MTBF)- Type II < 1 failure per month, Mean Time To Repair (MTTR) – Type II < 5 minutes (weibull.com 2007).

Type III - Event: Events will include all other occurrences that do not fall into either of the categories above. This might include events that cannot directly be classified as failures, but whose frequency is of engineering interest and would be appropriate for statistical analysis. Examples include failures caused by ancillary equipment malfunction or operator error. Mean Time Between (MTBF)- Type III < 1 failure per day (weibull.com 2007).

c. Performance

The application will be designed to support a single user at a time.

The application will be designed to support multiple accounts.

The application will complete login process within 5 seconds.

Dial-Center Requirements

The application will initiate a dial tone within 2 seconds of the user taking the phone off-hook.

The application will initiate a call within 1 second after the user initiates the dial sequence.

The application will initiate a ring tone within 4 seconds after the user initiates the dial sequence.

The application will terminate a connection within 1 second of the user selecting the “Hang-up” Button.

The application will open the VTC window within one second of the user selecting the “VTC” button.

The application will initiate a call within 2 seconds of the user selecting the “REDIAL” button.

The application will mute a call with 1 second of the user selecting the “MUTE” button.

The application will open the Conference call window within one second after the user selecting the “TELECON” button.

Message Center

The application will initiate the “Check Messages Greeting” within 2 seconds of the user selecting the “Check Messages Button.”

The application will open the message list window/table within 2 seconds of the user selecting the “Check Message Button.

The application will play the current message within 5 seconds of the user selecting the “PLAY” button.

The application will delete the current message within 3 seconds of the user selecting the “DELETE” button.

The application will save the current message within 3 seconds of the user selecting the “SAVE” button.

The application will initiate the dial sequence for the current message within 2 seconds of the user selecting the “RETURN CALL” button.

Directory Requirements

The application will open the Directory window within 2 seconds of the user selecting the “DIRECTORY” button.

The application will complete the Directory search within 5 seconds of the user selecting the “SEARCH” button.

The application will switch the tabular view of the directory within 1 second of the user selecting the “A, B,C,…Z, 1,2,3…” tab.

The application will initiate the dialing sequence of the selected Directory entry within 2 seconds of the user selecting a directory entry.

VTC Requirements

The application will open the VTC window within one second after the user selecting the “VTC” button.

The application will open the directory window within 3 seconds of the user selecting the “INVITE” button.

The application will complete the VTC Invite sequence within 3 seconds of the user selecting the directory entry.

The application will terminate all unanswered VTC invites within 10 seconds.

The application will close the current VTC connection within 2 seconds of the user selecting the “END VTC” button.

Teleconference Requirements

The application will open the Conference call window within one second after the user selecting the “TELECON” button.

The application will open the directory window within 3 seconds of the user selecting the “INVITE” button.

The application will complete the Telecon Invite sequence within 3 seconds of the user selecting the directory entry.

The application will terminate all unanswered Telecon invites within 10 seconds.

The application will close the current Teleconference connection within 2 seconds of the user selecting the “LEAVE TELECON” button.

d. Supportability

The application will require less than 1MB of system memory.

The application will run on any platform that includes Java Virtual Machine.

e. Implementation

The application is intended to be installed on IOW’s currently deployed, via network download and/or disk installation.

f. Interface

The Interface requirements document the Human Computer Interaction (HCI) features of the VoIPNET application. Initial usability requirements include the following:

User Interface: this is the basic Graphical User Interface (GUI) it will integrate the core functional areas of software into a single large button screen with limited windows.

Message Center Interface: this is a component of the GUI and provides a multi-button interface for the user to manage their message mailbox.

VTC Interface: Pop-up window component of the GUI. This window will include all of the VTC operations.

Telephone Book Interface: this component will mimic in appearance a standard telephone directory. It will include an alphabetic tabular representation as well as a search function.

Admin Mode: this mode of operation will allow a VoIPNET administrator to update the Telephone directory and set preemption priorities for critical users.

The application will utilize PC clock.

The application will interface with the PC file system.

4. System Models

Section 4 elaborates the operational expectations of the VoIPNET application. This section is the heart of the requirements document and should be used drive the development lifecycle.

a. Scenarios

The user will open the VoIPNET application and will be prompted to login. The GUI will open and the user will be prompted to set ring tone and light level (default is FLASH and Silent (low/red)). The user will then utilize the application to make and receive calls, initiate VTC meetings, initiate and join conference calls and search the directory for contact information.

b. Use Case Models

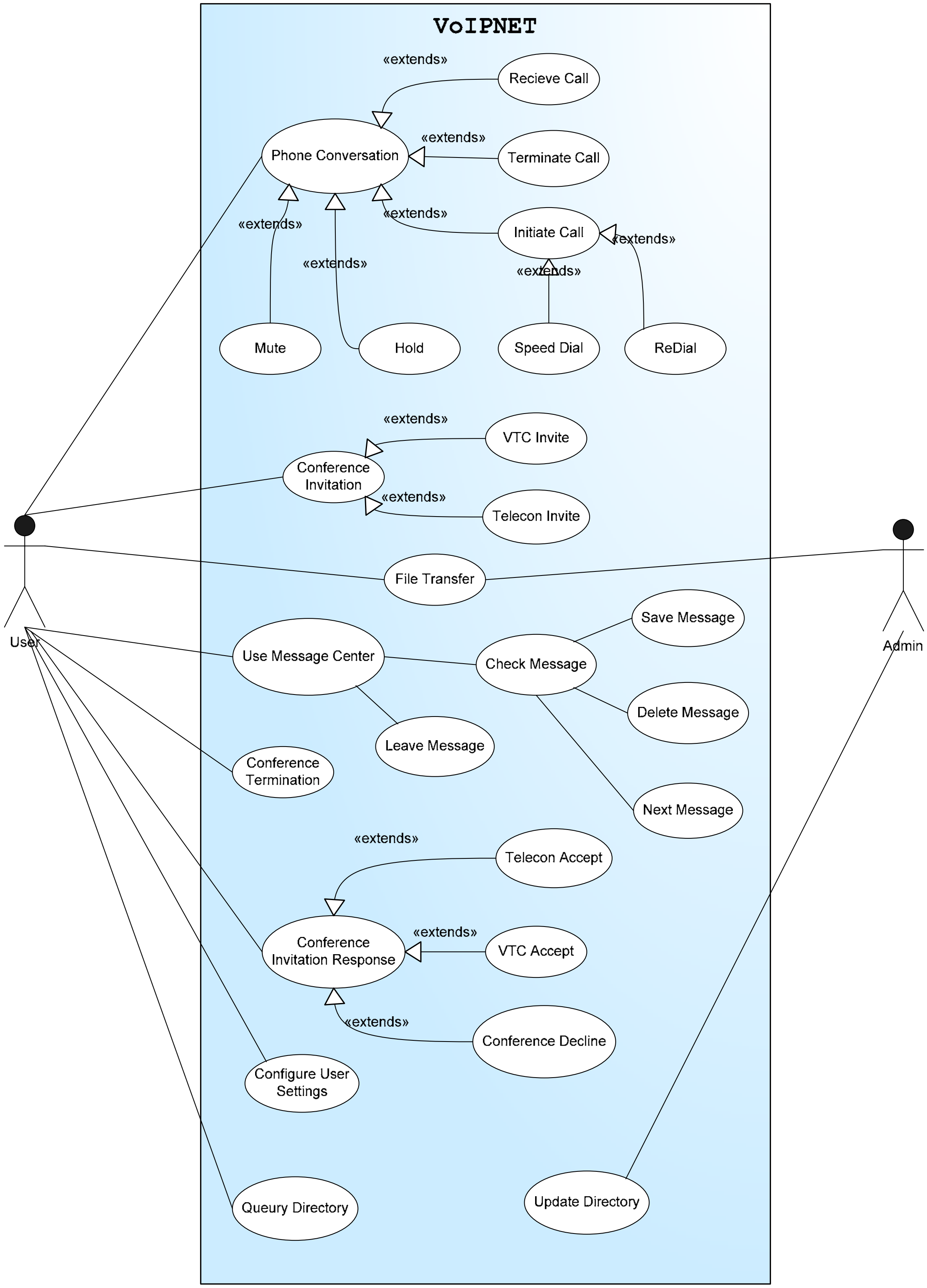


Figure 11. Use Case Model

Use case: UC-1 Phone Conversation
Primary Actor: User1
Other Actors: Data Network, User2
Description: User engages in a phone conversation

Stakeholders and Interest:

User wants a quick connection and a clear, error free conversation.

Entry conditions:

- Application is running.
- Network is up and stable.
- User logged in
- User not engaged in any other conversation or user has placed current call on hold to establish a teleconference.

Exit conditions:

- Either User terminates the call.
- System displays Main Menu.

Flow of events:

1. If User1 initiates a call
    a. Use UC-1a User Initiates Call

   If User1 receives a call
    a. Use UC-1b User Receives Call

   IF User1 terminates a call
    a. Use UC-1c User Terminates Call

Alternate Flows:

1a. User1 terminates the call before a connection is established.

Special Requirements:

None.

Use case: UC-1a Initiate Call
Primary Actor: User1

Other Actors: Data Network, User2
Description: Sub-use Case of UC-1 Phone Conversation.

Stakeholders and Interest:

User wants an intuitive easy to use interface with the Dialing mechanisms.

Entry conditions:

- Application is running.
- Network is up and stable.
- User not engaged in any other conversation or user has placed current call on hold to establish a teleconference.

Exit conditions:

- User “clears” the number or User “Dials” the number.
- Connection made or request terminated.
- System displays Main Menu.

Flow of events:

1. User1 enters number via keypad.
2. User1 “dials” the number.
3. The application plays the ring-back tone (RBT).
4. User2 answers call.
5. The application terminates the RBT.

Alternate Flows:

1a. If User1 selects Directory to search for a number.
    a. Use UC-6 Query Directory.
1b. User1 selects “Speed Dial” preset button.
    a. Use UC-1a.1 Speed Dial
1c. If User initiates “re-dial” function.
    a. Use UC-1a.2 Redial
3a. User1 terminates the call before a connection is established.
3b. User2 number not in service. Application plays “Number Not in Service” Message.
4c. User2 is “busy,” the application directs User1 to User2 mailbox. Use Leave Message UC.

Special Requirements:

None.

Use case: UC-1a.1 Speed Dial
Primary Actor: User1
Other Actors: Data Network, User2
Description: Sub-use Case of UC-1a Initiate Call.

Stakeholders and Interest:

User wants an intuitive easy to use interface with the Dialing mechanisms.

Entry conditions:

- Application is running.
- Network is up and stable.
- User not engaged in any other conversation or user has placed current call on hold to establish a teleconference.

Exit conditions:

- User “clears” the number or User “Dials” the number.
- Connection made or request terminated.
- System displays Main Menu.

Flow of events:

1. User1 selects speed dial number to call.
2. Application places the number in the Dial Center Dialog Box.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-1a.2 Redial
Primary Actor: User1
Other Actors: Data Network, User2

Description: Sub-use Case of UC-1a Initiate Call.

Stakeholders and Interest:

User wants an intuitive easy to use interface with the Dialing mechanisms.

Entry conditions:

- Application is running.
- Network is up and stable.
- User not engaged in any other conversation or user has placed current call on hold to establish a teleconference.

Exit conditions:

- User “clears” the number or User “Dials” the number.
- Connection made or request terminated.
- System displays Main Menu.

Flow of events:

1. The user selects “re-dial” function.
2. Last number called is placed in the Dial Center Dialog Box.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-1a.3 Abort Dial
Primary Actor: User1
Other Actors: Data Network, User2
Description: Sub-use Case of UC-1a Initiate Call.

Stakeholders and Interest:

User wants an intuitive easy to use interface with the Dialing mechanisms.
Entry conditions:

- Application is running.
- Network is up and stable.
- User not engaged in any other conversation or user has placed current call on hold to establish a teleconference.

Exit conditions:

- Connection request terminated.
- System displays Main Menu.

Flow of events:

1. User1 hangs up the call.
2. The application terminates the call.

Alternate Flows:

None.

Special Requirements:

A call must have been initiated.

Use case: UC-1b User Receives Call
Primary Actor: User2
Other Actors: Data Network, User1

Stakeholders and Interest:

User2 wants an intuitive easy to use interface with the Dialing Center mechanisms.

Entry conditions:

- Application is running.
- Network is up and stable.
- User not engaged in any other conversation or user has placed current call on hold to establish a teleconference.
- User2 has initiated a call to User1.

Exit conditions:

- Connection made or request terminated.

- System displays Main Menu.

Flow of events:

1. Application generates a ring tone for User1.
2. User1 answers the call.
3. User1 conducts UC-1 Phone Conversation

Alternate Flows:

1a. Silent mode is set so a Flashing screen signals the incoming call.
2a. User1 is directed to User2 Mailbox.

Special Requirements:

None.

Use case: UC-1c User Terminates Call
Primary Actor: User
Other Actors: Data Network

Stakeholders and Interest:

User wants a quick and simple way to terminate a call.

Entry conditions:

- Application is running.
- Network is up and stable.
- User engaged in a phone conversation.

Exit conditions:

- Connection terminated.
- System displays Main Menu.

Flow of events:

1. User terminates the connection.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-1d Mute
Primary Actor: User
Other Actors: Data Network

Stakeholders and Interest:

User wants a quick and simple way to mute a call.

Entry conditions:

- Application is running.
- Network is up and stable.
- User engaged in a phone conversation.

Exit conditions:

- Mute is terminated.
- System displays Main Menu.

Flow of events:

1. User selects “Mute” function
2. Application mutes voice transmission.
3. User selects “Un-mute” function.
4. Application un-mutes voice transmission.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-1e Hold
Primary Actor: User1
Other Actors: Data Network, User2

Stakeholders and Interest:

User wants a quick and simple way to place a call on hold.

Entry conditions:

- Application is running.
- Network is up and stable.
- User engaged in a phone conversation.

Exit conditions:

- Hold is terminated.
- System displays Main Menu.

Flow of events:

1. User selects “Hold” function
2. Application places connection in “hold status.”
3. User selects “Un-hold” function.
4. Application removes call from hold status.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-2 User Creates a Conference Invitation
Primary Actor: User1
Other Actors: Data Network, User2

Stakeholders and Interest:

User wants an intuitive easy to use interface with the Conference Center Mechanism.

Entry conditions:

- Application is running.
- Network is up and stable.
- User1 is in a phone conversation with User2.
- User1 has placed User2 call on hold.

Exit conditions:

- User1 rescinds the invite or User2 acknowledges receipt of invitation.
- System displays Conference Center Menu.

Flow of events:

1. User1 opens the Conference Center window.
2. User1 places User3 number in the Conference Center dialog box. Use UC-1a User Initiates Call
3. User1 sends the Conference invite to User2.
4. User2 application acknowledges receipt of invite
5. User1 application displays "Invite Received."

Alternate Flows:

3a. If Teleconference Invite use UC-2b Telecon Invite.
If VTC invite use UC-2a VTC Invite.
3b. User1 rescinds invite
3c. Invite time out. User2 did not receive invite. Resend invite x 2.

Special Requirements:

1. No more than one VTC per user.

Use case: UC-2a VTC Invite
Primary Actor: User1
Other Actors: Data Network, User2

Stakeholders and Interest:

User wants an intuitive easy to use interface with the Conference Center Mechanism.

Entry conditions:

- Application is running.
- Network is up and stable.
- User1 is in a phone conversation with User2.
- User1 has placed User2 call on hold.

Exit conditions:

- User1 rescinds the invite or User2 acknowledges receipt of invitation.

- System displays Conference Center Menu.

Flow of events:

1. User1 opens the VTC window.
2. User1 sends the VTC invite to User2.

Alternate Flows:

2a. User1 rescinds invite.
2b. Invite time out. User2 did not receive invite. Resend invite x 2.

Special Requirements:

1. No more than one VTC per user.

Use case: UC-2b Telecon Invite
Primary Actor: User1
Other Actors: Data Network, User2
Stakeholders and Interest:
User wants an intuitive easy to use interface with the Teleconference Center Mechanism.

Entry conditions:
- Application is running.
- Network is up and stable.
- User1 is in a phone conversation with User2.
- User1 has placed User2 call on hold.

Exit conditions:
- User1 rescinds the invite or User3 acknowledges receipt of invitation.
- System displays Conference Center Menu.

Flow of events:

1. User1 opens the Teleconference window.
2. User1 Initiates call to User3, use UC-1a
3. User1 sends the Telecon invite to User3.

Alternate Flows:

2a. User1 rescinds invite.

2b. Invite time out. User3 did not receive invite. Resend invite x 2.
2c. User3 unavailable, application displays " Number not In Service" message.

Special Requirements:

1. No more than one VTC connection at a time per user.

Use case: UC-3 Conference Invitation Response
Primary Actor: User2
Other Actors: Data Network, User1
Description: Handles conference invitation responses for both Telecon and VTC Invites.

Stakeholders and Interest:

User wants a quick, intuitive, easy to use interface with the Conference Center Mechanism.

Entry conditions:

- Application is running.
- Network is up and stable.
- Conference invitation received

Exit conditions:

- User has joined the Conference.

Flow of events:

1. User application opens the Conference Center window.
2. User application displays "Conference Invite" message.
3. User accepts the Conference invite.
4. User joins the conference.

Alternate Flows:

3a. If User accepts Telecon Invite use UC-3a Accept Telecon Invite.
If User accepts VTC Invite use UC-3b Accept VTC Invite
If User declines Telecon Invite UC-3c Decline Telecon Invite

If User declines VTC Invite use UC-3d Decline VTC Invite

Special Requirements:

1. A User may being engaged in no more than one Conference at a time.

Use case: UC-3a Accept Telecon Invite
Primary Actor: User2
Other Actors: Data Network, User1
Description: Extends Conference Invitation Response

Stakeholders and Interest:

User wants a quick, intuitive, easy to use interface with the Telecon Conference Center Mechanism.

Entry conditions:

- Application is running.
- Network is up and stable.
- User2 is engaged in a call with User1
- Telecon invitation received

Exit conditions:

- User has joined the Conference.

Flow of events:

1. User accepts the Telecon invite.
2. The Conference Center window is displayed.
3. The application sends “VTC Invite Accepted” response.

Alternate Flows:

None.

Special Requirements:

1. The user may be engaged in no more than one Conference at a time.

Use case: UC-3b Accept VTC Invite

Primary Actor: User2
Other Actors: Data Network, User1
Description: Extends Conference Invitation Response

Stakeholders and Interest:

User wants a quick, intuitive, easy to use interface with the VTC Conference Center Mechanism.

Entry conditions:

- Application is running.
- Network is up and stable.
- User2 is engaged in a call with User1
- VTC invitation received

Exit conditions:

- User has joined the Conference.

Flow of events:

1. User accepts the VTC invite.
2. The VTC Screen window is displayed.
3. The application sends “VTC Invite Accepted” response.
4. The Web Cam is activated
5. The application establishes the VTC connection.
6. User conducts VTC Conference.

Alternate Flows:

None.

Special Requirements:

1. The user may being engaged in no more than one Conference at a time.

Use case: UC-3c Decline Conference Invite
Primary Actor: User2
Other Actors: Data Network, User1
Description: Extends Conference Invitation Response

Stakeholders and Interest:

User wants a quick, intuitive, easy to use interface with the Conference Center Mechanism.

Entry conditions:

- Application is running.
- Network is up and stable.
- User2 is engaged in a call with User1
- Conference invitation received

Exit conditions:

- User2 has declined the Conference Invite.
- Conference Center window closed.

Flow of events:

1. User2 declines the Telecon invite.
2. Application sends "Conference Invite Declined" message.
3. Application closes the Conference Center Window.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-4 User Terminates a Conference
Primary Actor: User
Other Actors: Data Network

Stakeholders and Interest:

User wants a quick, intuitive, easy to use mechanism for disconnecting from a conference.

Entry conditions:

- Application is running.
- Network is up and stable.
- Conference in progress

Exit conditions:

* User has disconnected from the Conference.
* Conference window is closed.

Flow of events:

1. User selects “terminate conference” button.
2. The application disconnects the user from the conference connection.

Alternate Flows:

2a. The application terminates Teleconference.
2b. The application terminates a VTC.
    1. VTC screen closed.
    2. Teleconference remains open.

Special Requirements:

None.

Use case: UC-5 Configure Settings
Primary Actor: User
Other Actors: None.

Stakeholders and Interest:

User wants a quick, intuitive, easy to use mechanism for configuring the application settings.

Entry conditions:

* Application is running.
* User is logged in

Exit conditions:

* User has configured applications settings.
* Settings window is closed.
* Main window is displayed.

Flow of events:

1. User opens the settings window.
2. User configures the application settings

Alternate Flows:

2a. User configures headset volume.
2b. User configures ringer settings.
2c. User configures the windows theme.

Special Requirements:

None.

Use case: UC-6 Query Directory
Primary Actor: User
Other Actors: None.

Stakeholders and Interest:

User wants a quick, intuitive, easy to use mechanism for querying the VoIPNET Directory.

Entry conditions:

- Application is running.
- Directory file downloaded from Network Manager.
- User logged in

Exit conditions:

- User has queried the directory.
- Queried number is displayed in Dial Center Dialog box.
- Directory window is closed.

Flow of events:

1. User opens the Directory window.
2. User selects alphanumeric category tab from Directory.
3. User finds the queried entry.
4. User selects the entry.
5. Selected number displayed in Dial Center dialog box.

Alternate Flows:

2a. User initiates a text search for the user.

Special Requirements:

None.

Use case: UC-7 Update VoIPNET Directory
Primary Actor: Admin User
Other Actors: None.

Stakeholders and Interest:

Admin User wants a quick, intuitive, easy to use mechanism for updating the VoIPNET Directory.

Entry conditions:

- Application is running.
- Directory file available.
- Admin User logged in

Exit conditions:

- Admin User has updated the directory repository.
- Update Directory window is closed.

Flow of events:

1. Admin User opens the Update Directory window.
2. Admin User selects the new Directory file.
3. Admin User distributes the new directory.
4. Admin User closes the Update Directory window.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-8 File Transfer
Primary Actor: User
Other Actors: User2, Data Network

Stakeholders and Interest:

User wants a quick, intuitive, easy to use mechanism for transferring files.

Entry conditions:

- Application is running.
- User logged in
- File to be transferred is available.

Exit conditions:

- User has transferred the file.
- File Center window is closed.

Flow of events:

1. User opens the File transfer window.
2. User selects the file for transfer.
3. User sends the file transfer request.
4. Receiver accepts file transfer request.
5. User transfers the file.
6. User closes the File Transfer window.

Alternate Flows:

5a. Receiver declines the file transfer request.
   1. File Transfer terminated.
   2. Application sends "File Transfer Declined" message.

Special Requirements:

None.

Use case: UC-9 Leave a Message
Primary Actor: User1
Other Actors: Data Network, User2
Description: Sub-use Case of UC-1a Initiate Call.

Stakeholders and Interest:

User wants an intuitive easy to use mechanism for leaving messages when a call is unable to be established.

Entry conditions:

- Application is running.
- Network is up and stable.
- A call was initiated by the user.

- The call was “busy.”
- The user is prompted to leave a text message.
- User1 not engaged in any type of connection.

Exit conditions:

- Message is left in the distant user mailbox.
- System displays Main Menu.

Flow of events:

1. User1 chooses to “leave message.”
2. The user1 types the message.
3. The user1 sends the message.
4. User2 message center saves message in the mailbox queue.
5. User2 application posts “new Message” alert icon in GUI.

Alternate Flows:

None.

Special Requirements:

None.

Use case: UC-10 Check Messages
Primary Actor: User1
Other Actors: Data Network
Description: The user checks the mailbox for any messages left for them.

Stakeholders and Interest:

User wants an intuitive easy to use mechanism for checking and managing incoming messages.

Entry conditions:

- Application is running.
- Network is up and stable.
- User not engaged in any type of connection.

Exit conditions:

- Messages have been heard, saved, or deleted.

- System displays Main Menu.

Flow of events:

1. User1 selects “check messages.”
2. Application displays the message in the text area.
3. User selects “save,” “delete,” or “next.”
4. Application removes “new messages” alert icon.

Alternate Flows:

3a. User “saves” message.
 1. The message is moved to the back of the mailbox queue and marked as “old.”

3b. User “deletes” message.
 2. The message is deleted and the next message is played.

3c. User selects “next” message.
 3. The user is prompted to “save,” “replay,” or delete the last message. The next message is played.

Special Requirements:

None.

Use case: UC-11 Login
Primary Actor: User1
Other Actors: None.
Description: User logs into application

Stakeholders and Interest:

User wants a standardized, familiar log in procedure

Entry conditions:

- Application is running.
- ACL is installed.

Exit conditions:

- User logged in.
- System displays Main Menu.

Flow of events:

1. User starts application.
2. Application displays login screen.
3. User enters user name and password.
4. Application verifies the account credentials.
5. Application displays Main menu.

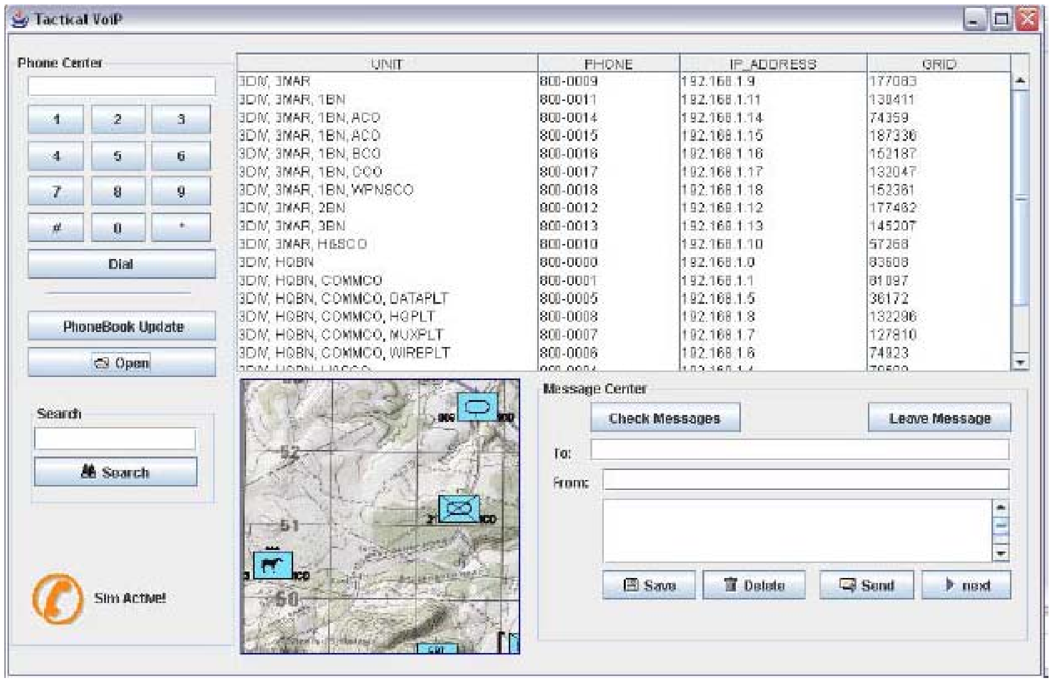
Alternate Flows:

3a. If the users supplies incorrect login credentials, the application will display the login screen again with error message.

Special Requirements:

None.

c. Prototype User Interface



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III. SOFTWARE RISK MANAGEMENT PLAN

A. RISK MANAGEMENT APPROACH

1. Overall Strategy

This section describes the high-level approach to risk management for this project. Risk Management will be a continuous activity on this project. A Risk Identification session will identify potential technical and non-technical risks. Each identified risk will be analyzed by likelihood of occurrence and impact. Each Risk will then be categorized and prioritized. Critical risks will have a contingency plan and a corresponding mitigation strategy developed. A monthly risk review will evaluate the status of identified risks. In addition, newly identified risks will be placed in the Risk Management Cycle (Texas Department of Information, 2007).

2. Roles Definition

The Matrix below identifies key roles within the project. Due to the size of the project all roles will be performed by a single person.

Risk Management Activity	**Project Manager**	**Requirements Engineer**	**Software Engineer**	**Test Engineer**	**Quality Control**	**Packaging Lead**	**Risk Manager**	
Develop and administer Risk Management Plan	P	S	S	S	S	S	J	
Determine if Risk Management Plan is ready for approval	S	S	S	S	S	S	P	

Risk Management Activity	**Project Manager**	**Requirements Engineer**	**Software Engineer**	**Test Engineer**	**Quality Control**	**Packaging Lead**	**Risk Manager**	
Assign and Coordinate Risk Management Activities	S	S	S	S	S	S	P	
Approve and authorize use of Contingency Plans	P	S	S	S	S	S	S	
Identify project risks	J	J	J	J	J	J	J	
Risk Analysis	P	S	S	S	S	S	J	
Risk Prioritization	P	S	S	S	S	S	S	
Contingency/Mitigation Plan Development	P	J	J	J	J	J	J	
Conduct Monthly Risk Review	J	S	S	S	S	S	J	
Modify Risk Management Plan	P	S	S	S	S	S	J	
Legend J = joint/shared responsibility P = primary/lead responsibility S = support/participatory responsibility								

Figure 12. Risk Roles Definition

B. RISK ASSESSMENT

1. Risk Identification

a. Methods and Techniques

Risk identification involves determining which risks might affect the project and documenting the characteristics of the risk. Risk identification will begin early in the planning phase and must continue throughout the life of the project. The following methods will be used to identify possible risks:

- Brainstorming
- Evaluations or inputs from project stakeholders
- Periodic reviews of project data
- Analysis of the Work Breakdown Structure (WBS)
- Risk Factor Checklist Templates

The process of risk identification is assisted by use of risk factor tables that capture indicators of commonly encountered risks. Risk factor tables are included in section D of this chapter. Each risk factor table is organized by categories with cues (characteristics) to help identify when the factor is considered low, medium, or high risk for the project. Risk factor tables will be used to prompt initial thoughts of risks for the project. Identify which risk factors are relevant to the project, and rate their potential for exposing risk to the project (low, medium, or high).

b. Project Risks

Identify and describe the project risks that will be used as a basis for risk analysis. Identify specific risks based on the defined methods and techniques.

Risk	**Risk Description**
The Application Prototype does not meet the Requirements.	The prototype is required to demonstrate basic call and VTC functionality. The inability to test these features renders the prototype useless.
The Requirements Analysis does not meet the user's needs.	The Requirements Analysis is critical and discrepancies here are costly to reengineer.
The Application Design Specification does not meet the Requirements as set forth in the Requirements Analysis Document.	The correct implementation of engineering solutions ensures that the product being built satisfies the User's Requirements and needs.
The Application Development Documentation does not meet the user's format requirements.	The Thesis requires a specific format for submission.
The Test Plan does not test a critical feature.	The test plan must thoroughly test all features for each release.
The Application Prototype is not ready on schedule.	A slippage in the delivery date of the prototype impacts the timely execution of the testing.
Testing Facility/Agency unavailable to support Operational Test & Evaluation.	The EPLRS radio is an asset held at MCTSSA. If they are unable to support then the application will not be tested on the target network.
Application does not perform as expected on the target network.	When the application is tested on the target networks it must communicate clearly and efficiently.
Loss of Data/Plan/Reports	The loss of electronic or hard copy materials.

Figure 13. Project Risks (Texas Department of Information, 2007).

2. Risk Analysis

a. Methods and Techniques

Describes how risks will be analyzed to establish the project exposure for each risk and to determine which risks are the most important ones to address. Risk analysis is the process of examining each risk to refine the risk description, isolate the cause, quantify the probability of occurrence, and determine the nature and impact of possible effects. The result of this process is a list of risks rated and prioritized according to their probability of occurrence, severity of impact, and relationship to other risk areas.

The process of risk analysis is assisted by determining the risk exposure (severity). The severity of a risk can be determined by multiplying the probability of the risk (event) actually occurring by the potential negative impact (consequence) to the project such as to cost, schedule, or performance. Risk analysis is assisted by use of a matrix that assigns risk ratings (very low, low moderate, high, very high) to risks based on combining probability and impact scales.

Once risks have been identified, and probability of occurrence and consequences assigned, the risk will be rated as to its severity. This facilitates ranking risks in priority order and deciding what level of resources to devote to each risk. Risk analysis is performed continuously throughout the life of the project as new risks are identified and as the profile of current risks change. Risks are analyzed and reviewed monthly. The decision to accept or reject a risk lies with the Project Manager.

Risk Probability Scale	Risk Impact Scale	RATING
.1	.1	Very Low
.3	.3	Low
.5	.5	Moderate
.7	.7	High
.9	.9	Very High

Figure 14. Risk Analysis Classification.

b. Risk Analysis and Prioritization

The rank and risk statement are provided for each project risk. The project risks are ranked in priority order based on the methods and techniques defined for risk analysis. The risk statement states clearly and concisely the context of the risk by identifying the event or condition (e.g., documentation format changes after documents are typed) and consequence (e.g., changes could extend project delivery completion date). Each risk statement is a refinement of the risk description defined during risk identification. The probability of occurrence, impact, and severity for each risk is identified.

Rank	Risk Statement		Probability (P), Impact (I), Severity (P*I)		
	Event	**Consequence**	**P**	**I**	**S**
1	Loss of Data/Plan/Reports	All documents must be recreated and testing redone.	.5	.9	.45
2	Testing Facility/Agency unavailable to support Operational Test & Evaluation.	OT&E testing will not be conducted.	.5	.7	.35
3	The Application Prototype is not ready on schedule.	Testing delayed/cancelled.	.5	.7	.35
4	Application does not perform as expected on the 100kbps network.	The requirements analysis must be started over.	.3	.9	.27
5	The Application Design Specification does not meet the Requirements as set forth in the Requirements Analysis Document.	The Design Specification document must be redone. The Prototype must be reengineered.	.2	.5	.10
6	The Application Development Documentation does not meet the user's format requirements.	The document must be reformatted.	.3	.3	.09

	Risk Statement		**Probability (P), Impact (I), Severity (P*I)**		
7	The Requirements Analysis does not meet the user's needs.	The Requirements engineering process must be done again. The Design Specification document must be redone. The Prototype must be reengineered.	.1	.7	.07
8	The Application Prototype does not meet the Requirements.	The Requirements analysis will have to be done again and the Prototype re-engineered.	.1	.5	.05
9	The Test Plan does not test a critical feature.	The test plan must be updated, and the testing redone.	.1	.3	.03

Figure 15. Risk Analysis.

3. Risk Response Actions

Risks may be addressed in different ways. Unique identifiers have been assigned to all risks. Actions have been identified and described (such as acceptance, transfer, avoidance, or mitigation) for how the project plans to provide the appropriate response strategies to address the risk events based on the level of prioritization defined. Only the highest ranked risk items will be included. Descriptions of these risk response actions follow:

• Accept the risk, with no investment of effort or cost. This is appropriate when the cost of mitigating exceeds the exposure, and the exposure is acceptable.

• Transfer the risk to someone else, or agree to share the risk. If a customer or partner is better able to handle the risk, this is probably the most effective approach.

• Avoid the risk by funding and staffing the efforts to reduce the probability that the risk will become a problem. Such mitigation tasks might include providing additional staff to help develop the product, getting special training for members of the team, or following a dual development path for the whole project.

• Mitigate the risk by funding and staffing the efforts to reduce the loss associated with the risk should it become a problem. Examples might include keeping a backup local area network (LAN) operational during the deployment of a new network.

• Establish contingency plans for significant risks that cannot be mitigated or otherwise resolved. Risk mitigation, the work required to handle the risk, may be small or significant; in either case, risk mitigation and costs assessment activities are included in the project schedule. Contingency management, the additional work required to handle the risk, must be budgeted and planned if the contingency event or condition occurs.

Risk Event	Application does not perform as expected on the 100kbps EPLRS network.
Risk ID	1
Risk Response Action	Conduct IOT&E and use a Network Emulator to test the application at 100kbps on a LAN.
Description	A software based network emulator can mimic network conditions that exist on the EPLRS network.
Trigger	None.
Assigned To	Capt Reiche.
Date	04/01

Risk Event	Loss of Data/Plan/Reports.
Risk ID	2
Risk Response Action	Avoid.
Description	Establish a SharePoint account for all Development Documents and Code. Conduct nightly uploads or as documents/code is changed. Backup w/ Jump Drive and Hard drive storage.
Trigger	None.
Assigned To	Capt Reiche.
Date	01/10

Risk Event	The Application Prototype is not ready on schedule.
Risk ID	3
Risk Response Action	Mitigate.
Description	Monthly reviews will be conducted to ensure that prototype development remains on schedule.
Trigger	Prototype review will identify schedule slippages.
Assigned To	Capt Reiche.
Date	01/24

Risk Event	The Application Prototype does not meet the Requirements.
Risk ID	4
Risk Response Action	Mitigate.
Description	Monthly reviews will be conducted to ensure that the features being implemented are validated/verified user requirements.
Trigger	None.
Assigned To	Capt Reiche.
Date	01/14

Risk Event	Testing Facility/Agency unavailable to support Operational Test & Evaluation.
Risk ID	5
Risk Response Action	Mitigate.
Description	An alternate test site will be coordinated.

Trigger	MCTSSA Testing Coordinator has not confirmed test date within 2 Months of Schedule test commencement.
Assigned To	Capt Reiche.
Date	01/14

Risk Event	The Requirements Analysis does not meet the user’s needs.
Risk ID	6
Risk Response Action	Avoid.
Description	A review will be conducted to ensure that the user’s needs as outlined in the Vision Document are reflected in the Requirements Specification Document.
Trigger	None.
Assigned To	Capt Reiche.
Date	01/14

Risk Event	The Application Design Specification does not meet the Requirements as set forth in the Requirements Analysis Document.
Risk ID	7
Risk Response Action	Avoid.
Description	A system Design review will be conducted as each feature is developed to ensure that it meets the requirements.
Trigger	None.
Assigned To	Capt Reiche.
Date	01/14

Risk Event	The Application Development Documentation does not meet the user’s format requirements.
Risk ID	8
Risk Response Action	Mitigate.
Description	The Thesis template will be utilized from the beginning of document development. In addition, a draft development document will be submitted for review 8 weeks prior to final delivery.
Trigger	Format review uncovers >10 errors.
Assigned To	Capt Reiche.
Date	01/14

Risk Event	The Test Plan does not test a critical feature.
Risk ID	9
Risk Response Action	Mitigate.
Description	A traceability Matrix will map user needs>Requirements>Features>Test Scenario.
Trigger	A feature is discovered during review that a test is not developed for.
Assigned To	Capt Reiche.
Date	01/14

Figure 16. Risk Response Actions.

C. RISK MONITORING AND CONTROL

1. Risk Tracking

Describes how the project team will determine if effective risk management is performed throughout the life of the project. Risk Checklists are provided as Framework supplemental tools in section D. The Risk Initiation Checklist identifies items to consider when checking if risk management has been established appropriately. The Risk Initiation Checklist will be conducted prior to the Risk Management Plan approval. The Risk Progress Checklist identifies items requiring routine consideration to ensure the project remains focused on risk management, and new risk are identified and tracked. The Risk Progress Checklist will be completed at each monthly review. The Risk Completion Checklist identifies items to consider when a project completes, or when a major phase completes, to evaluate the risk management process and results. The Risk Completion Checklist will be completed after each iteration of the Requirements, Design, Development, and Testing phases.

2. Risk Reporting

a. Risk Items

The Risk Item Report is used to provide a status on each of the top 5 ranked risk items that are assigned a mitigation strategy. Only the highest ranked risk items for mitigation are included and reviewed. See section D for the Risk Item Template.

b. Risk Status

A Risk Status report template is provided as a Framework supplemental tool in section D. The Risk Status report is used to provide a status of all project risks, including the rank for the current reporting period, the rank for the previous reporting period, number of times the risk is on the project status list, and a description of the mitigation progress. A status will be provided for each of the top 3 ranked risk items. Only the highest ranked risk items for mitigation are included and reviewed. This report provides summary information for the risks that are placed under a mitigation strategy as well as those that have be assigned an “accept,” “transfer,” or “avoid” strategy, as well as those that require contingency plans.

D. RISK MANAGEMENT SUPPLEMENTAL TOOLS

1. Risk Assessment Tools

Generic Software Project Risk Factors
(Texas Department of Information, 2007)

Generic Software Risk Factors	
Project Name	VoIPNET
Prepared By	C.P. Reiche
Date	01/15/07
Version	V1.0

Factor ID	**Risk Factors**	**Low Risk Cues**	**Medium Risk Cues**	**High Risk Cues**	**Low**	**Medium**	**High**	**Not applicable**	**Need info**	**TBD**	**Notes**
Mission and Goals											
1	Project Fit to Customer Organization	directly supports customer organization mission and/or goals	indirectly impacts one or more goals of customer	does not support or relate to customer organization mission or goals	x						
2	Project Fit to Provider Organization	directly supports provider organization mission and/or goals	indirectly impacts one or more goals of provider	does not support or relate to provider organization mission or goals	x						

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
3	Customer Perception	customer expects this organization to provide this product	organization is working on project in area not expected by customer	project is mismatch with prior products or services of this organization	x						
4	Work Flow	little or no change to work flow	will change some aspect or have small affect on work flow	significantly changes the work flow or method of organization	x						
Program Management											
5	Goals Conflict	goals of projects within the program are supportive of or complimentary to each other	goals of projects do not conflict, but provide little direct support	goals of projects are in conflict, either directly or indirectly	x						
6	Resource Conflict	projects within the program share resources without any conflict	projects within the program schedule resources carefully to avoid conflict	projects within the program often need the same resources at the same time (or compete for the same budget)		x					
7	Customer Conflict	multiple customers of the program have common needs	multiple customers of the program have different needs, but do not conflict	multiple customers of the program are trying to drive it in very different directions	x						
8	Leadership	program has active program manager who coordinates projects	program has person or team responsible for program, but unable to spend enough time to lead effectively	program has no leader, or program manager concept is not in use	x						
9	Program Manager Experience	program manager has deep experience in the domain	program manager has some experience in domain, is able to leverage subject matter experts	program manager is new to the domain		x					

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
10	Definition of the Program	program is well-defined, with a scope that is manageable by this organization	program is well-defined, but unlikely to be handled by this organization	program is not well-defined or carries conflicting objectives in the scope	x						
Decision Drivers											
11	Political Influences	no particular politically-driven choices being made	project has several politically motivated decisions, such as using a vendor selected for political reasons, rather than qualifications	project has a variety of political influences or most decisions are made behind closed doors	x						
12	Convenient Date	date for delivery has been set by reasonable project commitment process	date is being partially driven by need to meet marketing demo, trade show, or other mandate not related to technical estimate	date is being totally driven by need to meet marketing demo, trade show, or other mandate; little consideration of project team estimates	x						
13	Attractive Technology	technology selected has been in use for some time	project is being done in a sub-optimal way, to leverage the purchase or development of new technology	project is being done as a way to show a new technology or as an excuse to bring a new technology into the organization	x						
14	Short Term Solution	project meets short term need without serious compromise to long term outlook	project is focused on short-term solution to a problem, with little understanding of what is needed in the long term	project team has been explicitly directed to ignore the long term outlook and focus on completing the short term deliverable	x						
Organization Management											

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
15	Organization Stability	little or no change in management or structure expected	some management change or reorganization expected	management or organization structure is continually or rapidly changing	x						
16	Organization Roles and Responsibilities	individuals throughout the organization understand their own roles and responsibilities and those of others	individuals understand their own roles and responsibilities, but are unsure who is responsible for work outside their immediate group	many in the organization are unsure or unaware of who is responsible for many of the activities of the organization	x						
17	Policies and Standards	development policies and standards are defined and carefully followed	development policies and standards are in place, but are weak or not carefully followed	no policies or standards, or they are ill-defined and unused		x					
18	Management Support	strongly committed to success of project	some commitment, not total	little or no support	x						
19	Executive Involvement	visible and strong support	occasional support, provides help on issues when asked	no visible support; no help on unresolved issues	x						
20	Project Objectives	verifiable project objectives, reasonable requirements	some project objectives, measures may be questionable	no established project objectives or objectives are not measurable	x						
Customer/User											
21	User Involvement	users highly involved with project team, provide significant input	users play minor roles, moderate impact on system	minimal or no user involvement; little user input		x					
22	User Experience	users highly experienced in similar projects; have specific ideas of how needs can be met	users have experience with similar projects and have needs in mind	users have no previous experience with similar projects; unsure of how needs can be met		x					

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
23	User Acceptance	users accept concepts and details of system; process is in place for user approvals	users accept most of concepts and details of system; process in place for user approvals	users do not accept any concepts or design details of system	x						
24	User Training Needs	user training needs considered; training in progress or plan in place	user training needs considered; no training yet or training plan is in development	requirements not identified or not addressed	x						
25	User Justification	user justification complete, accurate, sound	user justification provided, complete with some questions about applicability	no satisfactory justification for system	x						
Project Parameters											
26	Project Size	small, non-complex, or easily decomposed	medium, moderate complexity, decomposable	large, highly complex, or not decomposable	x						
27	Hardware Constraints	little or no hardware-imposed constraints or single platform	some hardware-imposed constraints; several platforms	significant hardware-imposed constraints; multiple platforms	x						
28	Reusable Components	components available and compatible with approach	components available, but need some revision	components identified, need serious modification for use	x						
29	Supplied Components	components available and directly usable	components work under most circumstances	components known to fail in certain cases, likely to be late, or incompatible with parts of approach	x						
30	Budget Size	sufficient budget allocated	questionable budget allocated	doubtful budget is sufficient	x						

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
31	Budget Constraints	funds allocated without constraints	some questions about availability of funds	allocation in doubt or subject to change without notice	x						
32	Cost Controls	well established, in place	system in place, weak in areas	system lacking or nonexistent	x						
33	Delivery Commitment	stable commitment dates	some uncertain commitments	unstable, fluctuating commitments	x						
34	Development Schedule	team agrees that schedule is acceptable and can be met	team finds one phase of the plan to have a schedule that is too aggressive	team agrees that two or more phases of schedule are unlikely to be met	x						
Product Content											
35	Requirements Stability	little or no change expected to approved set (baseline)	some change expected against approved set	rapidly changing or no agreed-upon baseline	x						
36	Requirements Complete and Clear	all completely specified and clearly written	some requirements incomplete or unclear	some requirements only in the head of the customer	x						
37	Testability	product requirements easy to test, plans underway	parts of product hard to test, or minimal planning being done	most of product hard to test, or no test plans being made	x						
38	Design Difficulty	well defined interfaces; design well understood	unclear how to design, or aspects of design yet to be decided	interfaces not well defined or controlled; subject to change	x						
39	Implementation Difficulty	algorithms and design are reasonable for this team to implement	algorithms and/or design have elements somewhat difficult for this team to implement	algorithms and/or design have components this team will find very difficult to implement	x						

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
40	System Dependencies	clearly defined dependencies of the software effort and other parts of system (hardware, process changes, documentation, ...)	some elements of the system are well understood and planned; others are not yet comprehended	no clear plan or schedule for how the whole system will come together	x						
Deployment											
41	Hardware Resources for Deliverables	mature, growth capacity in system, flexible	available, some growth capacity	no growth capacity, inflexible	x						
42	Response or other Performance Factors	readily fits boundaries needed; analysis has been done	operates occasionally at boundaries	operates continuously at boundary levels	x						
43	Customer Service Impact	requires little change to customer service	requires minor changes to customer service	requires major changes to customer service approach or offerings	x						
44	Data Migration Required	little or no data to migrate	much data to migrate, but good descriptions available of structure and use	much data to migrate; several types of databases or no good descriptions of what is where	x						
45	Pilot Approach	pilot site (or team) available and interested in participating	pilot needs to be done with several sites (who are willing) or with one who needs much help	only available pilot sites are uncooperative or in crisis mode already	x						
46	External Hardware or Software Interfaces	little or no integration or interfaces needed	some integration or interfaces needed	extensive interfaces required		x					
Development Process											

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
47	Alternatives Analysis	analysis of alternatives complete, all considered, assumptions verifiable	analysis of alternatives complete, some assumptions questionable or alternatives not fully considered	analysis not completed, not all alternatives considered, or assumptions faulty	x						
48	Commitment Process	changes to commitments in scope, content, schedule are reviewed and approved by all involved	changes to commitments are communicated to all involved	changes to commitments are made without review or involvement of the team	x						
49	Quality Assurance Approach	QA system established, followed, effective	procedures established, but not well followed or effective	no QA process or established procedures	x						
50	Development Documentation	correct and available	some deficiencies, but available	nonexistent	x						
51	Use of Defined Engineering Process	development process in place, established, effective, followed by team	process established, but not followed or is ineffective	no formal process used	x						
52	Early Identification of Defects	peer reviews are incorporated throughout	peer reviews are used sporadically	team expects to find all defects with testing	x						
53	Defect Tracking	defect tracking defined, consistent, effective	defect tracking process defined, but inconsistently used	no process in place to track defects	x						
54	Change Control for Work Products	formal change control process in place, followed, effective	change control process in place, not followed or is ineffective	no change control process used	x						
Development Environment											

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
55	Physical Facilities	little or no modification needed	some modifications needed; some existent	major modifications needed, or facilities nonexistent	x						
56	Hardware Platform	stable, no changes expected, capacity is sufficient	some changes under evolution, but controlled	platform under development along with software	x						
57	Tools Availability	in place, documented, validated	available, validated, some development needed (or minimal documentation)	unvalidated, proprietary or major development needed; no documentation		x					
58	Vendor Support	complete support at reasonable price and in needed time frame	adequate support at contracted price, reasonable response time	little or no support, high cost, and/or poor response time							
60	Disaster Recovery	all areas following security guidelines; data backed up; disaster recovery system in place; procedures followed	some security measures in place; backups done; disaster recovery considered, but procedures lacking or not followed	no security measures in place; backup lacking; disaster recovery not considered	x						
Project Management											
61	PM Approach	product and process planning and monitoring in place	planning and monitoring need enhancement	weak or nonexistent planning and monitoring	x						
63	PM Experience	PM very experienced with similar projects	PM has moderate experience or has experience with different types of projects	PM has no experience with this type of project or is new to project management	x						
64	PM Attitude	strongly committed to success	willing to do what it takes	cares very little about project	x						

Factor ID	Risk Factors	Low Risk Cues	Medium Risk Cues	High Risk Cues	Low	Medium	High	Not applicable	Need info	TBD	Notes
65	PM Authority	has line management or official authority that enables project leadership effectiveness	is able to influence those elsewhere in the organization, based on personal relationships	has little authority from location in the organization structure and little personal power to influence decision-making and resources	x						
Technology											
76	Technology Match to Project	technology planned for project is good match to customers and problem	some of the planned technology is not well-suited to the problem or customer	selected technology is a poor match to the problem or customer	x						
77	Technology Experience of Project Team	good level of experience with technology	some experience with the technology	no experience with the technology		x					
78	Availability of Technology Expertise	technology experts readily available	experts available elsewhere in organization	will need to acquire help from outside the organization		x					
79	Maturity of Technology	technology has been in use in the industry for quite some time	technology is well understood in the industry	technology is leading edge, if not "bleeding edge" in nature	x						
Maintenance											
80	Design Complexity	structurally maintainable (low complexity measured or projected)	certain aspects difficult to maintain (medium complexity)	extremely difficult to maintain (high complexity)	x						
81	Support Personnel	in place, experienced, sufficient in number	missing some areas of expertise	significant discipline or expertise missing		x					
82	Vendor Support	complete support at reasonable price and in needed time frame	adequate support at contracted price, reasonable response time	little or no support, high cost, and/or poor response time	x						

2. Risk Monitoring and Control

RISK MANAGEMENT INITIATION CHECKLIST
(Texas Department of Information, 2007)

Risk Management Initiation Checklist	
Project Name	VoIPNET
Prepared By	C.P. Reiche
Date	01/15/07

ID	Yes/No	Items to be considered
Consider these when initiating the overall process		
1	n/a	Has funding been allocated to support a risk management?
2	Y	Have resources been assigned for risk identification?
3	Y	Are the following organizations represented on the risk identification team? Project Team Support Groups (SQA, CM, test, documentation, training, etc.) Representatives from other elements of the program, if the project is part of a larger program Partner or supplier representative User representative
4	Y	Has time been made available for the risk identification team to perform their tasks?
5	Y	Have risk factors been selected for use by the identification team? Have they included the following? General risk table (or one tailored to the organization) Specific risk factor table for this type of project Lessons learned on previous projects Use these items when reviewing the results of risk identification
6	Y	Has relevant risk factor been rated?
7	Y	For each factor rated high, has a specific risk statement been written?
8	Y	For each specific risk statement, have the conditions and consequences to the project been stated?
9	Y	Have the specific risks been organized into sets that support the analysis of impact and the development of mitigation actions?
10	Y	Have the risks been reviewed to determine which require further analysis?
Use these items when reviewing the results of analysis of specific risks		
11	Y	Has each risk statement been assigned a probability of occurrence?
12	Y	Has each risk statement been assigned an impact if risk occurs?
13	Y	Has the risk severity (probability x impact) been calculated for each risk statement?
14	Y	Have the risks been ranked in order of severity and agreed to by the team?
15	N	Have other project members and stakeholders reviewed and commented on the list?
16	N	Has the risk identification team reviewed and incorporated comments from other project members and stakeholders?

17	Y	With the risks as identified, should the project proceed as planned?
Use these items when reviewing the results of planning risk handling actions		
18	Y	Is there a mitigation action plan for each risk that is to be mitigated?
19	Y	For each risk to be mitigated, has an effort and/or cost been estimated for the mitigation action plan?
20	Y	Has a contingency plan been identified for the appropriate risks?
21	Y	Does the work breakdown structure for the project include risk management and mitigation actions?
22	Y	Have all the contingency plans been documented and do they include anticipated cost and effort?
23	Y	Has an agreement with management been made on when and if to authorize the use of a contingency plan?

Risk Management Progress Checklist
(Texas Department of Information, 2007)

Risk Management Progress Checklist	
Project Name	VoIPNET
Prepared By	C.P. Reiche
Date	1/15/07

ID	Yes/No	Items to be considered
1	Y	Is there a regular status review and update of key risks to assure they are under control?
2	Y	Is the Top Risk List reviewed and updated? (weekly, monthly, quarterly)
3	Y	Has the Top Risk List been disseminated to the appropriate people within the organization?
4	Y	For each scheduled risk mitigation action, is there progress in mitigating the risk as planned?
5	Y	For any risk exceeding defined trigger values, has the appropriate level of management approved the implementation of the contingency plan?
6	Y	Has any required risk status report been prepared for disseminating information at progress (and any other appropriate) reviews?
7	Y	Has the project schedule been undated to reflect the implementation of any approved risk contingency plans?
8	Y	Has the Project Team been reviewing the project for other risks that have appeared?
9	Y	Has the process to accept additional risks from project members and outside stakeholders been followed?

Risk Management Completion Checklists

(Texas Department of Information, 2007)

Risk Management Completion Checklist	
Project Name	VoIPNET Requirements Analysis
Prepared By	C.P. Reiche
Date	2/15/07

ID	Yes/No	Items to be considered
1	Y	Was it identified in the Project Plan when the effectiveness of a risk management process would be evaluated? (phase completion, periodically, project completed or terminated)
2	Y	Were review session(s) organized with appropriate people invited to attend?
3	Y	Were the results of the risk management activities reviewed? The results should have included at least the following: Risks that were detected initially and successfully handled Risks that were detected during the project, but not identified at the start Problems that arose during the project, but were not detected as risks at any point Cost and effort of the risk management activities Cost and effort of risk mitigation activities Cost and effort of contingency plans that were implemented
4	Y	Did the review session identify any implementation problems from the participants?
5	Y	Were any lessons for future risk management processes identified? Items of interest should have included: Mitigation activities that were effective Contingency actions that were successful Changes to the ineffective mitigation activities
6	N	Were changes identified to risk factors for use in the future? Items of interest should have included: New factors to include in the appropriate risk factor table Factors that can be removed from the table Changes in the cues provided in the chart for high, medium, and low risks
7	Y	Were the results of the analysis incorporated into risk factor tables and the risk management process?
8	n/a	Were the results of the analysis disseminated to other projects that were using the risk management process at that time?

Risk Management Completion Checklist	
Project Name	VoIPNET Design Phase
Prepared By	C.P. Reiche
Date	3/15/07

ID	Yes/No	Items to be considered
1	Y	Was it identified in the Project Plan when the effectiveness of a risk management process would be evaluated? (phase completion, periodically, project completed or terminated)
2	Y	Were review session(s) organized with appropriate people invited to attend?

3	Y	Were the results of the risk management activities reviewed? The results should have included at least the following: Risks that were detected initially and successfully handled Risks that were detected during the project, but not identified at the start Problems that arose during the project, but were not detected as risks at any point Cost and effort of the risk management activities Cost and effort of risk mitigation activities Cost and effort of contingency plans that were implemented
4	Y	Did the review session identify any implementation problems from the participants?
5	Y	Were any lessons for future risk management processes identified? Items of interest should have included: Mitigation activities that were effective Contingency actions that were successful Changes to the ineffective mitigation activities
6	N	Were changes identified to risk factors for use in the future? Items of interest should have included: New factors to include in the appropriate risk factor table Factors that can be removed from the table Changes in the cues provided in the chart for high, medium, and low risks
7	Y	Were the results of the analysis incorporated into risk factor tables and the risk management process?
8	n/a	Were the results of the analysis disseminated to other projects that were using the risk management process at that time?

Risk Management Completion Checklist	
Project Name	VoIPNET Prototype Development Phase
Prepared By	C.P. Reiche
Date	4/15/07

ID	Yes/No	Items to be considered
1	Y	Was it identified in the Project Plan when the effectiveness of a risk management process would be evaluated? (phase completion, periodically, project completed or terminated)
2	Y	Were review session(s) organized with appropriate people invited to attend?
3	Y	Were the results of the risk management activities reviewed? The results should have included at least the following: Risks that were detected initially and successfully handled Risks that were detected during the project, but not identified at the start Problems that arose during the project, but were not detected as risks at any point Cost and effort of the risk management activities Cost and effort of risk mitigation activities Cost and effort of contingency plans that were implemented
4	Y	Did the review session identify any implementation problems from the participants?
5	Y	Were any lessons for future risk management processes identified? Items of interest should have included: Mitigation activities that were effective Contingency actions that were successful Changes to the ineffective mitigation activities

6	N	Were changes identified to risk factors for use in the future? Items of interest should have included: New factors to include in the appropriate risk factor table Factors that can be removed from the table Changes in the cues provided in the chart for high, medium, and low risks
7	Y	Were the results of the analysis incorporated into risk factor tables and the risk management process?
8	n/a	Were the results of the analysis disseminated to other projects that were using the risk management process at that time?

Risk Management Completion Checklist	
Project Name	VoIPNET Prototype Testing Phase
Prepared By	C.P. Reiche
Date	5/20/07

ID	Yes/No	Items to be considered
1	Y	Was it identified in the Project Plan when the effectiveness of a risk management process would be evaluated? (phase completion, periodically, project completed or terminated)
2	Y	Were review session(s) organized with appropriate people invited to attend?
3	Y	Were the results of the risk management activities reviewed? The results should have included at least the following: Risks that were detected initially and successfully handled Risks that were detected during the project, but not identified at the start Problems that arose during the project, but were not detected as risks at any point Cost and effort of the risk management activities Cost and effort of risk mitigation activities Cost and effort of contingency plans that were implemented
4	Y	Did the review session identify any implementation problems from the participants?
5	Y	Were any lessons for future risk management processes identified? Items of interest should have included: Mitigation activities that were effective Contingency actions that were successful Changes to the ineffective mitigation activities
6	N	Were changes identified to risk factors for use in the future? Items of interest should have included: New factors to include in the appropriate risk factor table Factors that can be removed from the table Changes in the cues provided in the chart for high, medium, and low risks
7	Y	Were the results of the analysis incorporated into risk factor tables and the risk management process?
8	n/a	Were the results of the analysis disseminated to other projects that were using the risk management process at that time?

RISK ITEM REPORTS

(Texas Department of Information, 2007)

Risk Mitigation Reporting	
Project Name	VoIPNET

Prepared By	C.P. Reiche
Date	01/15/07

Risk Item Description	
Risk ID	4
Last Update	03/15/07
Current Rank in Top 3 Risks	2
Risk Statement Condition	The Application Prototype is not ready on schedule.
Risk Statement Consequence	The IOT&E testing will be delayed or cancelled.
Probability	.5
Impact	.7
Severity	.35
Original Rank	3
Current Mitigation Plan	Monthly Prototype reviews.
Plan Owner	C.P. Reiche
Date Mitigation Started	02/15/07
Date to Complete Mitigation	03/18/07
Mitigation Plan Status	Completed.
Trigger and Value for Contingency Plan	Prototype Development started.
Contingency Plan	Conduct monthly reviews of Prototype progress. Re-scope prototype as required.
Revision History	V1.0
Point of Contact	C.P. Reiche
Date Closed	03/15/07

Risk Mitigation Reporting	
Project Name	VoIPNET
Prepared By	C.P. Reiche
Date	01/15/07

Risk Item Description	
Risk ID	2
Last Update	03/15/07
Current Rank in Top 3 Risks	2
Risk Statement Condition	Loss of Data/Plans/Reports
Risk Statement Consequence	All documents must be recreated and testing redone.
Probability	.5

Impact	.9
Severity	.45
Original Rank	1
Current Mitigation Plan	Offsite storage and nightly builds.
Plan Owner	C.P. Reiche
Date Mitigation Started	01/15/07
Date to Complete Mitigation	06/18/07
Mitigation Plan Status	SharePoint account established. Jump Drive used for working Documents. Hard Drive local storage.
Trigger and Value for Contingency Plan	Requirements Analysis started.
Contingency Plan	All working documents and code are kept on a USB portable hard drive. Nightly uploads to the SharePoint of site storage. Semi-Daily uploads are made to a Local Hard drive.
Revision History	V1.0
Point of Contact	C.P. Reiche
Date Closed	04/15/07

Risk Mitigation Reporting	
Project Name	VoIPNET
Prepared By	C.P. Reiche
Date	01/15/07

Risk Item Description	
Risk ID	1
Last Update	03/15/07
Current Rank in Top 3 Risks	2
Risk Statement Condition	Testing Facility/Agency unavailable to support Operational Test & Evaluation.
Risk Statement Consequence	Testing will be delayed or cancelled.
Probability	.5
Impact	.7
Severity	.35
Original Rank	1
Current Mitigation Plan	Purchase of network emulator to mimic 100kbps network environment. Coordination of alternate test site.
Plan Owner	C.P. Reiche
Date Mitigation Started	03/15/07
Date to Complete Mitigation	04/18/07
Mitigation Plan Status	Network Emulator software on order. Pending return call from alternate test site.

Trigger and Value for Contingency Plan	First date slide from Testing agency.
Contingency Plan	Purchase network emulator to do 100Kbps testing. Contact I&I center in San Jose, CA to coordinate an alternate test site.
Revision History	V1.0
Point of Contact	C.P. Reiche
Date Closed	04/15/07

Risk Status Report
(Texas Department of Information, 2007)

Risk Status Report	
Project Name	**VoIPNET**
Prepared By	**C.P. Reiche**
Date	**04/23/07**

Rank	**Risk Statement**		**Rank This Time**	**Rank Last Time**	**# Times on List**	**Mitigation Progress**
	Condition	**Consequence**				
1	Application does not perform as expected on the 100kbps network.	Project Failure.	2	2	6	In progress
2	Loss of Data/Plan/Reports	Project Failure. All Documentation must be recreated. Data lost.	3	3	6	Completed
3	The Application Prototype is not ready on schedule.	Testing delayed/cancelled.	1	1	4	In progress

IV. SOFTWARE DESIGN SPECIFICATIONS

A. INTRODUCTION

1. Purpose

The purpose of this design document in this first iteration is to provide sufficient documentation to produce a working prototype and test it against the system requirements document

2. Application Scope

The intended functionality of VoIPNET is defined in Section II. The scope of the first iteration prototype is to model the interface and make a simple User Agent (UA)–to- UA call.

3. Definitions, Acronyms, and Abbreviations

An updated copy of definitions, acronyms, and abbreviations are maintained in Section I. paragraph h.

4. References

[1] “VoIPNET Project Proposal.” Reiche.

[2] “VoIPNET Requirements Specification Section I.” Reiche.

[3] “Software Design: From Programming to Architecture.” Braude.

[4] “MJSip Mini-Tutorial version 0.1..” Veltri.

5. VoIPNET Design Specification Overview

This section provides detailed technical data, system information, and other relevant information to VoIPNET’s development. This document includes an architecture diagram, a design class diagram, interaction diagrams, state diagrams, operation contracts and a Domain Model.

B. SYSTEM ARCHITECTURE

VoIPNET is organized into a 3-tier closed architecture composed of a presentation layer, logic layer and a services layer. This organization is intended to provide modularity and minimal manipulation of code should updates be required.

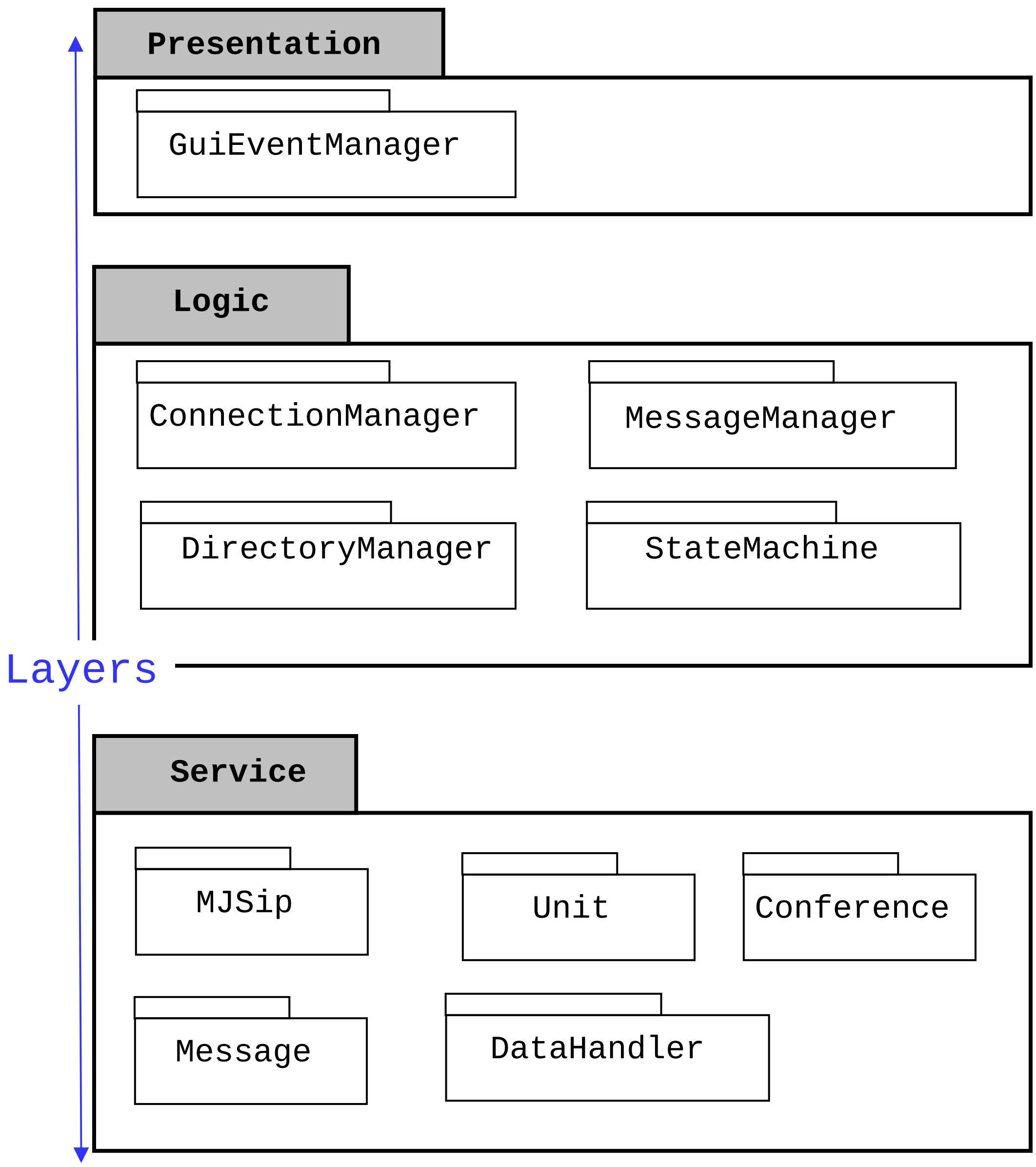

Figure 17. VoIPNET Architecture

1. Architecture Subsystems

The presentation layer is represented by the common graphical user interface (GUI) the GuiEventHandler class, The Logic layer is composed of the ConnectionManager,

MessageManager, and DirectoryManager classes. The Services

Layer is composed of the MJSip, Unit, Directory, Conference and Message classes.

a. Presentation Layer

The GuiEventHandler class is one frame with an array of content panes for VTC viewing, message center operations, dial center functionality and directory services. GuiEventHandler calls the appropriate logic class when an event is detected.

b. Logic Layer

The Logic Layer provides the appropriate functionality for each GuiEventHandler event. Each event calls the appropriate logic Class, which in turns invokes the required services to perform specific tasks or functions.

The ConnectionManager class initiates and receives voice calls from or to other UA’s. It calls the appropriate service level classes to initiate, terminate, and conduct a call. It also receives requests from both Service and Presentation Layers. It calls the appropriate service level classes in response to UA’s conference invitation or conference invitation response. In addition, the ConnectionManager Class communicates with the Presentation Layer to activate the appropriate graphical interface for VTC or Telecon calls.

The MessageManager class utilizes the appropriate service level classes to create, save and delete a text message for UA’s.

The DirectoryManager class receives requests from both the Presentation and Service Layers in response to directory queries. The class passes directory information to the Presentation Layer for graphical presentation. In addition, it creates, deletes, and updates Unit information in the Common Distributed Directory.

c. Service Layer

The service layer is the heart of the application. It contains several classes that provide functionality to the Logic Layer classes.

The MJSip Package is a combined API and SIP protocol stack implementation (Veltri 2005). It is a four-tier architecture that provides Call, Dialog, Transaction, and Transport services/management.

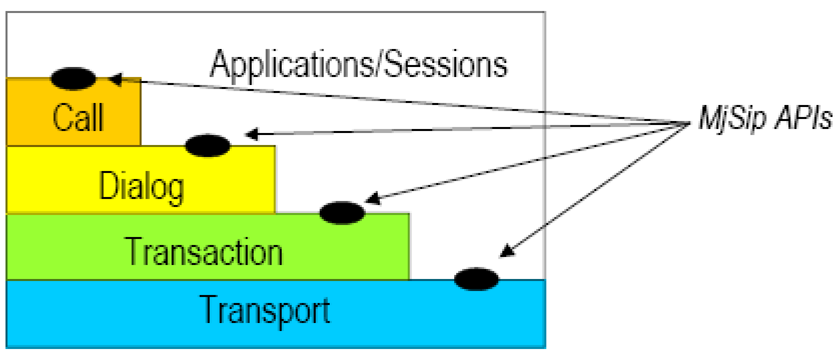

Figure 18. MJSip Architecture (Veltri 2005).

The *Transport* layer is the lowest layer in the MJSip stack. The SipProvider is the MJSip Object that provides the transport service to all upper and lower layers. It is also responsible for the de-multiplexing and direction of all incoming messages toward the appropriate upper layer entity. All SIP elements must use the

SipProvider’s API if they require access to the MJSip transport service (Veltri 2005).

The second layer is the *Transaction* layer. In SIP a transaction is a request sent by a client (a transaction client) to a transaction server along with all responses to that request sent from the transaction server back to the client. The transaction layer handles upper-layer retransmissions, matching of responses to requests, and timeouts. The Transaction layer sends and receives messages through the transport layer (Veltri 2005).

In SIP any task that a user agent client (UAC) accomplishes takes place using a series of transactions. As already introduced, the transaction layer has a client component (referred to as a transaction client) and a server component (referred to as a transaction server), each of which are constructed to process a particular request. There are two kinds of transactions:

two-way transactions, and *three-way transactions*

The third layer (above the transaction layer) is the *Dialog* layer that binds different transactions within the same “session.” A dialog is a peer-to-peer SIP relationship between two user agents that persists for some time. The dialog facilitates sequencing of messages and proper routing of requests between the user agents (Veltri 2005).

The upper SIP-layer is the *Call Control* layer and it implements a complete SIP call. The Call Control layer is implemented via the Call API. The Call API offers a simple-to-use interface for handling incoming and outgoing

SIP calls. The MJSip APIs for the four layers are implemented respectively by:

- class Call (and class ExtendedCall)
- class InviteDialog
- classes ClientTransaction, ServerTransaction, InviteClientTransaction, and InviteServerTransaction
- class SipProvider

The Unit class is a fundamental object that contains Directory information for each unit. It contains information regarding name, IP address, and telephone number.

The Message class is a fundamental object containing the message contents. It is created by and passes its contents to the Logic layer (MessageManager class).

The Conference class is the abstract class for the creation and management of the conference functionality. The class opens and closes required media streams and makes appropriate updates to the GuiEventHandler and StateMachine.

C. OBJECT / CLASS DESCRIPTION

Figure 12 shows the VoIPNET Domain Model. It illustrates the basic context in which the VoIPNET system operates. The user can initiate and receive calls, VTC’s and Teleconferences via the GUI. The user also has the ability to search the directory for Unit information and leave messages for unreachable UA’s. The admin user has the ability to update and distribute the Directory file.

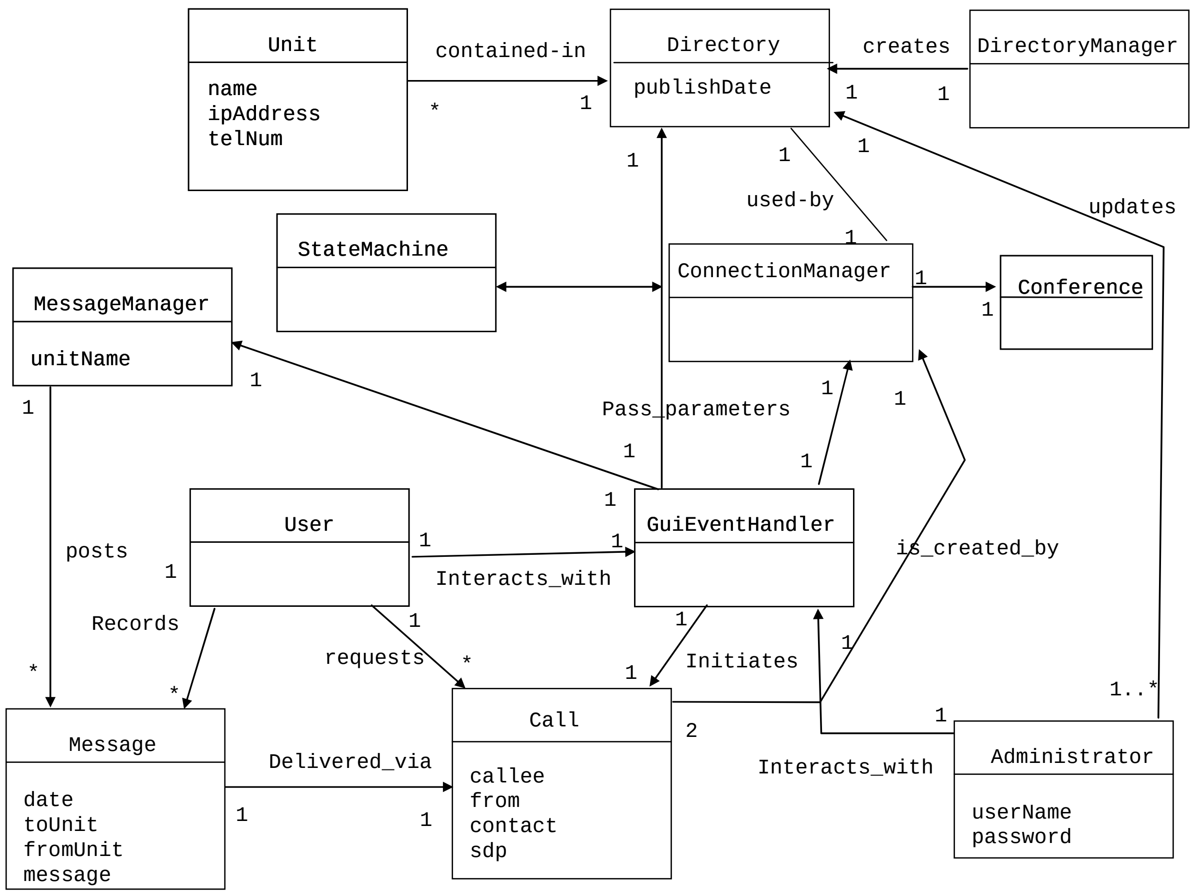

Figure 19. VoIPNET Domain Model.

1. GuiEventHandler Class Description

The GuiEventHandler class is the user interface class located in the presentation layer. Its main function is to provide the user with the correct interface when options are selected.

a. Attribute Descriptions

JPanel[]: phoneCenterPanel

Holds the dialing text fields and buttons.

Image[]: statusIcon

A flashing icon that alerts the user when a call, Telecon, or VTC is in progress.

JButton[]: JButton0, JButton1…JButton9, JButton#, JButton*

Phone dialing buttons for manual input of Telephone Numbers.

JButton[]: dialButton

Establishes a connection by searching the directory by Telephone number and initiates a call to the corresponding IP address of the user. Used for telephone number search only!

JButton[]: redialButton

Sends a message to the Connection Manager to redial the last number called.

JButton[]: muteButton

Sends a message to the system to “Mute” the microphone.

JButton[]: volumeButton

Sends a message to the system to adjust the earpiece/speaker volume.

JButton[]: phoneBookUpdateButton

Compares the timestamp of the current Directory with the Timestamp of other Directories on the network. If the current directory is older it requests an update from that UA.

JField[]: NumField

Telephone input text field. Number buttons pushed will place digit text into this field.

JPanel[]: messageCenterPanel

Contains the message center buttons and text fields.

JField[]: fromfield

Contains the logged in user’s user name when a message is sent from the message center.

JLabel[]: fromLabel

Labels the “From” JField.

JField[]: tofield

Contains the logged in user’s user name when a message is sent from the message center.

JLabel[]: toLabel

Labels the “To” JField.

JButton[]: checkMessageButton

Checks the Message Center for messages left for the user whom is currently logged in.

JButton[]: leaveMessageButton

Places the logged in user name into the “From” field and activates the message text area.

JButton[]: deleteButton

Deletes the current message from the Message Center Mailbox.

JButton[]: saveButton

Saves the message to the back of the logged in user’s mailbox queue.

JButton[]: sendButton

If the distant UA is “Busy” it posts the message into the mailbox of the “To” UA. If the distant UA is “unreachable” then the message is added to the “retransmit queue,” and resent to the user based on message priority.

JButton[]: nextButton

Advances the current message to the next message in the user’s message center mailbox queue.

JCheckbox[]: highPriority

Sets priority for message retransmission attempts to 240 at a one (1) minute interval.

JCheckbox[]: lowPriority

Sets priority for message retransmission attempts to 24 at a ten (10) minute interval.

JCheckbox[]: medPriority

Sets priority for message retransmission attempts to 48 at a five (5) minute interval.

JTextArea[]: messageTextArea

Text area for message.

JPanel[]: unitDirectoryPanel

Contains the table with the Directory information.

JTable[]: unitDirectoryTable

Contains the directory fields, Unit Name, PhoneNumber, and IP address. It is scrollable and selecting an entry initiates the dialing sequence for that unit.

JPanel[]: searchPanel

Contains the search text field and search button.

JField[]: searchField

Text box for entering unit name to search for.

JButton[]: searchButton

Executes the search on the searchField entry. On success it highlights the entry in the directory. User is notified of failure.

JPanel[]: VTCPanel

Contains the VTCDisplay, VTCConnectButton, and the VTCDisconnectButton.

JFrame[]: VTCDisplay

Displays the video for the VTC else displays the Unit crest of the current call.

JButton[]: VTCConnectButton

Sends a “VTC Connect” message and the current call IP address to the Conference Manager to initiate a VTC invite to the current call.

JButton[]: VTCDisconnectButton

Sends a “Disconnect” message and the current call IP address to the Conference Manager to terminate the VTC.

DirectoryManager[]: d

Instance created in GUIEventHandler to access methods of this class.

TableModel[]: model

Creates the model for the unitDirectoryTable.

String[]: searchValue

String value of the searchField.

Unit[]: unit

Basic Directory Object.

String[]: messageTo, MessageFrom, messageText

String values for Message Center “To, From, Message” fields.

MessageManager[]: manager

Instance created in GUIEventHandler to access methods of this class.

Message[]: message

Basic message manager object. Contains the “To, From, and Text attributes.

String[]: loginName

String representation of the user login name. Passed to password module.

b. Method Descriptions

private main(): void

Main method of the program.

public statusIcon(): void

Updates the status Icon to reflect the current state of the application.

private login(): void

Executes the authentication process for the application. When successful the logged-in state is updated in the instance of StateMachine.

private initialize(): void

Calls login(), creates instances of StateMachine, ConnectionManager, DirectoryManager and MessageManager.

2. ConnectionManager Class Description

The ConnectionManager class is the communications handler class located in the logic layer. Its function is to apply program logic to the application, based on connection service requests from the Presentation Layer. In addition, specific service layer requests must be evaluated by the ConnectionManager class’s collection of business rules. This class initiates, monitors, and terminates Call, VTC, and Telecon connections made on behalf of the GuiEventManager (User).

a. Attribute Descriptions

String: lastNumberDialed
Stores the last number dialed for the redial function.

String: state

Stores the current state as retrieved from the StateMachine.

b. Method Descriptions

public void makeCall()

Initiates a call on behalf of the user. It pulls state from the StateMachine, initiates the search for the corresponding IP address, updates StateMachine, initiates the connection, and updates the status Icon.

public string getLastNumber()

When re-dial is called it accesses the lastNumber variable and returns the number to the caller.

public void hangup()

Initiates the termination process on behalf of the user. It retrieves the State, terminates the call/VTC/Telecon, updates State, and updates the status Icon.

public void incomingCall()

Initiates the incoming call handling process. It retrieves State, assigns a call identification number, and returns the users intent to the caller, and updates the status Icon.

private assignCallID()

Creates a timestamp based ID number for each call.

public void autoAccept()

Routes the caller to the users Message Center mailbox, prompts to leave a message, terminates the call when the message is completed, and updates the new message icon.

public void initiateConference()

Starts the conference invite process on behalf of the user. Retrieves state, if a conference is already in progress it throws a conference rule violation method.

private ruleViolation()

This method is executed when a business rule is violated. For example; attempting to simultaneously engage in more than on VTC or Teleconference or attempting to engage in a VTC prior to establishing a call.

public incomingConferenceInvite()

This method initiates the process for accepting or declining a conference invitation. It also retrieves state, executes the autoDecline method (if engaged in a conference), and initiates the appropriate VTC/Telecon invite method.

public terminateVTC()

Disconnects the user from an established VTC, WITHOUT disconnecting the call itself. Destroys the instance of VTC, closes the VTC multimedia window, sets the state to call in progress, and updates the status icon.

public terminateTelecon()

Disconnects the user from an established teleconference. Destroys the instance of Telecon, disconnects the call, sets the state, and updates the status icon.

private autoDecline()

Automatically declines an invitation due to a conference rule violation.

public vtcInvite()

This method processes an incoming VTC invitation. It retrieves state, displays a JOptionDialogBox for user input regarding accept or decline VTC. If user accepts then state is set, the Local SDD is retrieved, an instance of Telecon is created and the user conducts the video teleconference.

public teleconInvite()

This method processes an incoming teleconference invitation. It retrieves state, displays a JOptionDialogBox for user input regarding accept or decline Telecon. If user accepts then state is set, the Local SDD is retrieved, an instance of Telecon is created, the status icon is updated and the user conducts the teleconference.

3. StateMachine Class Description

The StateMachine class is the primary tool for the business rules enforced in the ConnectionManager class. This class is composed Boolean variables for each state and corresponding accessor/mutator methods (set/get) for the state of the application.

a. Attribute Descriptions

private Boolean[]: shuttingDown

Application is closing.

private Boolean[]: establishingConnection

Call initiation in progress.

private Boolean[]: loggingIn

User in process of logging in to application.

private Boolean[]: loggedIn

User has successfully logged in to application.

private Boolean[]: initializing

Application is starting up, prior to initiation of logic layer classes.

private Boolean[]: stateMachineCreated

Logic layer class is initialized.

private Boolean[]: ready

Application is in listen mode and ready to initiate, receive calls.

private Boolean[]: connectionFailed

The attempted connection has failed.

private Boolean[]: connectionEstablished

The attempted connection has succeeded.

private Boolean[]: establishingTelecon

Application is creating an instance of the conference and opening the required multimedia streams.

private Boolean[]: teleconInProgress

User currently engaged in a teleconference.

private Boolean[]: teleConFailed

Attempt to establish a teleconference has failed.

private Boolean[]: callTerminated

Current call has been closed.

private Boolean[]: msgInProgress

Caller is leaving a message in the callee’s mailbox.

private Boolean[]: establishingCall

User Agent is attempting to coordinate a call with the callee.

private Boolean[]: callFailed

User Agent was unable to establish a call.

private Boolean[]: callInProgress

User is engaged in a call.

private Boolean[]: establishingVTC

Application is in the process of creating, inviting and connecting a video teleconference.

private Boolean[]: vtcInProgress

User is engaged in a video teleconference.

private Boolean[]: vtcFailed

Application failed to establish the video teleconference.

private Boolean[]: vtcTerminated

User/Application has closed the current video teleconference.

private Boolean[]: teleconTerminated

User/Application has closed the current teleconference.

b. Method Descriptions

public getState(): String

Returns the current state of the application.

public setState(String: state)

Sets the current state of the application.

4. MessageManager Class Description

The MessageManager class is the logic layer class responsible for the management of the message handling procedures. It receives messages from the GUIEventHandler, retrieves state from StateMachine, and creates/checks/deletes/saves messages to the users or callees mailbox.

a. Attribute Descriptions

private Object[]: message

Fundamental Message class object.

private Object[]: unit

Instance of Unit class.

private Object[]: data

Instance of DataHandler class

private String[]: inputFrom

Holds the name of the Unit that the message is from.

private String[]: inputTo

Holds the name of the Unit the message is for.

private String[]: unitName

Name of the unit for searching the appropriate data structure.

private String[]: inputText

Field for holding the text to be left in the message.

private GuiEventHandler[]: g

Instance of GUIEventHandler.

b. Method Descriptions

public checkMessage(): void

Accesses the mailbox of the user and plays the first message in the mailbox.

public nextMessage(): void

Advance to and plays the next message in the users mailbox.

public deleteMessage(): void

Deletes the current message in the user’s mailbox.

public leaveMessage(): void

Places the “To:” and “From” information into the text fields and sends a message to the user’s mailbox.

public saveMessage(): void

Saves the last message and places it at the end of the message queue in the mailbox.

5. DirectoryManager Class Description

The DirectoryManager class is logic layer class for creating, manipulating, searching, and modifying the directory. The Directory File I/O is handled here and several TreeMaps are utilized to expedite searches based on multiple criteria. The class receives search parameters from the GuiEventHandler, retrieves state from the StateMachine, and retrieves unit information from the instances of Unit.

a. Attribute Descriptions

private BufferedReader[]: bufReader

For reading the contents of the fileReader.

private File[]: inFile

Variable for the name of the directory file.

private FileReader[]: fileReader

Variable for reading the inFile.

private int[]: status

JFileChooser input status variable.

private int[]: count

Counts the lines read from file into the unitArray.

private int[]: fileSize

Control variable for iteration while building the unitArray.

private String[]: str

Variable for storing the contents of the line from bufReader.

private String[]: unitName

Variable to hold unit name and place into the appropriate unit object in the unitArray.

private String[]: unitTel

Variable to hold unit telephone number and place into the appropriate unit object in the unitArray.

private String[]: unitIP

Variable to hold unit IP address and place into the appropriate unit object in the unitArray.

private String[]: dialNumber

Holds the number/URL to be dialed by the application.

private TableModel[]: model

Sets the format for the display of the Directory.

private JFrame[]: frame

Contains the directory.

private StringBuffer[]: document

Assist in the opening and reading of the directory file.

private Unit[]: unit

Object created when read from the directory file and placed into the unit Array.

private GuiEventHandler[]: g

Instance of g passed to the constructor of the DirectoryManager to give access to presentation layer methods.

private String[][]: unitArray

Array created to hold the unit objects that are created from the directory file.

private Unit[]: searchResult

Unit object returned when a search is successful.

private JFileChoser[]: chooser

Selects the directory file to open with the user agent.

public TreeMap[]: dialMap

Maps a unit telephone number to its unit name.

public TreeMap[]: phoneNum

Maps a unit’s name to its telephone number

public TreeMap[]: URL

Maps a unit’s name to its URL.

public TreeMap[]: unitMap

Maps a unit name to its Unit object.

public Boolean[]: treeSet

Tracks the status of the creation of the directory table and the creation of the TreeMaps.

public Final Static String[]: HEADER

Header for the directory table.

b. Method Descriptions

public open()

Initiates the open resource method to open the directory file and calls refreshList() to repaint the director table in the GuiEventHandler

public openResource()

Opens the directory file.

public refreshList()

Passes the model parameters for the Directory table to the GuiEventHandler when a directory update occurs.

public createTree()

Creates an unsorted array (unitArray) from the inFile that is opened as the directory file, creates the Unit objects for each directory entry, and creates the TreeMaps for searches.

public displayDirectory()

Refreshes the directory table in the GuiEventHandler.

public search()

Searches the directory for the searchValue and if found displays the telephone number or URL in the numberDisplay field of the GuiEventHandler.

6. MJSip Class Description

The MJSip class implements the call, transaction, dialog, and transport services for the application. Access to these services is provided through the UserAgent class of the MJSip package.

a. Attribute Descriptions

Log: log

Event logger.

protected UserAgentProfile: user_profile

Variable for the UserAgentProfile

protected SipProvider sip_provider

Variable for the SipProvider

protected ExtendedCall: call

Fundamental functionality is derived from this class instance.

protected MediaLauncher: audio_app

Audio application variable.

protected MediaLauncher: video_app

Video application

protected String: local_session

Local SDP.

protected UserAgentListener: listener

Variable for the required instance of UserAgentListener.

final String: MEDIA_PATH

Sets the path for media needed by the User Agent.

final String: CLIP_ON

Appends the MEDIA_Path + the file name for the on.wav file.

final String CLIP_OFF

Appends the MEDIA_Path + the file name for the off.wav.

final String: CLIP_RING

Appends the MEDIA_Path + the file name for the ring.wav file.

AudioClipPlayer clip_ring

Sets the ring sound.

AudioClipPlayer: clip_on;

Sets the on sound.

AudioClipPlayer: clip_off

Sets the off sound.

b. Method Descriptions

public call(): void

Makes a new call

public accept(): void

Accepts an incoming call.

public hangUp(): void

Closes an ongoing, incoming, or pending call.

public getSessionDescriptor(): void

Gets the local SDP.

public getStatus(): void

Returns the status of the call.

public listen(): void

Waits for an incoming call

public printLog(): void

Prints events to the log.

public setLocalSessionDescriptor(): void

Sets the local SDP.

7. Unit Class Description

The Unit class is the fundamental information object in the application. Each Unit object contains information to aid identification and communication with each respective unit.

a. Attribute Descriptions

String[]: unitName

The unit’s name.

String[]: unitTel

The unit’s telephone number.

String[]: unitURL

The unit’s URL.

Boolean[]: isBusy

LinkedList[]: messageQueue The unit’s mailbox.

b. Method Descriptions

public getURL(): String

Returns the unit’s URL.

public getMessageQueue(): LinkedList

Returns the unit’s mailbox.

public getName(): String

Returns the unit’s name.

public getStatus(): Boolean

Returns the unit’s call status.

public getTel(): String

Returns the unit’s telephone number.

public setStatus(): void

Sets the unit’s call status.

8. Message Class Description

The Message class is the fundamental object of the MessageCenter operation. It contains the addressing and content elements of each message. Each instance of Message is stored in the corresponding mailbox(messageQueue) for the respective unit.

a. Attribute Descriptions

String[]: from

Whom the message is from.

String[]: to

Whom the message is to.

String[]: text

Content of the message.

b. Method Descriptions

public getTo(): String

Returns the contents of the To variable.

public getFrom(): String

Returns the contents of the "From" variable.

public getText(): String

Returns the contents of the text variable.

9. Conference Class Description

The Conference class is an abstract class for the creation and management of user conferences.

a. Attribute Descriptions

String[]: conferenceID

Unique identification number for each conference.

String[]: sdp1

Session Descriptor for conference participant 1

String[]: sdp2

Session Descriptor for conference participant 2.

b. Method Descriptions

public invite(): String

Sends an invitation to a caller on behalf of the user.

public displayInviteMessage(): void

Displays the invitation and solicits an accept/decline option.

public terminateConference(conferenceID): void

Terminates the requested conference.

public cancelConference(conferenceID):

Cancels the conference request prior to connection.

10. VTC Class Description

The class extends the Conference class. It requires an additional multimedia stream as well as synchronization between the audio and video streams.

a. Attribute Descriptions

Boolean[]: vtcStatus

Maintains the status of video connection.

VICLauncher[]: media_app

Video media object.

b. Method Descriptions

public cancelConference()

Cancels the VTC and destructs the instance.

11. Telecon Class Description

The Telecon class extends the Conference class.

a. Attribute Descriptions

None.

b. Method Descriptions

public leaveConference()

Disconnects the user from the current conference, but does not destroy the conference object. This allows other connected users to remain in conference.

D. BOUNDARY USE CASE

Use case: UC-12 Application Startup

Primary Actor: User

Other Actors: VoIPNET Application

Stakeholders and Interest:
User wants the application to initialize quickly and without error.

Entry conditions:
- Application is installed on the hardware

Exit conditions:
- Application GUI is displayed
- Application is listening for incoming call requests

Flow of events:

1. The application initializes.
2. The application initializes the StateMachine class
3. The application displays the login screen.
4. The user logs in.
5. The application initializes the Logic layer classes
6. The application displays the GUI

Alternate Flows:

5.a. User logs in incorrectly.
    1. Application displays login screen

Special Requirements:

None.

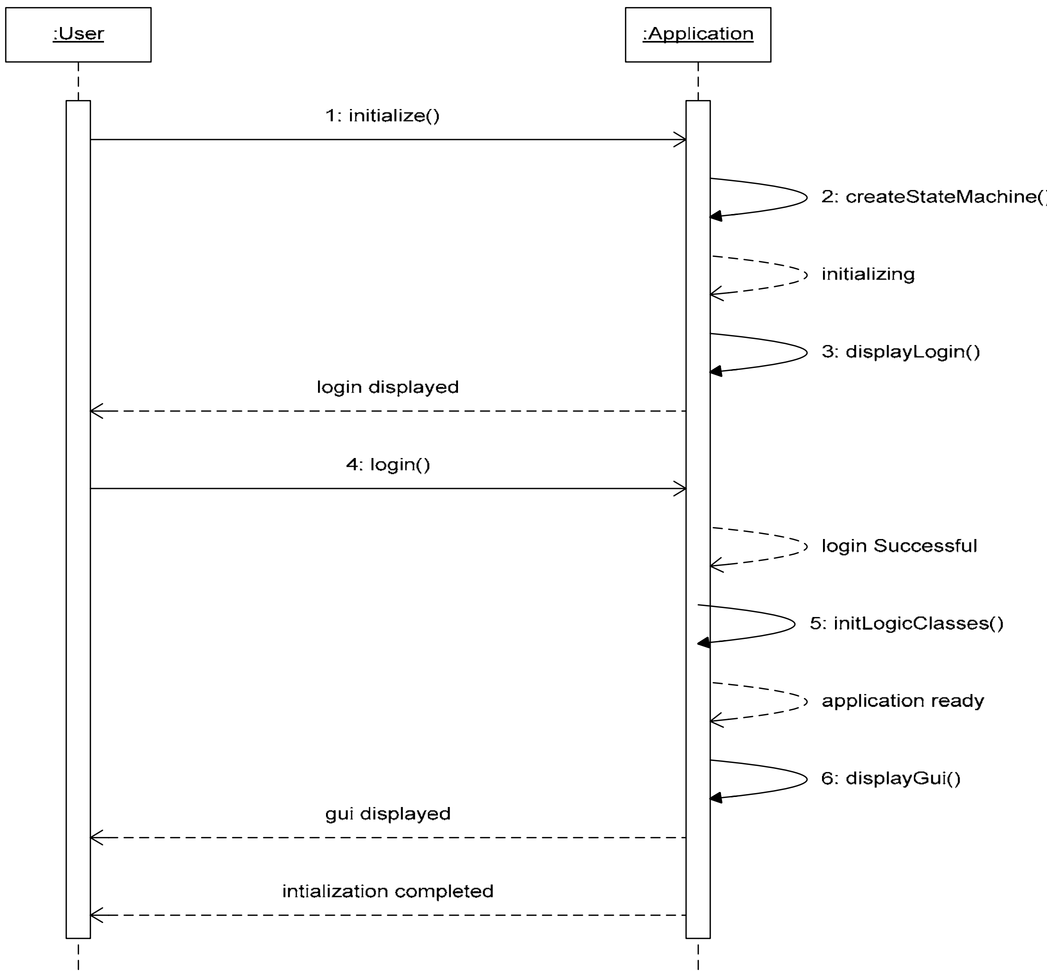

Figure 20. VoIPNET Start up Sequence Diagram.

Contract: C12: start
Operation: start()
Cross Reference: UC-12: Application Startup

Preconditions: None.

Postconditions:

1. A new instance ***s*** of StateMachine was created.
2. A new instance ***g*** of GuiEventHandler was created and displayed.
3. A new instance ***d*** of DataHandler was created.
4. A new instance ***c*** of ConnectionManager was created.
5. The application ready to process calls.

E. CLASS DIAGRAMS

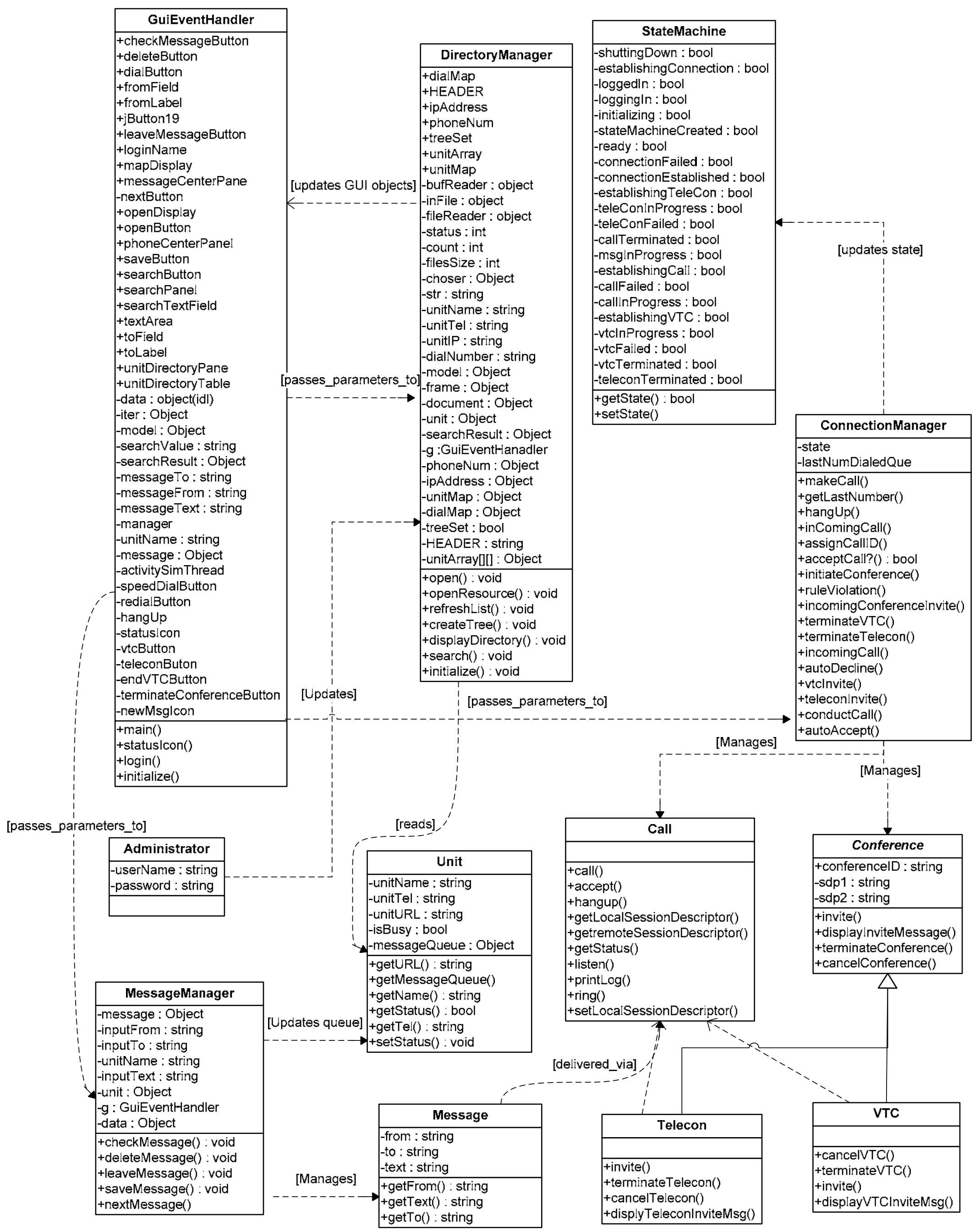

Figure 21. VoIPNET Class Diagram.

F. INTERACTION DIAGRAMS

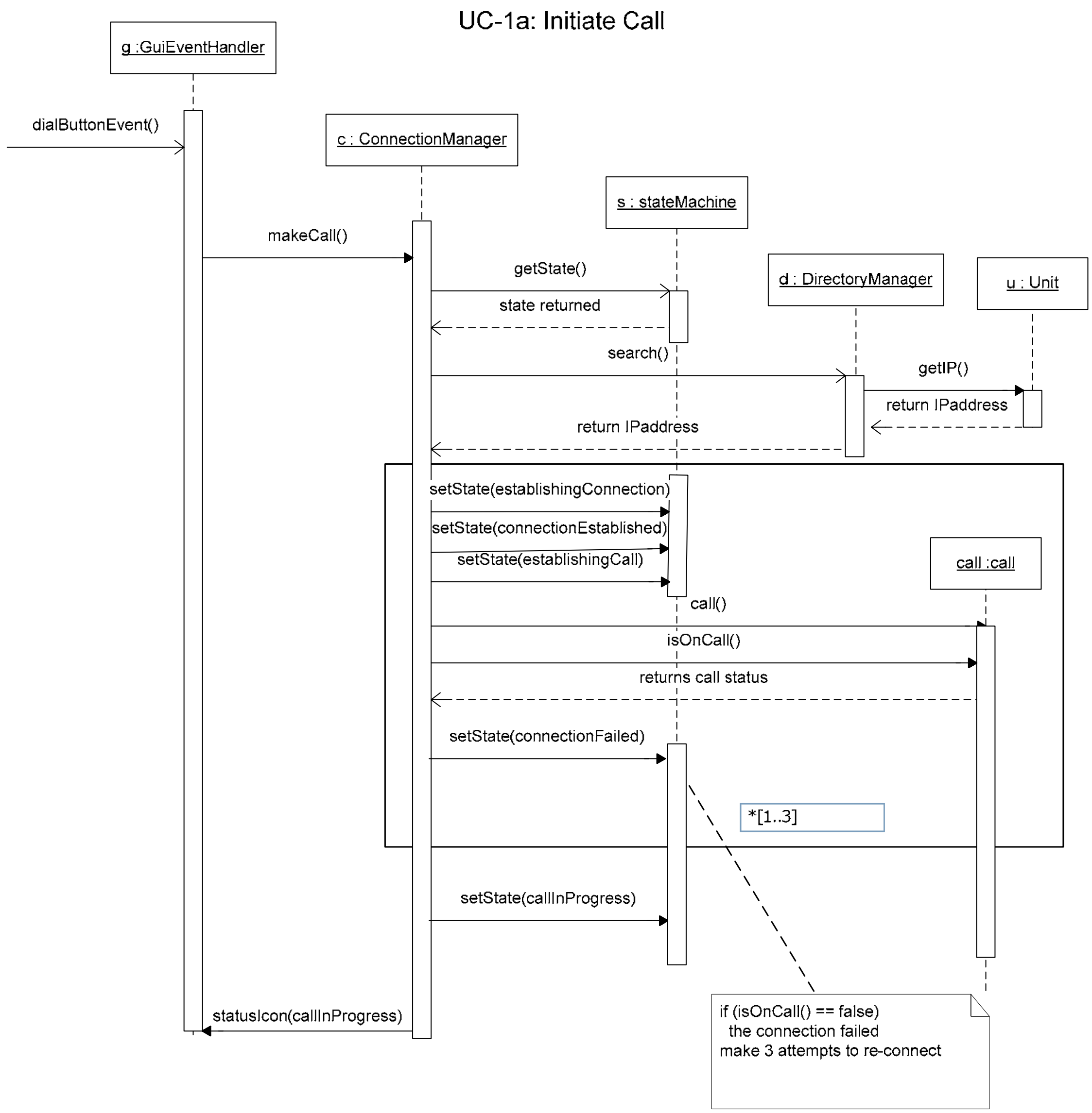

Figure 22. VoIPNET UC-1a: Initiate Call Interaction Diagram.

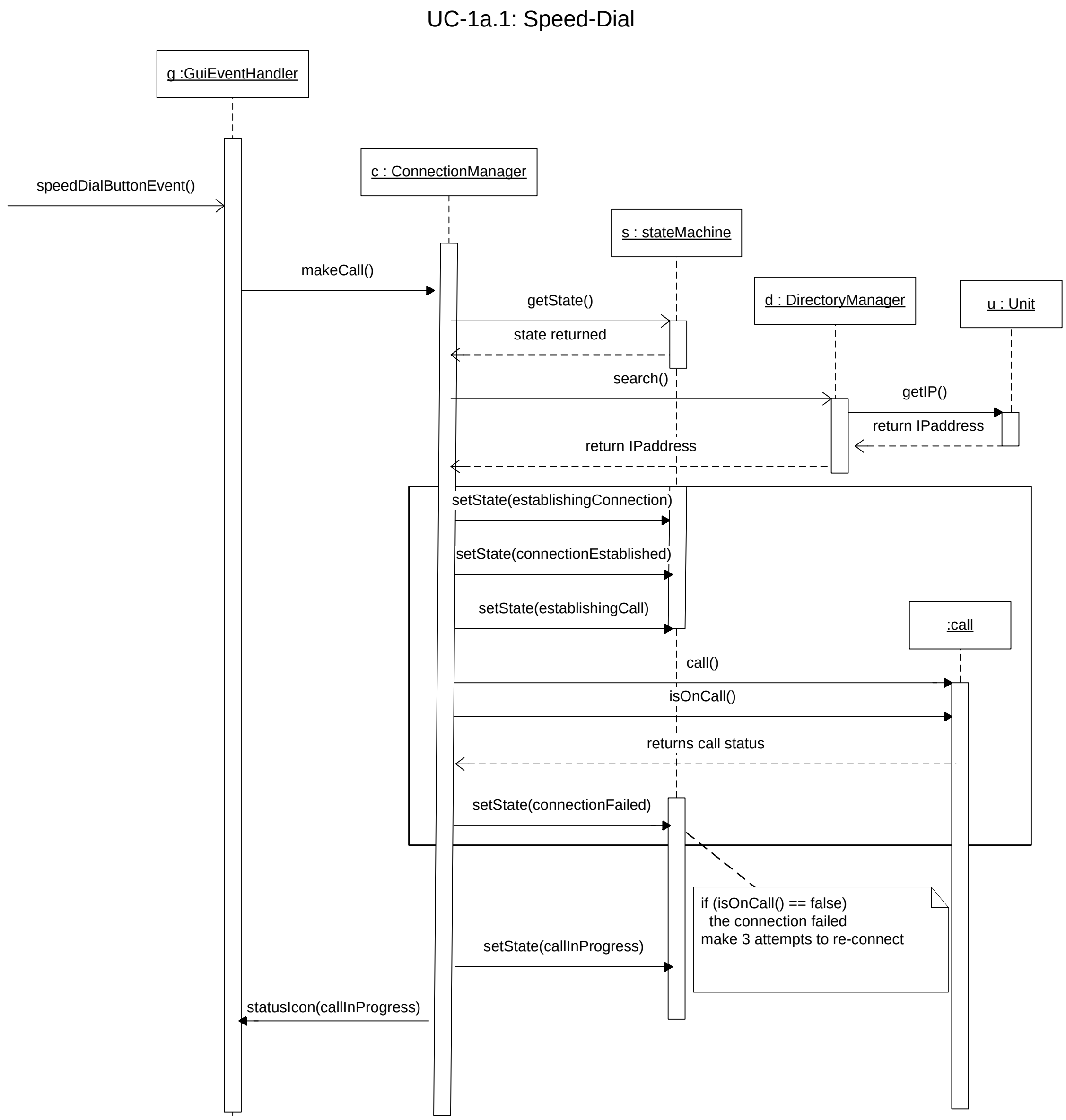

Figure 23. VoIPNET UC-1a.1 Speed-Dial Interaction Diagram.

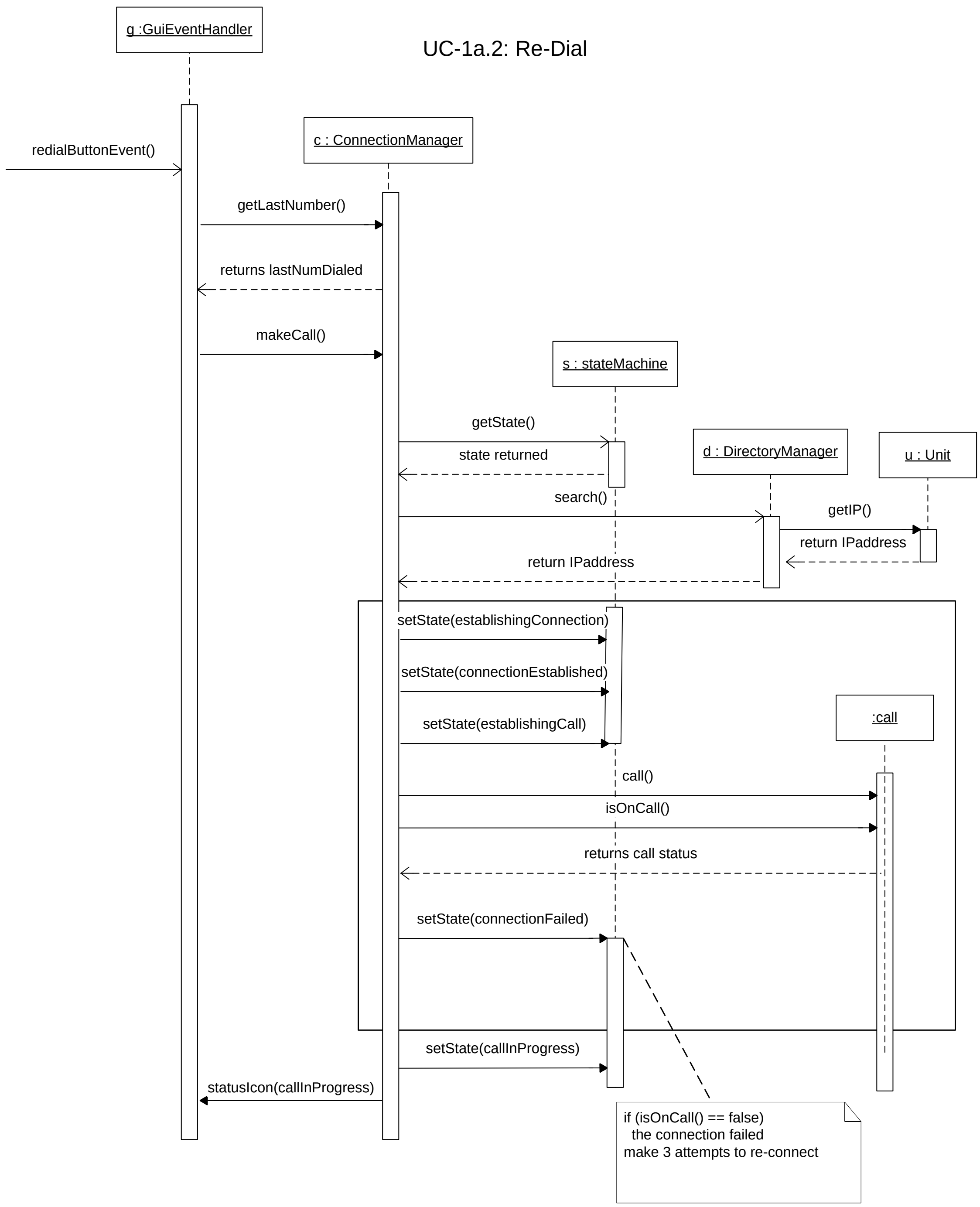

Figure 24. VoIPNET UC-1a.2 Re-Dial Interaction Diagram.

UC-1a.3: Abort Connection

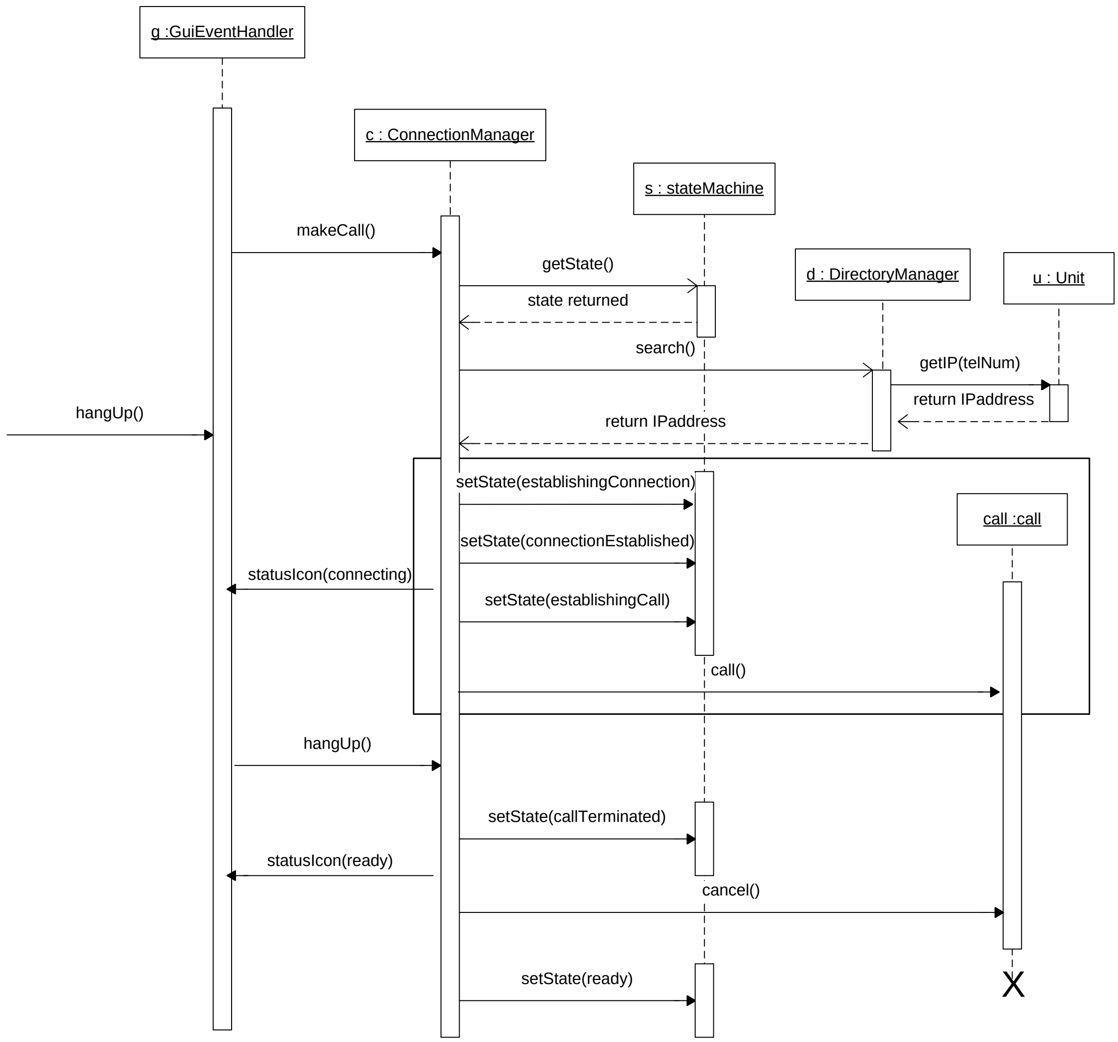

Figure 25. VoIPNET UC-1a.3 Abort Connection Interaction Diagram.

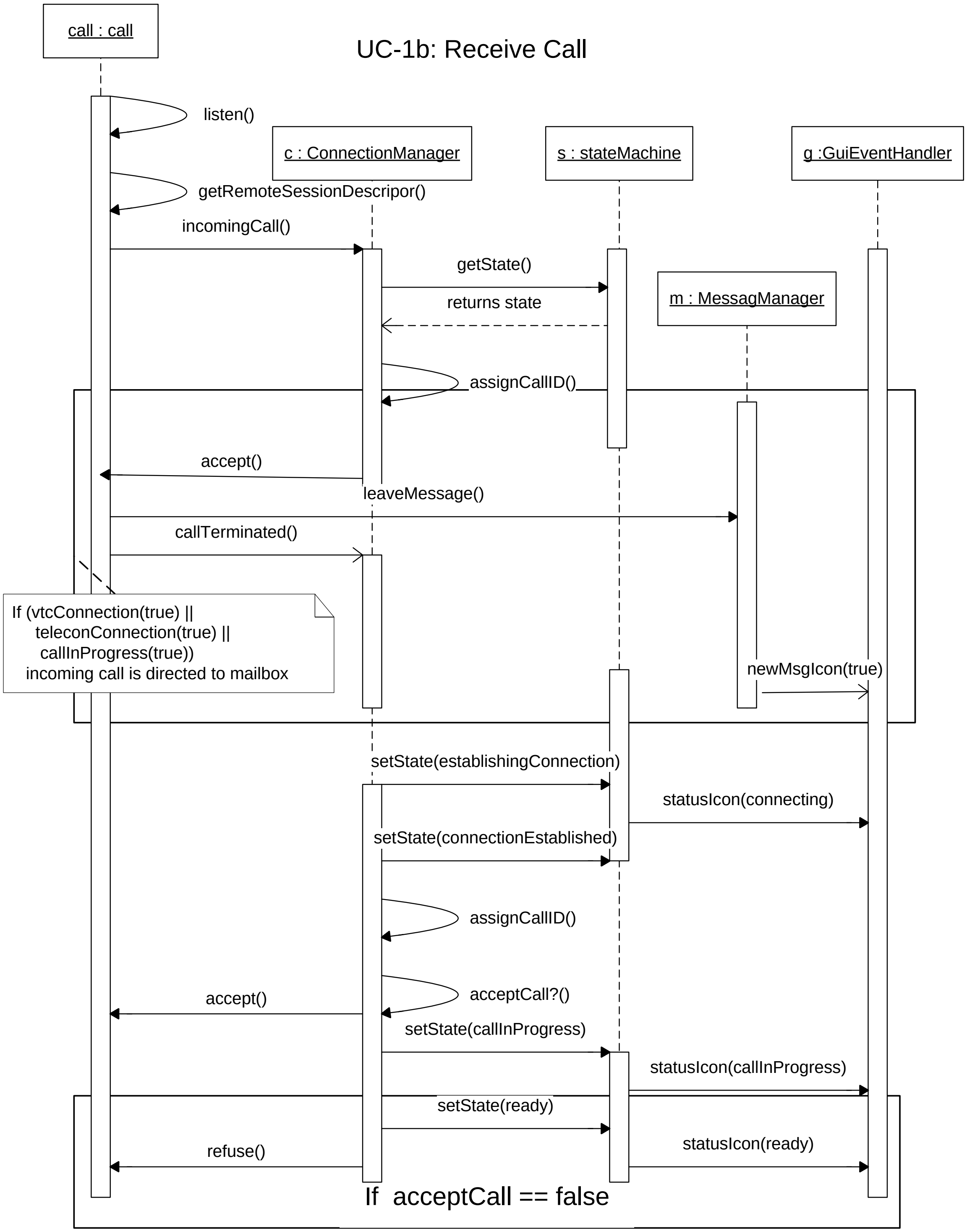

Figure 26. VoIPNET UC-1b Receive Call Interaction Diagram.

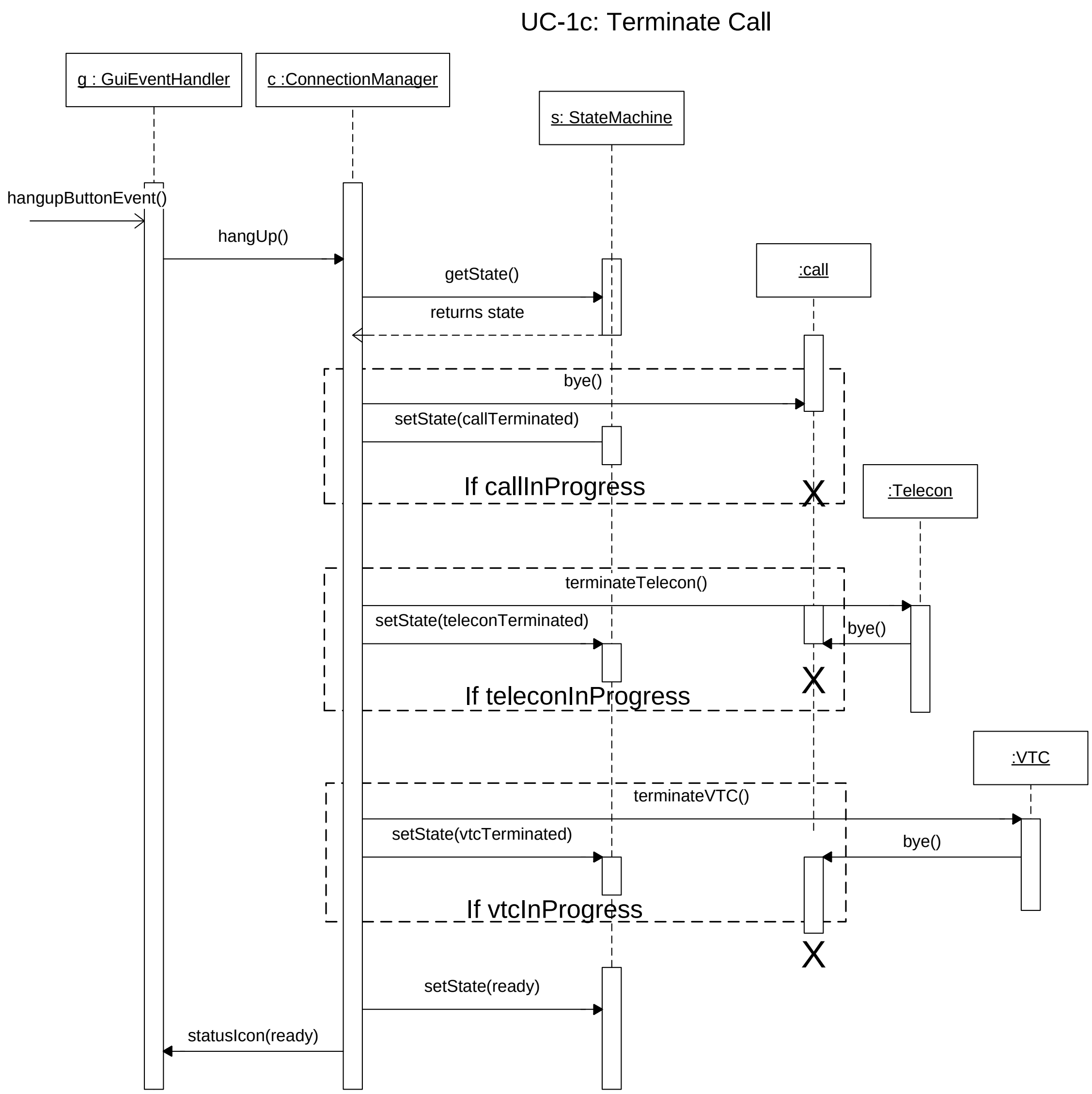

Figure 27. VoIPNET UC-1c Terminate Call Interaction Diagram.

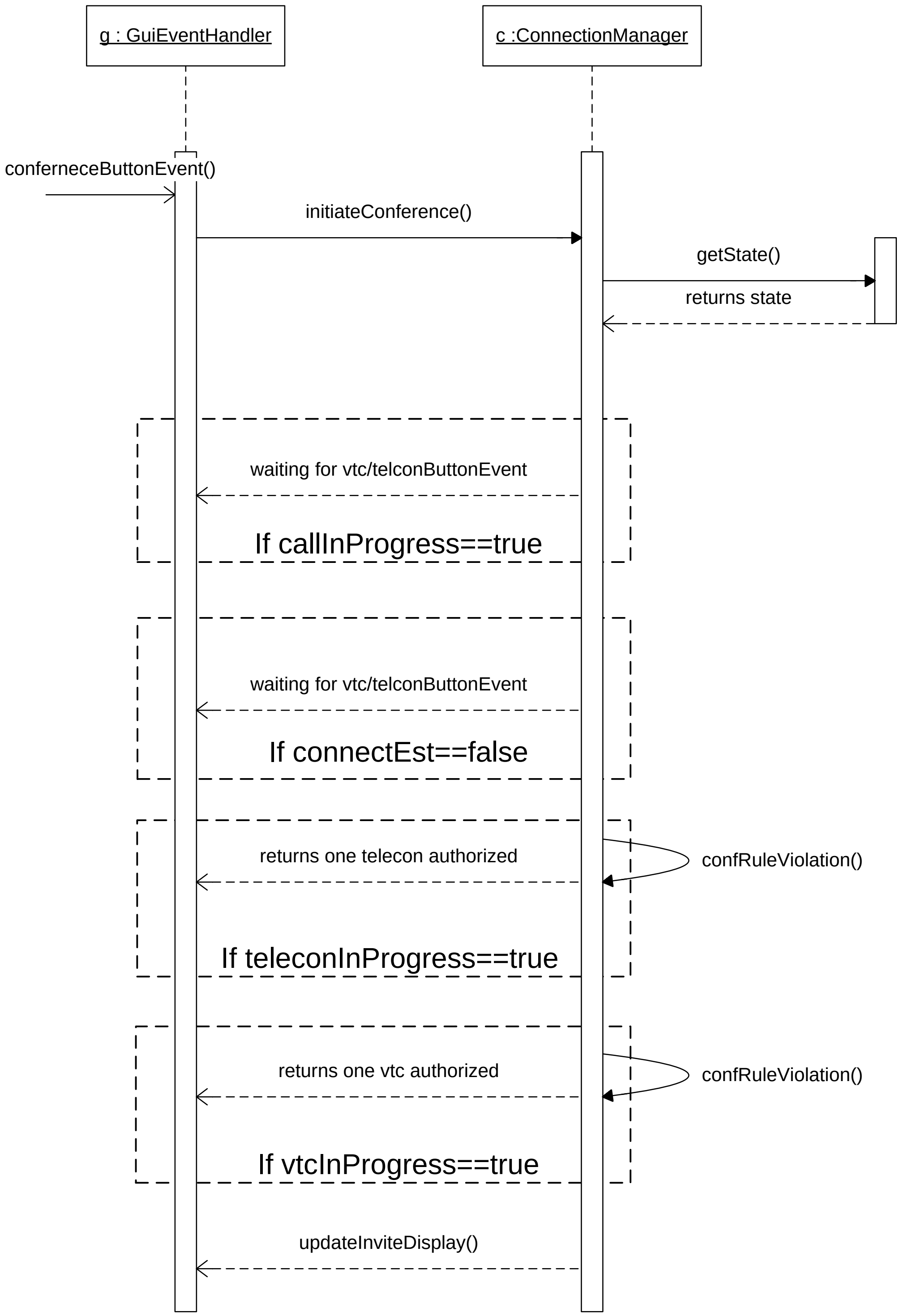

Figure 28. VoIPNET UC-2 Create Conference Invitation Interaction Diagram.

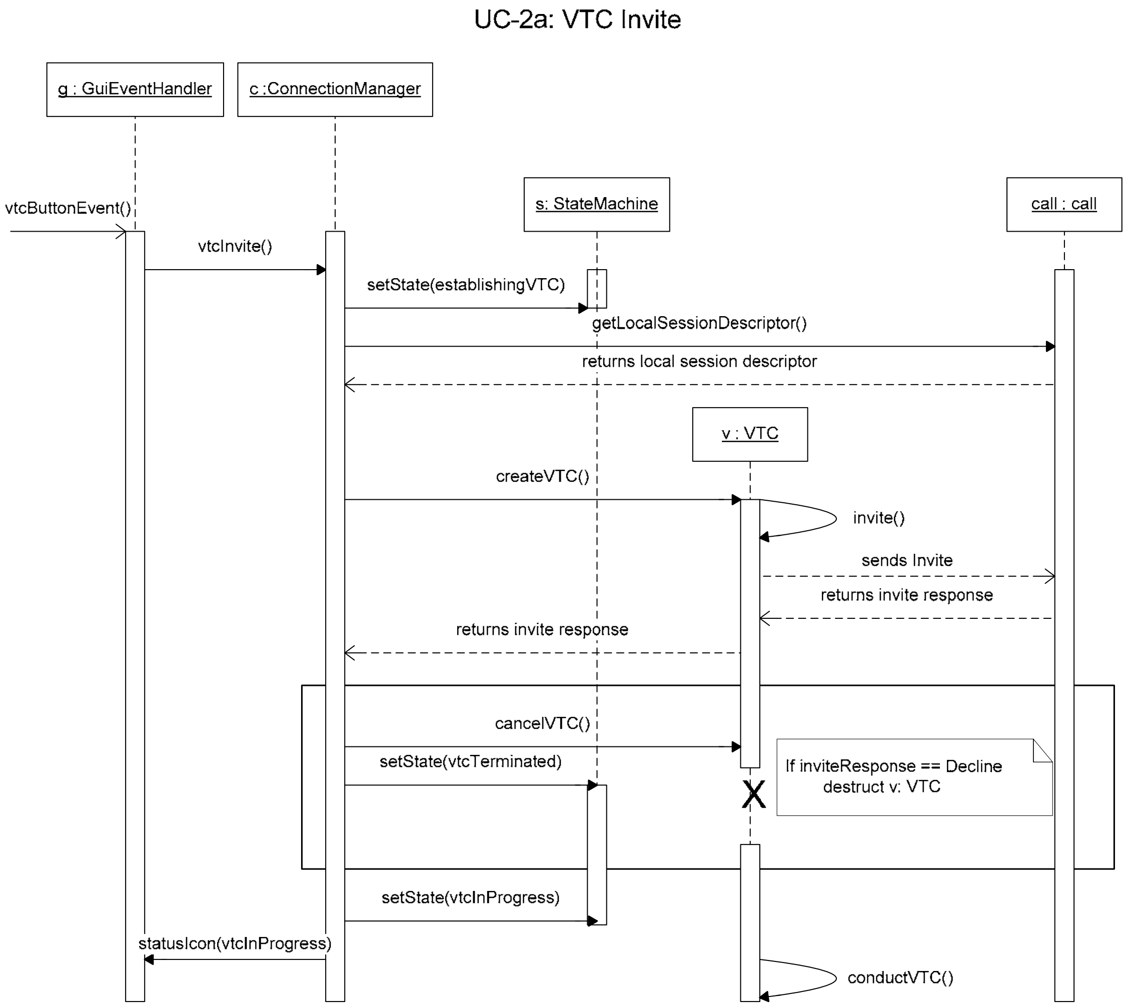

Figure 29. VoIPNET UC-2a VTC Invite Interaction Diagram.

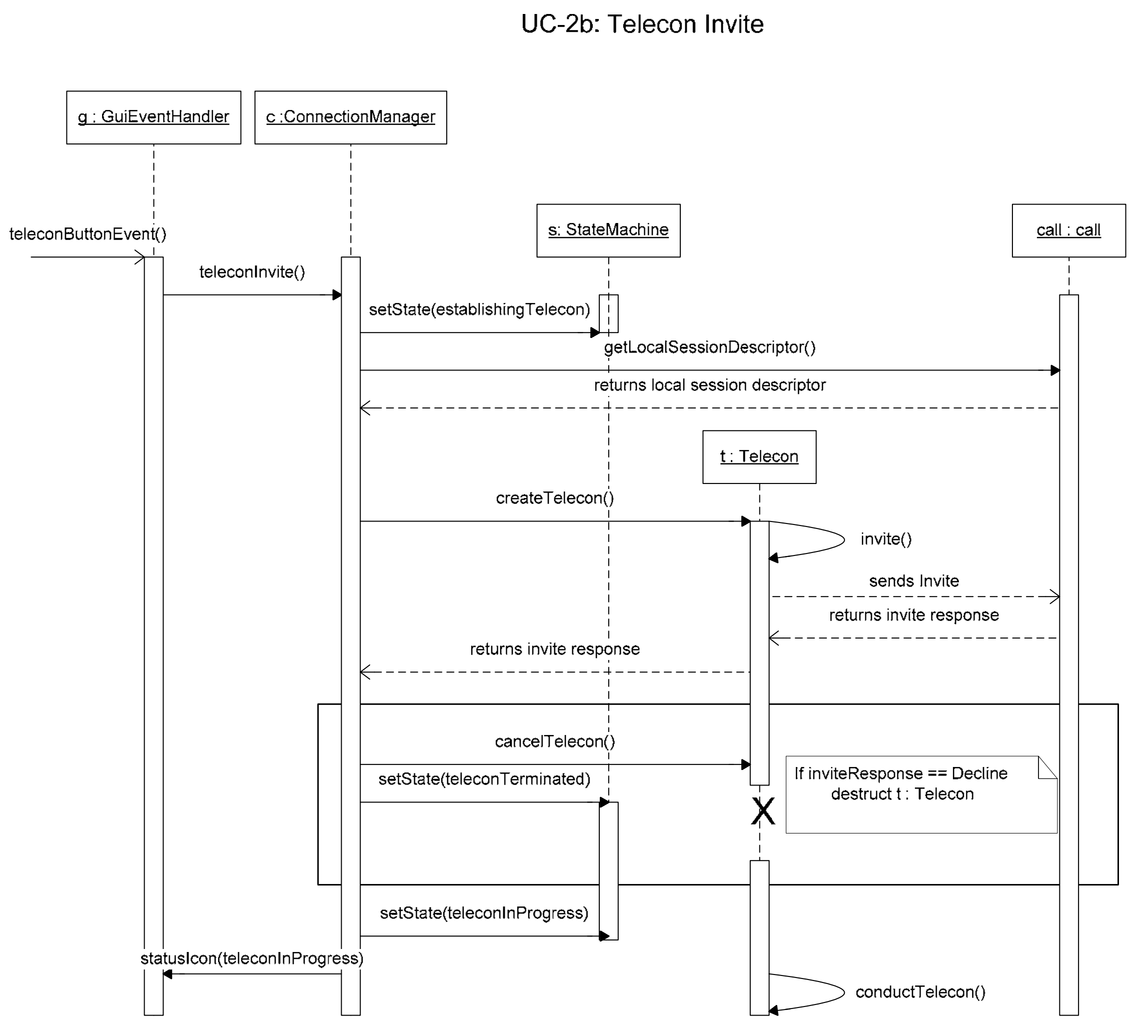

Figure 30. VoIPNET UC-2b Telecon Invite Interaction Diagram.

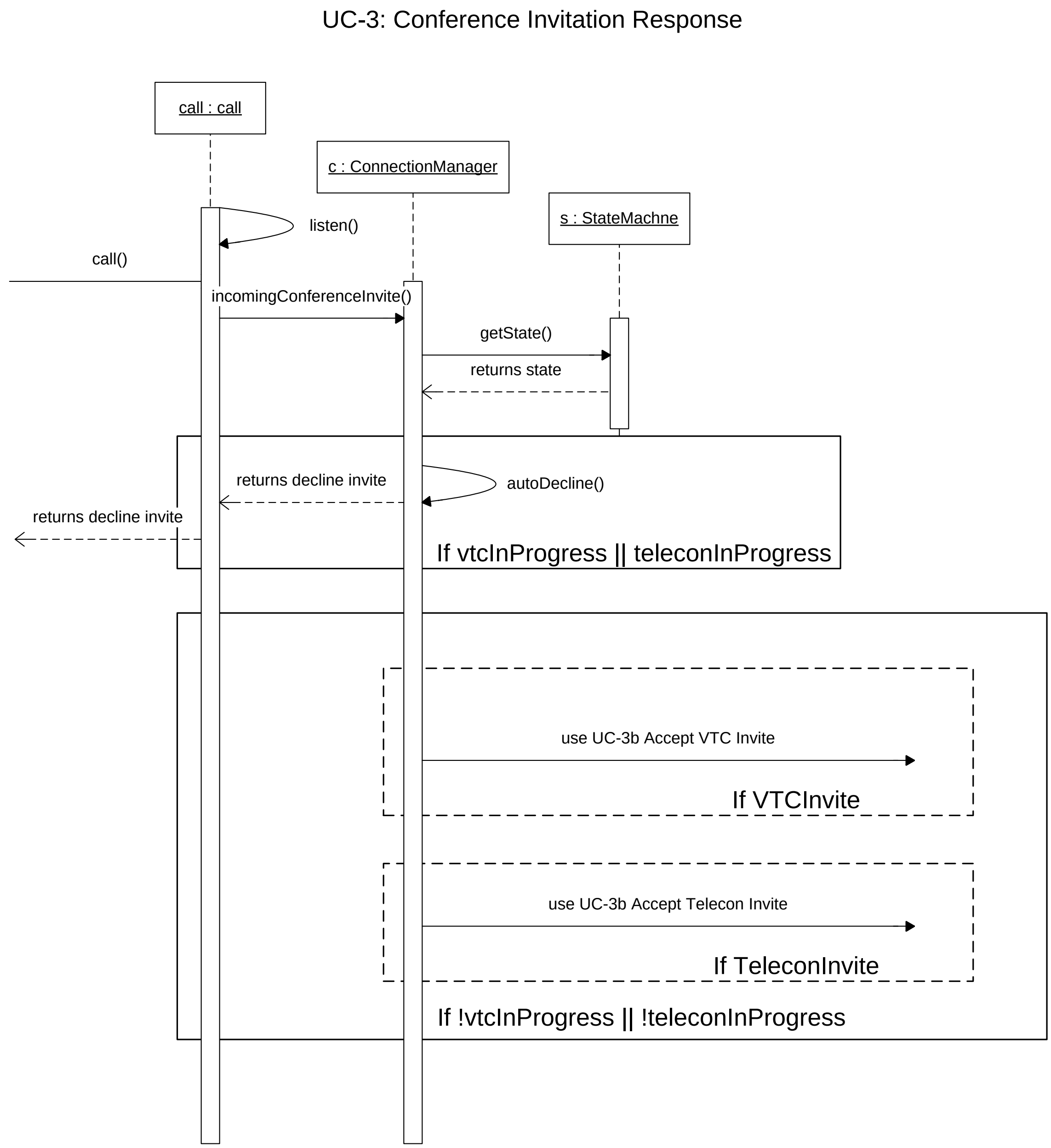

Figure 31. VoIPNET UC-3 Conference Invitation Response Interaction Diagram.

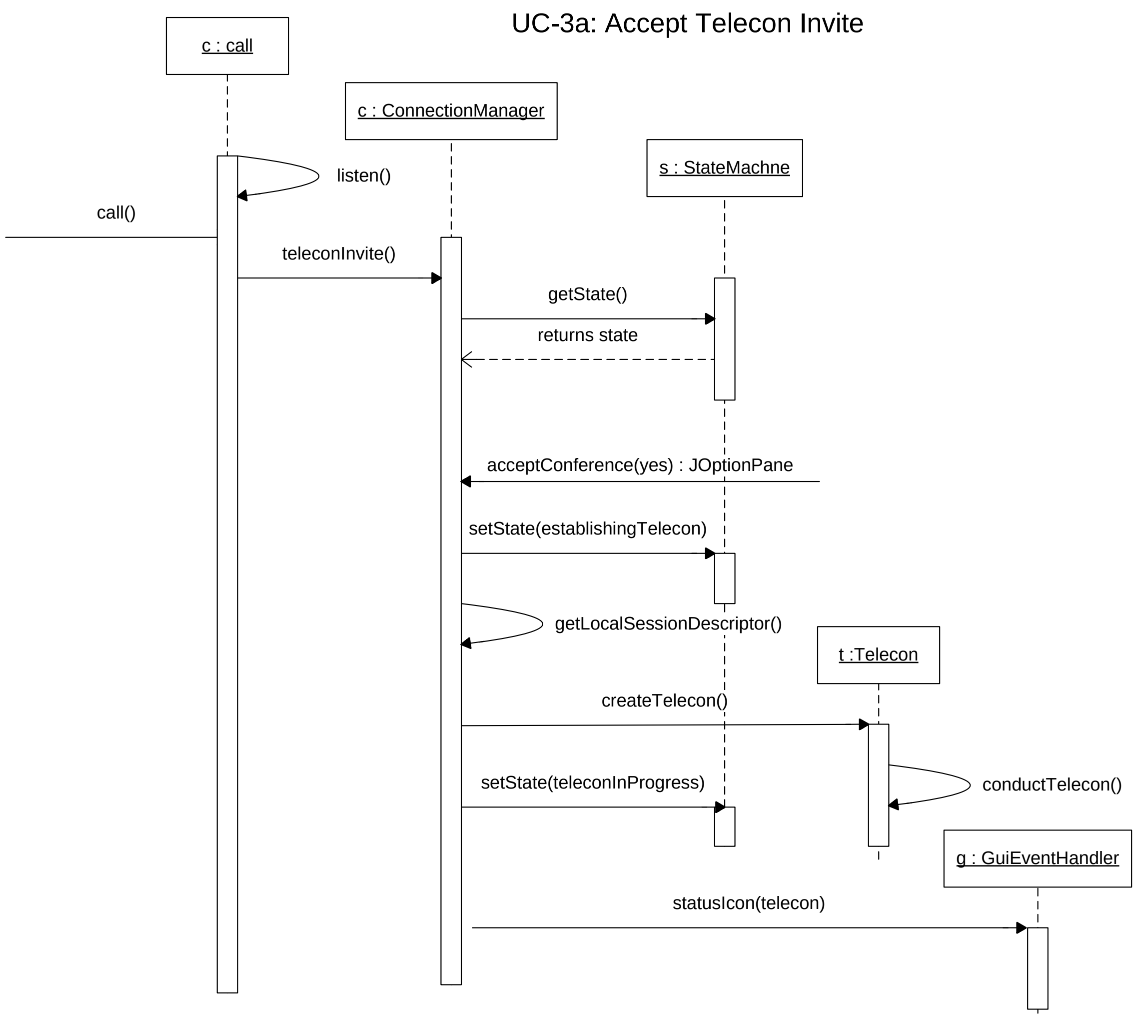

Figure 32. VoIPNET UC-3a Accept Telecon Invite Interaction Diagram.

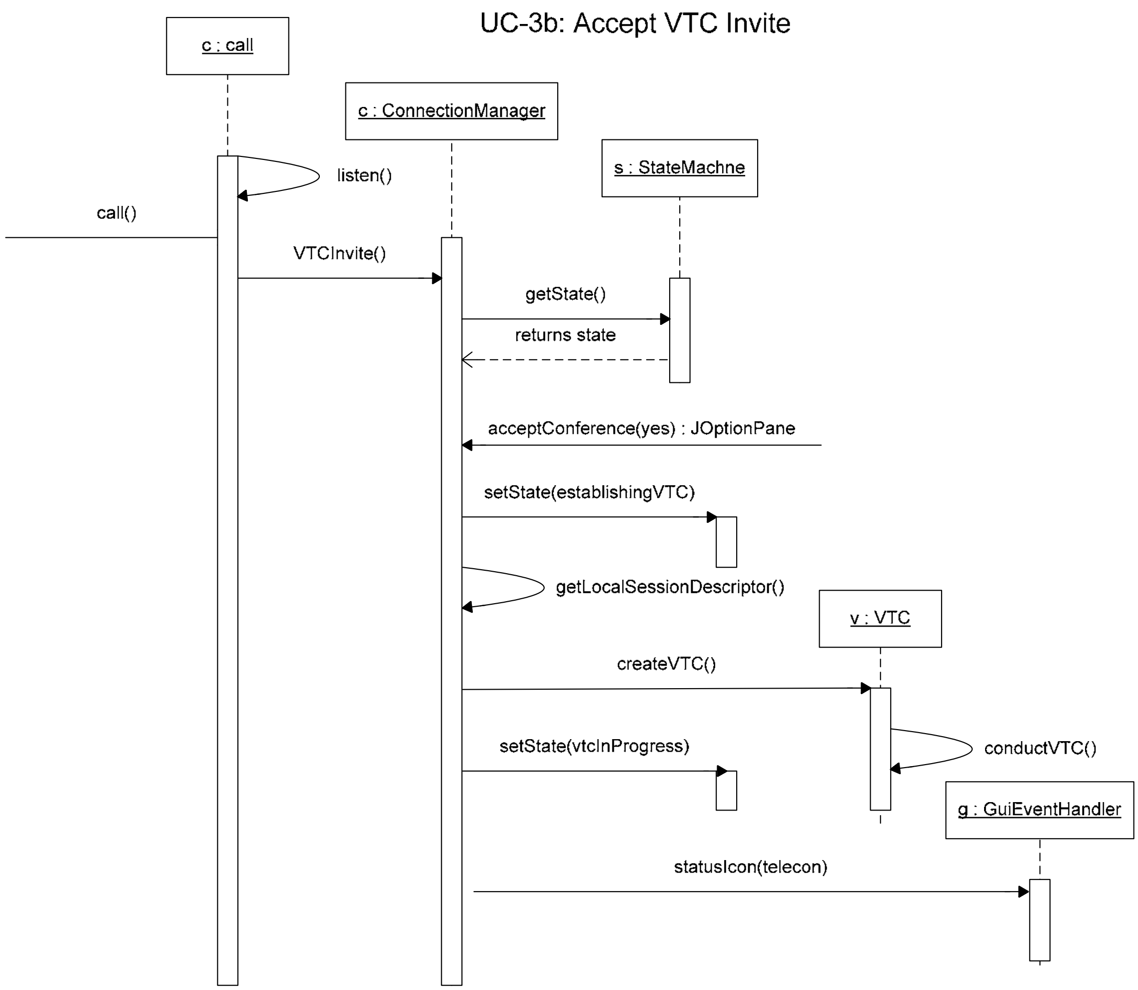

Figure 33. VoIPNET UC-3b Accept VTC Invite Interaction Diagram.

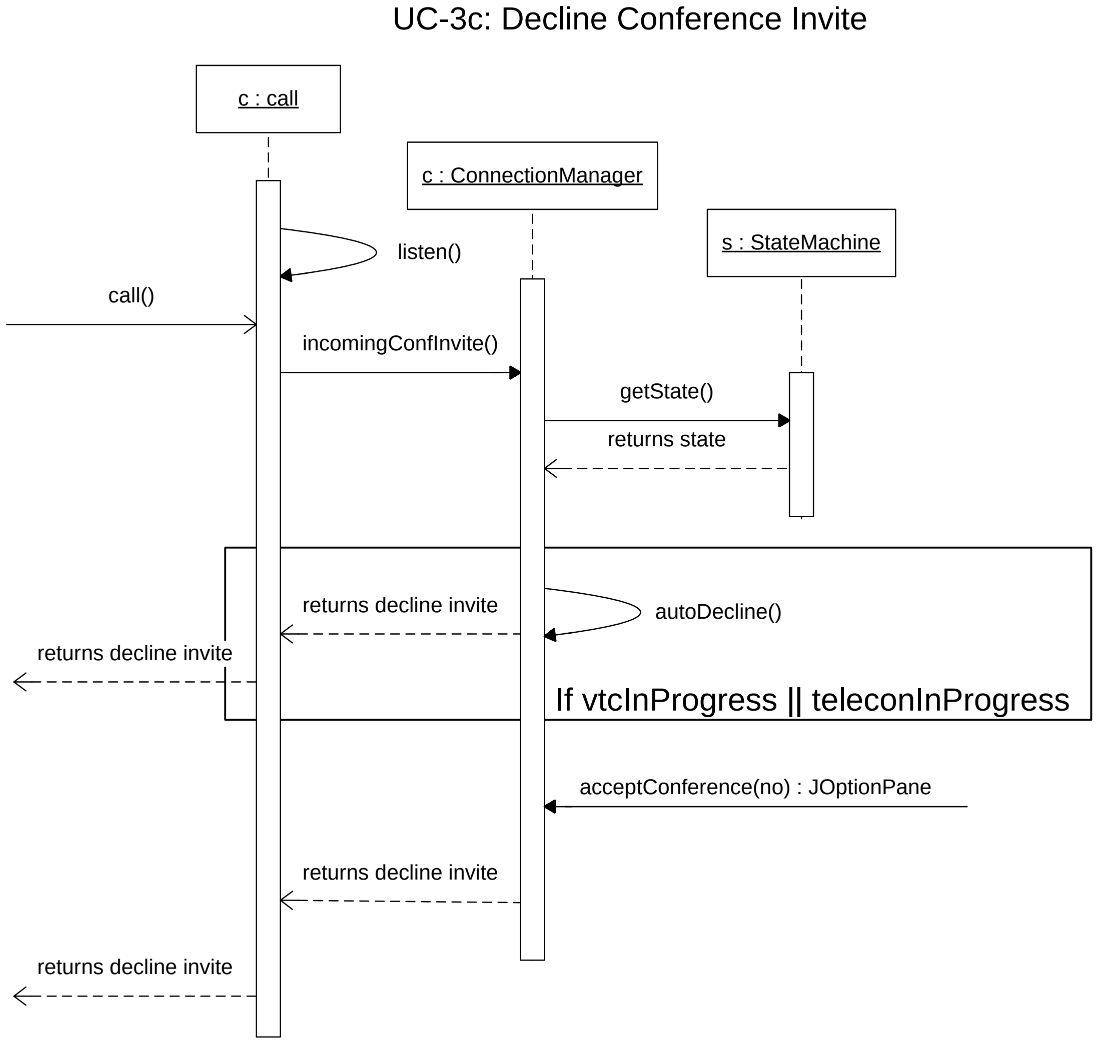

Figure 34. VoIPNET UC-3c Decline Conference Invite Interaction Diagram.

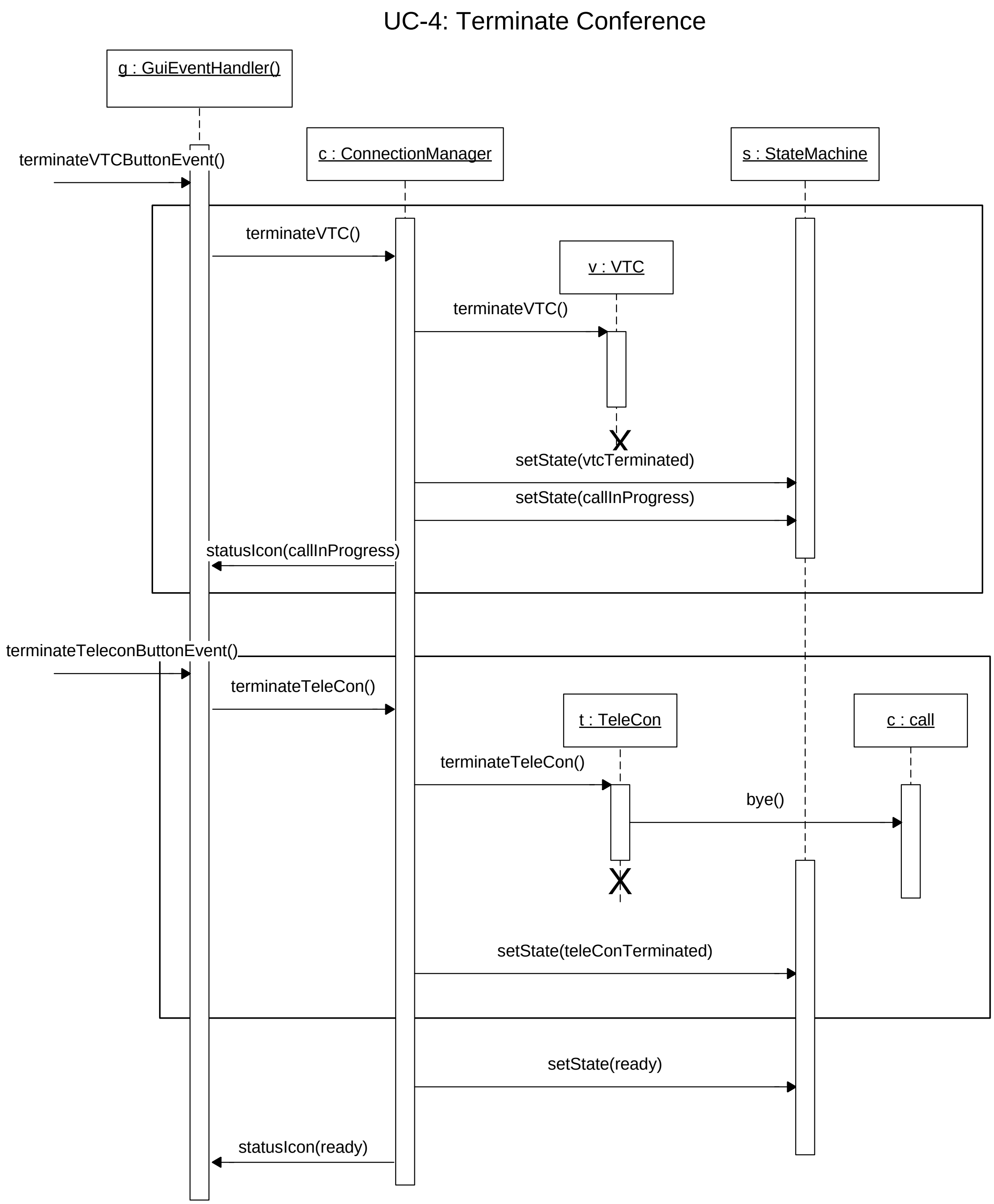

Figure 35. VoIPNET UC-4 Terminate Conference Interaction Diagram.

UC-6: Query Directory

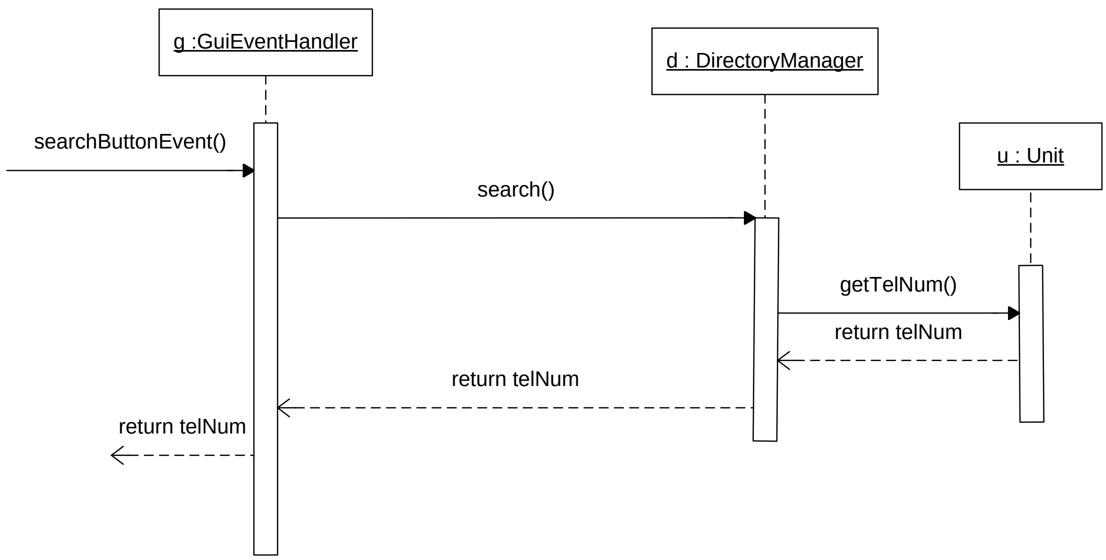

Figure 36. VoIPNET UC-6 Terminate Conference Interaction Diagram.

G. OPERATIONAL CONTRACTS

Contract: C1 makeCall(telNumber : string)

Cross Reference: UC-1a: Initiate Call

Preconditions:

Figure 37. Application running.

Figure 38. User logged in.

Figure 39. An instance ***c*** of ConnectionManager was created.

Figure 40. An instance ***s*** of StateMachine was created.

Figure 41. An instance ***d*** of DirectoryManager was created.

Figure 42. An instance ***dir*** of Directory was created.

Figure 43. ***s.getState()*** == **ready**.

Figure 44. User enters a telephone number.

Postconditions:

1. Application state is returned from ***s.***
2. ipAddress is returned from ***d.***
3. A new instance ***c*** of Call is created.
4. ***s.getState()*** == callInProgress.
5. A call request is sent to the callee **call()**.
6. g.callInProgressIcon == TRUE.

Contract: C2 getState() : string

Cross Reference: UC-1a: Initiate Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.

Postconditions:

1. Application state is returned from ***s.***

Contract: C3 getIP(telNumber : string) : string

Cross Reference: UC-1a: Initiate Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***dir*** of Directory was created.
7. ***s.getState()*** == ready.
8. User enters a telephone number.

Postconditions:

1. ipAddress is returned from ***d.***

Contract: C4 getIP(telNumber : string) : string

Cross Reference: UC-1a: Initiate Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of **Unit** was created.
7. ***s.getState()*** == **ready**.
8. User enters a telephone number.
9. ***d.getIP()*** has been called.

Postconditions:

1. ipAddress is returned to ***d*** from ***u.***

Contract: C5 setState(state : string)

Cross Reference: UC-1a: Initiate Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. A ***c*** event triggers a state change.

Postconditions:

s.getState() == **this.state.**

Contract: C6 call(callee,from,contact,sdp : string)

Cross Reference: UC-1a: Initiate Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. ***s.getState()*** == **establishingCall**.

Postconditions:

1. A new instance ***call*** of Call is created
2. ***s.getState()*** == **callInProgress.**
3. ***g.callInProgressIcon*** == TRUE.

Contract: C7 speedDialButtonEvent()

Cross Reference: UC-1a.1: Speed-Dial

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of unit was created.
7. Speed-Dial presets are configured.
8. ***s.getState()*** == ready.

Postconditions:

1. Application state is returned from ***s.***
2. ipAddress is returned from ***d.***
3. A new instance ***call*** of Call is created.
4. ***s.getState()*** == **callInProgress.**
5. A call request is sent to the callee.
6. ***g.callInProgressIcon*** == TRUE.

Contract: C8 reDialButtonEvent()

Cross Reference: UC-1a.2: Re-Dial

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. ***c.lastNumDialedQue***!= nil.
5. An instance ***s*** of StateMachine was created.
6. An instance ***d*** of DirectoryManager was created.
7. An instance ***u*** of Unit was created.
8. ***s.getState()*** == ready.

Postconditions:

1. Application state is returned from ***s.***
2. ipAddress is returned from ***d.***
3. A new instance ***call*** of Call is created.
4. ***s.getState()*** == callInProgress.
5. A call request is sent to the callee.
6. ***g.callInProgressIcon*** == TRUE.

Contract: C9 getLastNumber() : string

Cross Reference: UC-1a.2: Re-Dial

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.

4. ***c.lastNumDialedQue***!= nil.

5. An instance ***s*** of StateMachine was created.

6. An instance ***d*** of DirectoryManager was created.

7. An instance ***u*** of unit was created.

8. ***s.getState()*** == ready.

Postconditions:

1. last Number dialed is returned from Que.

Contract: C10 hangUp()

Cross Reference: UC-1a.3: Abort Connection

Preconditions:

1. Application running.

2. User logged in.

3. An instance ***c*** of ConnectionManager was created.

4. An instance ***s*** of StateMachine was created.

5. An instance ***d*** of DirectoryManager was created.

6. An instance ***u*** of Unit was created.

7. ***s.getState()***==establishingConnection|| connectionEstablished||establishingCall.

Postconditions:

1. The call is canceled.

2. ***s.getState()*** == callTerminated.

3. ***s.getState()*** == ready.

Contract: C11 cancel()

Cross Reference: UC-1a.3: Abort Connection

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. ***s.getState()***==establishingConnection|| connectionEstablished || establishingCall.
8. ***call.call()*** has been called.

Postconditions:

1. The call is canceled.
2. ***s.getState()*** == callTerminated.
3. ***s.getState()*** == ready.

Contract: C12 listen()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.

5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. ***s.getState()*** == ready.

Postconditions:

1. An incoming call is identified.
2. ***call.getRemoteSessionDescriptor()*** is called.

Contract: C13 getRemoteSessionDescriptor()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. An instance ***call*** of Call has been created.
8. ***call.listen()*** has been called.
9. Incoming call has been identified.

Postconditions:

1. Remote Session Descriptor is retrieved from caller.

Contract: C14 incomingCall()

Cross Reference: UC-1b: Receive Call

Preconditions:

2. Application running.
3. User logged in.
4. An instance ***c*** of ConnectionManager was created.
5. An instance ***s*** of StateMachine was created.
6. An instance ***d*** of DirectoryManager was created.
7. An instance ***u*** of Unit was created.
8. An instance ***call*** of call was created
9. An instance ***m*** of MessageManager is created.
10. ***call.listen()*** has been called.
11. ***call.getRemoteSessionDescriptor()*** has been called.

Postconditions:

1. If(vtcInProgress||teleconInProgress)
   a. Auto accept the call ***call.accept()***.
   b. Direct caller to the callee mailbox ***m.leaveMessage().***
   c. Update state ***s.setState(msgInProgress).***
   d. The message is completed so call ***c.terminateCall().***
   e. Set the new message Icon ***g.NewMsgIcon(true).***
   f. Assign new call ID number ***c.assignCallID().***
   g. Prompt accept/refuse call ***c.acceptCall?().***

2. If the call is refused call ***call.refuse().***
    a. Update the state by calling ***s.setState(callTerminated).***
3. If call is accepted call ***call.accept().***
    a. Update the state by calling ***s.setState(callInProgress).***

Contract: C15 assignCallID(sdp : string) :string

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. An instance ***call*** of call was created
8. ***s.getState()*** == ready.
9. ***listen.call()*** has been called
10. ***c.incomingCall()*** has been called.

Postconditions:

1. call ID number is returned to ***c***

Contract: C16 acceptCall?()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. An instance ***call*** of call was created.
8. ***c.incomingCall()*** has been called.

Postconditions:

1. If the call is accepted ***call.accept()***.
2. ***s.getState()*** == callInProgress.
3. If the call is refused *call.refuse()*.
4. ***s.getState()*** == callTerminated.

Contract: C17 accept()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.

6. An instance ***u*** of Unit was created.
7. An instance ***call*** of call was created.
8. ***s.getState()*** == callInProgress.

Postconditions:

1. Call is accepted.
2. The caller/callee conduct the call ***c.conductCall().***

Contract: C18 refuse()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit was created.
7. An instance ***call*** of call was created
8. ***s.getState()*** == establishingConnection.
9. ***c.incomingCall()*** has been called.

Postconditions:

1. The call is refused.
2. ***s.getState()*** == callTerminated.
3. ***s.getState()*** == ready.

Contract: C19 leaveMessage()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.
2. User logged in.
3. An instance ***c*** of ConnectionManager was created.
4. An instance ***s*** of StateMachine was created.
5. An instance ***d*** of DirectoryManager was created.
6. An instance ***u*** of Unit for all entries in ***directory*** is created.
7. An instance ***call*** of call was created.
8. An instance ***m*** of MessageManager was created.
9. ***s.getState()*** == msgInProgress.

Postconditions:

1. A new instance ***msg*** of Message is created.
2. Unit mailbox updated with message ***u.messageQueue(msg).***
3. ***s.getState()*** == **callTerminated**
4. ***s.getState()*** == **ready.**

Contract: C20 setMsgIcon()

Cross Reference: UC-1b: Receive Call

Preconditions:

1. Application running.

2. User logged in.
3. A new instance **msg** of Message is created
4. Unit mailbox updated with message ***u.messageQueue(msg).***
5. ***s.getState()*** == callTerminated.
6. ***s.getState()*** == **ready.**

Postconditions:

1. The new message Icon is displayed.

Contract: C21 bye()

Cross Reference: UC-1c: Terminate Call

Preconditions:

1. Application running.
2. User logged in.
3. A new instance **msg** of Message is created
4. Unit mailbox updated with message ***u.messageQueue(msg).***
5. ***s.getState()*** == callTerminated**.**
6. ***s.getState()*** == ready.

Postconditions:

1. The new message Icon is displayed.

Contract: C22 terminateTelecon()

Cross Reference: UC-1c: Terminate Call

Preconditions:

1. Application running.
2. User logged in.
3. A new instance **msg** of Message is created
4. Unit mailbox updated with message ***u.messageQueue(msg).***
5. ***s.getState()*** == callTerminated.
6. ***s.getState()*** == ready.

Postconditions:

1. The new message Icon is displayed.

Contract: C23 terminateVTC()

Cross Reference: UC-1c: Terminate Call

Preconditions:

1. Application running.
2. User logged in.
3. A new instance **msg** of Message is created
4. Unit mailbox updated with message ***u.messageQueue(msg)***.
5. ***s.getState()*** == callTerminated.
6. ***s.getState()*** == ready.

Postconditions:

1. The new message Icon is displayed.

Contract: C24 isOnCall()

Cross Reference: UC-1c: Terminate Call

Preconditions:

1. Application running.
2. User logged in.
3. A new instance **msg** of Message is created
4. Unit mailbox updated with message ***u.messageQueue(msg).***
5. ***s.getState()*** == callTerminated.
6. ***s.getState()*** == ready.

Postconditions:

The new message Icon is displayed.

Contract: C25 initiateConference()

Cross Reference: UC-2a/b: VTC Invite/Telecon Invite

Preconditions:

1. Application running.
2. User logged in.
3. vtc/teleconButtonEvent() has occurred.

Postconditions:

1. The "VTC/Telecon Authorized" message is displayed.
2. ***c.vtcInvite()*** or ***c.teleconInvite()*** is called.
3. –or- call ***c.ruleViolation()*** and display error message.

Contract: C26 vtcInvite()

Cross Reference: UC-2a: VTC Invite

Preconditions:

1. Application running.
2. User logged in.
3. ***initiateConference()*** has been called.
4. ***s.getState()*** == ready.

Postconditions:

1. A new instance ***v*** of **VTC** is created ***createVTC()***.
2. The "Invite sent" message is displayed.
3. The invite is sent to the callee.

Contract: C27 createVTC(sdp)

Cross Reference: UC-2a: VTC Invite

Preconditions:

1. ***vtcInvite()*** has been called.
2. ***getLocalSessionDescriptor()*** has been called.

Postconditions:

1. A new instance ***v*** of **VTC** is created.
2. The "Invite sent" message is displayed.
3. The invite is sent to the invitee ***v.invite()***.

Contract: C28 getLocalSessionDescriptor() :sdp

Cross Reference: UC-2a: VTC Invite

Preconditions:

1. ***vtcInvite()*** has been called.

Postconditions:

2. The sdp is returned to ***c.***

Contract: C29 v.invite()

Cross Reference: UC-2a: VTC Invite

Preconditions:

1. ***createVtc()*** has been called.

Postconditions:

2. The invite is sent to the invitee.

Contract: C30 teleconInvite()

Cross Reference: UC-2b: Telecon Invite

Preconditions:

1. Application running.
2. User logged in.
3. ***initiateConference()*** has been called.
4. ***s.getState(ready).***

Postconditions:

1. A new instance ***t*** of **Telecon** is created ***createTelecon()***.
2. The “Invite sent” message is displayed.
3. The invite is sent to the callee.

Contract: C31 createTelecon(sdp)

Cross Reference: UC-2b: Telecon Invite

Preconditions:

1. ***teleconInvite()*** has been called.
2. ***getLocalSessionDescriptor()*** has been called.

Postconditions:

1. A new instance ***t*** of **Telecon** is created.
2. The “Invite sent” message is displayed.
3. The invite is sent to the callee ***t.invite()***.

Contract: C32 t.invite()

Cross Reference: UC-2b: Telecon Invite

Preconditions:

1. ***createTelecon()*** has been called.

Postconditions:

1. The invite is sent to the callee.

Contract: C33 incomingConfInvite() :inviteResponse

Cross Reference: UC-3: Conference Invitation Response

Preconditions:

1. ***isOnCall()***== TRUE.
2. An invite is sent from caller.

Postconditions:

1. The conference display message is displayed.
2. The invite response is sent to the caller.

Contract: C34 displayVTCInviteMsg()

Cross Reference: UC-3: Conference Invitation Response

Preconditions:

1. ***incomingConfenceInvite()*** has been called.

Postconditions:

1. The “Join VTC?” message is displayed.
2. The user accepts the invite ***c.acceptConfButtonEvent()*** or declines the invite ***c.declineConfButtonEvent()***.

Contract: C35 displayTeleconInviteMsg()

Cross Reference: UC-3: Conference Invitation Response

Preconditions:

incomingConfenceInvite() has been called.

Postconditions:

1. The “Join Telecon?” message is displayed.
2. The user accepts invite ***c.acceptConfButtonEvent()*** or declines the invite ***c.declineConfButtonEvent()***.

Contract: C36 autoDecline ()

Cross Reference: UC-3: Conference Invitation Response

Preconditions:

1. ***incomingConfenceInvite()*** has been called.
2. ***s.getState*** == teleconInProgress || vtcInProgress.

Postconditions:

1. The "Conference Declined" message is automatically sent to inviter.

Contract: C37 terminateConferenceButtonEvent()

Cross Reference: UC-4: Terminate Conference

Preconditions:

1. ***s.getState()*** == confInProgress.

Postconditions:

1. ***c.terminateVTC()*** or ***c.terminateTelecon()*** is called.
2. ***s.getState*** == ready.
3. The "Conference Terminated" message is displayed.
4. The VTC/Telecon Icon is removed ***g.VTCIcon()*** == FALSE or ***g.teleconICON()*** == FALSE.

Contract: C38 c.terminateVTC()

Cross Reference: UC-4: Terminate Conference

Preconditions:

1. ***terminateConferenceButtonEvent()*** has been called.

Postconditions:

1. ***v.terminateVTC()*** is called.
2. ***s.getState*** == vtcTerminated.

Contract: C39 c.terminateTelecon()

Cross Reference: UC-4: Terminate Conference

Preconditions:

1. ***terminateConferenceButtonEvent()*** has been called.

Postconditions:

1. ***t.terminateTelecon()*** is called.
2. ***s.getState*** == teleconTerminated.

Contract: C40 v.terminateVTC()

Cross Reference: UC-4: Terminate Conference

Preconditions:

1. ***c.terminateVTC()*** has been called.
2. ***s.getState*** == vtcTerminated.

Postconditions:

1. ***v*** is destroyed.

Contract: C41 t.terminateTelecon()

Cross Reference: UC-4: Terminate Conference

Preconditions:

1. ***c.terminateTelecon()*** has been called.
2. ***s.getState*** == teleconTerminated.

Postconditions:

1. The call is disconnected ***call.bye().***
2. ***t*** is destroyed.

<u>Contract:</u> C42 searchButtonEvent(unitName) :Unit

<u>Cross Reference:</u> UC-6: Query Directory

<u>Preconditions:</u>

1. An instance ***d*** of **DirectoryManager** exists.
2. An instance ***u*** of **Unit** exists.

<u>Postconditions:</u>

1. ***d.search(unitName)*** is called.
2. The appropriate entry is highlighted in the Directory table.
3. The unit's telephone Number is placed in the dial textbox.

<u>Contract:</u> C43 search(unitName) :void

<u>Cross Reference:</u> UC-6: Query Directory

<u>Preconditions:</u>

a. An instance ***d*** of **DirectoryManager** exists.
b. An instance ***u*** of **Unit** exists.

<u>Postconditions:</u>

1. ***u.getTelNum(unitName)*** is called.
2. The appropriate entry is highlighted in the Directory table.
3. The unit's telephone Number is placed in the dial textbox.

<u>Contract:</u> C44 getTel(unitName) :Unit

Cross Reference: UC-6: Query Directory

Preconditions:

1. An instance ***d*** of **DirectoryManager** exists.
2. An instance ***u*** of **Unit** exists.

Postconditions:

1. The unit’s telephone number is returned.

Contract: C45 open() :void

Cross Reference: UC-11 Login

Preconditions:

1. Application is running.

Postconditions:

1. GUI display format is set and visible.

Contract: C46 openResource() :void

Cross Reference: UC-11 Login

Preconditions:

1. Login is complete and GUI is displayed.

Postconditions:

1. Contacts list file is opened.

Contract: C47 refreshList() :void

Cross Reference: UC- 7 Update Directory

Preconditions:

1. GUI is displayed.
2. Contact list is opened.

Postconditions:

1. Directory entries in the GUI are refreshed.

Contract: C48 createTree() :void

Cross Reference: UC-7 Update Directory

Preconditions:

1. Contact list is opened.

Postconditions:

1. Unsorted array f units created for map construction.
2. phoneNum, ipAddress, unitMap, and dialMap TreeMaps created.

Contract: C49 displayDirectory() :void

Cross Reference: UC-6 Query Directory

Preconditions:

1. Contact list is open.
2. unitArray created.

Postconditions:

1. The directory is displayed in the GUI.

Contract: C50 initialize() :void

Cross Reference: UC-11 Login

Preconditions:

1. Login successful.

Postconditions:

1. An instance ***c*** of ConnectionManager is created.
2. An instance ***d*** of DirectoryManager is created.
3. An instance ***g*** of GuiEventHandler is created.
4. An instance ***s*** of StateMachine is created.
5. An instance ***call*** of Call is created.
6. call.Listen().
7. s.setState(ready).

Contract: C51 ruleViolation() :void

Cross Reference: UC-2 Initiate Conference

Preconditions:

1. User attempts to initiate a conference.
2. User already engaged in a teleconference or VTC.

Postconditions:

1. Error message displayed.
2. No conference initiated.

Contract: C52 incomingCall() :void

Cross Reference: UC-1b. Receive Call

Preconditions:

1. Call request received from caller.
2. Session Descriptor is retrieved.

Postconditions:

1. Call is accepted or refused.
2. A new instance ***m*** of Message is created if the callee is in a conference.

Contract: C53 callTerminated() :void

Cross Reference: UC-1.c

Preconditions:

1. s.GetState(callInProgress).

Postconditions:

1. s.setState(callTerminated).
2. call.hangup().
3. s.setState(ready)

Contract: C54 getName() :void

Cross Reference: UC-6 Directory Query

Preconditions:

1. initialize() has been called.
2. An instance ***u*** of Unit exists.

Postconditions:

1. unitName is returned.

Contract: C55 hangUp() :void

Cross Reference: UC-1.c

Preconditions:

1. ***s.GetState(callInProgress)***

Postconditions:

1. call is terminated.
2. ***call.listen()***.
3. ***s.setState(ready).***

Contract: C56 printLog() :void

Cross Reference: UC-1,2,3

Preconditions:

1. Call/VTC/Telecon request is received or call/VTC/Telecon invitation is sent.

Postconditions:

1. Call log is updated with call information.

Contract: C57 ring() :void

Cross Reference: UC-1

Preconditions:

1. A call has been initiated.

Postconditions:

1. Ring wav file is played.

Contract: C58 cancelTelecon() :void

Cross Reference: UC-2b

Preconditions:

1. An instance ***t*** of Telecon exists.

Postconditions:

1. ***s.setState(teleconTerminated).***
2. ***t*** is destroyed.

Contract: C59 conductTelecon() :void

Cross Reference: UC-3a Accept Telecon Invite

Preconditions:

1. An instance ***t*** of Telecon exists.
2. isOnCall(true)

Postconditions:

1. ***s.getState(teleconInProgress).***

Contract: C60 cancelVTC() :void

Cross Reference: UC-2a VTC Invite

Preconditions:

1. An instance ***v*** of VTC exists.
2. ***isOnCall(true).***

Postconditions:

1. ***s.setState(vtcTerminated).***

2. ***v*** is destroyed.

Contract: C61 conductVTC() :void

Cross Reference: UC-3a Accept VTC Invite

Preconditions:

1. An instance ***v*** of VTC exists.
2. isOnCall(true)

Postconditions:

1. s.getState(vtcInProgress).

H. STATEMACHINE STATE DIAGRAM

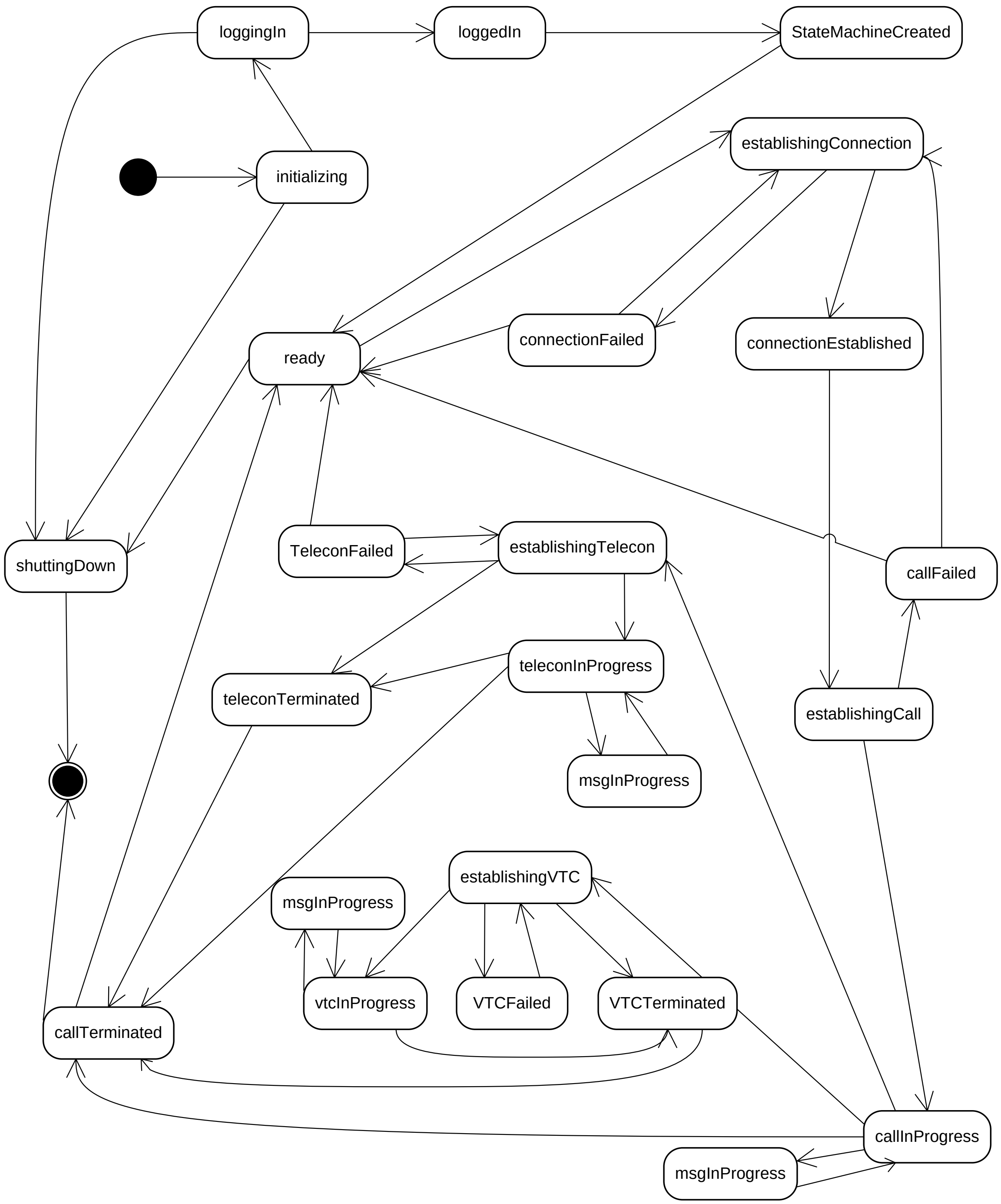

Figure 45. VoIPNET State Diagram.

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V. SOFTWARE TESTING PLAN

A. INTRODUCTION

1. Objectives

This section describes, at a high level, the scope, approach, resources, and schedule of the testing activities. It provides a concise summary of the test plan objectives, the products to be delivered, major work activities, major work products, major milestones, required resources, and master high-level schedules, and effort requirements.

2. Testing Strategy

Testing is the process of analyzing a software item to detect the differences between existing and required conditions and to evaluate the features of the software item.

3. Scope

This section specifies the plans for producing both scheduled and unscheduled updates to the Software Test Plan (change management). Methods for distribution of updates will be specified along with version control and configuration management requirements. Testing will be performed at several points in the life cycle as the product is constructed. Testing is a very 'dependent' activity. As a result, test planning is a continuing activity performed throughout the system development life cycle. Test plans will be developed for each level of product testing (IOT&E/OT&E). All updates to the test plan must be approved by the Test Coordinator and distributed to all testing agencies. If an impact to the Project Schedule is anticipated then the change must be proposed to all stakeholders for consideration and approval. The Project Manager will update the schedule and distribute in a timely fashion to all stakeholders.

4. Reference Material

[1] "VoIPNET Thesis Proposal." Reiche.

[2] "VoIPNET Vision Document." Reiche.

[3] "VoIPNET Requirements Specification Document." Reiche.

[4] "VoIPNET Design Specification Document." Reiche.

[5] "IEEE Software Test Plan Template." IEEE 829-1998 Format.

5. Definitions and Acronyms

See Glossary.

B. TEST ITEMS

This section outlines the testing to be performed for VoIPNET.

1. Component Testing

The RAT, VIC, GraphicalUA, NAD, and Rider components will be tested for suitability and functional correctness. Component testing will be conducted during IOT&E testing.

2. Integration Testing

The integration of RAT, VIC, Graphical UA, NAD and Rider will be tested. Integration testing will be conducted during IOT&E testing.

3. Recovery Testing

A fully integrated VoIPNET application will be tested under network and peripheral failure conditions. Recovery Testing will be conducted during both IOT&E and OT&E testing.

4. Performance Testing

A fully integrated VoIPNET prototype will be tested under multiple network configurations and capacity. Performance Testing will occur during OT&E testing.

C. APPROACH

This section describes the overall approaches to testing. Testing will composed of two major test evolutions, IOT&E and OT&E. The Initial Operational Test and Evaluation (IOT&E) will be conducted in a laboratory environment, at Naval Postgraduate School, on a 100Mbs network with three (3) users. During IOT&E the Component, Integration, Recovery, and Performance tests will be conducted. Immediately upon completion of Testing the Analysis of all test data will begin. Prior to the execution of OT&E the IOT&E report must be completed and reviewed. Any recommended changes to the OT&E plan must then be submitted to the Test Coordinator and or Project Manager. All approved changes must be installed prior to the commencement of OT&E.

Operational Test and Evaluation (OT&E) will be conducted, at Marine Corps Tactical Systems Support Activity (MCTSSA), in both a laboratory and field environment on a low bandwidth (<150Kbs) network (EPLRS Radio Network). During OT&E the Recovery and Performance tests will be conducted. Upon completion of OT&E all test results will be analyzed and the OT&E Report will be generated. Following the review of the OT&E report, the IOT&E and OT&E reports will be included and summarized as part of the Comprehensive Test Report.

D. PROCEDURAL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

1. Test Suspension Criteria

The criteria used to suspend all or a portion of the testing activity associated with a test item is as follows:

Network becomes unstable.

If two or more User Agents drop off of the network.

Software Testing Tools are inoperable.

Improper Activity Log Entry prior to commencement of testing event.

2. Test Resumption Criteria

The criteria used to resume all or a portion of the testing activity associated with a suspended test item is as follows:

The Network Administrator must certify the network is stable and provide the Test Coordinator with the Network Configuration.

The Test Coordinator must enter the corrective actions taken in the Test Activity Log and create a new entry for the next test.

3. Test Result Approval Criteria

The test result approval process requires that all test results be verified in writing by the Network Administrator, Test Coordinator, and Test Approval Authority. Each test result must be accompanied by the network configuration and baseline capacity measurements.

E. TESTING PROCESS

This section identifies the methods and criteria used in the performance of test activities. It defines the specific tests and procedures for each type of test. In addition, it provides the detailed criteria for evaluating test results.

1. Primary Tasks

This section identifies the primary testing tasks during the test process.

- Conduct a Comprehensive Test of VoIPNET and provide deployment recommendations.
- Conduct IOT&E Testing in order to determine suitability, ensure integration quality, record fault tolerance and compile performance characteristics. Detailed Test Plans can be found in Section I of this chapter.
    - Conduct Component Testing
    - Conduct Integration Testing
    - Conduct Recovery Testing
    - Conduct Performance Testing

- Conduct OT&E Testing in order to record fault tolerance and compile performance characteristics on an EPLRS network. Detailed OT&E Test Plans can be found in section J of this chapter.
    - Conduct Recovery Testing
    - Conduct Performance Testing

2. Supplementary Tasks

Identifies all tasks, skills and dependencies necessary to prepare for and conduct testing activities. Supplementary tasks include:

Software Tool Identification: identify the tools required to measure performance metrics.

Software Packaging: consolidate all testable software packages, supporting components, and test tools into an easily deployable media.

Facilities Coordination: coordinate the physical location for testing and storage of hardware.

Special Skills required: An EPLRS network manager is required to install, operate and maintain the test network platform.

3. Responsibilities

Identify the groups responsible for managing, designing, preparing, executing, witnessing, checking, and resolving test activities. These groups may include developer, testers, operations staff, technical support staff, data administration staff, and user staff. The NPS Test Coordinator has overall responsibility for the synthesis of all planning documents, test reports, and recommendations. In addition, the NPS Test Coordinator will supervise and/or conduct the preparation and execution of all test activities.

4. Resources

Identifies the resources allocated for the performance of testing tasks. Identifies the organizational elements or individuals responsible for performing testing activities. Assigns specific responsibilities and specifies resources by category.

Testing Responsibility

Initial Operational Test and Evaluation (IOT&E)- IOT&E will be conducted at the Naval Post Graduate School under the supervision of the NPS Testing Coordinator.

Operational Test and Evaluation (OT&E)- OT&E will be conducted at MCTSSA, Camp Pendleton, CA, under the joint supervision of the NPS Test Coordinator and the MCTSSA EPLRS Coordinator. The below resources and the corresponding responsible authority are required to conduct the testing.

Resource Categories

Infrastructure:

Storage/Lab Space- MCTSSA

Operator Support- MCTSSA

Hardware:

EPLRS Systems- MCTSSA

Break-out Boxes- MCTSSA

Net Manager IOW and Software- MCTSSA

Personal Computers- NPS

Routers- NPS

USB Hub- NPS

USB Headsets- NPS

USB WebCams- NPS

Cat-5- NPS

Software:

VoIPNET Software package- NPS

Robust Audio Tool Software package- NPS

Video Conference Software package- NPS

Rider Software Package- NPS

Net Activity Diagram- NPS

MCTSSA Support Coordinator: Captain Jeffery Wrobel/USMC, MCTSSA EPLRS Coordination Officer.

MCTSSA Network Administrator: Mr. Pedro Zenquis

NPS Testing Coordinator Captain Charles P. Reiche, Jr/USMC

5. Schedule

Identifies the high level schedule for each testing task. Establishes specific milestones for: 1) initiating and completing each type of test activity, 2) for the completion of the comprehensive test plan, and 3) for the delivery of test reports. Estimates of the time required to do each test activity are provided in the respective test plan.

High-Level Testing Milestones:

- Comprehensive Testing Started
    - IOT&E Testing Started
    - IOT&E Testing Complete
    - OT&E Testing Started
    - OT&E Testing Complete
- Comprehensive Test Complete

6. Test Deliverables

This section identifies the deliverable documents from the test process.

IOT&E Testing Activity Report

OT&E Testing Activity Report

Component Testing Activity Report

Integration Testing Activity Report

Recovery Testing Activity Report

Performance Testing Activity Report

Comprehensive Testing Summary Report

F. ENVIRONMENTAL REQUIREMENTS

1. Hardware

The computer and network requirements to complete testing activities include:

EPLRS Radio (3-5) and associated SL-3 Components.

Net Manager Intelligence Operations Workstation (1)

PC-RT connection box (Breakout Box) (3-5)

Windows configured Personal Computers (3-5)

USB Plantronics DSP-400 VoIP headset (3-5)

USB WebCamera (3-5)

Belkin USB Hub (1)

Belkin/Cisco 4-6 port Router (1)

Cat-5 w/rj-45 connectors (10 @ 20’ length)

2. Software

The Software requirements to complete testing activities include:

VoIPNET (MJSIP GraphicalUA)

Robust Audio Tool v4.2.24

Video Component v2.8ucl1.1.6

EPLRS Net Manager (ENM)

Rider v11.50.0.45560

Net Activity Diagram v2.3.0.293

3. Security

Testing environment security and asset protection requirements include a securable room for overnight storage of hardware.

4. Tools

There are no special software tools, techniques and/or methodologies employed in the testing effort.

5. Publications

Identify the documents and or publications required to conduct testing.

EPLRS RS Technical Manuals

EPLRS Network Manager Technical Manual

EPLRS Network Planner Technical Manual

MCTSSA EPLRS Network Standard Operating Procedure

Robust Audio Tool (RAT) Users Manual

Video Component (VIC) Users Manual

Codec Comparison Charts

VoIPNET Comprehensive Test Plan

6. Risks and Assumptions

Identifies significant resource constraints, such as test item availability, testing facility availability, test resource availability, and time constraints. See the Risk Management Plan in Chapter IV. for detailed Risk assessments.

MCTSSA Test Facility availability is only available for one business week. We have limited influence over test date adjustments.

MCTSSA resource provisions are under the control of the EPLRS representative. Assets may be reallocated to support MCTSSA efforts.

Testing must be completed by May 15, 2007, to allow for the synthesis and analysis of test reports.

G. CHANGE MANAGEMENT PROCEDURE

All changes to the test plan require approval of the corresponding Approval Authority as identified in Section H of this document. All approved changes will be incorporated into the corresponding individual test plan and the Comprehensive test plan. Changes will be submitted to all participants in the testing process.

H. TEST PLAN APPROVAL AUTHORITY

The below personnel are authorized to approve changes to the VoIPNET Comprehensive Test Plan:

Captain Charles P. Reiche, Jr/USMC

The below personnel are authorized to approve changes to the VoIPNET Individual Test Plans:

Captain Charles P. Reiche, Jr/USMC

I. IOT&E TEST PLAN

1. Component Test Plan

VERSION: 1.1

DATE: April 7, 2007

TEST COORDINATOR: Capt C.P. Reiche. JR

PURPOSE

The purpose of Component Testing is to ensure that each component meets the functional and non-functional requirements as defined in the Requirements Specification Document.

ITEMS TO BE TESTED

The RAT, VIC, GraphicalUA, NAD, and Rider components will be tested for suitability and functional correctness. Component testing will be conducted during IOT&E testing.

FEATURES TO BE TESTED

This section identifies all features and specific combinations of features that will be tested under each test plan.

GraphicalUA: The Call, Hangup, Directory Services, and Configuration File features will be tested.

RAT: Talk checkbox, Options>Transmission>Audio Encoding, Options>Audio>Audio Device, Options>Audio>Sample Rate, Options>Audio>Channels, and Options>Security features will be tested.

VIC: The Menu>Transmission Rate Slider, Menu>Frame Rate Slider, Menu>Transmit/Release Button, Menu>Global Statistics, Menu>Members, and Image View options will be tested.

NAD: The General, Appearance, Filters, and Notifications Options will be tested for suitability.

Rider: The Response, Bandwidth, VoIP, and Traceroute features will be tested for suitability.

FEATURES NOT TO BE TESTED

All features that will not be tested are identified.

GraphicalUA: None.

RAT: Options>Channel Encoding, Options>Reception, Options>Interface, Options>Codec Mapping, and Reception Quality Matrix.

VIC: Menu>Encoder, Menu>Display, and Menu>Session.

NAD: None.

Rider: None.

MANAGEMENT AND TECHNICAL APPROACH

Each component will be tested independently for suitability and functionality during IOT&E.

PASS / FAIL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

VoIPNET: The call feature must connect to the correct addressee. The Hangup feature must disconnect the current call and return both UA’s (User Agents) to the listen state. The Directory Services feature must place the selected sip address into the dial text field. The Configuration File feature must configure the Graphical US with the correct settings. The contact list feature must install the correct contact list as identified in the configuration file.

RAT: The CODEC selection feature correctly replaces the current CODEC with the selected CODEC. The Audio Options feature correctly adjusts the audio configuration settings.

VIC: The Transmission Rate Slider correctly adjusts the transmission rate. The Frame Rate Slider correctly adjusts the video frame rate. The view image option correctly adjusts the viewing window.

NAD: The settings feature correctly adjusts the General, Appearance, Filters, and Notifications settings.

Rider: The Response feature provides accurate response times as verified with a standard ping test. The Bandwidth feature provides bandwidth estimates comparable to known standard technical capabilities. The VoIP test feature provides test results that coincide with observable performance characteristics. The Traceroute feature returns a traceroute that matches the physical configuration of the network.

INDIVIDUAL ROLES AND RESPONSIBILITIES

The NPS Test Coordinator is responsible for application collection, Laboratory establishment, Testing and Report Generation.

MILESTONES

- Tested Software installed
- Component Testing Started
    - GraphicalUA Test Started
    - GraphicalUA Test Complete
    - RAT Test Started
    - RAT Test Complete
    - VIC Test Started
    - VIC Test Complete
    - NAD Test Started
    - NAD Test Complete
    - Rider Test Started
    - Rider Test Complete
    - Test Result Analysis Started
    - Test Result Analysis Complete
    - Test Report Generation Started
    - Test Report Generation Complete
- Component Testing Complete

SCHEDULES

Day 1- Component Testing

RISK ASSUMPTIONS AND CONSTRAINTS

See Chapter IV.

2. Integration Test Plan

VERSION: 1.2

DATE: April 7, 2007

TEST COORDINATOR: Capt C.P. Reiche. JR

PURPOSE

The purpose of Integration Testing is to ensure that the individual components perform as expected when fully integrated into VoIPNET.

ITEMS TO BE TESTED

The integration of RAT, VIC, Graphical UA, NAD and Rider will be tested. Integration testing will be conducted during IOT&E testing.

FEATURES TO BE TESTED

The Component Tests will be executed again after the individual components are integrated into a single application (VoIPNET).

FEATURES NOT TO BE TESTED

GraphicalUA: None.

RAT: See Component Test plan.

VIC: See Component Test Plan.

NAD: None.

Rider: None.

MANAGEMENT AND TECHNICAL APPROACH

Incremental development/testing will be utilized. As each component is integrated into the application, regression testing will done to ensure all features continue to function as required. Integration/Integration testing will occur in the following order: RAT, VIC, NAD, Rider.

PASS / FAIL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

Each component must pass the respective Component Test following integration into the VoIPNET application without causing errors in the VoIPNET application and the integrated components.

INDIVIDUAL ROLES AND RESPONSIBILITIES

See Section G.

MILESTONES

- 512kbps network established
- Integration Testing Started
    - GraphicalUA Test Started
    - GraphicalUA Test Complete
    - RAT Test Started
    - RAT Test Complete
    - VIC Test Started
    - VIC Test Complete
    - NAD Test Started
    - NAD Test Complete
    - Rider Test Started
    - Rider Test Complete
    - Test Result Analysis Started
    - Test Result Analysis Complete
    - Test Report Generation Started
    - Test Report Generation Complete
- Integration Testing Complete

SCHEDULES

Day 1- Integration Testing

RISK ASSUMPTIONS AND CONSTRAINTS

See Chapter IV.

3. Recovery Test Plan

VERSION: 1.1

DATE: April 7, 2007

TEST COORDINATOR: Capt C.P. Reiche. JR

PURPOSE

The purpose of Recovery Testing is to evaluate VoIPNET's ability to recover from a variety of potential system errors.

ITEMS TO BE TESTED

A fully integrated VoIPNET application will be tested under network and peripheral failure conditions. Recovery Testing will be conducted during both IOT&E and OT&E testing.

FEATURES TO BE TESTED

The Dropped call recovery, Dropped VTC recovery and Network failure procedures will be tested.

FEATURES NOT TO BE TESTED

Nat, Firewall Traversal.

MANAGEMENT AND TECHNICAL APPROACH

The test technician will induce peripheral, Call, VTC and/or network faults. If the application fails to handle the errors in a graceful manner, the condition will be logged and a recommendation for correction must be provided in the Testing Report.

PASS / FAIL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

A dropped call/VTC should return both UA's to the listen state without error. A terminated VTC should not terminate the connected voice call. Network errors must not cause application errors aside from those expected communication errors due to network connectivity.

INDIVIDUAL ROLES AND RESPONSIBILITIES

See Section G.

MILESTONES

- 512kbps network established
- Recovery Testing Started
    - Peripheral Failure Tests Started
    - Peripheral Failure Tests Complete
    - Network Failure Test Started
    - Network Failure Test Complete
    - Call Failure Test Started
    - Call Failure Test Complete
    - VTC Failure Test Started
    - VTC Failure Test Complete
    - Test Result Analysis Started
    - Test Result Analysis Complete
    - Test Report Generation Started
    - Test Report Generation Complete
- Recovery Testing Complete

SCHEDULES

Day 2- Recovery testing

RISK ASSUMPTIONS AND CONSTRAINTS

See Chapter IV.

4. Performance Test Plan

VERSION: 1.2

DATE: May 7, 2007

TEST COORDINATOR: Capt C.P. Reiche. JR

PURPOSE

The purpose of the Performance Test is to evaluate VoIPNET's performance characteristics under various application and network configurations, in order to provide the optimal network and application configuration deployment recommendations.

ITEMS TO BE TESTED

A fully integrated VoIPNET prototype will be tested under multiple network configurations and capacity. Performance Testing will occur during OT&E testing.

FEATURES TO BE TESTED

The P2P Voice, VTC, VTC Conference, Call, and Hangup features will be tested.

FEATURES NOT TO BE TESTED

None.

MANAGEMENT AND TECHNICAL APPROACH

Performance will be tested and measured during IOT&E on a high bandwidth network as well as during OT&E on a low bandwidth network. Various CODEC, Transmission Rate, and Frame rate settings will be tested for both Low and High Bandwidth Networks.

PASS / FAIL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

Each Call must return a Quality of Service (QOS) Average –> Good at 100 Kbps, Good –> Very Good at 256 Kbps, and Very Good at 512 Kbps .

Each VTC must return a QOS Average -> Good at < 100kbs, Good -> Very Good at < 256kbs, and Very Good at 512Kbs.

INDIVIDUAL ROLES AND RESPONSIBILITIES

See Section G.

MILESTONES

- Network Established
- Performance Testing Started
    - Audio Test Started
    - 512kbps Network Emulation Established
        - Audio Package 1 Started
        - Audio Package 1 Complete
    - 256kbps Network Emulation Established
        - Audio Package 2 Started
        - Audio Package 2 Complete
    - 100kbps Network Emulation Established
        - Audio Package 3 Started
        - Audio Package 3 Complete
- Audio Test Result Analysis Started
- Audio Test Result Analysis Complete
- Audio Test Report Generation Started
- Audio Test Report Generation Complete
- Audio Test Complete
- VTC Test Started
    - 512kbps Network Emulation Established
        - VTC Package 4 Started
        - VTC Package 4 Complete
    - 256kbps Network Emulation Established
        - VTC Package 5 Started
        - VTC Package 5 Complete

        - 100kbps Network Emulation Established
            - VTC Package 6 Started
            - VTC Package 6 Complete
        - VTC Test Result Analysis Started
        - VTC Test Result Analysis Complete
        - VTC Test Report Generation Started
        - VTC Test Report Generation Complete
        - VTC Test Complete
- Performance Test Result Analysis Started
- Performance Test Result Analysis Complete
- Performance Test Report Generation Started
- Performance Test Report Generation Complete
- Performance Testing Complete

SCHEDULES

Day 2- Audio Package 1, 2, and 3

Day 3- Video Package 4, Video Package 5

Day 4- Video Package 6

RISK ASSUMPTIONS AND CONSTRAINTS

See Chapter IV.

J. OT&E TEST PLAN

1. Recovery Test Plan

VERSION: 1.1

DATE: April 7, 2007

TEST COORDINATOR: Capt C.P. Reiche. JR

PURPOSE

The purpose of Recovery Testing is to evaluate VoIPNET’s ability to recover from a variety of potential system errors.

ITEMS TO BE TESTED

A fully integrated VoIPNET application will be tested under network and peripheral failure conditions. Recovery Testing will be conducted during both IOT&E and OT&E testing.

FEATURES TO BE TESTED

The Dropped call recovery, Dropped VTC recovery and Network failure procedures will be tested.

FEATURES NOT TO BE TESTED

Nat, Firewall Traversal.

MANAGEMENT AND TECHNICAL APPROACH

The test technician will induce peripheral, Call, VTC and/or network faults. If the application fails to handle the errors in a graceful manner, the condition will be logged and a recommendation for correction must be provided in the Testing Report.

PASS / FAIL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

A dropped call/VTC should return both UA's to the listen state without error. A terminated VTC should not terminate the connected voice call. Network errors must not cause application errors aside from those expected communication errors due to network connectivity.

INDIVIDUAL ROLES AND RESPONSIBILITIES

See Section G.

MILESTONES

- 4 node EPLRS network established
- Recovery Testing Started
    - Peripheral Failure Test Started

    - Peripheral Failure Test Complete
    - Network Failure Test Started
    - Network Failure Test Complete
    - Call Failure Test Started
    - Call Failure Test Complete
    - VTC Failure Test Started
    - VTC Failure Test Complete
    - Test Result Analysis Started
    - Test Result Analysis Complete
    - Test Report Generation Started
    - Test Report Generation Complete
- Recovery Testing Complete

SCHEDULES

Day 4- Recovery Test.

RISK ASSUMPTIONS AND CONSTRAINTS

See Chapter IV.

2. Performance Test Plan

VERSION: 1.2

DATE: May 13, 2007

TEST COORDINATOR: Capt C.P. Reiche. JR

PURPOSE

The purpose of the Performance Test is to evaluate the VoIPNET prototype’s performance characteristics under various application and network configurations, in order to provide the optimal network and application configuration deployment recommendations and Requirement updates for development.

ITEMS TO BE TESTED

A fully integrated VoIPNET prototype will be tested under multiple network configurations and capacity. Performance Testing will occur during OT&E testing.

FEATURES TO BE TESTED

The P2P Voice, VTC Conference, Call, and Hangup features will be tested.

FEATURES NOT TO BE TESTED

None.

MANAGEMENT AND TECHNICAL APPROACH

A two phased approach to testing will be used. Configuration testing will be done to identify two candidate network configurations for follow on application testing. Secondly, various software CODEC, Transmission Rate, and Frame rate settings will be tested for both Low and High Bandwidth Networks. Once a suitable network is identified the configuration will remain unchanged. All configuration changes will then be VoIPNET software based.

PASS / FAIL CRITERIA

The criteria used to determine whether the tested item has passed/failed testing.

Each Call must return a Quality of Service (QOS) Average –> Good at < 100 Kbps.

Each VTC must return a QOS Average -> Good at < 100kbs.

INDIVIDUAL ROLES AND RESPONSIBILITIES

See Section G.

MILESTONES

- 4 node EPLRS Network Established

- Configuration Testing Started
- Configuration Testing Complete
- Performance Testing Started
    - Audio Test Started
        - Audio Package 1 Started
        - Audio Package 1 Complete
        - Audio Test Result Analysis Started
        - Audio Test Result Analysis Complete
        - Audio Test Report Generation Started
        - Audio Test Report Generation Complete
    - Audio Test Complete
    - VTC Test Started
        - VTC Package 1 Started
        - VTC Package 1 Complete
        - VTC Test Result Analysis Started
        - VTC Test Result Analysis Complete
        - VTC Test Report Generation Started
        - VTC Test Report Generation Complete
    - VTC Test Complete
    - Performance Test Result Analysis Started
    - Performance Test Result Analysis Complete
    - Performance Test Report Generation Started
    - Performance Test Report Generation Complete
- Performance Testing Complete

SCHEDULES

Day 1- Configuration Testing

Day 2- Configuration Testing

Day 3- Configuration Testing

Day 4- Audio Package 1, VTC Package 1, Recovery Test

RISK ASSUMPTIONS AND CONSTRAINTS

See Chapter IV.

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VI. SOFTWARE TEST REPORT

A. IOT&E REPORT

1. Test Network Overview

Network Emulator Software (Shunra) was utilized to mimic various Tactical Networks. Shunra possesses the ability to regulate bandwidth, inject packet loss and force latency into a network, in order to emulate actual network conditions. The test was conducted on an emulated 512kbps, 256kbps and 128kbps networks. The EPLRS CSMA, zero hop network profile (56msec Latency/1-2% dropped packets) was used to get preliminary data on prototype suitability.

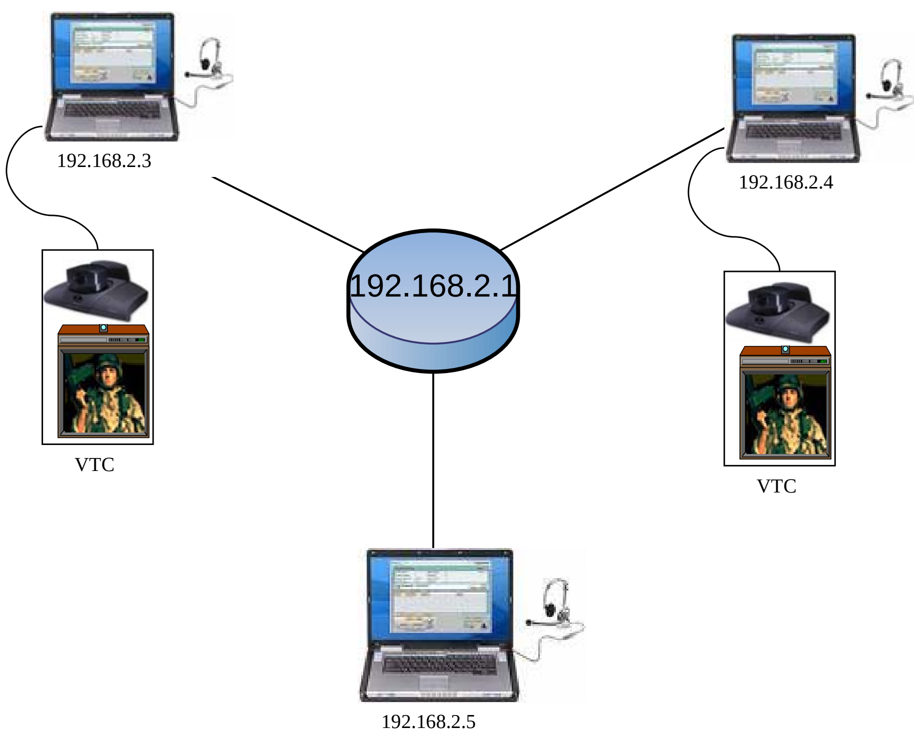

Figure 46. IOT&E Test Architecture.

2. Test Data

There were no network difficulties. The Shunra network emulation software performed as expected.

Component Test Report

		Configuration	**Observation**	
Package	**ID#**	**Description**	**Pass/Fail**	**Activity Log**
UA	1	Call Test	P	
UA	2	Hangup Test	P	
UA	3	Directory Services Test	P	
UA	4	Contact List Test	P	
UA	5	Configuration File Test	P	
Package	**ID#**	**Description**	**Pass/Fail**	**Activity Log**
RAT	1	Talk Checkbox	P	
RAT	2	Options>Transmission>	P	
RAT	3	…Audio Encoding	P	
RAT	4	Options>Audio	P	
RAT	5	>Audio Device	P	
RAT	6	>Sample Rate	P	
RAT	7	>Channels	P	
RAT	8	Options>Security	P	
Package	**ID#**	**Description**	**Pass/Fail**	**Activity Log**
VIC	1	Menu>	P	
VIC	2	>Transmit/Release	P	
VIC	3	>TX Rate Control	P	
VIC	4	>Frame Rate Control	P	
VIC	5	>Global Statistics	P	
VIC	6	>Members	P	
Package	**ID#**	**Description**	**Pass/Fail**	**Activity Log**
NAD	1	General Settings Test	P	
NAD	2	Appearance Settings Test	P	
NAD	3	Filters Settings Test	P	
NAD	4	Notifications Settings Test	P	
Package	**ID#**	**Description**	**Pass/Fail**	**Activity Log**
Rider	1	Response feature Test	**P**	
Rider	2	Bandwidth feature Test	**P**	
Rider	3	VoIP feature Test	**P**	
Rider	4	Traceroute feature Test	**P**	

512Kbps Baseline Data Measurements

Bandwidth: 512Kbps

Latency: 50ms

Packet Loss: 2%

256Kbps Baseline Data Measurements

Bandwidth: 512Kbps

Latency: 50ms

Packet Loss: 2%

Audio Test Report

Pkg	ID#	Network	CODEC	Sample Rate	Bandwidth	QOS	Pass/ Fail
1	1	512Kbps	Linear-16	8Khz	128	VERY GOOD	P
1	2	512Kbps	PCMU	8Khz	64	VERY GOOD	P
1	3	512Kbps	PCMA	8Khz	64	VERY GOOD	P
1	4	512Kbps	G.726-40	8Khz	40	VERY GOOD	P
1	5	512Kbps	G.726-32	8Khz	32	GOOD	P
1	6	512Kbps	G.726-24	8Khz	24	GOOD	P
1	7	512Kbps	G.726-16	8Khz	16	AVERAGE	P
1	8	512Kbps	DVI	8Khz	32	VERY GOOD	P
1	9	512Kbps	VDVI	8Khz	32	VERY GOOD	P
1	10	512Kbps	GSM	8Khz	13.2	VERY GOOD	P
1	11	512Kbps	LPC	8Khz	5.6	POOR	F

Pkg	ID#	Network	CODEC	Sample Rate	Bandwidth	QOS	Pass/ Fail
2	1	256Kbps	Linear-16	8Khz	128	GOOD	P
2	2	256Kbps	PCMU	8Khz	64	VERY GOOD	P
2	3	256Kbps	PCMA	8Khz	64	VERY GOOD	P
2	4	256Kbps	G.726-40	8Khz	40	VERY GOOD	P
2	5	256Kbps	G.726-32	8Khz	32	GOOD	P
2	6	256Kbps	G.726-24	8Khz	24	GOOD	P
2	7	256Kbps	G.726-16	8Khz	16	AVERAGE	P
2	8	256Kbps	DVI	8Khz	32	VERY GOOD	P
2	9	256Kbps	VDVI	8Khz	32	VERY GOOD	P
2	10	256Kbps	GSM	8Khz	13.2	VERY GOOD	P
2	11	256Kbps	LPC	8Khz	5.6	POOR	F

Pkg	ID#	Network	CODEC	Sample Rate	Bandwidth	QOS	Pass/ Fail
3	1	128Kbps	Linear-16	8Khz	128	GOOD	P
3	2	128Kbps	PCMU	8Khz	64	VERY GOOD	P
3	3	128Kbps	PCMA	8Khz	64	VERY GOOD	P
3	4	128Kbps	G.726-40	8Khz	40	VERY GOOD	P
3	5	128Kbps	G.726-32	8Khz	32	GOOD	P
3	6	128Kbps	G.726-24	8Khz	24	GOOD	P

Pkg	ID#	Network	CODEC	Sample Rate	Bandwidth	QOS	Pass/ Fail
3	8	128Kbps	DVI	8Khz	32	VERY GOOD	P
3	9	128Kbps	VDVI	8Khz	32	VERY GOOD	P
3	10	128Kbps	GSM	8Khz	13.2	VERY GOOD	P
3	11	128Kbps	LPC	8Khz	5.6	POOR	F

VTC Test Report

Pkg	ID#	Network	CODEC	Sample Rate	TX/Frame Rate (Kbps/fps)	Actual fps	QOS	Pass/ Fail
4	1	512Kbps	Linear-16	8Khz	512/15	11	VG	P
4	2	512Kbps	Linear-16	8Khz	512/8	8	VG	P
4	3	512Kbps	Linear-16	8Khz	512/4	4	GOOD	P
4	4	512Kbps	Linear-16	8Khz	128/15	4	GOOD	P
4	5	512Kbps	Linear-16	8Khz	128/8	4	GOOD	P
4	6	512Kbps	Linear-16	8Khz	128/4	4	GOOD	P
4	7	512Kbps	Linear-16	8Khz	64/4	2-3	AVG	P
4	8	512Kbps	PCMU	8Khz	512/15	12	VG	P
4	9	512Kbps	PCMU	8Khz	512/8	8	VG	P
4	10	512Kbps	PCMU	8Khz	512/4	4	GOOD	P
4	11	512Kbps	PCMU	8Khz	128/15	4	GOOD	P
4	12	512Kbps	PCMU	8Khz	128/8	4	GOOD	P
4	13	512Kbps	PCMU	8Khz	128/4	4	GOOD	P
4	14	512Kbps	PCMU	8Khz	64/4	2-3	AVG	P
4	15	512Kbps	PCMA	8Khz	512/15	12	VG	P
4	16	512Kbps	PCMA	8Khz	512/8	8	VG	P
4	17	512Kbps	PCMA	8Khz	512/4	4	GOOD	P
4	18	512Kbps	PCMA	8Khz	128/15	4	GOOD	P
4	19	512Kbps	PCMA	8Khz	128/8	4	GOOD	P
4	20	512Kbps	PCMA	8Khz	128/4	4	GOOD	P
4	21	512Kbps	PCMA	8Khz	64/4	2-3	AVG	P
4	22	512Kbps	G.726-40	8Khz	512/15	12	VG	P
4	23	512Kbps	G.726-40	8Khz	512/8	8	VG	P
4	24	512Kbps	G.726-40	8Khz	512/4	4	GOOD	P
4	25	512Kbps	G.726-40	8Khz	128/15	4	GOOD	P
4	26	512Kbps	G.726-40	8Khz	128/8	4	GOOD	P
4	27	512Kbps	G.726-40	8Khz	128/4	4	GOOD	P
4	28	512Kbps	G.726-40	8Khz	64/4	2-3	AVG	P
4	29	512Kbps	G.726-32	8Khz	512/15	12	VG	P
4	30	512Kbps	G.726-32	8Khz	512/8	8	VG	P
4	31	512Kbps	G.726-32	8Khz	512/4	4	GOOD	P
4	32	512Kbps	G.726-32	8Khz	128/15	4	GOOD	P
4	33	512Kbps	G.726-32	8Khz	128/8	4	GOOD	P
4	34	512Kbps	G.726-32	8Khz	128/4	4	GOOD	P
4	35	512Kbps	G.726-32	8Khz	64/4	2-3	AVG	P

Pkg	ID#	Network	CODEC	Sample Rate	TX/Frame Rate (Kbps/fps)	Actual fps	QOS	Pass/ Fail
4	36	512Kbps	G.726-24	8Khz	512/15	12	VG	P
4	37	512Kbps	G.726-24	8Khz	512/8	8	VG	P
4	38	512Kbps	G.726-24	8Khz	512/4	4	GOOD	P
4	39	512Kbps	G.726-24	8Khz	128/15	4	GOOD	P
4	40	512Kbps	G.726-24	8Khz	128/8	4	GOOD	P
4	41	512Kbps	G.726-24	8Khz	128/4	4	GOOD	P
4	42	512Kbps	G.726-24	8Khz	64/4	2-3	AVG	P
4	43	512Kbps	G.726-16	8Khz	512/15	12	VG	P
4	44	512Kbps	G.726-16	8Khz	512/8	8	VG	P
4	45	512Kbps	G.726-16	8Khz	512/4	4	GOOD	P
4	46	512Kbps	G.726-16	8Khz	128/15	4	GOOD	P
4	47	512Kbps	G.726-16	8Khz	128/8	4	GOOD	P
4	48	512Kbps	G.726-16	8Khz	128/4	4	GOOD	P
4	49	512Kbps	G.726-16	8Khz	64/4	2-3	AVG	P
4	50	512Kbps	DVI	8Khz	512/15	12	VG	P
4	51	512Kbps	DVI	8Khz	512/8	8	VG	P
4	52	512Kbps	DVI	8Khz	512/4	4	GOOD	P
4	53	512Kbps	DVI	8Khz	128/15	4	GOOD	P
4	54	512Kbps	DVI	8Khz	128/8	4	GOOD	P
4	55	512Kbps	DVI	8Khz	128/4	4	GOOD	P
4	56	512Kbps	DVI	8Khz	64/4	2-3	AVG	P
4	57	512Kbps	VDVI	8Khz	512/15	12	VG	P
4	58	512Kbps	VDVI	8Khz	512/8	8	VG	P
4	59	512Kbps	VDVI	8Khz	512/4	4	GOOD	P
4	60	512Kbps	VDVI	8Khz	128/15	4	GOOD	P
4	61	512Kbps	VDVI	8Khz	128/8	4	GOOD	P
4	62	512Kbps	VDVI	8Khz	128/4	4	GOOD	P
4	63	512Kbps	VDVI	8Khz	64/4	2-3	AVG	P
4	64	512Kbps	GSM	8Khz	512/15	12	VG	P
4	65	512Kbps	GSM	8Khz	512/8	8	VG	P
4	66	512Kbps	GSM	8Khz	512/4	4	GOOD	P
4	67	512Kbps	GSM	8Khz	128/15	4	GOOD	P
4	68	512Kbps	GSM	8Khz	128/8	4	GOOD	P
4	69	512Kbps	GSM	8Khz	128/4	4	GOOD	P
4	70	512Kbps	GSM	8Khz	64/4	2-3	AVG	P
4	71	512Kbps	LPC	8Khz	512/15	12	POOR	F
4	72	512Kbps	LPC	8Khz	512/8	8	POOR	F
4	73	512Kbps	LPC	8Khz	512/4	4	POOR	F
4	74	512Kbps	LPC	8Khz	128/15	4	POOR	F
4	75	512Kbps	LPC	8Khz	128/8	4	POOR	F
4	76	512Kbps	LPC	8Khz	128/4	4	POOR	F
4	77	512Kbps	LPC	8Khz	64/4	2-3	POOR	F
5	1	256Kbps	Linear-16	8Khz	256/15	8	POOR	P
5	2	256Kbps	Linear-16	8Khz	256/8	8	POOR	P

Pkg	ID#	Network	CODEC	Sample Rate	TX/Frame Rate (Kbps/fps)	Actual fps	QOS	Pass/ Fail
5	4	256Kbps	Linear-16	8Khz	128/15	4	AVG	P
5	5	256Kbps	Linear-16	8Khz	128/8	4	AVG	P
5	6	256Kbps	Linear-16	8Khz	128/4	4	AVG	P
5	7	256Kbps	Linear-16	8Khz	64/4	2-3	GOOD	P
5	8	256Kbps	PCMU	8Khz	256/15	8	BAVG	P
5	9	256Kbps	PCMU	8Khz	256/8	8	AVG	P
5	10	256Kbps	PCMU	8Khz	256/4	4	AVG	P
5	11	256Kbps	PCMU	8Khz	128/15	4	AVG	P
5	12	256Kbps	PCMU	8Khz	128/8	4	AVG	P
5	13	256Kbps	PCMU	8Khz	128/4	4	AVG	P
5	14	256Kbps	PCMU	8Khz	64/4	2-3	GOOD	P
5	15	256Kbps	PCMA	8Khz	256/15	8	BAVG	P
5	16	256Kbps	PCMA	8Khz	256/8	8	AVG	P
5	17	256Kbps	PCMA	8Khz	256/4	4	AVG	P
5	18	256Kbps	PCMA	8Khz	128/15	4	AVG	P
5	19	256Kbps	PCMA	8Khz	128/8	4	AVG	P
5	20	256Kbps	PCMA	8Khz	128/4	4	AVG	P
5	21	256Kbps	PCMA	8Khz	64/4	2-3	AVG	P
5	22	256Kbps	G.726-40	8Khz	256/15	8	GOOD	P
5	23	256Kbps	G.726-40	8Khz	256/8	8	VG	P
5	24	256Kbps	G.726-40	8Khz	256/4	4	GOOD	P
5	25	256Kbps	G.726-40	8Khz	128/15	4	GOOD	P
5	26	256Kbps	G.726-40	8Khz	128/8	4	GOOD	P
5	27	256Kbps	G.726-40	8Khz	128/4	4	AVG	P
5	28	256Kbps	G.726-40	8Khz	64/4	2-3	VG	P
5	29	256Kbps	G.726-32	8Khz	256/15	8	GOOD	P
5	30	256Kbps	G.726-32	8Khz	256/8	8	GOOD	P
5	31	256Kbps	G.726-32	8Khz	256/4	4	VG	P
5	32	256Kbps	G.726-32	8Khz	128/15	4	VG	P
5	33	256Kbps	G.726-32	8Khz	128/8	4	VG	P
5	34	256Kbps	G.726-32	8Khz	128/4	4	VG	P
5	35	256Kbps	G.726-32	8Khz	64/4	2-3	VG	P
5	36	256Kbps	G.726-24	8Khz	256/15	8	GOOD	P
5	37	256Kbps	G.726-24	8Khz	256/8	8	GOOD	P
5	38	256Kbps	G.726-24	8Khz	256/4	4	VG	P
5	39	256Kbps	G.726-24	8Khz	128/15	4	VG	P
5	40	256Kbps	G.726-24	8Khz	128/8	4	VG	P
5	41	256Kbps	G.726-24	8Khz	128/4	4	VG	P
5	42	256Kbps	G.726-24	8Khz	64/4	2-3	VG	P
5	43	256Kbps	G.726-16	8Khz	256/15	8	GOOD	P
5	44	256Kbps	G.726-16	8Khz	256/8	8	GOOD	P
5	45	256Kbps	G.726-16	8Khz	256/4	4	VG	P
5	46	256Kbps	G.726-16	8Khz	128/15	4	VG	P
5	47	256Kbps	G.726-16	8Khz	128/8	4	VG	P

Pkg	ID#	Network	CODEC	Sample Rate	TX/Frame Rate (Kbps/fps)	Actual fps	QOS	Pass/ Fail
5	49	256Kbps	G.726-16	8Khz	64/4	2-3	VG	P
5	50	256Kbps	DVI	8Khz	256/15	8	BAVG	P
5	51	256Kbps	DVI	8Khz	256/8	8	GOOD	P
5	52	256Kbps	DVI	8Khz	256/4	4	VG	P
5	53	256Kbps	DVI	8Khz	128/15	4	AVG	P
5	54	256Kbps	DVI	8Khz	128/8	4	AVG	P
5	55	256Kbps	DVI	8Khz	128/4	4	AVG	P
5	56	256Kbps	DVI	8Khz	64/4	2-3	VG	P
5	57	256Kbps	VDVI	8Khz	256/15	8	BAVG	P
5	58	256Kbps	VDVI	8Khz	256/8	8	GOOD	P
5	59	256Kbps	VDVI	8Khz	256/4	4	VG	P
5	60	256Kbps	VDVI	8Khz	128/15	4	AVG	P
5	61	256Kbps	VDVI	8Khz	128/8	4	AVG	P
5	62	256Kbps	VDVI	8Khz	128/4	4	AVG	P
5	63	256Kbps	VDVI	8Khz	64/4	2-3	VG	P
5	64	256Kbps	GSM	8Khz	256/15	8	GOOD	P
5	65	256Kbps	GSM	8Khz	256/8	8	GOOD	P
5	66	256Kbps	GSM	8Khz	256/4	4	VG	P
5	67	256Kbps	GSM	8Khz	128/15	4	VG	P
5	68	256Kbps	GSM	8Khz	128/8	4	VG	P
5	69	256Kbps	GSM	8Khz	128/4	4	VG	P
5	70	256Kbps	GSM	8Khz	64/4	2-3	VG	P
5	71	256Kbps	LPC	8Khz	256/15	8	POOR	F
5	72	256Kbps	LPC	8Khz	256/8	8	POOR	F
5	73	256Kbps	LPC	8Khz	256/4	4	POOR	F
5	74	256Kbps	LPC	8Khz	128/15	4	POOR	F
5	75	256Kbps	LPC	8Khz	128/8	4	POOR	F
5	76	256Kbps	LPC	8Khz	128/4	4	POOR	F
5	77	256Kbps	LPC	8Khz	64/4	2-3	POOR	F
6	1	128Kbps	Linear-16	8Khz	128/15	5-6fps	POOR	F
6	2	128Kbps	Linear-16	8Khz	128/8	5-6fps	POOR	F
6	3	128Kbps	Linear-16	8Khz	128/4	4fps	POOR	F
6	4	128Kbps	Linear-16	8Khz	64/4	2-3fps	POOR	F
6	5	128Kbps	PCMU	8Khz	128/15	5-6fps	POOR	F
6	6	128Kbps	PCMU	8Khz	128/8	5-6fps	POOR	F
6	7	128Kbps	PCMU	8Khz	128/4		AVG	P
6	8	128Kbps	PCMU	8Khz	64/4		GOOD	P
6	9	128Kbps	PCMA	8Khz	128/15	5-6fps	POOR	F
6	10	128Kbps	PCMA	8Khz	128/8	5-6fps	POOR	F
6	11	128Kbps	PCMA	8Khz	128/4	4 fps	AVG	P
6	12	128Kbps	PCMA	8Khz	64/4	2-3 fps	GOOD	P
6	13	128Kbps	G.726-40	8Khz	128/15	5 fps	BAVG	F
6	14	128Kbps	G.726-40	8Khz	128/8	4 fps	BAVG	F
6	15	128Kbps	G.726-40	8Khz	128/4	4fps	BAVG	F

Pkg	ID#	Network	CODEC	Sample Rate	TX/Frame Rate (Kbps/fps)	Actual fps	QOS	Pass/ Fail
6	17	128Kbps	G.726-32	8Khz	128/15	4-5fps	POOR	F
6	18	128Kbps	G.726-32	8Khz	128/8	4-5fps	POOR	F
6	19	128Kbps	G.726-32	8Khz	128/4	4fps	GOOD	P
6	20	128Kbps	G.726-32	8Khz	64/4	2-3fps	VG	P
6	21	128Kbps	G.726-24	8Khz	128/15	6-7fps	BAVG	F
6	22	128Kbps	G.726-24	8Khz	128/8	5-6fps	BAVG	F
6	23	128Kbps	G.726-24	8Khz	128/4	4fps	AVG	P
6	24	128Kbps	G.726-24	8Khz	64/4	2-3fps	VG	P
6	25	128Kbps	G.726-16	8Khz	128/15	6-7fps	POOR	F
6	26	128Kbps	G.726-16	8Khz	128/8	5-6fps	POOR	F
6	27	128Kbps	G.726-16	8Khz	128/4	4fps	BAVG	P
6	28	128Kbps	G.726-16	8Khz	64/4	2-3fps	AVG	P
6	29	128Kbps	DVI	8Khz	128/15	6-7fps	POOR	F
6	30	128Kbps	DVI	8Khz	128/8	4-6fps	BAVG	F
6	31	128Kbps	DVI	8Khz	128/4	4fps	BAVG	F
6	32	128Kbps	DVI	8Khz	64/4	2fps	VG	P
6	33	128Kbps	VDVI	8Khz	128/15	6-7fps	POOR	F
6	34	128Kbps	VDVI	8Khz	128/8	4-6fps	POOR	F
6	35	128Kbps	VDVI	8Khz	128/4	4fps	BAVG	P
6	36	128Kbps	VDVI	8Khz	64/4	2fps	VG	P
6	37	128Kbps	GSM	8Khz	128/15	5-6fps	POOR	F
6	38	128Kbps	GSM	8Khz	128/8	6-7fps	POOR	F
6	39	128Kbps	GSM	8Khz	128/4	4fps	AVG	P
6	40	128Kbps	GSM	8Khz	64/4	2-3fps	VG	P
6	41	128Kbps	LPC	8Khz	128/15	5-6fps	POOR	F
6	42	128Kbps	LPC	8Khz	128/8	6-7fps	POOR	F
6	43	128Kbps	LPC	8Khz	128/4	4fps	POOR	F
6	44	128Kbps	LPC	8Khz	64/4	2-3fps	POOR	F

3. Findings

The VoIPNET Prototype was tested on each emulated network. No issues or surprises during Component, Integration or Recovery testing. However, during recovery testing when a network failure was induced, there was no notification of a dropped call/VTC. This will be addressed in the next proposed release of VoIPNET. The 512kbps and 256kbps networks supported all the CODECS tested. However, the LPC CODEC failed due to very poor voice quality and was excluded from further testing. VTC tests showed Video Transmit rates that approached or exceeded capacity greatly degraded service quality. Therefore, it is recommended that each OT&E test use the following formula:

TX rate = .95(Network Capacity – CODEC Bandwidth)

The .95 allows a 5% overhead “buffer.” Network Capacity is the available network bandwidth. Finally, CODEC Bandwidth is the bandwidth each codec consumes at its corresponding sample rate.

In addition, the 30, 15 and 8 fps settings proved to be less efficient on the 512/256kbps network configurations. When data rates approach 512 and 256kbps, the maximum frame rates available peek at 12 and 8 fps respectively. Data showed no increase in quality at the higher frame rate. Therefore, it is recommended that 30fps settings be removed from OT&E tests.

The Linear-16 (128kbps) and U/A-Law(64kbps) CODECS were successful but the increase in quality was not sufficient to justify the excess bandwidth required. The GSM CODEC was the most efficient and effective CODEC tested. On the lower bandwidth networks it was the only CODEC that provided clear voice at low bandwidth consumption (18kbps).

The 128kbps network test proved to be the most useful. When bandwidth is this restricted, it is critical that the VTC transmit and frame rates be regulated to maximize the quality. At 128kbps the 4fps setting was the preferred setting. When VTC transmit rates approached network capacity the video quality degraded quickly. At this data rate I found that it was most efficient and effective to restrict the VTC down to 64Kbps and 2 fps. At this low transmit rate an increase in frame rate does not have a corresponding increase in video quality.

4. Recommendations

Audio

- Use the GSM CODEC only for OT&E testing
- LPC CODEC is unacceptable. Drop the LPC CODEC from OT&E testing.
- Use 8khz sampling only. 16 and 32khz will only serve to double and quadruple the bandwidth required.

VTC

- To successfully regulate the bandwidth the maximum VTC transmit rate should be:

  VTC Tx rate =.95(Capacity – CODEC Bandwidth).

- On larger capacity tactical networks (512-256kbps) use only the 128kbps/4fps or 64kbps/2fps VTC settings.
- On smaller capacity tactical networks (<128kbps) use on the 64kbps/2fps VTC settings.
- If a high quality VTC circuit is established follow these setting guidelines for efficiency and quality:
  - 512Kbps yields a maximum of 12 fps
  - 256Kbps yields a maximum of 8fps

        - 128 kbps yields a maximum of 4fps
        - 64kbps yields a maximum of 2-3 fps
- Use 50kbps @2fps for OT&E testing

Requirements Update

None.

B. OT&E REPORT

1. Test Network Overview

The OT&E testing was conducted on multiple EPLRS network configurations. CSMA and MSG networks were established to mimic an Infantry Battalion organization. The networks were composed of a CSMA or MSG needline, 5 RSs, 4 host computers, an EPLRS Network Monitor(ENM) and audio/video peripherals for each 3 hosts. The ENM was connected to an RS and on its own needline. This ensured it was not connected to the test network but was still able to make configuration changes to the tested Network.

The CSMA and MSG LTS, waveform and hop settings were then changed to identify the best network configuration. The Rider software component was used to take measurements for latency, packet loss, and available bandwidth. The Network activity Diagram software was also used to confirm real-time bandwidth consumption. In addition, once an effective network was discovered, a CSMA multi-hop test was conducted at 0, 2 and 4 hops respectively. No MSG multi-hop test was conducted do to SME knowledge gap of applicable settings and configurations. Below are the specifics of the network architecture:

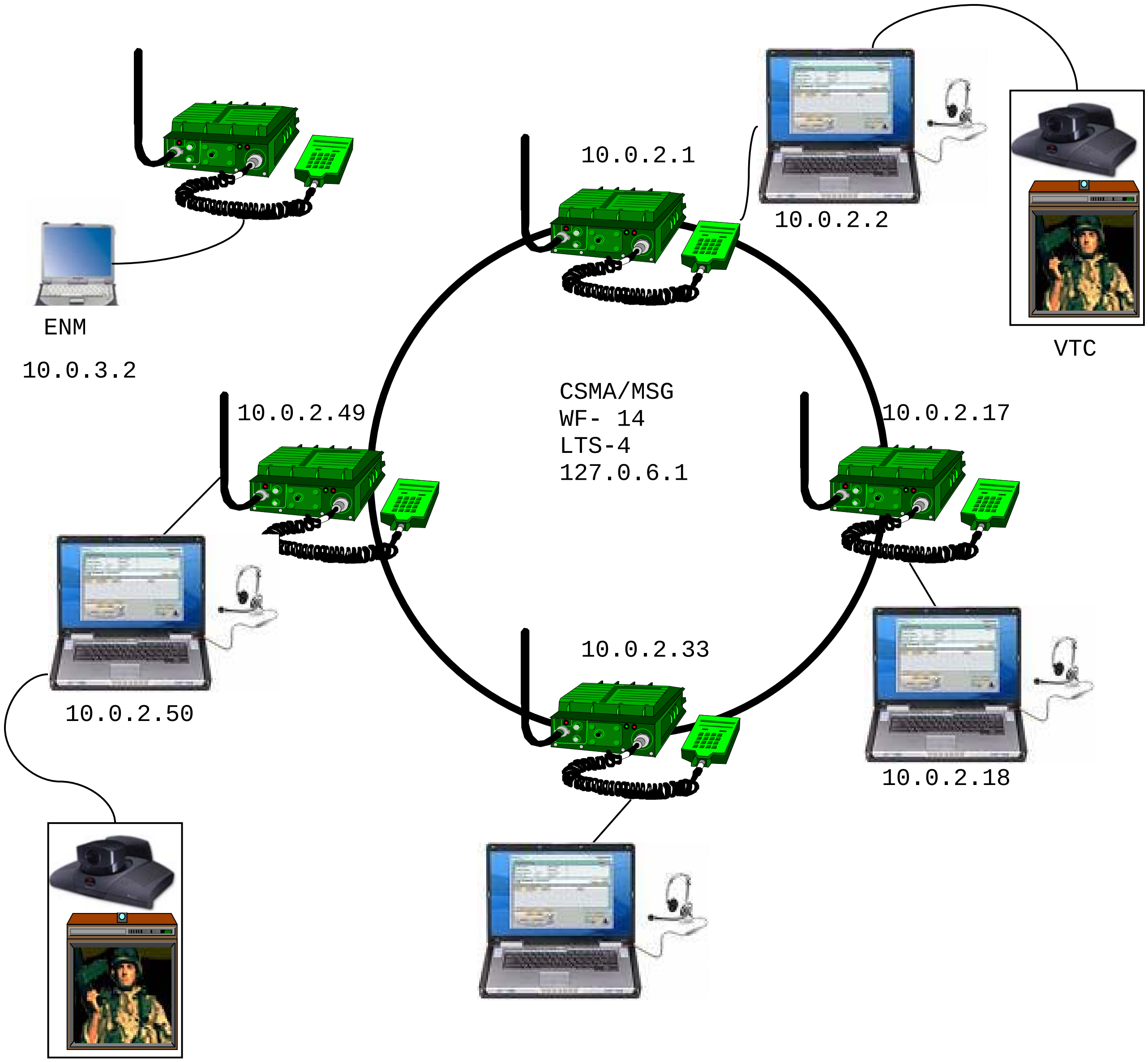

Figure 47. OT&E Test Architecture.

2. Test Data

Configuration Test 1:

Network Type: MSG

LTSs: 2

Waveform Mode: 9

hop/relay: 2/1

Baseline Measurements:

Bandwidth: not measured

Latency: not measured

Packet Loss: not measured

Result: **Fail.** Network would not establish. Not all RSs could be brought into net. No Host was brought into net, therefore a baseline measurements could not be taken.

Configuration Test 2:

Network Type: MSG/CSMA

LTSs: 4

Waveform Mode: 9

hop/relay: 2/1

Baseline Measurements:

Bandwidth: 64Kbps

Latency: 52ms

Packet Loss: 1%

Result: **PASS.** All RSs and hosts brought onto the network. Successful VTC conducted with 50Kbps @2fps transmit rate. Static on calls was traced to RF interference from the RSs. The RS power setting was changed from Med-high to low. This resolved the static issue and produced clear calls. Call behaved as half-duplex. Echo suppression was turned off and this resolved the problem.

Configuration Test 3:

Network Type: MSG/CSMA

LTSs: 7

Waveform Mode: 9

hop/relay:2/1

Baseline Measurements:

Bandwidth: not measured

Latency: not measured

Packet Loss: not measured

Result: **Fail.** Failed to establish network.

Configuration Test 4:

Network Type: MSG/CSMA

LTSs: 4

Waveform Mode: 14

hop/relay: 2/1

Echo Suppression: OFF

Baseline Measurements:

Bandwidth: 64kbps

Latency: 21ms

Packet Loss: 1%

Result: **PASS.** Echo suppression was turned off and this resolved the half-duplex problem. Successful VTC slightly pixilated with 50Kbps @ 2fps.

Configuration Test 5:

Network Type: MSG/CSMA

LTSs: 2/4

Waveform Mode: 14

hop/relay: 2/1

Baseline Measurements:

Bandwidth: 18kbps

Latency: 177ms

Packet Loss: 5%

Result: **FAIL.** Very Poor VTC and voice. 5-8 second delay on voice and 30 second delay on video.

Configuration Test 6:

Network Type: MSG/CSMA

LTSs: 4/2

Waveform Mode: 14

hop/relay: 2/1

shares: 4/4/1/1

Baseline Measurements:

Bandwidth: 64kbps

Latency: 104ms

Packet Loss: 2%

Result: **PASS.** Successful VTC with TX rate 40kbps @ 2fps.

Configuration Test 7:

Network Type: MSG/CSMA

LTSs: 4/4

Waveform Mode: 14

Routes: MSG-Programmed/CSMA-default

hop/relay: 2/1

Baseline Measurements:

Bandwidth: 64kbps

Latency: 53ms

Packet Loss: 1%

Result: **PASS.** Successful VTC with TX rate 50kbps @ 2fps. Excellent quality VoIP and VTC. Best MSG configuration tested.

Configuration Test 8:

Network Type: MSG/CSMA

LTSs: 8/4

Routes: MSG-programmed/CSMA-default

Waveform Mode: 14

hop/relay:2/1

Baseline Measurements:

Bandwidth: 64kbps

Latency: 49ms

Packet Loss: 1-2%

Result: **PASS.** Network established. VTC resulted 40kbps variation in transmit rate and a 2fps variation in frame rate. 2-3 second delay in audio and 2 second delay in pixilated video.

Configuration Test 9:

Network Type: MSG/CSMA

LTSs: 8/4

Routes: MSG-programmed/CSMA-programmed

Waveform Mode: 14

hop/relay: 2/1

Baseline Measurements:

Bandwidth: 64kbps

Latency: 54ms

Packet Loss: 2%

Result: **PASS.** Same results as test #8. Video transmit rat had 15kbps variation.

Configuration Test 10:

Network Type: MSG/CSMA

LTSs: 4/2

Routes: MSG-programmed/CSMA-default

Waveform Mode: 14

hop/relay: 2/1

shares: 2/2/1/1

Baseline Measurements:

Bandwidth:18.6

Latency: 159ms

Packet Loss: 2%

Result: **Fail.** Insufficient bandwidth to conduct VTC test.

Configuration Test 11:

Network Type: MSG/CSMA

LTSs: 4/2

Routes: MSG-programmed/CSMA-default

Waveform Mode: 14

hop/relay: 2/1

shares: 4/4/1/1

Baseline Measurements:

Bandwidth: 64kbps

Latency: 137ms

Packet Loss: 1%

Result: **FAIL.** No successful VTC. 30 Second Video delay. No audio with the VTC.

Configuration Test 12:

Network Type: CSMA/MSG

LTSs: 4/4

Routes: CSMA-programmed/MSG-default

Waveform Mode: 14

hop/relay: 1/0

Baseline Measurements:

Bandwidth: 142Kbps

Latency: 5ms

Packet Loss:.17%

Result: **PASS.** Best network yet. Observed bandwidth as high as 185kbps. Conducted successful VTC. Will conduct detailed package tests on this network.

Configuration Test 13:

Network Type: CSMA

LTSs:4

Waveform Mode: 9

hop/relay: 1/0

Baseline Measurements:

Bandwidth: 52kbps

Latency: 21ms

Packet Loss: 1%

Result: **PASS.** Conducted successful VTC. No noticeable difference in quality or metrics from Test # 12.

Configuration Test 14:

Network Type: CSMA

LTSs: 8

Waveform Mode: 9

hop/relay: 1/0

Baseline Measurements:

Bandwidth: not measured

Latency: not measured

Packet Loss: not measured

Result: **FAIL.**Could not establish network.

Configuration Test 15:

Network Type: CSMA

LTSs: 4

Waveform Mode: 9

hop/relay: 2/1

Baseline Measurements:

Bandwidth: 30kbps

Latency: 190ms

Packet Loss: 2.1%

Result: **PASS.** Successful VTC. Demonstrates multi-hop capability with CSMA.

Configuration Test 16:

Network Type: CSMA

LTSs:4

Waveform Mode: 9

hop/relay: 4/3

Baseline Measurements:

Bandwidth: 16Kbps

Latency: 227ms

Packet Loss: 2.3%

Result: **FAIL.** Voice was choppy with 5 second delay. Performance commensurate with CSMA relay specifications.

Summary

Several configurations successfully conducted a VTC but only one CSMA and one MSG configuration will be tested. The networks were chosen based on capacity, packet loss, latency, and video transmit rate stability.

CSMA - Test Configuration #12

MSG – Test Configuration #7.

CSMA – Test Configuration #15 (GSM CODEC ONLY)

Test Coordinator: **Captain C.P. Reiche, Jr** **Date: 05/18/07**
Network Administrator: Mr. Pedro Zenquis

Audio Test Report

Pkg	ID#	Network	CODEC	Sample Rate	QOS CSMA	QOS MSG	Pass / Fail CSMA	Pass / Fail MSG
1	1	EPLRS	Linear-16	8Khz	GOOD	POOR	P	F
1	2	EPLRS	PCMU	8Khz	GOOD	POOR	P	F
1	3	EPLRS	PCMA	8Khz	GOOD	POOR	P	F
1	4	EPLRS	G.726-40	8Khz	VG	VG	P	P
1	5	EPLRS	G.726-32	8Khz	VG	VG	P	P
1	6	EPLRS	G.726-24	8Khz	VG	VG	P	P
1	7	EPLRS	G.726-16	8Khz	AVG	AVG	P	P
1	8	EPLRS	DVI	8Khz	GOOD	GOOD	P	P
1	9	EPLRS	VDVI	8Khz	GOOD	GOOD	P	P
1	10	EPLRS	GSM	8Khz	VG	VG	P	P
~~1~~	~~11~~	~~EPLRS~~	~~LPC~~	~~8Khz~~				

Video Test Report

Pkg	ID#	Network	CODEC	TX/Frame Rate (Kbps/fps)	F/S CSMA/MSG	QOS CSMA	QOS MSG	Pass/Fail CSMA/MSG
2	1	EPLRS	Linear-16	128/15	4 / 4	POOR	POOR	F/F
2	2	EPLRS	Linear-16	128/8	4 / 4	POOR	POOR	F/F
2	3	EPLRS	Linear-16	128/4	4 / 4	POOR	POOR	F/F
2	4	EPLRS	Linear-16	64/4	2-3/2-3	BAVG	POOR	F/F
2	5	EPLRS	PCMU	128/15	4 / 4	POOR	POOR	F/F

2	6	EPLRS	PCMU	128/8	4 / 4	POOR	POOR	F/F
2	7	EPLRS	PCMU	128/4	4 / 4	POOR	POOR	F/F
2	8	EPLRS	PCMU	64/4	2-3/2-3	GOOD	POOR	P/F

Pkg	**ID#**	**Network**	**CODEC**	**TX/Frame Rate (Kbps/fps)**	**F/S CSMA/MSG**	**QOS CSMA MSG**	**Pass/Fail CSMA/MSG**	**Pkg**
2	10	EPLRS	PCMA	128/8	4 / 4	POOR	POOR	F/F
2	11	EPLRS	PCMA	128/4	4 / 4	POOR	POOR	F/F
2	12	EPLRS	PCMA	64/4	2-3/2-3	GOOD	POOR	P/F
2	13	EPLRS	G.726-40	128/15	4 / 4	GOOD	POOR	P/F
2	14	EPLRS	G.726-40	128/8	4 / 4	GOOD	POOR	P/F
2	15	EPLRS	G.726-40	128/4	4 / 4	GOOD	POOR	P/F
2	16	EPLRS	G.726-40	64/4	2-3/2-3	VG	POOR	P/F
2	17	EPLRS	G.726-32	128/15	4 / 4	GOOD	POOR	P/F
2	18	EPLRS	G.726-32	128/8	4 / 4	GOOD	POOR	P/F
2	19	EPLRS	G.726-32	128/4	4 / 4	GOOD	POOR	P/F
2	20	EPLRS	G.726-32	64/4	2-3/2-3	VG	BAVG	P/F
2	21	EPLRS	G.726-24	128/15	4 / 4	GOOD	POOR	P/F
2	22	EPLRS	G.726-24	128/8	4 / 4	GOOD	POOR	P/F
2	23	EPLRS	G.726-24	128/4	4 / 4	GOOD	POOR	P/F
2	24	EPLRS	G.726-24	64/4	2-3/2-3	VG	BAVG	P/F
2	25	EPLRS	G.726-16	128/15	4 / 4	GOOD	POOR	P/F
2	26	EPLRS	G.726-16	128/8	4 / 4	GOOD	POOR	P/F
2	27	EPLRS	G.726-16	128/4	4 / 4	GOOD	POOR	P/F
2	28	EPLRS	G.726-16	64/4	2-3/2-3	GOOD	GOOD	P/P
2	29	EPLRS	DVI	128/15	4 / 4	AVG	POOR	P/F
2	30	EPLRS	DVI	128/8	4 / 4	AVG	POOR	P/F
2	31	EPLRS	DVI	128/4	4 / 4	AVG	POOR	P/F
2	32	EPLRS	DVI	64/4	2-3/2-3	VG	BAVG	P/F
2	33	EPLRS	VDVI	128/15	4 / 4	AVG	POOR	P/F
2	34	EPLRS	VDVI	128/8	4 / 4	AVG	POOR	P/F
2	35	EPLRS	VDVI	128/4	4 / 4	AVG	POOR	P/F
2	36	EPLRS	VDVI	64/4	2-3/2-3	VG	BAVG	P/F
2	37	EPLRS	GSM	128/15	4 / 4	VG	BAVG	P/F
2	38	EPLRS	GSM	128/8	4 / 4	VG	BAVG	P/F
2	39	EPLRS	GSM	128/4	4 / 4	VG	BAVG	P/F
2	40	EPLRS	GSM	64/4	2-3/2-3	VG	VG	P/P
~~2~~	~~41~~	~~EPLRS~~	~~LPC~~	~~128/15~~				
~~2~~	~~42~~	~~EPLRS~~	~~LPC~~	~~128/8~~				
~~2~~	~~43~~	~~EPLRS~~	~~LPC~~	~~128/4~~				
~~2~~	~~44~~	~~EPLRS~~	~~LPC~~	~~64/4~~				

VII. CONCLUSION

A. FINDINGS

A variety of MSG and CSMA network configurations were utilized to find the most efficient and effective network and application settings. Networks were tested at waveform modes 9/14 and 2/4/8 LTSs. Both waveforms provided maximum available bandwidth for the respective burst rate (2/4 ms).

The MSG network is the theoretically preferred network configuration. Its membership restriction, guaranteed bandwidth, and “no bandwidth cost” multi-hop capability make it the most suitable for small tactical networks. The MSG network is not a network configuration that is frequently used by the U.S. Marine Corps. As a result the Subject Matter Experts (SMEs) were unfamiliar with the configuration or administration of the network. This prevented extensive testing of a sole MSG network and required the use a variety of CSMA and MSG combinations.

On all occasions regardless of network configuration, the attempt to configure an 8 LTS network failed. The network would not establish. According to published technical manuals this should have been an attainable configuration. In addition, any attempt to establish an MSG network without the inclusion of a CSMA on some variation of LTSs resulted in a failed network. On no occasion was the Network Administrator able to establish a viable network. I attribute this to lack of Network Administrator experience with the MSG network, and recommend that any follow-on researchers seek out an operator/planner with experience with this network configuration. In addition, the use of a 2 LTS network resulted in maximum bandwidth of only 18kbps regardless of needline type therefore subsequent network configurations were not conducted with this setting.

Most EPLRS technical publications recommend the 4ms waveform mode for streaming media. However, with regard to quality of VoIP services I found no discernible differences, excepting available bandwidth, between the 4ms and 2ms families of waveform modes.

Tests were conducted on a network plan that included a CSMA network on 4 LTSs with programmed routes in addition to an MSG network with default routes only. Here after referred to as the CSMA/MSG network. This configuration should route VoIP traffic over the CSMA only. In addition, an MSG network with the same LTS configuration was tested. Here after known as the MSG/CSMA Network. The final network tested was a CSMA network with no MSG network. Here after known as the CSMA network. On each occasion the CSMA/MSG combination configuration provided greater bandwidth. While the MSG/CSMA configuration provided dedicated bandwidth to each user.

Despite having no planned routes on the MSG portion of the network, some traffic was observed on the default routed network. The same observations were made on the CSMA portion of the MSG Network. Subject Matter Experts at MCTSSA could only speculate and not sufficiently explain this increase. I recommend follow-up researchers apply more sophisticated network monitoring tools in order to identify what traffic is traveling on which network and query Raytheon engineers regarding this observation.

The VoIPNET Prototype performed best when the CSMA/MSG and MSG/CSMA network was configured with 4 LTSs and 0 hops (CSMA only). However, 2 “voice only” calls were conducted via the CSMA/MSG 2 hop network with excellent results. The voice quality was excellent with no delay. Again, the 4Ms or 2Ms setting had no impact on quality. Baseline averages for both networks were:

Bandwidth: CSMA-180kbps MSG-64kbps.

Jitter: 50-70 ms

Packet loss: 1-2%

The CSMA network supported 4 simultaneous “voice only” calls utilizing the GSM CODEC. The GSM CODEC proved to be the most efficient CODEC (18kbps) that provided the highest quality of service. The DVI, VDVI, U-Law, A-Law and G-726 series CODECs were successful, but provided no noticeable increase in quality at a greater bandwidth cost. Calls were successfully conducted, with average quality when available bandwidth was as low as 16kbps. Based on call quality and baseline bandwidth measurements, the CSMA network configuration, in a laboratory environment, could support 6 simultaneous “voice only” calls. Due to limitations in SME proficiency no threshold testing was conducted on the MSG/CSMA network.

Both network configurations were able to support a single VTC with audio. VTC quality was best at a 50kbps and 2 frame/sec rate. Transmit/frame rates must be actively managed, as an unregulated VTC will consume excessive bandwidth and dramatically degrade network performance and capability. On the CSMA/MSG network 128kbps @ 2f/s video throughput rates could be achieved, but at no significant improvement in quality and leaving no bandwidth for other users. The MSG/CSMA network is not capable of supporting transmit/frame rates greater than 40-50 kbps/2fps unless a tactically undesirable share allocation configuration is utilized. This configuration allocates all MSG share to two users and is in essence a High Data Rate Duplex circuit. In addition, regardless of network configuration the actual Transmit/frame rate of video data was 5-10% less than the connection capacity. The manual transmit and frame rate sliders in the prototype were utilized to find an acceptable rate.

CODECs consuming more than 32kbps of bandwidth provided no measurable increase in quality. The GSM CODEC provided the best quality at the lowest bandwidth. On this EPLRS network, silence suppression was critical. Calls made without silence suppression used approximately 30% more bandwidth. Peripheral audio devices are subject RF interference when high power settings are used. When multiple radios were used in close proximity, a significant amount of static and interference can be heard. Excellent quality VoIP calls were supported using a 2 hop/ 1 relay network configuration. No VoIP calls were supportable when hops exceeded this configuration.

CSMA needlines can support more voice only users than an MSG network. However an MSG network guarantees bandwidth to its users. Use CSMA if no user requires dedicated bandwidth and many users require the ability to transmit and receive. Use MSG if all users require dedicated bandwidth and only a few must transmit and receive.

B. CONCLUSION

VoIP is an acceptable voice option for tactical networks. However, special care must be paid to ensure that the software employed has the capability to automatically adjust transmit and frame rates based on network health. This feature would optimize bandwidth utilization and increase quality. While the current version of EPLRS firmware supports VoIP, the current MSG network knowledge base prevents the use of an MSG network. The CSMA network does not provide sufficient bandwidth to take advantage of the multi-hop requirement in compartmentalized terrain (urban/mountain/jungle).

The VoIPNET prototype test results support the concept of VoIP via radio. While the existing firmware version requires detailed MSG network planning and highly efficient VoIP applications, it is feasible but not optimal. In a sparse CSMA network, the advantages gained by automatic relay and routing may be nullified. As the CSMA network increases in density, relays are more plentiful, but so are contentious users. I would not recommend a CSMA implementation for anything other than a small static network.

In contrast, the MSG network has the potential to provide precisely the services required for VoIP. Circuit size may be tailored depending upon services required. Share allocation provides increased bandwidth to high priority users, while guaranteeing minimal bandwidth to lower priority shares. The “no-cost” multi-hop capability is ideal for compartmentalized terrain. The ability to automatically route communications is invaluable and will greatly increase the reliability of infantry communications. VoIP is supportable using an MSG network on this version of firmware. However, MSG network configuration expertise is not mature enough to support the advanced configuration requirements needed. Coordination with experienced MSG network planners is a requirement for future use of VoIP via the EPLRS tactical network.

VoIPNET Requirement Updates

The current prototype sends a ”BYE” message when one User Agent terminates the call. However, when the call is dropped by network or video peripheral failure, no status is sent. A VTC/Call status notification display would greatly increase the user’s situational awareness. Transmit and frame rate sliders must be included to efficiently manually adjust VTC quality based on network characteristics. The prototype tested proved the value of a graphically updateable CODEC library. The ability for an operator to manually change the CODEC based on Network conditions increases the efficiency of the application and utilization of network resources.

An Automated continuous network health test will evaluate the health of the network for VoIP and automatically adjust the call variables, such as transmit rate, frame rate, and CODEC choice. This feature will ensure the efficient use of network resources.

Deployment Recommendations

VoIPNET should be deployed as an independent EPLRS network until the new firmware is fielded and interoperability is tested with existing Command and

Control applications. Use CSMA if no user requires dedicated bandwidth and many users require the ability to transmit and receive. Maximum supportable is Bandwidth dependant. Use MSG if all users require dedicated bandwidth and only a few must transmit and receive.

Use silence suppression and the mute button. It conserves bandwidth. In sparse networks use a dedicated 2 hop/1 relay site with good LOS to the other RSs. For a CSMA network limit relays to 1 hop /0 relays or 2 hops/1 relay. Use shielded Audio/VTC peripheral components (headset/camera) or remote the RS to avoid RF interference when high power settings are used.

To successfully regulate the bandwidth the VTC transmit rate should be:

TX rate < .95 *(CAPCITY – CODEC Bandwidth

On Low bandwidth networks the overhead “buffer” may need to be increased to 10%, depending on network health.

TX rate < .90(Capacity – CODEC Bandwidth)

On larger capacity tactical networks (512-256kbps) keep the video transmit rates < 128kbps @ 4fps or < 40-50kbps @ 2fps respectively. On smaller capacity tactical networks (<128kbps) Video transmit rate should be 40-50kbps @ 2fps.

Keep VTCs to a minimum. For every VTC (@50Kbps/2fps) 2.5 voice calls can be supported. If a high quality VTC circuit is required use these guidelines for efficiency and quality:

512Kbps yields a maximum of 12 fps

256Kbps yields a maximum of 8fps

128 kbps yields a maximum of 4fps

64kbps yields a maximum of 2-3 fps

C. AREAS OF FUTURE STUDY

The MSG network requires further study. VoIP service compatibility with the current version of EPLRS firmware requires that the MSG network be employed. Coordination with an MSG knowledgeable planner is required to complete the testing of VoIP services on the current firmware version. In addition, threshold testing and field testing would provide greater insight into the capabilities in a tactical environment. When the MSG network issues are resolved the next phase of testing should be in a field environment.

Performance of EPLRS Radio Software Version 11.4.0.9.5

The most recent release of EPLRS Firmware has specific VoIP support characteristics, to include 1Mbps needlines, Tactical Ad-Hoc Multiple Access Protocol (TAMA), weighted Node Activation Multiple Access (NAMA) Protocol, Multicast, and Fast Attach/Fast Release Service (SMSG).

Multicast Capabilities

Session management tools support multicast functionality. It may be possible to use a tool like SDR from UCL Media to broadcast a VTC to many users. I attempted to run SDR on the EPLRS networks but was unable to get the session advertisement to propagate through the network. The EPLRS firmware currently supports 15 multicast routes. The new firmware supports 30. It would be a useful addition to the suite of audio and video tools if this could be resolved.

EPLRS VoIP Threshold Testing

Resource Limitations prevented the conduct of threshold/VoIP capacity testing on the EPLRS Network. The deployment recommendations found in this Thesis are preliminary and therefore must be followed up with comprehensive field testing. Of interest are the effects of topography/obstacles and maximum supportable VoIP users.

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