Investigation of a Packet-Switched Inter-System Interface for Land Mobile Radio Systems

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(ABSTRACT)

Traditionally, and up to this date, Land Mobile Radio (LMR) systems have been interconnected via leased lines and microwave links across circuit-switched networks. With the recent deployment of digital LMR standards such as the Association of Public and Communications Officials (APCO) Project 25 and the Terrestrial Trunked Radio (TETRA), traffic exchange has become more bursty and non-uniform, and as such, less suitable for circuit-switched networks.

This thesis proposes a framework for a packet-switched Inter-System Interface (ISI) for LMR systems. Packet-switched networks have the advantage of supporting traffic integration, utilize capacity efficiently, scale easily and seamlessly, and eliminate single points of failure by providing a distributed architecture. Session Initiation Protocol (SIP) signaling messages are defined for setting up and tearing down unit-to-unit calls across the ISI. The Session Description Protocol (SDP) is used to describe how the voice calls are encoded. Voice packets are exchanged between LMR users using the Real-Time Transport Protocol (RTP).

Based on the proposed framework, we develop a simulation model to investigate the performance of the ISI when different numbers of LMR users try to establish unit-to-unit calls across the packet-switched ISI. Three packet transport technologies providing Wide Area Network (WAN) connectivity are considered, IP, ATM, and Frame Relay. The results indicate that a packet-switched ISI can take advantage of statistical multiplexing techniques to distribute network resources more efficiently. Quantitative results are obtained for throughput and link utilization. When using an access link providing T1 service, we show that the End-To-End (ETE) delay, and delay variation can be controlled at levels capable of supporting the timely delivery of real-time voice packets. Assuming link utilization is maintained below 100%, the maximum ETE delay experienced in all three packet transport technologies considered is 58 ms and the maximum call setup time is less than 300 ms.

An ATM WAN provides the best performance for all time-dependent metrics considered, i.e., ETE delay, delay variation, and call setup time. An IP WAN provides the highest bandwidth efficiency. Selecting the appropriate packet transport technology for the WAN is a tradeoff between the delay that can be tolerated by the voice packets traversing the LMR network and the cost of bandwidth on the access link.

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Chapter 1

Introduction

1.1 Overview

Land Mobile Radio (LMR) systems are the communications backbone of public safety organizations. The service offered by LMR systems in the day-to-day activities of public safety organizations is imperative and the communications they provide is often the only lifeline in mission-critical and life-threatening situations. In addition to public safety organizations, many commercial businesses and transit companies rely heavily on private LMR systems to manage and coordinate their daily activities. In this thesis, we associate most of the discussion on LMR systems with public safety communications.

LMR systems evolved as autonomous two-way radio systems servicing the specific needs and interests of different public safety organizations [DSD01]. This led to the development of "stovepipe" systems that were confined to a local scope. Each organization deployed their own independent LMR system based on individual initiatives that were often driven by complex techno-political objectives. As a result, radio communications between the different public safety organizations lacked interoperability even within the same coverage zones.

The drawbacks of these initial LMR systems and the increasing awareness of the operational necessity for interoperability led to the unified concept of an integrated LMR system. Such a system must be able to provide the required communications where (distance-independence) and when (time-independence) needed. Technological advancements in the area of radio communications, the clear trend over the last few years to reinvent public safety communications, and the mandated reallocation of spectrum resources have all contributed towards the restructuring of LMR communications. Ultimately, these developments are encouraging the migration from independent to shared LMR systems [DSD01].

Standardization bodies, both in North America and Europe, have developed digital LMR communication systems based on open standards that are replacing proprietary and outdated analog LMR systems. The Association of Public and Communications Officials (APCO) Project 25 and the Terrestrial Trunked Radio (TETRA) are the two most prominent open, digital, LMR systems. The availability of open standards ensures vendor competition during the life-cycle of the LMR system and promotes interoperability between manufacturer products.

Typically, LMR systems use Very High Frequencies (VHF) and Ultra High Frequencies (UHF) for their wireless communications. Depending on the physical topography and foliage of the coverage area and the level of interference from local Radio Frequency (RF) sources, the communications range of LMR base stations is usually less than one hundred kilometers. Hence, extended regional and national coverage is only possible if multiple LMR base stations are inter-connected to form a Wide Area Network (WAN). Both TETRA and APCO Project 25 define interfaces that facilitate the interconnection of multiple LMR systems. These interfaces can support services and features such as mobility management, group calls, individual calls, text messaging, and security. A combination of these services allows interoperability, seamless wide-area coverage, and resource sharing at multiple levels.

Currently, networking of digital LMR systems is implemented using proprietary circuit-switched technology inherited from the telephony world. In fact, this is the same technology as that used for legacy analog LMR systems. Using a circuit-switched infrastructure to deploy an LMR network with extended coverage has some fundamental disadvantages. Circuit-switched networks are vulnerable to single points of failure and utilize network capacity inefficiently. Moreover, with the recent deployment of digital LMR systems, traffic exchange has become more bursty and non-uniform, and as such, less suitable for circuit-switched networks.

The ability to digitize voice and send it as packets over the network has enabled the provisioning of integrated services over a single data network. Traffic integration can be achieved by implementing a packet-switched LMR network. In this thesis, we investigate the performance of a packet-switched Inter-System Interface (ISI) for LMR systems.

1.2 Problem Statement

This research has two main objectives. The first objective is to develop a framework for an LMR network that uses packet-switched technology to interconnect base stations across an extended coverage area. Based on this framework, a simulation model is developed. An ISI for packet-switched LMR networks has not yet been standardized for any digital LMR technology. Consequently, the simulation model developed should be useful for performance evaluation, both in this study and in subsequent research efforts in this area.

The second objective of this study is to use the simulation model to characterize the loading effects on the LMR network and investigate the performance of the packet-switched ISI. We focus our study on unit-to-unit voice calls between LMR users. Different underlying packet transport technologies are considered for backhauling voice packets across the LMR network. The effects of using Internet Protocol (IP), Asynchronous Transfer Mode (ATM), and Frame Relay WANs are examined based on Service Level Agreements (SLAs) that conform to present commercial offerings.

To characterize the loading effects on the LMR network, packet flows are generated that represent network traffic in a typical unit-to-unit call across the ISI. The Session Initiation

Protocol (SIP) is used for signaling and the Real-Time Transport protocol (RTP) is used to transport the voice packets. Based on these packet flows, several scenarios are considered and analyzed. The effects of varying the number of active LMR users in the network and increasing the background utilization on the access link are examined by observing the link utilization, throughput, call setup time, end-to-end (ETE) delay, and delay variation.

1.3 Organization of thesis

The thesis is organized in six chapters as follows.

Chapter 2 provides enough background information to facilitate a more detailed discussion of the ISI, the focus of this study. In particular, the APCO Project 25 and TETRA digital LMR standards are introduced. The network architecture of the two LMR systems is presented and compared. The chapter concludes with an overview of SIP, a signaling protocol that is likely to replace the circuit-switched signaling protocols currently being implemented at the ISI.

Chapter 3 defines the research objectives and motivation of this thesis. An overview of circuit-switched and packet-switched LMR networks is first presented. Selection of the appropriate signaling protocol for a packet-switched ISI is also discussed and justified. To elaborate on the research objectives, the problem statement, the performance metrics, and the performance evaluation technique are formalized.

Chapter 4 describes the model used to investigate the performance of the packet-switched ISI. The methodology followed to characterize and simulate the network traffic in different scenarios is presented in detail and the simulation experiments are defined. To verify and analyze the results of the model, controlled simulation experiments are performed.

Chapter 5 presents the results of the simulation experiments. Based on these results, the performance of the packet-switched ISI proposed in Chapter 4 is assessed.

Chapter 6 provides a summary of the entire thesis that outlines the work completed. Final conclusions and opportunities for further research are described.

Appendix A provides a list of acronyms used. Data collected from the simulation experiments is provided in Appendix B.

Chapter 2

Background

This chapter introduces two digital LMR communication systems based on open standards, TETRA and APCO Project 25. For each of the standards, a brief overview is presented followed by a description of their network architecture. This provides enough background information to facilitate a more detailed discussion of the inter-system interface, the focus of this thesis. A comparison of TETRA and APCO Project 25 is included to emphasize the key differences of the two LMR standards considered. The chapter concludes with an overview of SIP, a signaling protocol that is likely to replace the current circuit switched signaling protocols being implemented at the inter-system interface.

2.1 TETRA

TETRA is a digital, trunked, LMR communication standard offering voice and data services. Since the inception of its standardization process by the European Telecommunications Standards Institute (ETSI) in 1989 it has experienced increasing popularity as a digital LMR solution, particularly in Europe. Estimates based on current indicators claim that approximately 80% of the digital LMR market in Europe will be operating using TETRA technology

[IMS01]. Recently, the North American TETRA Forum (NATF) has been formed to promote, discuss, and enhance TETRA technology in North America [TET02].

2.1.1 System Overview

TETRA has been developed as an open standard ensuring vendor competition during the life cycle of an LMR system and permitting interoperability between manufacturer products. Despite involving a high degree of complexity in its design, it offers a high degree of flexibility to its users as a tradeoff. TETRA supports both direct and trunked mode communications between mobile stations with the ability of “dual watch” operation for a range of services including packet mode data, circuit mode data, short data messages, and voice.

Radio access is based on a four-slot Time Division Multiple Access (TDMA) frame occupying a bandwidth of 25 kHz. The frame format of the TETRA air interface is shown in Figure 2.1. In Voice Plus Data (V+D) trunked mode, each user can be accommodated by a single time slot resulting in an effective bandwidth of 6.25 kHz per user. This makes TETRA an attractive solution for efficient, narrowband, operation as mandated by the National Telecommunications and Information Administration (NTIA). In Direct Mode Operation (DMO) TETRA can take advantage of situations where the mobile stations are outside the base station coverage area. DMO can also be used within the coverage area of a base station if additional security is required between two users.

One useful feature offered by TETRA that is essential for all public LMR systems that handle emergencies and mission-critical communications is the ability to prioritize important traffic. Signaling information can be given higher priority than user information and if necessary it can “steal” sub-slots from traffic channels controlled by the TETRA system. A good example of this feature is implemented by the Short Data Service (SDS). SDS specifications define both “access priority” and “traffic stealing” fields that can be set to give a message maximum priority [EN01].

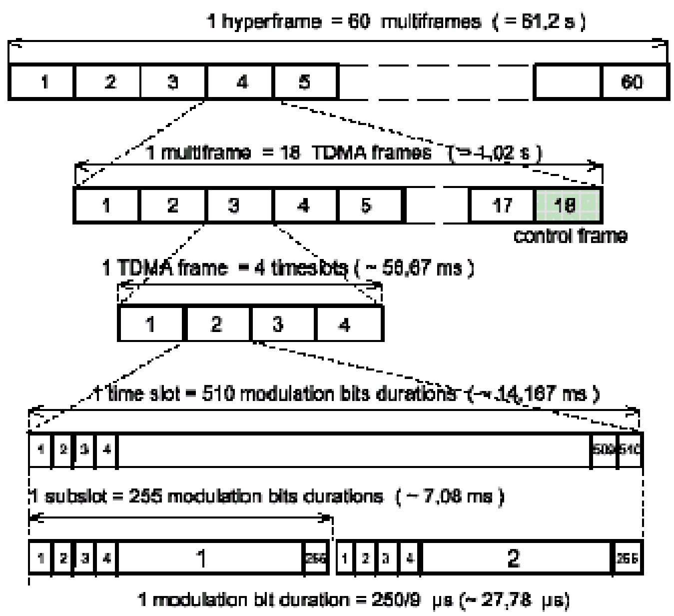

Figure 2.1: TETRA frame format from [EN01].

With the emergence of multimedia services in recent years, the need for data capable radios has become important. TETRA offers both connection oriented (X.25) and connectionless (IP) packet and circuit mode data services. Furthermore, it can combine up to four TDMA slots (channels) per user providing a maximum data transfer rate of 28.8 Kbps. The ability to combine multiple TDMA slots to support different user services running concurrently is one of the strengths of TETRA. Envision the implications of a scenario where a TETRA terminal uses one slot for voice communication, two slots for video imaging transfer and the remaining slot for a data connection to a central database. Even in simple terminals that can support only single-channel operation it is still possible to use the signaling channel for control messages, SDS, or location updates while simultaneously carrying out a voice conversation on the traffic channel.

Security of information is of paramount importance in every public safety LMR system. Many of the existing analog systems have done little if anything at all to address this issue. TETRA implements a number of security mechanisms that provide different levels of security depending on user needs. Due to the way digital systems are implemented, it is relatively easy and inexpensive to integrate robust security mechanisms into the radio infrastructure. Some key security features available by TETRA are mutual authentication of end user and LMR network, signaling level confidentiality, secure functions for air interface key management, and air interface encryption. In addition, an interface for end-to-end encryption is specified in the standards for users seeking a high level of security.

2.1.2 Network Architecture

The network architecture implemented by TETRA is entirely defined by six standardized interfaces, I1 through I6, as illustrated in Figure 2.2. Collectively, these interfaces provide interoperability, interworking, and network management for the TETRA system. No definition is required for the functional sub-entities within a TETRA system (i.e., MSC, BTS, OMC) and their implementation is left to proprietary solutions of the different manufacturers. Our main focus will be to examine the inter-system interface, I3, that is responsible for the interworking of different TETRA networks.

2.1.3 Inter-System Interface (ISI)

The performance of the ISI is critical for all Wide Area Network (WAN) services offered in a TETRA system. The effectiveness of this interface can be measured in terms of the bandwidth requirements and the signaling efficiency of its inter-working protocols. The ISI can support services and features such as mobility management, authentication, group calls, individual calls and messaging. A combination of these services allows roaming, interoperability, and resource sharing at multiple levels between different TETRA systems. When talking about the ISI, we can define the service users as the Switching and Management Infrastructure (SwMI) entities. The SwMI represents all the TETRA equipment for a Voice plus Data (V+D) network [ETS99].

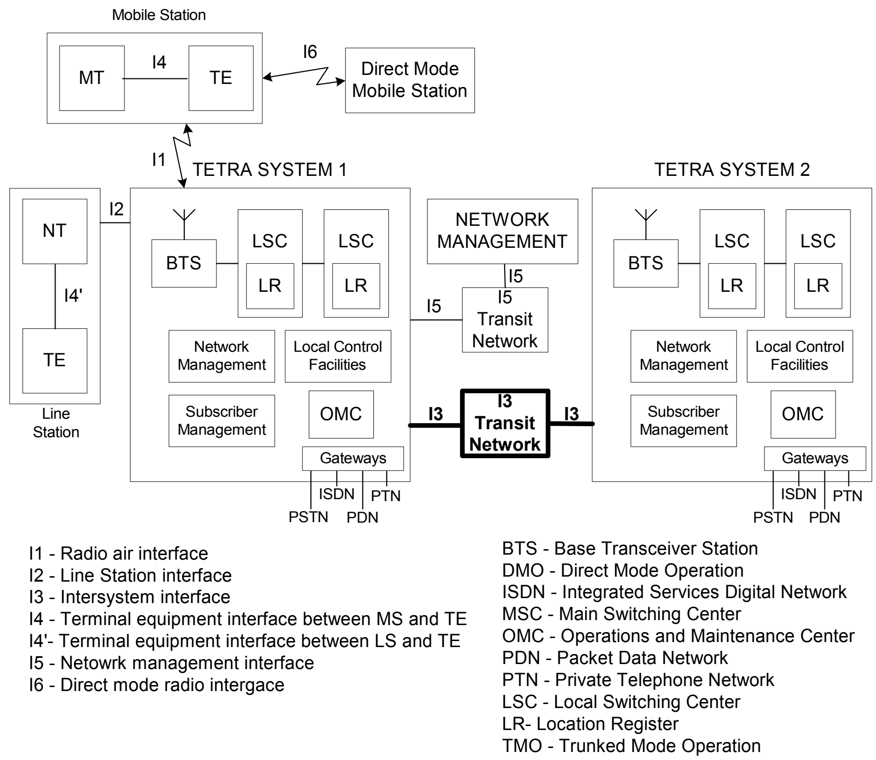

Figure 2.2: Functional network architecture, adapted from [DGI99].

The current TETRA standards define a Private Integrated Service Network (PISN) configuration to support the ISI [IPN99, ETS94]. Essentially, the PISN is a corporate communication network developed using networking equipment called Private Integrated Network Exchange (PINX). The PINXs are capable of supporting a range of networking services. Connection between different PINXs can be achieved through dedicated links or Virtual Private Networks (VPNs) [IPN99]. In the TETRA ISI parlance, the SwMI can be considered as a PINX. Figure 2.3 shows how TETRA systems might fit into the context of a PISN.

TETRA ISI Protocol Stack

To understand how basic services such as individual calls are implemented using the TETRA ISI, we first need to examine the protocol stack that is defined in the standards. All the ser-

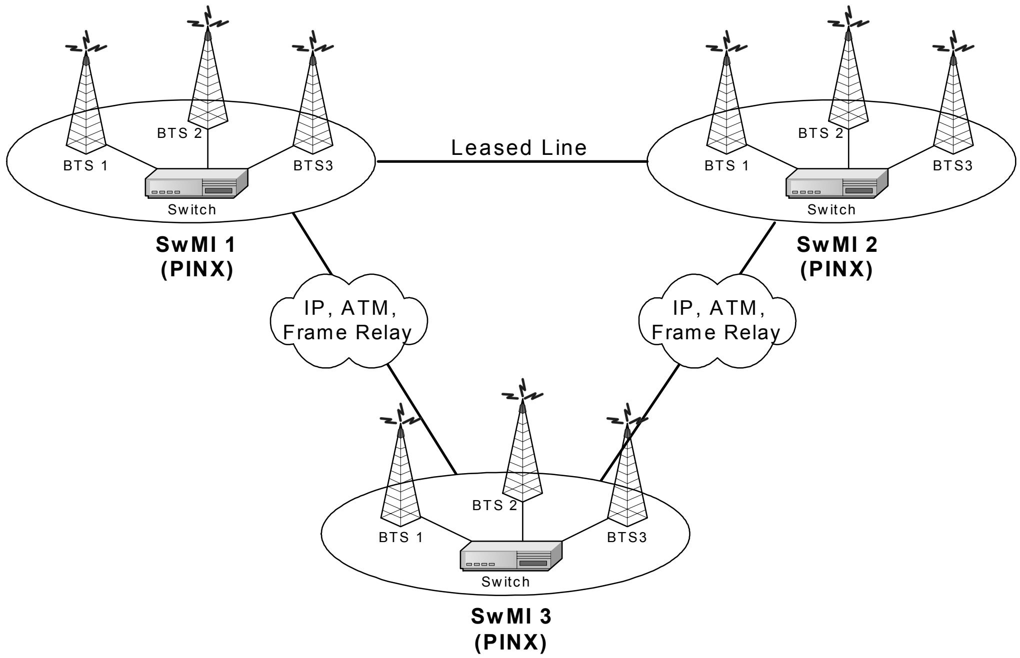

Figure 2.3: TETRA system in a PISN context.

vices supported by the TETRA ISI are standardized as application layer Additional Network Features (ANFs) [CCI88].

As shown in the Figure 2.4, TETRA ISI is based on the QSIG [IPN99] protocol stack which allows the TETRA SwMIs to be interconnected via transit PISNs. A brief overview of the different protocols implemented at the ISI is given below.

TETRA ANFs: Five ANFs are supported by the ISI [ETS99].

- Additional Network Feature - ISI Mobility Management (ANF - ISIMM). Enables SwMIs implementing the ISI to support air interface Mobility Management (MM), authentication and OTAR services.
- Additional Network Feature - ISI Individual Call (ANF - ISIIC). Enables calls to be setup between users of different SwMIs.
- Additional Network Feature - ISI Group Call (ANF - ISIGC). Supports point-to-

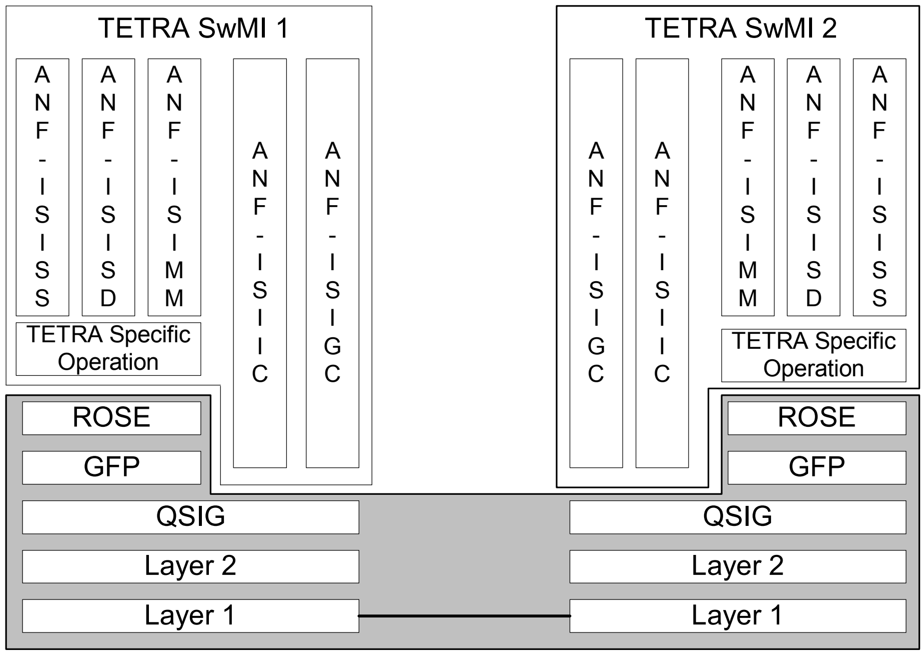

Figure 2.4: Protocol stack for TETRA ANFs at the ISI [ETS99].

multipoint connections between users residing in multiple SwMIs.

- Additional Network Feature - ISI Short Data Service (ANF - ISISD). Supports both point-to-point and point-to-multipoint short data message exchanges between users residing in multiple SwMIs.

- Additional Network Feature - ISI Supplementary Service (ANF - ISISS). Provides a transport mechanism to support the exchange of signaling information for the control of TETRA supplementary services.

Remote Operation Service Element (ROSE): ROSE is a common application service element. Its role is to provide a framework for requesting operations to be performed by the remote systems and for the results to be returned to the local system [ITU97]. In the TETRA ISI, it is used to transport ANF-ISI Application Protocol Data Units (APDUs). The APDUs are conveyed in ROSE facility information elements via a coordination function.

Generic Function Protocol (GFP): The GFP provides all the signaling requirements of the TETRA ISI not supported by the QSIG protocol (see below). It does not directly control the ANF-ISI APDUs but instead provides a generic service to convey them [ETS93].

QSIG: QSIG is a global signaling system based on the ISO reference model. It is used for inter-PINX communications in the PISN. It has many advantages that make it a suitable candidate for the TETRA ISI signaling system [IPN99]:

- Vendor independence
- Guaranteed interoperability (through MoU facilitating interoperability tests)
- Topology Independence
- Unrestricted in number of SwMIs (nodes) supported
- Flexible interconnection (supports any suitable physical layer)
- Support for supplementary services
- Multiple application domain

Since it is defined for a logical reference point, the QSIG protocol stack can be implemented over different physical layer technologies.

2.1.4 Message Sequences for Individual Call using TETRA ISI

A typical scenario of a unit-to-unit call is considered next to investigate the message sequences and associated Packet Data Units (PDUs) that are exchanged over the TETRA ISI. To enable an individual call to be setup between users located in different SwMIs, the ANF-ISIIC application layer entity needs to be invoked. For the duration of each call, the ANF-ISIIC is responsible for passing all the required signaling information between TETRA SwMIs, and takes care of call restoration when users migrate to different SwMIs during the course of a call [EN00]. This is achieved by accessing call control applications and databases (i.e., HLR, VLR) at the SwMIs.

By definition, ANF-ISIIC establishes a QSIG Basic Call (BC) when invoked [EN00]. As mentioned earlier, ANF-ISIIC is an extension of the QSIG BC that enables features required by the TETRA network but not provided by QSIG signaling. Hence, we can construct the message sequences that define an individual call setup over the TETRA ISI by first looking at the QSIG BC message sequence between two PINXs (Figure 2.5) and then adding the ANF-ISIIC TETRA-specific extensions. The standards define all possible scenarios that can result from setting up and tearing down individual calls including call restoration, call migration, call routing, call modification, call maintenance, interaction of services, and transmission control [EN00]. For the purposes of our study, we assume an individual call based on normal procedures, no collisions, and no interactions with other TETRA supplementary services and ANFs. This allows a clearer understanding of the TETRA ISI without having to examine complicated "what-if" scenarios.

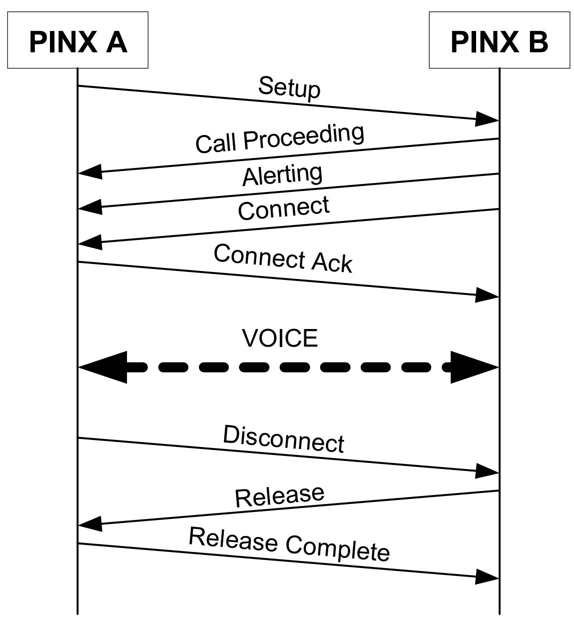

Figure 2.5: QSIG BC message sequence.

A single ANF-ISIIC is invoked for each individual call. The first step in establishing an individual call is to send a QSIG "SETUP" message. The TETRA-specific information will be included in an "ISI-SETUP" PDU by the originating SwMI. After being encoded into a ROSE APDU, it is carried as a facility information element in the QSIG "SETUP" message. Figure 2.6 illustrates how an "ISI-SETUP" PDU is encapsulated in a QSIG "SETUP" message [ETS99]. The terminating SwMI receives the "ISI-SETUP" PDU and replies by sending an "ISI-CALL PROCEEDING" PDU informing the originating SwMI of any characteristics requested in the "ISI-SETUP" that it cannot support. In addition, during the setup phase it includes information such as call time-out that allows the originating SwMI to adjust its timers. A QSIG "FACILTY" message is used to send the "ISI-CALL PROCEEDING" PDU. The next step in the BC sequence is for the terminating SwMI to send a call confirmation indication and call connect message. Depending whether or not on/off hook or direct call setup signaling is used, an "ISI-ALERTING" or "ISI-CONNECT" PDU is sent respectively. The terminating SwMI is informed of the type of signaling used by the air interface at the originating SwMI. To acknowledge that the call establishment is successful, an "ISI-CONNECT ACK" PDU is sent by the originating SwMI. Voice or data can then be exchanged between the SwMIs.

After the voice call or data transfer is complete, a connection clearing sequence is initiated for the invoked ANF-ISIIC. There are a few possible scenarios that may cause a connection to be cleared, i.e., originating SwMI disconnecting, terminating SwMI disconnecting, user moving out of range etc. Furthermore, a call may be cleared prior-to or after a call establishment is completed. For our basic scenario, we will assume that the call is cleared by the originating SwMI after a call connection has been successfully established. In this case, an "ISI-DISCONNECT" TETRA PDU is sent in a QSIG "DISCONNECT" message to the terminating SwMI. "ISI-DISCONNECT" includes the disconnect cause represented by a six bit binary number. Twenty-six different disconnect cause messages are currently defined in the TETRA standard [EN00].

- **QSIG Setup Message**
  - Protocol discriminator, Call reference, Message type, Sending complete, Bearer capability, Channel Identification, Progress Indicator, Calling party number, Calling party sub-address, Called party number, Called party sub-address, Low layer compatibility, High layer comp.
  - **Facility Information Element**
    - Identifier, Length, Protocol profile (Networking Extensions), Network Facility Extension (NFE), Network Protocol Profile, Interpretation APDU
    - **Service APDU: ROSE INVOKE**
      - Identifier, InvokeId, LinkedId, OperationValue
      - **ARGUMENT: tetraIsiMessage**
        - SourceEntity, DestinationEntity
        - **TetraMessage**
          - PDU Type
          - Originating SwMI MNI
          - Routing method choice
          - Call time-out, set-up phase
          - Call time-out
          - Hook method selection
          - Simplex/duplex selection
          - Basic service information
          - Speech service requested
          - Security level at calling user air interface
          - Transmission grant
          - Transmission request permission
          - Call priority
          - Called party address SSI
          - Called party extension
          - SS-CLIR invoked for calling party
          - Calling party address SSI
          - Calling party extension
          - External subscriber number length
          - External subscriber number digits
          - Speech services supported
          - Notification indicator
          - Proprietary
        - **END of TetraMessage**
        - Extension
      - **END of ARGUMENT**
    - **END of Service APDU**
  - **END of Facility Information Element**
- Notification indicator
- **END of QSIG SETUP message**

Figure 2.6: Example of encoding "ISI-SETUP" complementary TETRA ANF-ISIIC information element in QSIG "SETUP" message [ETS99].

2.2 APCO Project 25

APCO Project 25 is an open, digital, LMR standard based on Frequency Division Multiple Access (FDMA) trunked radio operation. It is a consolidated effort by United States federal, state and local government supported by the Telecommunications Industry Association (TIA) [APC99]. The standards themselves are called Project 25 and their development began in 1989. At the time, state and local official represented by National Association of State Telecommunications Directors (NASTD) and APCO respectively had a meeting to address issues in public safety communications. Developments such as the emergence of digital radio, FCC frequency re-farming rules and a lack of interoperability between the different agencies prompted the government officials with the help of industry representatives to begin a standard developing process that later came to be known as Project 25.

2.2.1 System Overview

The main objectives of the Project 25 Phase I standards are to:

- provide LMR digital service that accommodates for a smooth migration path from the existing analog service,
- achieve spectral efficiency in accordance with the NTIA narrowband mandate requirement by moving from 25 kHz channels to 12.5 kHz channels,
- satisfy all the user needs for public safety communications,
- provide interoperability to support inter-agency communications, and
- develop an open standard to promote competition throughout the life cycle of the systems.

Additional objectives have been established for Project 25 Phase II that will reduce channel spacing even further to 6.25 kHz and consider alterative medium access technologies for the common air interface (CAI). Primary candidates are two- and four-slot TDMA systems [DSD00].

The advantage of Project 25 being developed as a new LMR technology standard is that a well structured systems engineering approach can be applied. User requirements are developed by the Project 25 Steering Committee, defined in the Statement Of Requirement (SOR) [Pro99], approved by different subcommittees, and classified as "Mandatory", "Standard Option", or "Option." "Standard Option" has been termed to define options offered by Project 25 systems that have to comply with the Project 25 specifications if implemented by a manufacturer. "Option" defines those options that are proprietary to a specific manufacturer and do not have to comply with the Project 25 specifications. Based on the SOR, a system architecture has been proposed based on six open interface standards as shown in Figure 2.7.

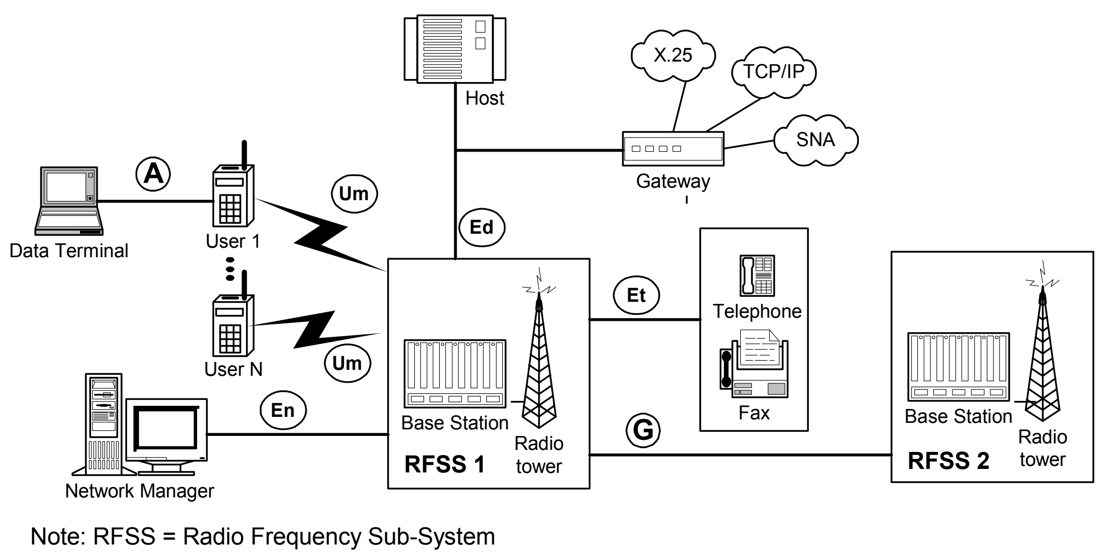

Figure 2.7: Project 25 open system interfaces.

2.2.2 Network Architecture

The Project 25 network architecture consists of a common air interface "Um," a telephone interconnect interface "Et," a data port interface "A," a host and network data interface "Ed," a network management interface "En," and an inter-subsystem interface "G" [TSB95]. As part of Project 25 Phase II at least two more interfaces are being standardized, the open console interface and the open base station/repeater interface [DSD01].

One of the most important interfaces in the Project 25 standard is the CAI interface, represented by reference point "Um" in Figure 2.7. This interface makes it possible for mobile and portable radio equipment from different manufacturers to interoperate on any Project 25 system. Both conventional and trunked communications are supported for voice and data services. Reference point "A" defines the connection of data terminals (laptops, PDAs, etc.) to the subscriber units through a common port. This is intended to act as a transparent pipe connecting the wireless subscriber units to Fixed Network Equipment (FNE) implementing X.25, TCP/IP, or SNA networking protocols. The "Ed," "En," and "Et" reference points define the open interfaces between the Radio Frequency Sub-System (RFSS) and external FNE. Services such as connection to the PSTN, network management and connectivity to host computers (using native open interface) are provided through these interfaces. Access to existing networking technologies is also provided though a gateway supporting the "En" reference point. Our main focus for this study is to examine the Project 25 inter-system interface ("G" reference point) that is responsible for the inter-working of different Project 25 systems.

2.2.3 Inter-RF Subsystem Interface

The Inter-RF Subsystem Interface (ISSI) makes it possible for RFSSs to be connected with each other to form a WAN when extended service coverage is required for an LMR communication system. Furthermore, the ISSI supports an interface that is capable of connecting devices using different frequency bands and technologies given that they implement the same suite of industry standard protocols. All services required to support the free roaming of users between the RFSS are defined in the ISSI standard [TSB96a]. These include location tracking, attach and detach procedures, data management, end-to-end security, group voice call, and individual voice call.

No particular network configuration is required for the ISSI. A selection of either a private or public network links can be used implementing different networking protocols. The main constraint for any chosen implementation is that the ISSI messaging packets need to be preserved in their original format. The voice traffic is transmitted as coded speech representations of the original analog voice [TSB96a].

Security mechanisms for the ISSI are defined based on indications from the TIA WG8.3 Encryption Subcommittee [TSB96b]. Security features supported include authentication of the ISSI ingress and egress points, end-to-end encryption of the information traversing the interface and secure transfer of Over The Air Re-keying (OTAR) information.

Currently, no specific protocols for signaling, mobility management, and WAN service support have been defined. The ISSI standard defines primitives based on the Mobile Application Part (MAP) messaging format used in GSM. MAP-like extensions are suggested for messaging requirements not supported directly by MAP, i.e., group call mobility management [TSB96a]. All messages that traverse the ISSI are included in a general message format outlined in Table 2.1.

2.2.4 Message Sequences for Individual Call using Project 25 ISSI

A typical scenario of a unit-to-unit call is considered next to explain the message sequences and associated PDUs that are exchanged over the Project 25 ISSI. Since the standard does not define the whole protocol stack implemented at the ISSI, only the exchange of MAP-based application layer PDUs is considered. This information is sufficient to provide an overall understanding of how Project 25 standards envisage a unit-to-unit call message sequence.

Table 2.1: Project 25 ISSI General Message Format

Field	Length(bits)	Remarks
Source Address	48	Address of originating RFSS [Network (20 bits), System (12 bits), RFSS (8 bits), HLR/RFSS (8 bits)]
Destination Address	48	Address of terminating RFSS (field structure similar to Source Address)
Sequence Number	16	First bit - Origination = 0, Response =1, followed by 15 bit numeric value
Size of Message	16	Size in bytes (Size of Message[bytes] = Payload + 20)
Manufacturers ID	8	Manufacturer specific
Opcode	8	Code representing all supported ISSI messages (26 defined)
Payload	Variable	Messages representing each supported opcode
Checksum	8	CRC checksum (all fields included in calculation)

We assume that no collisions will occur during the setup of the call, and that the HLR databases are physically co-located at the same site as the RFSS. Figure 2.8 illustrates the unit-to-unit call message sequence used [TSB96b].

Consider the scenario where a subscriber unit registered in RFSS 1 originates a unit-to-unit call directed to a subscriber unit registered in RFSS 2. The first message RFSS 1 sends across the ISSI requests routing information for the subscriber unit in RFSS 2 ("Send Routing Information" message). The subscriber unit is identified by its Subscriber Unit ID (SUID). The HLR at RFSS 2 replies with a "Routing Information" message providing location information to RFSS 1. To test the routing information available and to make sure that the target unit is capable of participating in the unit-to-unit call, RFSS 1 sends an

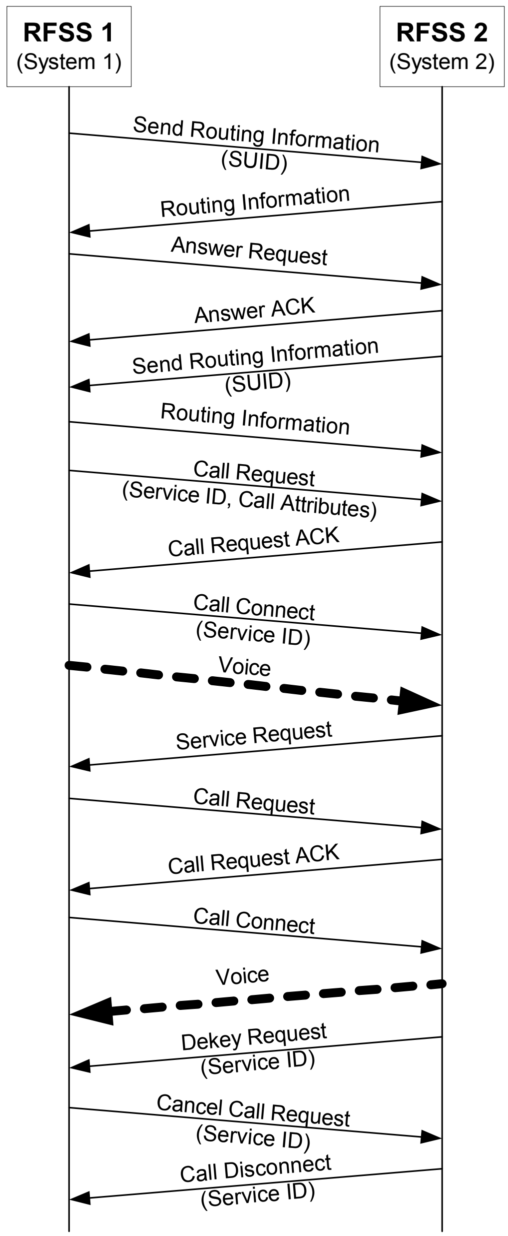

Figure 2.8: Project 25 unit-to-unit call message sequence.

"Answer Request" message. RFSS 2 replies with an "Answer ACK" message signaling that both the target unit and the RFSS are capable of receiving the unit-to-unit call.

Before any voice packets can be exchanged over the ISSI, RFSS 2 also sends routing information messages to RFSS 1 in order to establish a path to the originating subscriber unit. In our scenario, the HLR is co-located with the RFSS and, hence, the routing process is trivial since we know where each of the subscriber units is registered. In some cases however, the HLR may not be physically co-located with the RFSS to which a particular unit is registered. It may be located in a different physical site of the same system or even in a different system altogether. Once all the routing information is exchanged and the subscriber unit availability is confirmed, RFSS 1 sends a "Call Request" message to start a call. All the required service attributes to initiate the call are included in the "Call Request" message. RFSS 2 responds with a "Call Request ACK" message assuming that it can accommodate the service level requested.

A "Call Connect" message is sent by RFSS 1 to inform RFSS 2 that the call may begin and voice packets are sent over the ISSI. Before the subscriber unit at RFSS 2 can send any voice packets back to RFSS 1, a service request message exchange takes place that will grant transmit permission to RFSS 2. After permission is received, RFSS 2 can send voice packets to the originating subscriber unit at RFSS 1. The end of voice transmission is marked by a "Dekey Request" message from RFSS 2.

A call ends with a "Cancel Call Request." Both the controlling and participating RFSS can invoke this message. In our scenario, RFSS 1 that is controlling the call invokes the "Cancel Call Request" message. The last message to complete the call is "Call Disconnect" by RFSS 2 that takes care of confirming the call termination and freeing up any resources used for the unit-to-unit call.

2.3 Comparison of APCO Project 25 and TETRA

Table 2.2 compares the main parameters and features between Project 25 and TETRA. There have been long debates among public safety officials and manufacturers as to which one of these standards best suits the needs for a robust, interoperable LMR system. The answer to this question from an engineering standpoint is often hard to quantify since most of the comparisons between the two systems seem to be obscured by techno-political interests that are associated with such decisions.

Project 25 was designed based on the requirements of the North American public safety LMR networks while TETRA was designed based on the demands of the European LMR networks (both public and private). Despite these design objectives, one cannot eliminate the possibility of implementing a TETRA system in North America. This is particularly true if we consider that TETRA is a proposed solution for Phase II of Project 25.

2.3.1 Backward Compatibility and Migration

The primary advantage of Project 25 over TETRA is its backward compatibility with existing analog systems. This is instrumental in the deployment of any new digital LMR system since it will allow for a smooth migration between the analog and digital systems. In addition, it will potentially minimize the financial and operational impact during the transition. TETRA is based entirely on digital technology and as such offers no compatibility with analog systems. Any implementation of a TETRA system will essentially require an entirely new network deployment independent of the existing analog systems. Even though it is possible to design TETRA mobile terminals with added functionality to support compatibility with analog systems, no such development is foreseen in the near future unless there is a significant market demand for such a unit.

Table 2.2: Basic Parameters and Features for Project 25 and TETRA

Parameters/Features	**Project 25**	**TETRA**
Carrier spacing	12.5 kHz (Phase I)	25 kHz
Time slots per carrier	N/A	4
Traffic channel bandwidth	12.5 kHz	6.25 kHz for V+D and DMO, 25 kHz for PDO
Gross bit rate	9.6 Kbps in 12.5 kHz channel	36 Kbps in 25 kHz channel
Net bit rate	5.8 Kbps in 12.5kHz channel	28.8 Kbps unprotected data, 19.2 Kbps protected data (low), 9.6 Kbps protected data (high) in 25 kHz channel
Access technology	FDMA	TDMA
Operation modes	Digital (trunked, conventional), analog	Digital (trunked, conventional)
Present frequency allocation	136-162 MHz; 146-174 MHz; 403-433 MHz, 438-470 MHz; 450-482 MHz; 482-512 MHz; 806-870 MHz	380-400 MHz; 410-430 MHz; 806-825 MHz, 851-870 MHz (designed for operation in 150-900 MHz range)
Digital modulation format	C4FM (Phase I)	$\pi/4$-DQPSK
Vocoder	IMBE (Improved Multi-Band Excitation)	ACELP (Algorithmic Code Excited Linear Predictive)
Backward compatibility	Yes, APCO 16 (analog)	No (analog mode is not supported)
OTAR support	Yes	Yes
Short Data Service (SDS)	No	Yes, up to 2047 bits (Type 4)
Standardization of CAI based on OSI protocol stack)	Layer 2 (data link)	Layer 3 (network)
Mobile/Portable operation	Half-duplex	Full-duplex, half-duplex

2.3.2 Carrier Spacing

The carrier spacing in Phase I of Project 25 is 12.5 kHz.This meets the narrowband requirements set by the NTIA. TETRA on the other hand uses 25 kHz channels that are shared between four users resulting in an effective bandwidth of 6.25 kHz per user (in Packet Data Optimized [PDO] mode the full 25 kHz bandwidth is required). Considering the carrier spacing, with everything else being equal, TETRA can theoretically support twice the capacity of Project 25. However, in situations where a service area requires only a single channel to satisfy its operational needs, a TETRA system would still require the full 25 kHz bandwidth as opposed to a 12.5 kHz channel required by a Project 25 system.

2.3.3 Data Rates Supported

Packet and circuit data transmission is supported by both standards. The advantage of TETRA is that it can:

- offer higher data rates, and
- support concurrent voice and data communications.

TETRA can offer a maximum net bit rate of 28.8 Kbps compared to 5.8 Kbps supported by Project 25. In addition, TETRA can offer a range of bit rates to users (2.4-28.8 Kbps) depending on the application requirements, the desired level of protection, and the equipment capabilities.

2.3.4 Short Data Messaging

One useful feature supported by TETRA is SDS. It supports point-to-point and point-to-multipoint data services up to a maximum of 2047 bits (Type 4). SDS can be used effectively to communicate both group and individual messages (user-defined and predefined) without having to establish a dedicated connection. This is achieved by utilizing the extra capacity on the signaling channels rather than having to reserve a traffic channel when sending SDS messages. Consequently, SDS messages can be exchanged while users are engaged in a voice or data call. Project 25 does not offer a similar service even though it supports two-way pre-programmed messaging as a "Standard Option."

2.3.5 Standardization of Common Air Interface (CAI)

The air interface of TETRA is defined up to OSI layer 3 (network layer) where as APCO Project 25 is only defined up to OSI layer 2 (data link layer). The lack of a standardized network layer in Project 25 has potential interoperability issues. Recent experience has shown that leading LMR manufacturers have defined their own proprietary network layer protocols making it difficult for smaller manufacturers to join the market competition. This is contrary to the objectives of the Project 25 standardization group for ensuring a competitive product through the development of an open standard.

2.3.6 Power Levels

Table 2.3 shows a comparison between power levels required for Project 25 and TETRA equipment. When compared to TETRA, Project 25 equipment is typically designed to operate at higher power. From an infrastructure point of view, this means that for a similar coverage area TETRA needs to have a higher number of base stations (and/or repeaters) than Project 25. The advantage of Project 25 power levels is that they preserve the range-repeater siting that is currently in use by the analog systems.

Furthermore, Project 25 mobile and portable subscriber units can communicate over longer distances in "talkaround" mode (direct mobile-mobile communication) and can access base stations from further away in trunked mode. This is of particular significance to the United States where large coverage areas are often required. The tradeoff for supporting high powered transmissions is that Project 25 mobile and portable units need to have larger batteries compared to similar TETRA units. In effect, this implies higher costs and larger form factors for Project 25 mobile and portable units.

Table 2.3: Power Levels for Project 25 and TETRA

Equipment	Project 25 Equipment Power	TETRA Equipment Power
Base Station	6-350 Watts	20-25 Watts
Mobile	10-110 Watts	3-10 Watts
Portable	1-5 Watts	1-2 Watts

2.3.7 Speech Coder

Project 25 radios use an Improved Multi-Band Excitation (IMBE) speech coder for digital processing. The IMBE speech coder is used to digitally encode the speech signal at 4.4 Kbps or decode the speech signal back to analog audio. 2.8 Kbps of forward error correcting codes (FEC) are added increasing the signaling rate to 7.2 Kbps.

TETRA uses an Algebraic Code Excited Linear Prediction (ACELP) based speech coder. The coding process involves two steps. First, 30 ms samples of speech are used to calculate the synthesis filter parameters. Then, the excitation sequence is calculated by dividing the 30 ms speech frames into 7.5 ms subframes. Excitation parameters are calculated for each subframe. The ACELP algorithm produces 137 bits per 30 ms of speech that is equivalent to 4.567 Kbps. When channel coding is added, 216 bits are required for 30 ms of speech resulting in a gross bit rate of 7.2 Kbps [ETS98].

2.4 Overview of SIP

The Session Initiation Protocol (SIP) is an application layer signaling protocol used for creating, managing and terminating multimedia sessions over packet networks. It has recently experienced strong support over H.323 [ITU99] due to its efficiency and simplicity. SIP can support unicast and multicast communication involving one or more participants. Furthermore, it has a lot in common with ubiquitous Internet protocols such as HTTP and SMTP making SIP text-based and highly extensible. The SIP Working Group within the Internet Engineering Task Force (IETF) is responsible for the protocol development [HSS99].

2.4.1 SIP Logical Entities

Two basic classes of network entities are defined in SIP, clients and servers. The client, known as User Agent Client (UAC), is an application that initiates SIP requests. A server responds to the requests. Four types of logical entities have been defined for a SIP network. Each logical entity has particular functions in a SIP call acting as client, server, or both a client and a server. Multiple logical SIP entities can be, and often are, implemented in a single physical device.

User Agent (UA): The UA represents the originating and terminating points of a SIP call. It is defined as an application that supports both a UAC and a User Agent Server (UAS). Hence, a UA can initiate and respond to SIP requests.

Proxy Server: The proxy server is an intermediary entity that makes requests on behalf of other SIP clients. It acts both as a server and a client. Requests received by the proxy can be either serviced internally or passed on to other servers after making any necessary packet translations. The proxy server is placed between the UAC and the far-end UAS.

Redirect Server: The redirect server maps the destination address of the SIP requests received into zero or more new addresses and returns them to the client. It differs from a proxy server as it does not create its own requests. In addition, it differs from a UAS since it does not accept calls.

Registrar: The registrar receives "REGISTER" requests. This provides the server with an address it can use to reach a user during a SIP session. The registrar can also support location services such as updating location databases

2.4.2 SIP Signaling and Call Flow

SIP supports two types of messages, requests (from client to server) and responses (from server to client). The clients are identified by telephone numbers or e-mail type addresses assigned to them when they sign up for service. As mentioned earlier, SIP messages are text-based (using the ISO 10646 character set). Each message consists of a start-line, followed by a header and an optional message body. The start line identifies the message type (response or request) and the protocol version. Each request is associated with a request method and each response is associated with a response code as shown in Tables 2.4 and 2.5, respectively. The message header conveys additional attributes related to the request or response and can expand to several lines by accepting multiple comma separated values. The message body describes the type of session that will be initiated and includes information on the media to be exchanged.

Table 2.4: SIP Request Methods

Method	**Description**
INVITE	Initiates call signaling.
ACK	Confirms that a final response for "INVITE" has been received
OPTIONS	Queries the server on its capabilities
BYE	Used by UAC to release/terminate call
CANCEL	Cancels request in progress
REGISTER	Registers address listed in the "To: <>" header field with SIP server

Table 2.5: Response Codes

Response Code Prefix	**Function**
1xx	Provisional, searching, ringing, queuing
2xx	Success
3xx	Redirection, forwarding
4xx	Client mistakes
5xx	Server failures
6xx	Global failure (busy, refuse, not available anywhere)

To examine SIP signaling further, a simple session establishment and call termination sequence are examined. Figure 2.9 illustrates the message flow between two clients. We assume they have email-type SIP addresses `UserA@xyz.edu` and `UserB@xyz.edu`, respectively. The call establishment begins with the SIP "INVITE" message sent to UserB. Once UserB receives the "INVITE" message, he or she replies using a "100 Trying" response message. At the same time, the UAS sends a "180 Ringing" response message to the UAC and starts ringing to notify UserB that a call has been received. When UserB picks up the call, the UAS sends a "200 OK" response message. The UAC confirms the final response for the "INVITE" request by sending an "ACK" message. After this, media can be exchanged (i.e., voice, data). If UserA decides to hang-up a "BYE" request message is sent to UserB's UAS at SIP address `UserB@xyz.edu`. Finally, the UAS sends a "200 OK" response message to confirm the end of the call.

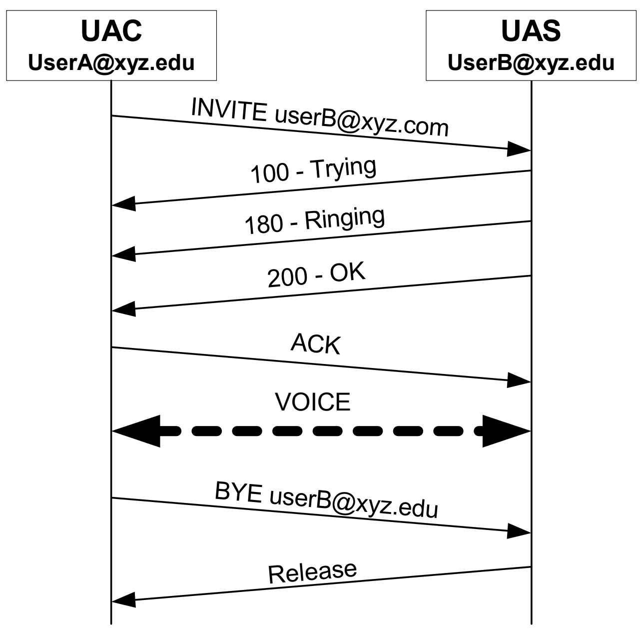

Figure 2.9: SIP session establishment and call termination sequence.

We have described a typical SIP call from a high level. To get a better idea of the actual information exchanged, we examine the structure of the "INVITE" request message used in the call.

```
(1)    INVITE sip:UserB@xyz.edu SIP/2.0
(2)    Via: sip/2.0/udp server1.xyz.edu
(3)    From: sip:UserA@xyz.edu
(4)    To: sip:UserB@xyz.edu
(5)    Call-ID: 12345@server1.xyz.edu
(6)    CSeq:1 INVITE
(7)    Subject: Example
(8)    Content-Type: application/SDP
(9)    Content-Length: 99
(10)   (message body)
```

The numbers in front of each line are added for reference purposes and are not used in the actual SIP message. The start line (1) identifies that the method type is "INVITE", the SIP address of the called party is `UserB@xyz.edu`, and version 2.0 of the SIP protocol is used. The header `Via:` (2) includes the address of the previous hop while the `From:` (3) and `To:` (4) fields include the SIP users engaged in the call. The `Call-ID:` (5) includes a globally unique identification of the call between UserA and UserB. The command sequence `CSeq: 1` (6) identifies the transaction as being an "INVITE" message. Information describing the call can be added in the call `Subject:` (7) field. The `Content-Type` (9) identifies the type of message body used (if a message body is used). In the scenario we considered a Session Description Protocol (SDP) body type is likely to be used. The size in bytes of the message body is included in the `Content-Length` (9) field. The SDP-specific messages are included in the final part of the message (10).

2.5 Summary

This chapter reviewed and compared the two dominant digital LMR communication standards available today, TETRA and APCO Project 25. TETRA was originally developed for the European market, but its deployment has gradually been expanded to countries outside of Europe. APCO Project 25 has been developed in the U.S.A. and is quickly becoming the *de facto* LMR standard for public safety communication systems in the U.S.A. Particular attention has been focused on the discussion of the inter-system interface for each standard since this is the theme of subsequent chapters. To better understand the interactions at the inter-system interface, a message sequence associated with a typical unit-to-unit call has been examined for each of the two standards considered. Finally, we provided an overview of SIP, an application layer signaling protocol that provides features suitable for next generation packet switched inter-system interfaces.

Chapter 3

Research Objectives and Motivation

As we have seen in Chapter 2, Project 25 and TETRA LMR networks requiring wide-area coverage were initially implemented using a circuit-switched infrastructure following the convention of other digital cellular technologies, such as GSM and CDMA. One of the main problems associated with circuit-switched networks is their vulnerability to single points of failure and their inefficient utilization of network capacity. This is especially true with the increasing proportion of data traffic experienced in recent years. To address these issues, a number of vendor solutions are becoming available that support Project 25 and TETRA using a packet-switched infrastructure. In these systems switching is achieved by using packet-switched routers, a technology leveraged from the Internet and other packet-switched networks, rather than conventional telecom switches.

To better understand the operation and performance of this new infrastructure, we develop a model that examines the behavior of a packet-switched LMR network. Our research objective is to characterize the loading effects on the network and investigate the performance of the ISI. Due to the dynamic behavior and complexity of such a model, its performance can only be evaluated using simulation experiments. We identify throughput, call setup time, ETE delay, and delay variation as the most suitable metrics for characterizing the performance of the ISI.

In Section 3.1 we give an overview of circuit-switched LMR networks and in Section

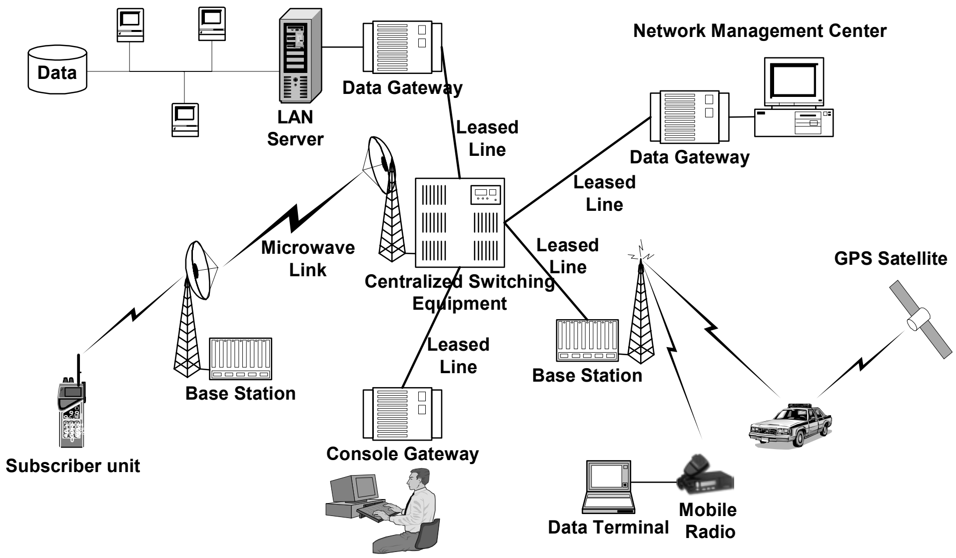

Figure 3.1: Circuit-switched LMR network [EFJ01].

3.2 we introduce the concept of a packet-switched LMR network that forms the foundation of our study. In Section 3.3 we make the case for using SIP rather than H.323 as our signaling protocol due to its simplicity and efficiency. In Section 3.4 we define the problem statement and identify the performance metrics. The system boundaries and the performance evaluation technique are also discussed.

3.1 Circuit-Switched LMR Network

Traditionally, and up to this date, LMR networks have been connected using circuit- switched technology inherited from the telephony world. This same technology has been used since the days of analog systems. Connections are established via leased telephone lines and dedicated microwave links, both of which are inefficiently used and are capital-intensive. With the recent deployment of digital LMR systems, traffic exchange has become more bursty and non-uniform, and as such, less suitable for circuit-switched networks. Figure 3.1 indicates how present LMR networks are inter-connected.

Base stations, dispatch consoles, network management equipment, and gateways to other networks are all connected to a centralized switch using microwave links and leased lines, typically providing T1 or E1 service. These dedicated connections require a high fixed cost to install and a high monthly recurring cost to maintain. In addition, most of the equipment uses proprietary technology that hinders interoperability, restricts the users to a single vendor for the life-cycle of the LMR system, and requires specially trained technicians to repair and maintain. With the advent of new digital wireless technologies and the explosion of the Internet, open standards have been developed that can accommodate voice transport over packet-switched networks.

3.2 Packet-Switched LMR Network

The ability to digitize voice and send it as data packets over a network has enabled the provisioning of integrated services over a single data network. Traffic integration is, in fact, the main business driver for packet voice [Wri01]. Rather than having two disparate networks, one circuit-switched to carry voice and another packet-switched to carry data, many companies have moved to a single packet-switched backbone network to accommodate all of their needs. Besides voice traffic, such a network could be used to carry internal data traffic, operations administration and maintenance (OAM) traffic, video, images, and short messages. Figure 3.2 indicates how a packet-switched LMR network can be interconnected [Mar99].

New digital LMR systems such as TETRA and APCO Project 25 packetize voice and compress it before transmitting it over the air interface. Therefore, when the voice comes to the wired portion of the network it is already packetized. It follows that it would be more suitable to use packet technology in the wired portion of the network as well.

A packet-switched LMR network makes use of a distributed architecture that essentially eliminates single points of failure. In the event that a network component fails, alternative communication paths are likely available. In addition, dispatch stations and gateways can

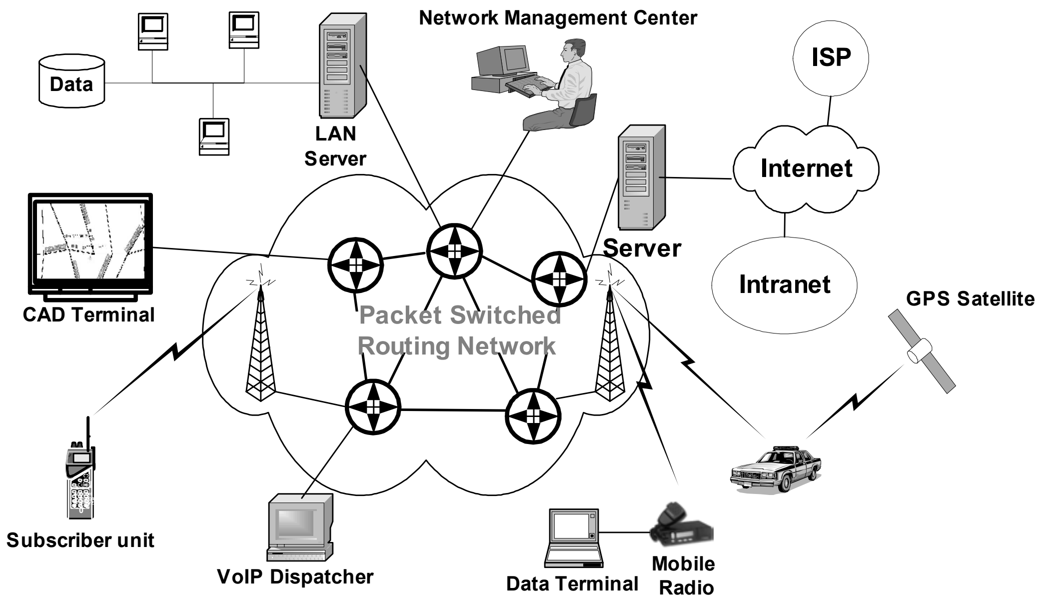

Figure 3.2: Packet-switched LMR network [Mar99].

be attached to any point in the network infrastructure without the need for any specialized gateways. This level of robustness and flexibility is beneficial for mission-critical and emergency LMR communications.

From a financial standpoint, a packet-switched network has many advantages. It can scale easily and seamlessly to accommodate a larger user base or an extended coverage area. Higher capacity can be added incrementally without having to restructure the whole switching hierarchy. Integration with information technology systems and applications requires no additional interfacing since a common IP platform can be used. Due to the widespread availability of packet-switched devices, commercial off the shelf (COTS) solutions, both hardware and software, including routers, gateways and network management applications, can be readily used.

3.3 Selecting a Signaling Protocol

Apart from selecting the appropriate network technology, a suitable signaling protocol is required for setting up and tearing down multimedia sessions. The main competing signaling protocols for packet-switched networks are the Session Initiation Protocol (SIP), developed by the IETF [SCF96] and H.323, an ITU recommendation under the title "Packet-based Multimedia Communication Systems" [ITU00a].

H.323 was initially developed to provide video conferencing capabilities over a local area network (LAN) and later extended to provide telephony capabilities over the Internet. Many companies developing IP telephony networks in the mid-1990s adopted H.323 primarily due to a lack of any other analogous standard. With the introduction of SIP, the IETF aimed at creating a more powerful signaling protocol that would follow the Internet paradigm and would be simple to implement [HSS99].

It is important to realize that the IETF and ITU follow different philosophies when developing their protocol specifications. The ITU tries to anticipate everything that anyone would ever want to do and defines all possible scenarios in its protocol specification. This can take several years to realize and typically results in lengthy specifications. The IETF follows a "bottom-up" approach. Rather than defining the architecture first, and then implementing the various protocols between the nodes, it provides protocols that solve specific needs of the user community. SIP is no exception to this philosophy and its simplicity can be attributed to this fact. Figure 3.3 shows the basic call setup procedure using SIP and H.323.

Without going into any details about the structure of the individual messages, Figure 3.3 shows that ten messages need to be exchanged in order to setup a basic call using H.323 (assuming "FastStart" call setup) as opposed to five messages required by SIP. Even though SIP messages are inherently longer than H.323 messages (due to the additional information carried in each message and the text-based encoding used) it is generally preferable to send a few larger packets rather than many smaller packets [WP01]. Because SIP is a simpler

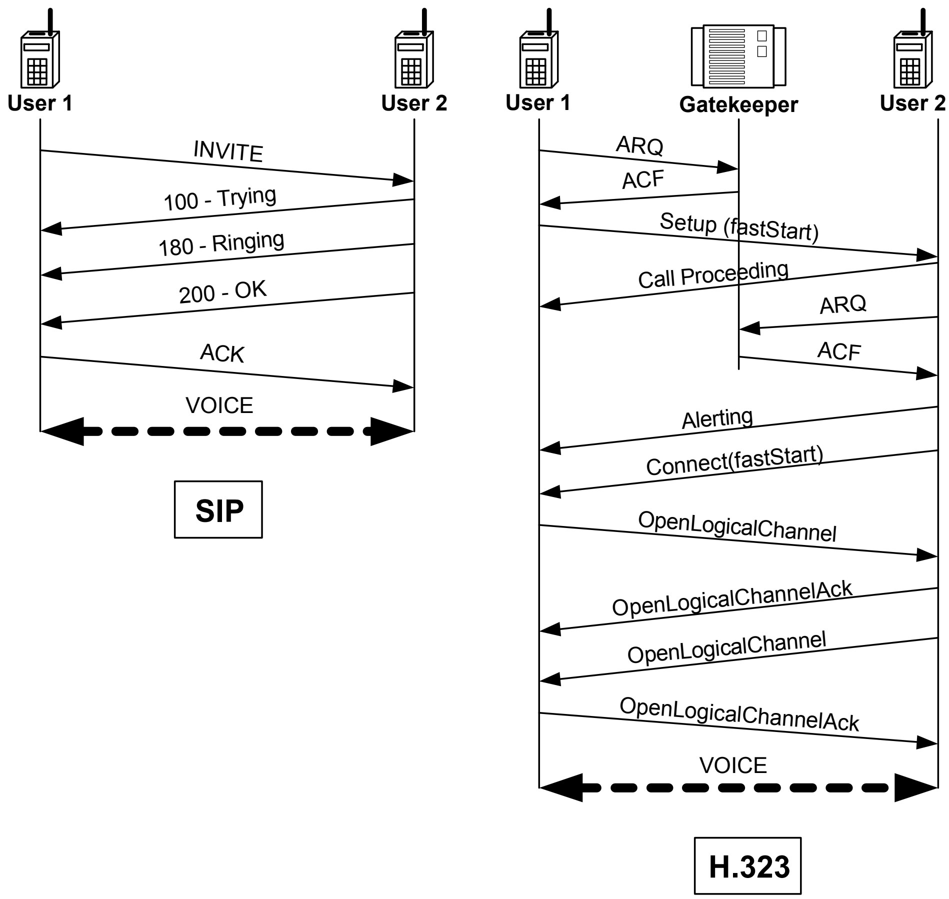

Figure 3.3: SIP and H.323 call setup message flows.

protocol, it can be implemented using less code, uses less dynamic memory and can achieve shorter call setup times, a critical factor in LMR communications. Furthermore, H.323 requires TCP for reliable message transport since it has no timers or message retransmission schemes. This introduces additional round-trip delay partly due to the initial handshake required to set up a TCP connection [Joh01]. SIP can use UDP, TCP, or any other transport layer protocol available even though UDP is preferred due to its lower latency and higher efficiency [Cam02]. Table 3.1 provides a comparison of the main features of of SIP and H.323.

Table 3.1: Comparison of SIP and H.323 Main Features [Joh01][WP01]

Feature	**SIP**	**H.323**
Transport	TCP, UDP, other	TCP
Complexity	Simple	Complex
Encoding	Text-based	Binary
Conferencing	IP Multicast	Multipoint Control Units (MCU) required
Philosophy	Horizontal architecture that re-uses Internet elements	Vertically-integrated suite of protocols
Specification	Signaling only	"Blanket" specification (defines how other protocols can be used to create a service)
Code Size	Small	Large
Dynamic Memory Usage	Small to medium	Large
Host Application	Simple	Much more complex
Debugging	Easy	Difficult

With this information in mind, and based on current trends in the LMR industry, SIP has been selected as the preferred signaling protocol for our study.

3.4 Research Objectives and Methodology

Having established that a packet-switched network using SIP is the preferred networking technology for interconnecting future LMR networks, we next outline the fundamental objectives of this thesis and define the basic simulation model.

3.4.1 Problem Statement

This research study has two main objectives. The first objective is to develop a simulation model for an LMR network that uses a packet-switched technology to interconnect base stations[1]. As we have seen in Chapter 2, the interface that connects the different RF systems to form a WAN is referred to as the ISI. An ISI for packet-switched LMR networks has not yet been standardized for any digital LMR technology. Hence, the simulation model developed should be useful for performance evaluation, both in this study and in any subsequent research efforts in this area.

The second objective of this study is to use the simulation model to characterize the loading effects on the LMR network and investigate the performance of the ISI. This is achieved by simulating the traffic traversing the wired portion of the network when users located in different LMR systems try to communicate[2] with each other. We consider different underlying packet transport technologies for carrying data over the LMR network. The effects of using IP, ATM, and Frame Relay are examined based on service level agreements (SLAs) that conform to present market offerings.

To characterize the loading effects on the LMR network, a detailed message flow is generated to represent network traffic in a typical unit-to-unit call across the ISI. SIP is used for signaling and RTP is used for transport of real-time audio. Based on these message flows, several scenarios are considered and analyzed. The effects of varying the number of LMR users in the network is examined by observing the throughput, call setup time, ETE delay,

[1] Hereon, the term base station refers to the RFSS in APCO Project 25, and the SwMI in TETRA.

[2] The term communicate is restricted to connote voice communications.

and delay variation. Each of these performance metrics is described in section 3.4.4.

3.4.2 Selecting an Evaluation Technique

Three techniques are generally available that may be used to evaluate the performance of the LMR network: (i) analytical models with closed form numerical solutions, (ii) empirical measurements, and (iii) simulation experiments.

The LMR network is defined by many parameters, both deterministic and stochastic, that collectively influence its performance. The task of developing an analytical model that captures the complex interaction of these parameters and predicts the performance of the LMR network is not feasible, even if the precise modelling parameters are known *a priori*. Alternatively, simulation models are capable of investigating dynamic behavior patterns of complex systems. In recent years, the availability of improved simulation tools and high speed processors have made it possible for researchers to study and evaluate complex systems. By taking advantage of these resources, we use simulation to evaluate the performance of the LMR network. OPNET Modeler is used as the primary simulation tool. Analytical techniques are only used to verify the simulation model by carrying out a set of well defined simulation experiments (extreme operating conditions are used to isolate the stochastic behavior inherent in the model).

Empirical measurements are not currently available and, therefore, this technique cannot be used to evaluate the performance of the LMR network. When empirical measurements eventually become available, i.e., after a digital packet-switched LMR WAN is deployed, this data can be used to validate the simulation model.

3.4.3 System Boundaries

Figure 3.4 delineates the boundaries of the system under consideration. In the remainder of this thesis, we refer to this system simply as the "LMR network."

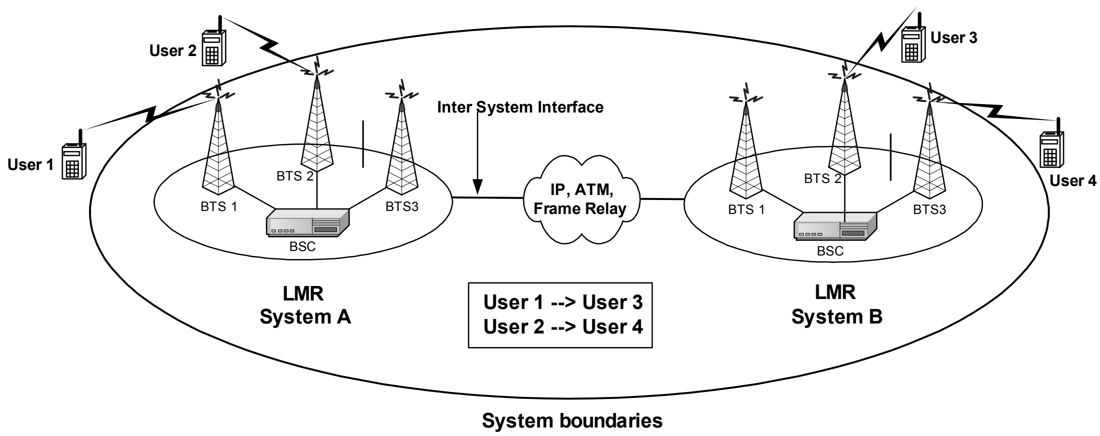

Figure 3.4: LMR network.

The LMR network is defined by two base stations with non-overlapping coverage zones connected together over a WAN. IP, ATM, and Frame Relay can all be used as the packet transport technology for the WAN.

The number of LMR users communicating across the ISI at any given time can be set between 2 and 256. Figure 3.4 shows one possible scenario in which four users are accessing the WAN. User 1 establishes communication with User 3 and User 2 establishes communication with User 4.

3.4.4 Performance Metrics

The metrics used to characterize the performance of the ISI are listed below. All metrics are defined within the context of the LMR network.

Throughput: This is defined as the amount of traffic traversing the ISI of the LMR network for a particular simulation time, measured in number of bits per second. It is a function of the number of users accessing the network at any given time.

Call setup time: This is defined as the time required to establish a call across the ISI using SIP. It measures the time between the SIP "INVITE" message and the SIP "ACK" message. Note that the call setup time does not include the time required to setup a call over the common air interface, i.e., between the base station and the LMR user.

ETE delay: This is defined as the time difference between when a voice packet is ready to be sent across the ISI and when the packet reaches it final destination.

Delay Variation: This is defined as the variation in the ETE delay. It is an important performance metric for the LMR network since a high delay variation has a negative impact on the quality of the voice communications between the LMR users.

3.5 Summary

Next generation LMR networks will make the transition from circuit-switched to packet-switched technology. Furthermore, SIP is gaining popularity as a signaling protocol in packet-switched networks and it is anticipated that it will become the standard for LMR networks. Under these postulations, we develop a simulation model to examine the loading effects on the ISI as a function of the number of users accessing the network. Details of the simulation methodology and results are provided in Chapters 4 and 5, respectively.

Chapter 4

Simulation Model and Methodology

This chapter describes the model used to investigate the performance of the packet-switched ISI. The methodology followed to characterize and simulate the network traffic in different scenarios is presented in detail and the simulation experiments are defined. To verify and analyze the results of the model, controlled simulation experiments are carried out.

4.1 Simulation Environment

OPNET Modeler version 8.1.A is used to build the simulation model. Developed in 1987, OPNET has become the leading network simulation environment for research and design of protocols, communication networks, and applications. While other simulation tools were considered, OPNET Modeler was found to be the most suitable for this study due to its flexibility, clear and simple modeling paradigm, hybrid simulation capabilities, and its comprehensive library of detailed standards-based protocols and application models. Customized packet exchanges are defined to characterize the traffic traversing the ISI of the LMR network. Furthermore, underlying packet transport technologies, including IP, ATM, and Frame Relay, are modeled to examine their influence on the performance of the ISI. OPNET Modeler provides the necessary tools required to configure customized network traffic and supports models required to simulate the underlying packet transport technologies of interest.

Network design in OPNET Modeler is based on a hierarchical approach comprised of three design levels, each one distinguished by its own editor. These are the "network," "node," and "process" levels. Collectively, these three levels define the simulation model [OPN01].

4.2 Network Model Design

At the network layer, we graphically represent the topology of the LMR network under consideration as shown in Figure 4.1. This topology is defined by node and link objects that simulate the behavior of each network element. The mobile terminals represent users trying to establish communication across the ISI. Figure 4.1 shows *user_A* and *user_B* engaged in a unit-to-unit call. At this point, it is important to realize that the model does not take into account the effects of the wireless link defined by the common air interface. Rather, it simulates the behavior of the communication across the ISI that is associated with the wired part of the LMR network.

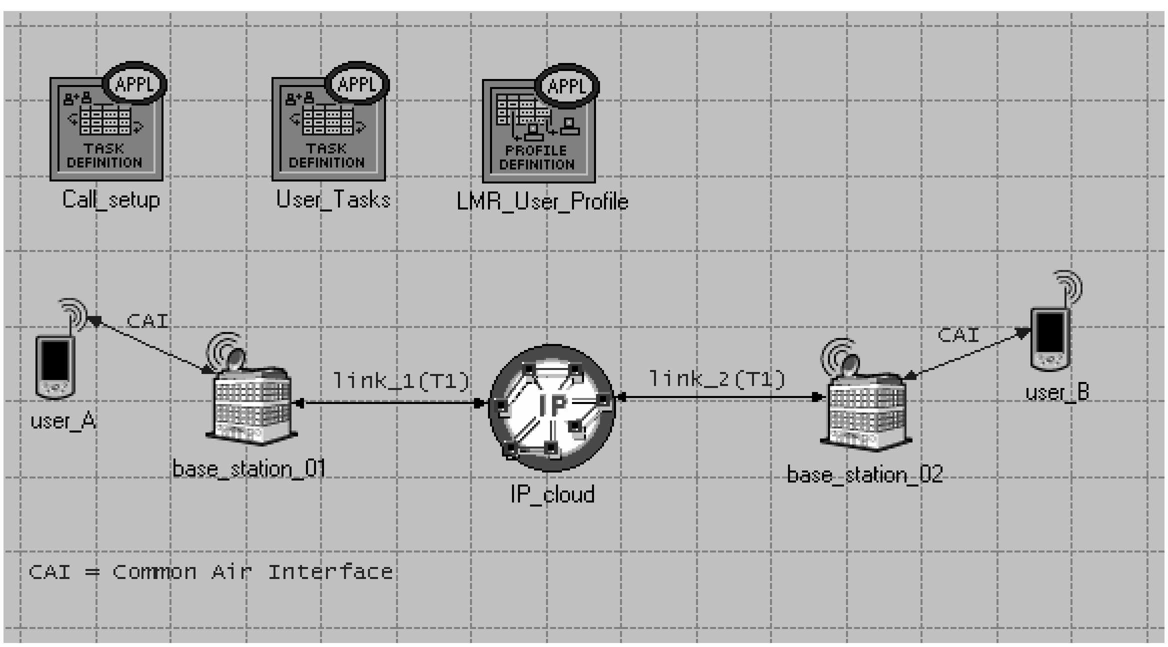

Figure 4.1: LMR network layer topology.

The traffic that traverses the ISI is generated by servers located at the base station closest to each active user (*base_station_01* and *base_station_02*). Nodes representing the mobile terminals are defined in order to: (i) distinguish the traffic patterns generated by each user and, (ii) provide a logical hierarchy to the simulation model.

The underlying packet transport technology used to connect the base stations in a wide area network (WAN) is simulated using a "cloud" object. An *IP_cloud* is shown in Figure 4.1. Using a similar approach, Frame Relay and ATM "cloud" objects are also considered as part of the network topology. We anticipate that most government organizations and private institutions interested in deploying large-scale LMR networks will purchase carrier services from a backbone service provider instead of deploying a private WAN. Consequently, no assumptions can be made about the infrastructure of the carrier network used, apart from information available in service level agreements (SLAs). In such a situation, a "cloud" object is ideal as it allows parts of the WAN infrastructure to be abstracted without losing any details.

The base station nodes used in the model, *base_station_01* and *base_station_02*, do not simulate the behavior of a complete LMR base station. For the purposes of this thesis, the base station node is configured to simulate the functionality of the proxy server generating SIP messages and the gateway used for interfacing base station equipment to the carrier network. Hereon, this gateway is referred to as the ISI gateway. Network equipment within the base station are interconnected via an Ethernet LAN.

Access links (*link_1* and *link_2*) are used to connect the base station nodes to the WAN. Each link provides T1 service at a bandwidth of 1.544 Mbps. The distance between the base stations is set to 500 km. In an actual LMR network, this distance can vary from about twenty kilometers all the way up to several thousand kilometers depending on the size and scope of the network.

The scenario illustrated in Figure 4.1 represents two users in a unit-to-unit call across the ISI. The simulation model is configured to include additional scenarios that represent 4, 8, 16, 32, 64, 128, and 256 users. The network level design uses the same topology in all scenarios even though some parameters might be configured to different values depending on the particular scenario being considered.

4.3 LMR User Profile Definition

At a high level, the network model specifies how LMR users establish communications across the ISI. As a result of this communication, traffic flows are generated. It is important to model these traffic flows as accurately as possible since they are used to characterize the loading effects on the LMR network. In this section, we examine the traffic load generated by LMR users engaged in a unit-to-unit call across the ISI. We then explain how this traffic is modeled in our network using OPNET.

4.3.1 Signaling Protocol Traffic Load

In Chapter 3, the argument was made that SIP is the preferred signaling protocol for call control and mobility management in a packet-switched LMR network. This is mainly due to the simplicity and efficiency of the protocol as well as its ability to meet the future requirements of LMR networks. Based on the protocol definition [HSS99], SIP messages required for setting-up and tearing-down LMR calls are created (basic message flows are presented in Chapter 2, Figure 2.9).

When an LMR user tries to establish communications with another user across the ISI he or she must first be granted privileges to communicate with the base station through the common air interface. This, of course, assumes that the user is in the coverage zone of the base station that is servicing him. Once the base station recognizes that a received call must be directed across the ISI, a SIP "INVITE" message is invoked by a SIP user agent client (UAC) located at the base station. This initiates the flow of signaling messages required to setup the inter-system unit-to-unit call.

To determine the size of the SIP messages, their content is examined in further detail. Even though we are not interested in the protocol specifics, SIP messages need to be represented as accurately as possible to model their loading effects on the LMR network. The assumption is made that LMR users are not roaming and that they are already registered with their home network.

4.3.1.1 INVITE

Recall that SIP supports two types of messages, requests, also known as methods, from client to server, and responses, from server to client. The "INVITE" request is used to initiate the media session between the users of the LMR network. It includes both a header, of the form `header:value`, and a message body that contains information on how voice calls between LMR users are coded. The discussion that follows in the remainder of this subsection refers to the "INVITE" request.

Definition of Message Headers

A SIP uniform resource locator (URL) is used to identify the LMR users in the network. The first line of the "INVITE" request, known as the start line, includes the SIP-URL of the LMR user being called and the version of the SIP protocol being used (currently 2.0): `sip:user_B@network_yy.gov SIP/2.0`. The `Via:` header records the route taken by the SIP request message. This information is then used to route the response message back to the originator. Whenever a SIP-enabled device originates or forwards a SIP message it inserts its address into the `Via:` header. The port number used may optionally be included. By default, port 5060 is used since this is a well-known port number for SIP. In addition, a branch tag is added by proxy servers processing the header. This is computed as a hash function of several mandatory SIP headers. Hence, the complete `Via:` header sent by the proxy server at the originating base station is of the form `Via: SIP/2.0/UDP base_station_01@network_xx.gov:5060;branch=1c9f1232416a1`.

Header

```
INVITE sip:user_B@network_yy.gov SIP/2.0
Via: SIP/2.0/UDP agent.network_xx.gov:5060
Via: SIP/2.0/UDP base_station_01@network_xx.gov:5060;branch=1c9f1232416a1
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>
Call-ID: 12345678@network_xx.gov
CSeq: 1 INVITE
Proxy-Authorization: pgp version=5.0;
 realm="ISI Gateway Password Required",
 nonce="1f4%647&ft$3$cde221q7986b",
 signature="9e4r5614rt$1acg673ers789"
Contact: John Doe <sip:user_A@network_xx.gov>
Content-Type: application/sdp
Content-Length: 190
```

Message body

```
v=0
o=User_A 1234 1 LMR IP4 base_station_01@network_xx.gov
s=UTU Voice
c=LMR IP4 network_xx.gov
t=0 0
m=audio 24000 RTP/AVP 100
a=rtpmap:100 X-IMBE_P25/8000
a=fmtp:100 bitrate=7200
```

The `From:` and `To:` headers identify the LMR users initiating and receiving the call requests respectively. An optional display name may be included followed by the SIP-URLs of the LMR users enclosed in <>. For completeness purposes, we arbitrarily identify *user_A* originating the call as John Doe (`From: John Doe` <`sip:user_A@network_xx.gov`>), and *user_B* receiving the call as Jane Doe (`Jane Doe` <`sip:user_B@network_yy.gov`>). The contents of the `From:` and `To:` headers remain unchanged in all messages exchanged during a particular session. To uniquely identify calls between users within the domain of the LMR network, the `Call-ID:` header is used. It includes a random identifier followed by the host name of the respective base station, i.e., `Call-ID: 12345678@network_xx.gov`.

The `Cseq:` header is used to determine out-of-sequence requests and to distinguish between new SIP requests and retransmissions. This is achieved by incrementing `Cseq:` for each new SIP request generated. The "CANCEL" and "ACK" requests are the only exceptions to this rule. These two requests use the sequence number of the "INVITE" request to which they refer [HSS99]. Assuming the "INVITE" request is the first SIP message sent, i.e., no “REGISTER” request precedes it, the `Cseq:` header is defined as `Cseq: 1 INVITE`.

The `Contact:` header includes the SIP-URL at which the LMR user invoking the SIP message can be reached directly. This feature makes it possible to offload redundant servers that do not have to be in the signaling path once the first “INVITE” request is sent [Cam02]. The final mandatory header in the “INVITE” request is `Content-Length` that indicates the number of bytes in the message body. The value calculated includes the carriage return (CR) and line feed (LF) characters at the end of each line. Using this method, the message body length of the “INVITE” request is 190 bytes, so the header is, `Content-Length: 190`. The main purpose of having the `Content-Length` header is to separate multiple messages sent in a TCP stream [Joh01].

Besides the mandatory headers, SIP defines thirty-two optional headers for the “INVITE” request. Of these, two are identified as being applicable to our implementation, the `Content-Type` and `Proxy-Authorization` headers. The `Content-Type` header is used to identify the media type in the message body. SDP [HJ98] is used to describe the multimedia sessions across ISI, hence, the `Content-type:` header is of the from `Content-Type: application/sdp`.

The `Proxy-Authorization` header is used to verify the credentials of the LMR user trying to initiate a call across the ISI. In LMR communications where confidential and mission-critical information is often exchanged between users, the security of the LMR network must be at the highest possible level. The use of `Proxy-Authorization` headers is one way of providing such security. SIP is based on HTTP, therefore, it borrows from HTTP authentication mechanisms. More advanced authentication mechanisms that implement modern public-key cryptography are also available, i.e., PGP [Aki00]. The format for PGP version 5.0 authentication is included in the “INVITE” request.

Definition of Message Body

SDP is used to describe how calls between LMR users are encoded. This information is included in the message body of the SIP "INVITE" request. SDP fields described in Table 4.1 are mandatory and must be defined in the order shown [HJ98].

Table 4.1: Mandatory SDP Fields

Field	**Name and field syntax**	**Definition used**
v=	Protocol version	`v=0`
o=	Origin (o= <username> <session-id> <version> <network type> <address type> <address>)	`o=User_A 1234 1 LMR IP4 base_station_11@network_xx.gov`
s=	Session name	`s=UTU Voice`
c=	Connection information (c= <network-type> <address-type> <connection-address>)	`c=LMR IP4 network_xx.gov`
t=	Time session (t= <start-time> <stop-time>)	`t=0 0`
m=	Media information (m= <media> <port> <transport> <format list>)	`m=audio 24000 RTP/AVP 100`

The following assumptions are made when defining the SDP fields.

1. The LMR network uses IPv4-type addresses.

2. Session management is not defined at the SDP level. Therefore, the time session field is irrelevant. "t=0 0" indicating a permanent session is arbitrarily used.

3. The "media" parameter is set to "audio" to identify the voice communication between LMR users.

4. RTP is used to transport voice packets across the LMR network. This is defined by setting the "transport" parameter in the media information to "RTP/AVP."

In addition to the mandatory SDP fields, two additional fields are defined that describe the speech coder and data rate of the LMR standard considered.

Table 4.2: Additional SDP Fields

Field	Name and field syntax	Definition used
a=rtpmap:	RTP/AVP list (a=rtpmap: <payload type> <encoding name>/<clock rate>)	`a=rtpmap:100 X-IMBE_P25/8000`
a=fmtp:	Format transport (a=fmtp: <format> <format specific parameters>)	`a=fmtp:100 bitrate=7200`

In Table 4.2 a possible syntax for an audio profile is suggested that can be used by Project 25. `X-IMBE_P25/8000` represents the <encoding name> parameter. Since this audio profile has not yet been registered as a standard format, the profile name is preceded by `X-` [HJ98]. With this syntax, additional audio profiles can be defined as `X-<vocoder standard>_<LMR technology>/<clock rate>`. A TETRA audio profile is then defined as `X-ACELP_TETRA/8000`. The bit rate used by the speech coders is specified as a format transport attribute. Project 25 and TETRA use 7.2 Kbps speech coders. Hence, the format transport attribute is defined as `a=fmtp:100 bitrate=7200`.

4.3.1.2 Trying

When the SIP proxy server located at the base station servicing the LMR user being called receives an “INVITE” request, it replies with a “100 Trying” response message. This message informs the originator that some action is being taken but it does not indicate that an LMR user being called has been found [HSS99]. Since the header definitions have already been defined, the contents of the “100 Trying” response message are provided without further explanation. Most of the header information required by the “100 Trying” response can be readily duplicated from the “INVITE” request. This approach simplifies the processing required to create SIP response messages [Joh01].

```
SIP/2.0 Trying
Via: SIP/2.0/UDP agent.network_xx.gov:5060
Via: SIP/2.0/UDP base_station_01@network_xx.gov:5060;branch=1c9f1232416a1
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>;tag=123456
Call-ID: 12345678@network_xx.gov
CSeq: 1 INVITE
Contact: Jane Doe <sip:user_B@network_yy.gov>
```

4.3.1.3 Ringing

The “180 Ringing” response is sent when the base station begins alerting the LMR user over the common air interface. Apart from the start line, the message contents are the same as the “100 Trying” response.

```
SIP/2.0 Ringing
Via: SIP/2.0/UDP agent.network_xx.gov:5060
Via: SIP/2.0/UDP base_station_01@network_xx.gov:5060;branch=1c9f1232416a1
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>;tag=123456
Call-ID: 12345678@network_xx.gov
CSeq: 1 INVITE
Contact: Jane Doe <sip:user_B@network_yy.gov>
```

4.3.1.4 OK

When and if the base station receiving the call request is ready to accept the call, a “200 OK” response is sent. Unlike the previous two responses that are informative and, therefore, optional, the “200 OK” response is essential and indicates that the “INVITE” request has succeeded. The message body of the “200 OK” response includes the media properties supported by the base station receiving the call. All the SIP-servers within the domain of the LMR network are expected to be configured in the same way, therefore, the message body of the "200 OK" response should be identical to the message body of the "INVITE" request as shown below.

```
SIP/2.0 OK
Via: SIP/2.0/UDP agent.network_xx.gov:5060
Via: SIP/2.0/UDP base_station_01@network_xx.gov:5060;branch=1c9f1232416a1
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>;tag=123456
Call-ID: 12345678@network_xx.gov
CSeq: 1 INVITE
Contact: Jane Doe <sip:user_B@network_yy.gov>
Content-Type: application/sdp
Content-Length: 190

v=0
o=User_B 1234 1   LMR IP4 base_station_02@network_yy.gov
s=UTU Voice
c=LMR IP4 network_yy.gov
t=0 0
m=audio 24000 RTP/AVP 100
a=rtpmap:100 X-IMBE_P25/8000
a=fmtp:100 bitrate=7200
```

4.3.1.5 ACK

The "ACK" request confirms that the originating base station has received a final response to the "INVITE" request. This message completes[1] the call setup phase after which voice packets can be exchanged between LMR users across the ISI.

```
ACK sip:user_B@network_yy.gov> SIP/2.0
Via: SIP/2.0/UDP agent.network_xx.gov:5060
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>;tag=123456
Call-ID: 12345678@network_xx.gov
CSeq: 1 ACK
Contact: John Doe <sip:user_A@network_xx.gov>
```

[1] It is possible to start sending voice packets after the "200 OK" response message is received. However, for consistency purposes the assumption is made that voice packets are exchanged only after the "ACK" request is sent out by the originating base station.

4.3.1.6 BYE

A “BYE” request is used to end calls between LMR users. Given the possibility that either user can terminate the call, a “BYE” request can be sent by both the user initiating a call and the user responding to a call.

```
BYE sip:user_B@network_yy.gov> SIP/2.0
Via: SIP/2.0/UDP agent.network_xx.gov:5060
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>;tag=123456
Call-ID: 12345678@network_xx.gov
CSeq: 2 BYE
Contact: John Doe <sip:user_A@network_xx.gov>
```

4.3.1.7 Release

To confirm the “BYE” request, a “200 OK” response is sent. This releases all the resources used to establish communications across the ISI.

```
SIP/2.0 OK
Via: SIP/2.0/UDP agent.network_xx.gov:5060
From: John Doe <sip:user_A@network_xx.gov>
To: Jane Doe <sip:user_B@network_yy.gov>;tag=123456
Call-ID: 12345678@network_xx.gov
CSeq: 2 BYE
Contact: Jane Doe <sip:user_B@network_yy.gov>
```

4.3.2 SIP Message Sizes

In the previous section, the contents of the SIP messages used to setup and tear down unit-to-unit calls across the ISI were defined. Next, the size of each of each of these messages is calculated. SIP is a text-based protocol. Therefore, every ASCII character in the message is counted as one byte. In addition, two bytes are added per line of text to account for the carriage return (CR) and line feed (LF) characters. The total size S, in bytes, of each message is calculated using Equation 4.1.

$$S = (\text{number of characters in SIP message}) + (\text{number of lines}) \times 2 \tag{4.1}$$

The results obtained are summarized in Table 4.3. In an actual implementation of the SIP protocol, the specific contents of the SIP messages may vary depending on the characteristics of the particular session invoked. Ultimately, the message sizes will change somewhat compared to the values shown in Table 4.3. Note that this study assumes that the SIP message size is fixed at the size calculated here.

Table 4.3: SIP Message Sizes

SIP Request/Response	**Message Size (bytes)**
INVITE	744
100 Trying	329
180 Ringing	330
200 OK	569
ACK	275
BYE	275
200 OK (Release)	247

4.3.3 Voice Communication Traffic Load

Even though studies have been made in the past to analyze and characterize traffic patterns for full-duplex telephone conversations between two users [Bra65][Bra68][Bra69], no information is available on traffic patterns for half-duplex two-way radio communications between LMR users. To overcome this shortcoming, information from LMR users and radio user guide manuals [USCS99] is collected. Based on this information, a voice call flow is created detailing a typical voice communication between two LMR users (Figure 4.2). Due to the stochastic nature of the talkspurt and silence periods in voice communications, statistical distributions are used in the model to capture, as best as possible, this randomness. Additional information on the distributions used is provided in Section 4.4. The time in seconds shown in Figure 4.2 represents mean values for each phase of the voice call.

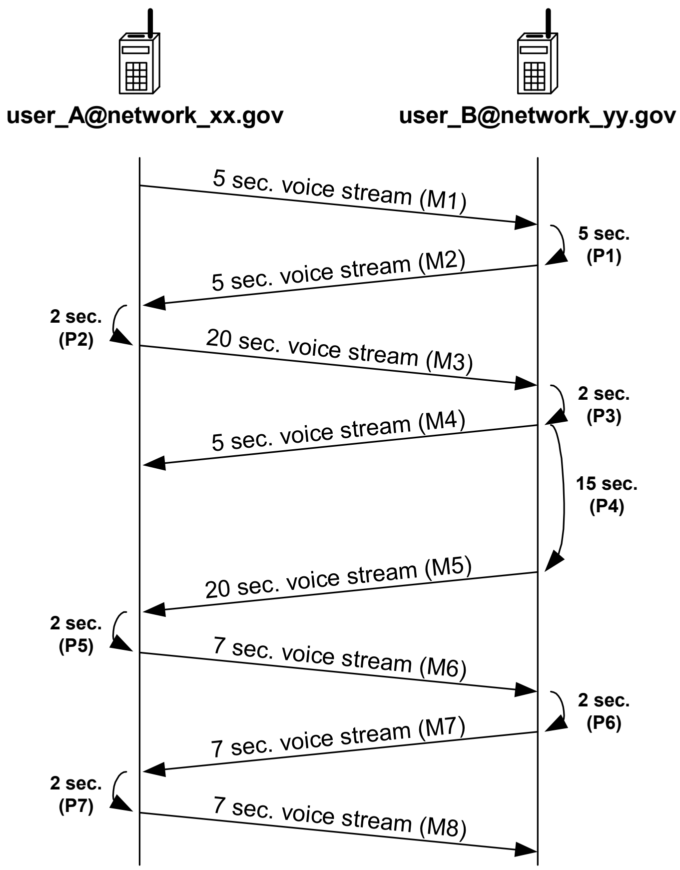

Figure 4.2: LMR unit-to-unit voice call flow.

Naturally, this voice call flow cannot capture all possible traffic patterns experienced in LMR voice communications between two users. Nevertheless, it provides a good starting point and some initial data that can be used in the simulation model. As shown in Figure 4.2, the typical response time for push-to-talk (PTT) activation once a voice stream is received by an LMR user is anticipated to be approximately two seconds. This includes the time required by the user to physically react to a call he or she receives as well as the PTT delay that can range between 50ms and 120ms [APC99a]. During the first voice exchange, an average response time of five seconds is used. This takes into account the extra time needed by an LMR user to respond when he or she is called the first time. Longer response times are possible if LMR users make requests that require additional processing before a reply can be given. In fact, such delays can be quite long, depending on the task. For our purposes, it is reasonable to treat a session with a long delay as separate sessions.

The two most prominent digital LMR standards, Project 25 and TETRA, use speech coders that have a gross bit rate of 7.2 Kbps. TETRA uses an algebraic code excited linear predictive (ACELP) speech coder. The ACELP process implemented by TETRA generates 137 bits per 30 ms of speech, equivalent to 4.6 Kbps [ETS98]. When channel coding is added, 216 bits are required per voice frame. This is equivalent to 7.2 Kbps. Project 25 uses an improved multi-band excitation (IMBE) speech coder that generates 88 bits per 20 ms of speech, equivalent to 4.4 Kbps [TSB98]. When forward error correction coding is added, 144 bits are required per voice frame. This is equivalent to 7.2 Kbps.

Since the bit rate of the speech coders is known, the traffic load imposed by the speech coders on the LMR network can be predicted. To accomplish this, a mechanism for transporting the voice frames across the ISI is first specified and then the overhead associated with this transport is analyzed. Overhead is added to each voice packet as it traverses the LMR network. The exact size of this overhead depends on the packet transport technology used.

4.3.3.1 RTP Considerations

RTP was developed to provide end-to-end delivery services for real-time data packets [SCF96]. It is an application protocol that conventionally uses UDP to transport multimedia data packets over IP networks. RTP is used in the model for transferring real-time voice packets across the LMR network.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|     one or more voice frames (IMBE or ACELP speech coders)    |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 4.3: Format of an RTP packet [IET96].

RTP can support multiple voice frames per packet. Consequently, one important decision that needs to be made is how many voice frames should each packet contain? Such a decision involves a tradeoff between packet overhead and packetization delay. By increasing the number of voice frames per RTP packet, the packet overhead and bandwidth utilization are improved. At the same time, however, more latency is introduced in the network by having to wait for several voice frames to accumulate before sending them across the network. This means that additional delay is introduced in the audio playback making it sound intermittent and distorted. In general, voice transmissions are resource intensive and as a result, generate a large number of small packets.

The format of an RTP packet is shown in Figure 4.3. Without going into too many details about the individual packet fields, notice that each RTP packet has a header of 12 bytes (assuming no contributing source identifiers) that carries a sequence number, timestamp, payload type, and synchronization source identifier. In addition to the RTP header, UDP (8 bytes) and IP (20 bytes) headers are added onto each voice packet by the time it is processed at the network layer. In total, 40 bytes of overhead are required for each voice packet. This does not include layer 2 overhead.

The Project 25 speech coder requires 18 bytes per voice frame (20 ms) and the TETRA speech coder requires 27 bytes per voice frame (30 ms). Figure 4.4 indicates the percent overhead (RTP/UDP/IP) that results from varying the number of voice frames per packet

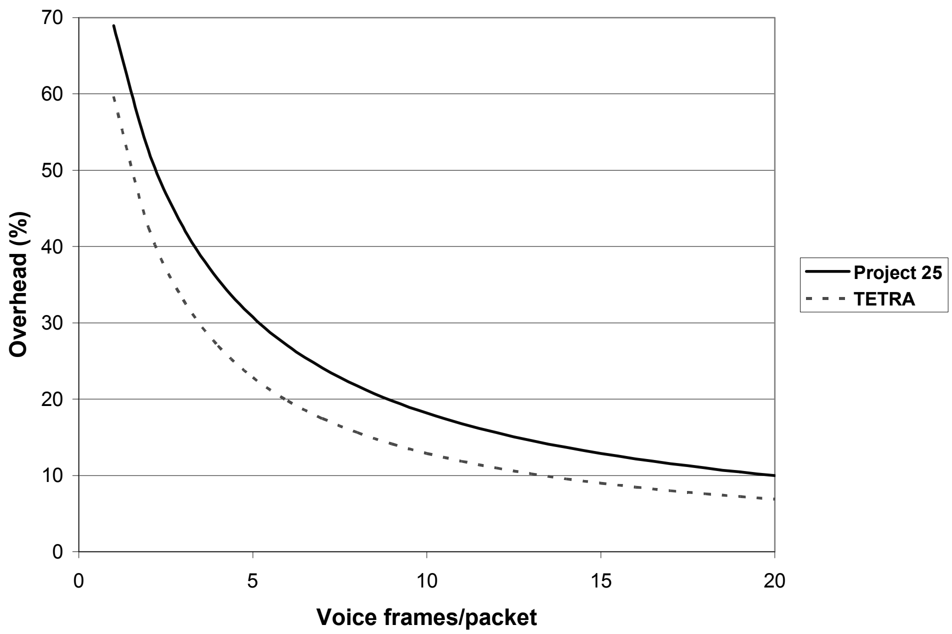

Figure 4.4: Packet overhead as a function of voice frames per RTP packet.

between 1 and 20. As more voice frames are added in a fixed packet header (40 bytes) the overhead decreases. It is critical to ensure that the maximum one-way latency of the voice packets be less than 150 ms so that the flow of the conversation can be maintained. The ITU considers network delay for voice applications in Recommendation G.114 [ITU00b]. According to the recommendation, delays of up to 150 ms are acceptable for most voice applications.

To obtain the optimal number of voice frames per packet, the packetization[2] delay must first be found as a function of the number of voice frames per packet. With this information available, the packetization delay curve can be constructed and superimposed onto Figure 4.4 after normalizing. The intersection point of the two curves jointly minimizes both packetization delay and packet overhead [ElG98]. Hence, this intersection point can be used as the

[2]Packetization delay is defined as the time taken to fill a packet payload with encoded speech. This includes the latency resulting from the voice packet preparation manager that is responsible for capturing the voice frames and synthesizing them into voice packets.

optimal value of voice frames per packet. This is ideal in situations where packet efficiency and low packetization delay are important. If bandwidth is not a limitation, more packet overhead can be tolerated to provide lower packetization delays.

Currently, no studies have been made to construct the packetization delay curve for the TETRA and Project 25 speech coders. Other speech coders with similar data rates, i.e., G.729 (8 Kbps) can be configured to use two voice frames per RTP packet [Sch01]. The assumption is made that two voice frames are used per RTP packet for TETRA, and three voice frames for Project 25 (Table 4.4). This combination is convenient since it generates the same RTP payload for both Project 25 and TETRA speech coders. It is possible to use more voice frames per packet, but first there needs to be some guarantee that the delay budget can support this.

Table 4.4: RTP Payload and Overhead using ACELP and IMBE Speech Coders

Speech Coder	**Voice frames/packet**	**RTP payload (bytes)**	**Overhead (%)**
Project 25 - IMBE	3	54 (3×18)	42.55
TETRA - ACELP	2	54 (2×27)	42.55

The effective bearer bandwidth, B, of a voice packet stream is calculated using Equation 4.2.

$$B(\text{Kbps}) = \left[\frac{\text{speech coder frame rate (s}^{-1})}{\text{voice frames per RTP packet}}\right] \times \frac{\text{packet size (bits)}}{1000} \tag{4.2}$$

The speech coder frame rate per second is $\frac{1000}{\text{voice frame duration (ms)}}$. The voice packet size is 66 bytes for RTP and 94 bytes for RTP/UDP/IP.

For Project 25,

$$
\begin{aligned}
B_{RTP}(Kbps) &= (\tfrac{50}{3} \times \tfrac{66\times 8}{1000}) \text{ Kbps} \\
&= 8.8 \text{ Kbps}
\end{aligned}
$$

$$
\begin{aligned}
B_{RTP/UDP/IP}(Kbps) &= (\tfrac{50}{3} \times \tfrac{94\times 8}{1000}) \text{ Kbps} \\
&= 12.53 \text{ Kbps}
\end{aligned}
$$

For TETRA,

$$
\begin{aligned}
B_{RTP}(Kbps) &= (\tfrac{33\frac{1}{3}}{2} \times \tfrac{66\times 8}{1000}) \text{ Kbps} \\
&= 8.8 \text{ Kbps}
\end{aligned}
$$

$$
\begin{aligned}
B_{RTP/UDP/IP}(Kbps) &= (\tfrac{33\frac{1}{3}}{2} \times \tfrac{94\times 8}{1000}) \text{ Kbps} \\
&= 12.53 \text{ Kbps}
\end{aligned}
$$

4.4 Representing Network Traffic

In the previous section, the traffic generated during a unit-to-unit call across the ISI was characterized. Traffic due to both signaling and voice communication is considered. Next, we explain how this traffic is represented in the simulation model.

OPNET can be used to represent traffic in a variety of ways. It can import traffic from specialized collection tools or the user can define the traffic patterns manually. Furthermore, network traffic can be defined explicitly or analytical models can be used to represent background traffic. In our model, customized traffic patterns are defined for maximum accuracy. Both, explicit and background traffic flows are represented. The traffic patterns generated by the LMR users are defined using OPNET's "Custom Application Model."

4.4.1 Custom Application Model

With the "Custom Application Model," it is possible to represent in detail the SIP and RTP packet flows that are exchanged between LMR users when they engage in a unit-to-unit call across the ISI. Figure 4.5 shows the process model used at the application layer. It is a modified version of the generic process model provided by OPNET that utilizes only the states required in the simulation study.

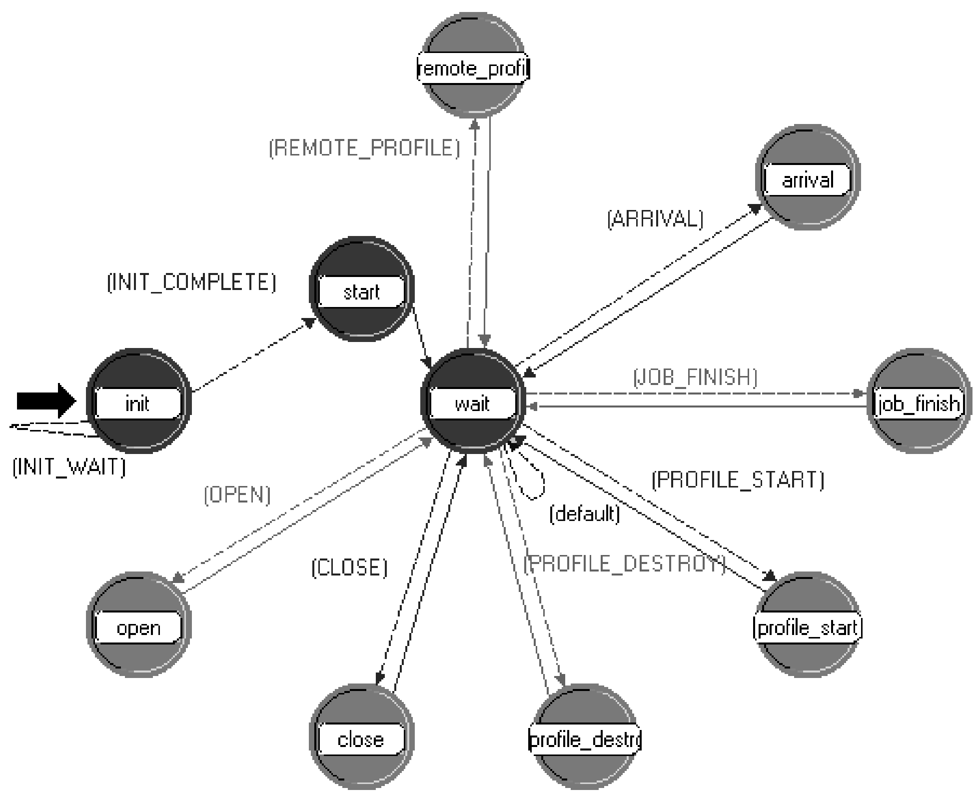

Figure 4.5: Application layer process model.

The application process model can service both clients and servers. By default, the process model remains in the *wait* state. Initialization procedures are performed in the *init* and *start* states. When a connection is established between base stations across the ISI, indicated by an interrupt received from the lower layers, the process enters the *open* state.

This state opens a session that enables application layer packets to be sent and received. If a packet is received, the process model transits to the *arrival* state. Using the information contained in the packet, the *arrival* state processes the packets received and records the statistics related to the packet's application. Responses associated with the received packet are also generated and sent. Once the received packet is completely processed, the application process model transitions back to the *wait* state.

Final handling of the packet when it completes service is carried out by the *job_finish* state. If a packet is destined for the upper layer protocols, it is forwarded to the appropriate node for final delivery to the LMR user. The *close* state is invoked when a session receives a close message confirmation from the *tpal* module. Its main purpose is to free all the memory associated with current session so that the system does run out of memory.

Details regarding the packet exchange are included in the user profile definition. When an LMR user initiates a call, the *profile_start* state is entered. This spawns the message sequence defined in the application profile. Once a particular profile is completed, it is destroyed by invoking the *profile_destroy* state.

Traffic patterns for each LMR user are modeled by defining user profiles. Each profile is made up of tasks, representing a basic unit of user activity, and phases, representing intervals of activity contained within a task. Collectively, the tasks and phases define the traffic flows generated by LMR users. Each profile represents the complete traffic exchange that takes place in a typical unit-to-unit call across the ISI. Figure 4.6 illustrates how tasks and phases can be associated with message flows. Part of a voice communication session is used as an example.

Unit-to-unit calls across the ISI are defined by three distinct tasks: (i) call setup using SIP signaling, (ii) voice communication using RTP, and (iii) call disconnect using SIP signaling. Each task is made up of distinct phases that represent the traffic flow at the bit-level. For each phase, the following seven attributes are configured in the model.

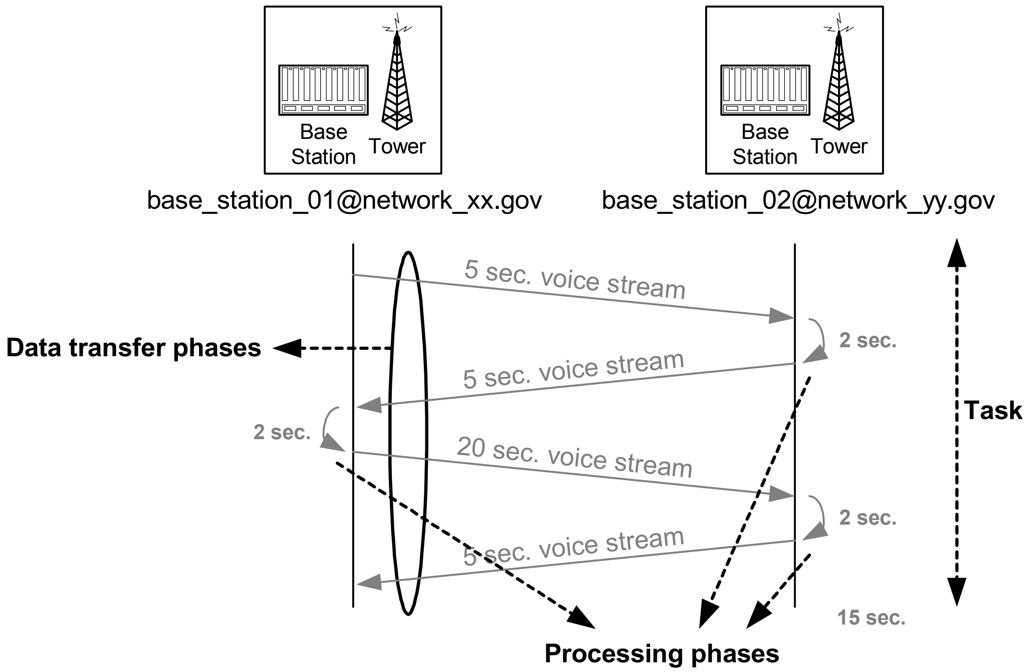

Figure 4.6: Tasks and phases representing message flow.

1. Phase start time
2. Source LMR network
3. Destination LMR network
4. Source-destination traffic characteristics (distribution type, number of packets sent, size of each packet, packet inter-request time, source initialization time)
5. Destination-source traffic characteristics
6. Request-response patterns
7. Transport connection (UDP is used in the model by default)

4.4.1.1 Call Setup Task

Each SIP message exchanged across the ISI is represented by a single data transfer phase. Since the size of each SIP message has already been calculated (Table 4.3), deterministic packet sizes are used to represent each of these messages. Figure 4.7 shows a snapshot of the task configuration table used to model the SIP messages. Each row in the table defines a single phase of the call setup task. To avoid any confusion, the phase names used correspond to the names of the SIP messages they represent.

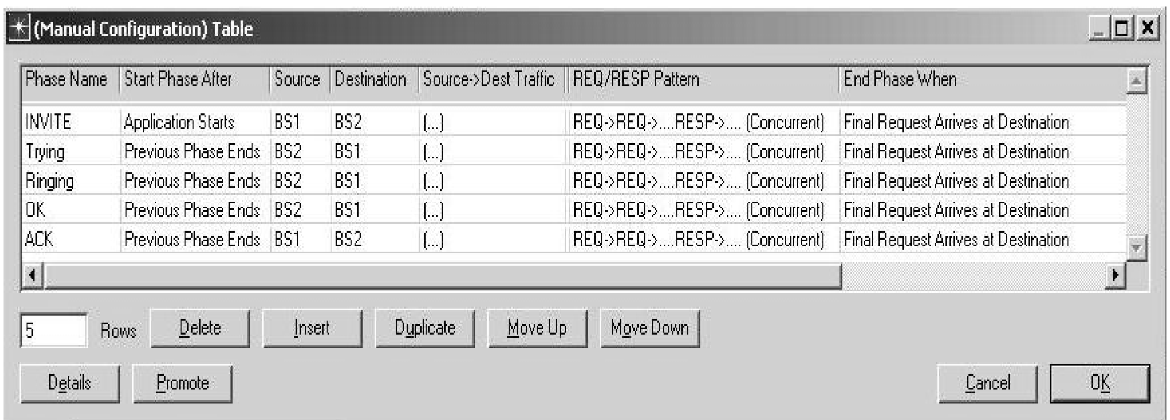

Figure 4.7: Call setup task.

In addition to the basic call setup illustrated above, two additional call setup tasks are defined using the same methodology to represent the following scenarios.

1. Call setup with registration

2. Call setup with registration and authentication

These scenarios are activated when a user is roaming in a visiting LMR network and needs to register back to his home network before beginning the call setup procedure. An authentication phase is included when a home LMR network requires its users to be authenticated prior to registering.

4.4.1.2 Voice Communication Task

Each data phase in the voice communication task represents a voice stream and each processing phase represents the response time for PTT activation. No voice packets are transferred during processing phases. Voice streams generated by LMR users are not deterministic by nature due to the random duration of talkspurt and silence periods. This randomness is modeled using distributions.

The primary objective is to use distributions that are able to capture the users' behavior as closely as possible. To achieve this, four distinct events that may occur during an LMR voice communication are identified. For each of these events, a suitable distribution is used [OPN02]. This allows a more refined representation of the voice communication traffic than if a single distribution was used to characterize all four events. Table 4.5 summarizes these events and the corresponding distributions used.

Table 4.5: Distributions Identifying Voice Communication Events

Event (refer to Figure 4.2)	**Distribution used**
User response time for PTT activation - initial reply to call (P1)	Exponential distribution - appropriate due to its memoryless[3]property.
User response time for PTT activation - no request processing required (P2, P3, P5-P7)	Uniform distribution - appropriate since the upper and lower bounds for the user response time are known
User response time for PTT activation - with some request processing required (P4)	Exponential distribution - appropriate for modeling user behavior that requires user think time
Voice stream (M1-M8)	Normal distribution - appropriate since the mean time for each voice stream is defined and both, positive and negative deviations from the mean are equally likely

For each stream of RTP voice packets, a bandwidth of 8.8 Kbps is required. This is the effective RTP bandwidth calculated in Section 4.3.3.1.

4.4.1.3 Call Disconnect Task

The call disconnect task is defined by two SIP messages, a “BYE” request and a “200 OK” response (Figure 4.8). These messages are represented using two distinct phases following the same methodology as that used in the call setup task (Section 4.4.1.1). Successful completion of this task also marks the end of the user profile for a particular LMR call.

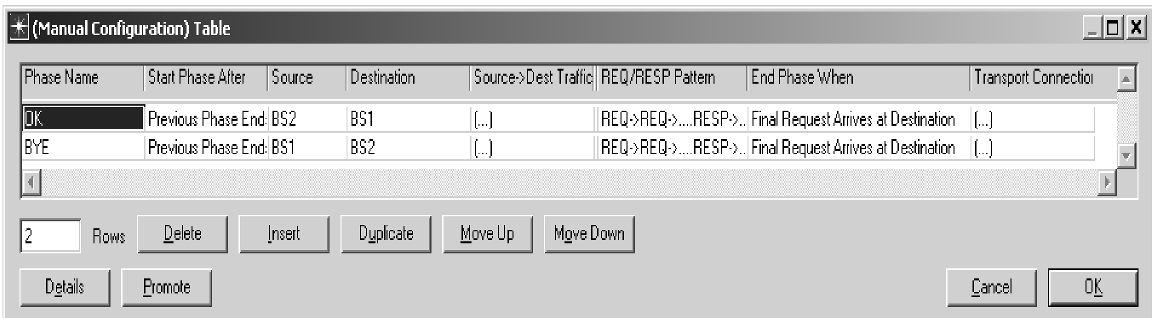

Figure 4.8: Call disconnect task.

4.5 Model of Underlying Packet Transport Technology

Information exchanged between users in an LMR network, be it voice or data, must arrive at its final destination according to some meaningful throughput, latency, and jitter guarantees. Several packet transport technologies are available that provide data services capable of supporting these guarantees over a WAN.

The effects of using Frame Relay, ATM, and IP packet transport technologies to interconnect the base stations in the LMR network are investigated. Current market trends indicate that there are advocates for all three technologies. Frame Relay and ATM have been available for several years now, each one presenting its own advantages and disadvantages. ATM is primarily used as a core network technology (Tier 1) and it is commonly used as the cell-switching backbone of Frame Relay. Frame Relay on the other hand, matured before

[3] The time taken by LMR users to respond to the initial call in the future is independent of the time taken by LMR users to respond to the initial call at the present time.

ATM. This allowed it to become widely available, well understood, and cost effective [Buc00]. Recently, service providers introduced SLAs for private IP service. Credits are offered for network availability, packet loss, and latency.

OPNET "cloud" objects use aggregation to model the WAN. Aggregation represents a collection of network entities as a single object in the modeling environment while still capturing the essence of the behavior of the collection [OPN02]. This allows us to abstract the details of the WAN yet model its behavior adequately enough for the objectives of this thesis. Published SLAs [Wil02][Spr02] are used to define the "cloud" objects. SLAs are treated as worst-case scenarios for the performance of the WAN.

For each "cloud" object, the packet (or cell) latency that specifies the one-way delay experienced by each packet (or cell), and the packet (or cell) discard ratio that specifies the ratio of packets dropped to packets submitted over the WAN are defined. Table 4.6 summarizes the values used in the model for IP, ATM and Frame Relay.

Table 4.6: "Cloud" Object Definitions Based on Service Level Agreements

Network Technology	**Packet Latency**	**Packet discard ratio**
IP	27.5 ms (average), 50 ms (maximum)	0.5%
Frame Relay	27.5 ms (Gold level - VFRrt)	0.01%
ATM	$\leq$ 20 ms (CBR)	$\leq$ 0.01% (CBR)

4.6 Background Utilization

Thus far in the definition of the model, only explicit simulation events defined by customized packets flows were considered. In this section, we introduce background utilization and discuss its relevance to our simulation model. OPNET uses queuing theory formulas to calculate the background traffic delay [OPN02].

Besides the voice packets exchanged between end users, it is possible to integrate data traffic from other sources in an LMR network. In fact, one of the advantages of migrating to a packet-switched LMR network is to enable the provisioning of integrated services over a single data network. Within the context of an LMR network, additional sources of data can be classified as follows:

- short data service (messaging),
- packet mode data (between data-enabled terminals),
- IT network traffic,
- dispatch consoles,
- authentication and registration of roaming users,
- network management protocol, and
- PTT arbitration (managing PTT activation across the ISI).

Additional data sources are incorporated into the model by considering different levels of background utilization. This method is used since the precise traffic load is not known. Scenarios that introduce 0, 40, and 80 percent background utilization are defined. This is accomplished by configuring every link and server in the network model to receive an additional amount of usage equal to the specified percentage relative to the maximum throughput. For example, 30% background utilization on a 1.544 Mbps link would imply an additional link usage of 463.2 Kbps. Similarly, 30% background utilization on a server would imply that the server performs at 70% of its normal processing speed. OPNET uses analytical techniques rather than discrete-event simulation to model background traffic [OPN02]. This introduces some limitations to the model. The burstiness of background traffic cannot be modeled and many protocol-specific effects cannot be captured since protocol mechanisms are usually difficult to represent analytically. Nevertheless, background utilization provides a means for investigating the performance of the LMR network when data from other sources is present.

4.7 Simulation Experiments

Simulation experiments are carried out to investigate the characteristics of the packet-switched ISI. In the previous sections, parameters of the simulation model were defined. In this section, issues related to the simulation experiments are considered.

4.7.1 Simulation Factors

Three factors are used to identify each simulation experiment: (i) the number of LMR users, (ii) the background utilization of the LMR network, and (iii) the underlying packet transport technology. Table 4.7 summarizes these simulation factors. The number of LMR users accessing the ISI at the same time can vary between 2 and 256. Background utilization is defined at three different levels, 0%, 40%, and 80%, and the packet transport technology used for the WAN can be ATM, IP, or Frame Relay.

Table 4.7: Simulation Experiment Factors and Options

Factor	**Options**
LMR users	2,4,8,16,32,64,128,256
Background utilization (%)	0,40,80
Underlying packet transport technology	ATM, IP, Frame Relay

To capture every possible combination at all levels of all three factors, a full factorial experiment is used. In total, 72 ($8 \times 3 \times 3$) different simulation experiments are performed.

4.7.2 Simulation Duration

The duration of the simulation needs to capture the time-span during which network traffic is present. This is sufficient since we are interested in investigating the traffic load on the network when different numbers of LMR users try to establish unit-to-unit calls across the ISI. Rather than simulating over a long period of time and have LMR users engage in multiple unit-to-unit calls, we simulate over a short period of time that is comparable to the length of a single unit-to-unit call. During this time, all LMR users present in the network are engaged in a unit-to-unit call and each LMR user is involved in a single call. This approach is preferred because: (i) it gives us a better understanding on how many LMR users can be simultaneously engaged in unit-to-unit calls across the ISI, and (ii) it still captures all the network traffic generated by the LMR users.

The maximum duration of a unit-to-unit call is about 180 seconds. This is an approximate value based on observations from preliminary simulation experiments. The exact maximum duration of a call cannot be determined since many of the parameters that define the call itself are random. For each simulation, a 10-second initialization period is defined. After initialization, an LMR user can potentially engage in a unit-to-unit call.

The exact time when the call is initiated is determined by a uniform distribution with a minimum value of 0, i.e., a call can begin immediately after initialization, up to a maximum value of 100. Figure 4.9 illustrates a scenario in which two users engage in two different unit-to-unit calls that are initiated 10 seconds and 110 seconds after the simulation begins. Since these two values represent the boundaries of the distribution that defines them, all users are expected to initiate their calls during this time interval.

Considering a worst-case scenario, a simulation time of 300 seconds, or 5 minutes is required to capture all of the traffic traversing the LMR network. This assumes a 10 second settle time after a call ends. Hence, the simulation experiments have a 5-minute duration.

4.7.3 Developing Confidence in Simulation Results

Each simulation experiment is repeated three times, each time using a different random seed. This makes it possible to establish confidence in the simulation results and eliminate any exceptional values (unless, of course, all three values obtained are exceptional). This results in needing to perform 216 (72×3) simulation runs.

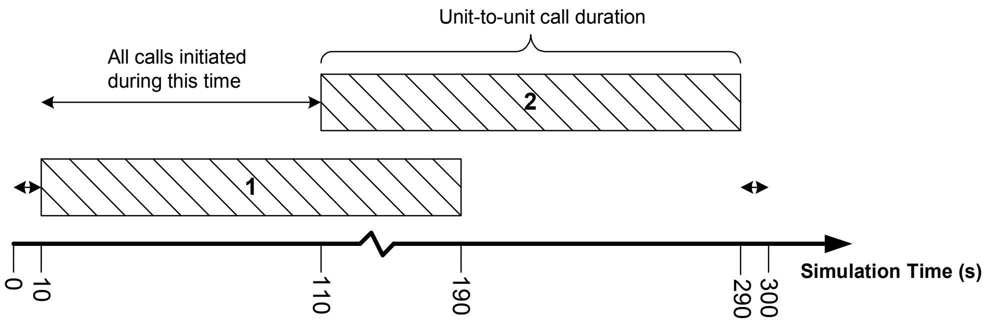

Figure 4.9: Unit-to-unit call simulation time.

4.8 Analysis and Verification of Model

To ensure that the LMR network model performs as designed, a number of controlled simulation experiments are carried out. A well defined scenario is developed to obtain results with a high degree of confidence. This is achieved by isolating the factors that introduce randomness and abstraction in the model. The "cloud" object representing the WAN is replaced by a well defined point-to-point (PPP) link. Furthermore, latency calculations are limited to deterministic SIP messages.

The following assumptions are made:

- the PPP link has a bandwidth of 64 Kbps,
- the distance between base stations is 500 km,
- the processing delay for each SIP message is 5 ms,
- the processing delay between SIP response messages is 10ms,
- UDP is used as the transport protocol, and
- two LMR users are accessing the network.

4.8.1 Verification of SIP Message Exchange

The call setup task presented in Section 4.4.1.1 is simulated using the LMR network model and the results are verified. Figure 4.10 shows a detailed call flow diagram identifying critical timing events.

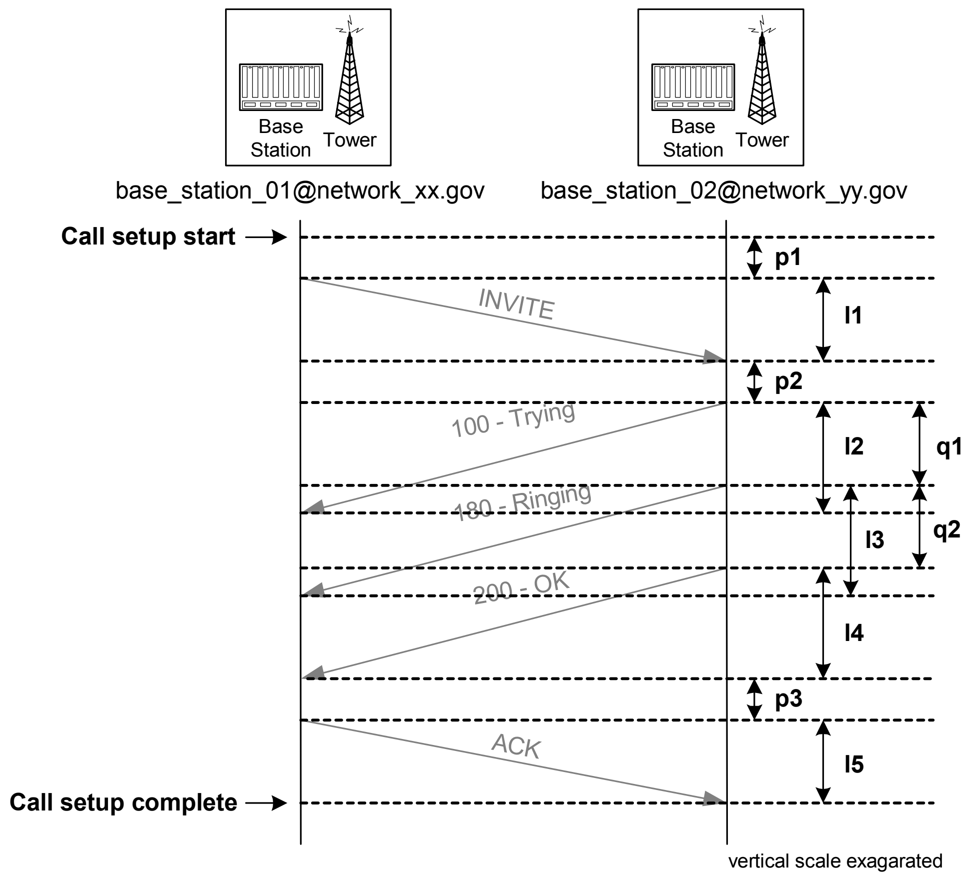

Figure 4.10: Timing of call setup message flow.

The processing delay for generating SIP messages is represented by p_i for i=1 to 3. For example, p_1 is the time from when the call setup request is received by the base station until the "INVITE" message is sent. The latency for each SIP message is represented by l_j for j=1 to 5. This includes transmit, propagation, queuing, and router processing delays. For example, l_1 corresponds to the latency for the "INVITE" message. This includes the time it takes between the first bit of the "INVITE" message being put onto the access link and the last bit of the "INVITE" message being put on the access link, the propagation delay for sending the "INVITE" message across the WAN, and the queuing and processing delays introduced at the packet switches. The processing delay between SIP responses is represented by q_k for k=1,2. For example, q_1 corresponds to the time between sending the "100-Trying" and "180-Ringing" SIP responses.

The total call setup time is given by Equation 4.3.

$$\text{Call setup time} = \sum_{i=1}^{3} p_i + \sum_{\substack{j=1 \\ j \neq 2,3}}^{5} l_j + \sum_{k=1}^{2} q_k \tag{4.3}$$

Using the assumptions made, $p_i = 5\ ms$ for all i and $q_k = 10\ ms$ for all k. The network latencies, l_i, are calculated using Equation 4.4. Note that l_2 and l_3 representing the latencies for the "100 - Trying" and "180 - Ringing" SIP responses are not used in the calculation of the call setup time. This is because 1xx SIP responses are informational messages used to indicate call progress and, therefore, do not influence the call setup timing.

$$Latency = \sum_{i=1}^{n} (\frac{s_i}{b} + \alpha) + \sum_{j=1}^{n} (\beta_j + \gamma_j) \tag{4.4}$$

Here, s_i is the packet size, b is the link bandwidth, α is the propagation delay, β_j is the queuing[4] delay, and γ_j is the processing delay.

Hence,

$$\begin{aligned} \alpha &= \tfrac{\text{distance}}{\text{speed of light}} \\ &= \tfrac{500{,}000\ m}{3 \times 10^8 m/s} \\ &= 1.67\ ms \end{aligned}$$

[4]Since the service time for a single SIP packet is less than the inter-arrival time, the SIP packets will not experience any queuing delay.

$$
\begin{aligned}
\frac{s_i}{b} &= \frac{\text{size of SIP messages (bits)}}{64{,}000} \\
&= \frac{(798+623+329)\times 8 \text{ bits}}{64{,}000 \text{ bps}} \\
&= 218.75\ ms
\end{aligned}
$$

$$
\begin{aligned}
\gamma_j &= p1 + p2 + p3 + q1 + q2 \\
&= 35\ ms
\end{aligned}
$$

$$
\begin{aligned}
\Rightarrow Latency &= 218.75\ ms + 3(1.67\ ms) + 35\ ms \\
&= 258.75\ ms
\end{aligned}
$$

The simulation model yields a call setup time of 249.72 ms. This value is obtained from the "task response time" statistic that calculates the total time required for the call setup task to complete. Compared to the analytical result, the simulation model is able to predict within acceptable error margins the call setup time. To further verify the correct simulation of the call setup task, the network traffic is examined. Figure 4.11 shows the packet exchange for the call setup task as a function of time. The size of each SIP message is given in Table 4.3. During simulation, an additional 35 bytes are added to each SIP message to account for protocol overhead (8 bytes UDP, 20 bytes IP, and 7 bytes PPP). This overhead is also considered in the analytical model.

Comparing Figure 4.11 to Figure 4.10 verifies that the simulation model represents the correct sequence and timing of the SIP messages for the call setup task.

4.8.2 Verification of Voice Communication

The modeling of the voice communication between the LMR users is based on stochastic processes and different distributions. Therefore, it is forbiddingly difficult to obtain analytical results that can verify the simulated results. To verify the correct representation of the voice streams in the simulation model, the throughput of the voice communication task (Section 4.4.1.2) is considered. The results obtained from the simulation model are illustrated in Figure 4.12. The simulated results indicate that each voice packet sent is 808 bits long

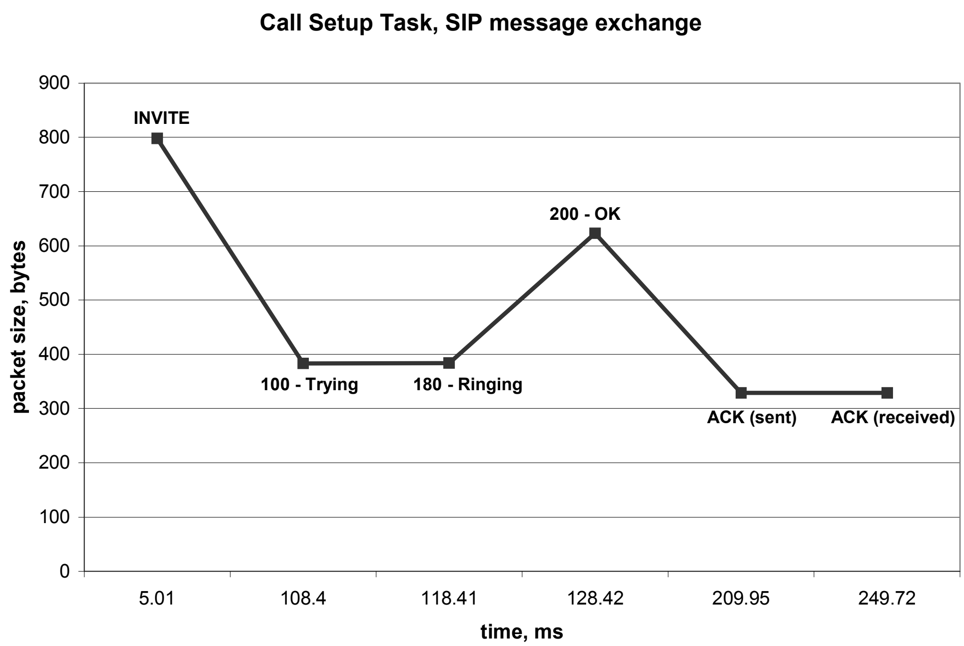

Figure 4.11: Packet exchange for call setup as a function of time.

(including overhead). Furthermore, 16.67 packets are sent every second. This detail cannot be observed in Figure 4.12 due to the restrictive scale on the time axis. Using the simulation results, the data rate imposed on the network by the voice communication task is found to be 13.5 Kbps (808 bits $\times$ 16.67 packets/s).

To verify the data rate of the voice packets, the following calculation is performed.

$$\text{Data rate } = \text{packet creation rate } \times \text{packet size} \tag{4.5}$$

Using the results from Section 4.3.3.1,

$$\begin{aligned} \text{packet size} &= \text{66 bytes RTP packet + 35 bytes overhead (UDP/IP/PPP)} \\ &= \text{101 bytes (808 bits)} \end{aligned}$$

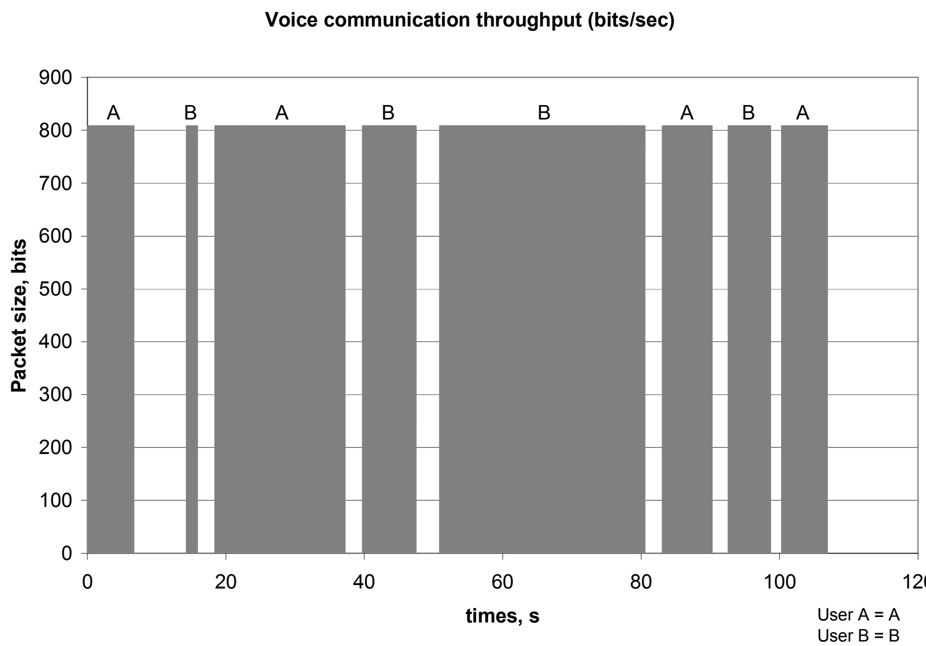

Figure 4.12: Voice communication throughput (bits/s).

$$\text{packet creation rate} = 16.67 \text{ packets/s}$$

$$\begin{aligned} \Rightarrow \qquad \text{Data rate} &= 808 \text{ bits/s} \times 16.67 \text{ packets/s} \\ &= 13.5 \text{ Kbps} \end{aligned}$$

As a final step in verifying the voice communication task, the timing of the voice call flow is considered. The results are summarized in Table 4.7. Data from a single simulation experiment is compared to the model definition of the voice communication task (Section 4.3.3). This information is then used to verify that each phase in the task is executed within the bounds of the distribution that defines it. Table 4.7 indicates that all simulated results fall within these bounds.

Latency calculations for more complex scenarios, i.e., multiple LMR users communicating across the ISI, are not easy to compute even if the precise traffic patterns are known *a priori*. This is because dynamics of queues are a function of traffic patterns in addition to traffic

Table 4.8: Timing of Voice Call Flow

	Duration (seconds)	
Event	**Model Definition**	**Simulation Results**
Voice stream 1	Normal distribution (μ=5s, σ=2)	6.689s
User response time - initial reply to call	Exponential distribution (μ=5s)	7.625s
Voice stream 2	Normal distribution (μ=5s, σ=2)	1.652s
User response time - no processing	Uniform Distribution (min=1.3s, max=2.7s)	2.571s
Voice stream 3	Normal distribution (μ=20s, σ=5)	18.753s
User response time - no processing	Uniform Distribution (min=1.3s, max=2.7s)	2.518s
Voice stream 4	Normal distribution (μ=5s, σ=2)	7.751s
User response time - with processing	Exponential distribution (μ=15s)	3.408s
Voice stream 5	Normal distribution (μ=20s, s.d=5)	29.629s
User response time - no processing	Uniform Distribution (min=1.3s, max=2.7s)	1.872s
Voice stream 6	Normal distribution (μ=7s, σ=2)	7.126s
User response time - no processing	Uniform Distribution (min=1.3s, max=2.7s)	1.423s
Voice stream 7	Normal distribution (μ=7s, σ=2)	6.063s
User response time - no processing	Uniform Distribution (min=1.3s, max=2.7s)	1.611s
Voice stream 8	Normal distribution (μ=7s, σ=2)	6.626s

volumes. Moreover, traffic patterns are shaped by the network in complex time-dependent ways that are often impossible to predict.

4.9 Simulation Validation

The standards for a packet-switched ISI for LMR systems are still in the development process. Consequently, the results of the simulation model cannot be confirmed with data collected from a real system. Confirmation of the model's "equivalence" with the characteristics of a real system is essentially obtained using common sense and intuition, a simplistic yet powerful validation tool. One would intuitively expect that the call setup time increases once the network starts becoming congested and it is common sense to realize that the throughput increases as the number of LMR users in the network increases.

4.10 Summary

This chapter describes the design of the model used to investigate the performance of the packet-switched ISI. The methodology followed to develop the model is explained and justified.

OPNET Modeler is used as the primary simulation tool. The topology of the LMR network is defined by node and link objects that simulate the behavior of each network element. Traffic flows generated by LMR users engaged in unit-to-unit calls are characterized. SIP messages are defined for setting up and tearing down calls and RTP packets are used for transferring real-time voice across the ISI. Traffic patterns for each LMR user are modeled in OPNET by defining profiles consisting of tasks and phases. Three packet transport technologies, ATM, IP, and Frame Relay, provide WAN connectivity for the LMR network. Each packet transport technology is modeled using a "cloud" object defined according to published SLAs.

In total 216, different simulation experiments are identified that capture all possible combinations of the factors considered. Controlled simulation experiments are carried out to verify the model prior to collecting final results. No formal validation procedures are analyzed as empirical data from real LMR systems are not yet available. In the next chapter, the results of the simulation experiments are presented.

Chapter 5

Simulation Results

This chapter presents the results of the simulation experiments. In total, 72 different simulation experiments are considered as described in Section 4.7. The number of LMR users accessing the ISI can vary between 2 and 256. To model the presence of additional data sources that may share the available bandwidth with the LMR network, background utilization is considered at three different levels, 0%, 40%, and 80%. In addition, three different packet transport technologies are considered for the WAN, IP, Frame Relay, and ATM. Each simulation experiment is repeated three times, with each repetition using a different random seed to establish confidence in the results. The values used to analyze the results represent the average outcome of the three simulation experiments. All the data collected is provided in Appendix B.

For the purpose of presenting results, different simulation scenarios are referred to using the convention <LMR users>_<packet transport technology>_<background utilization>. For example, 2_FR_0 refers to the scenario in which two LMR users are communicating in a unit-to-unit call across the ISI using a Frame Relay WAN with 0% background utilization on the access link.

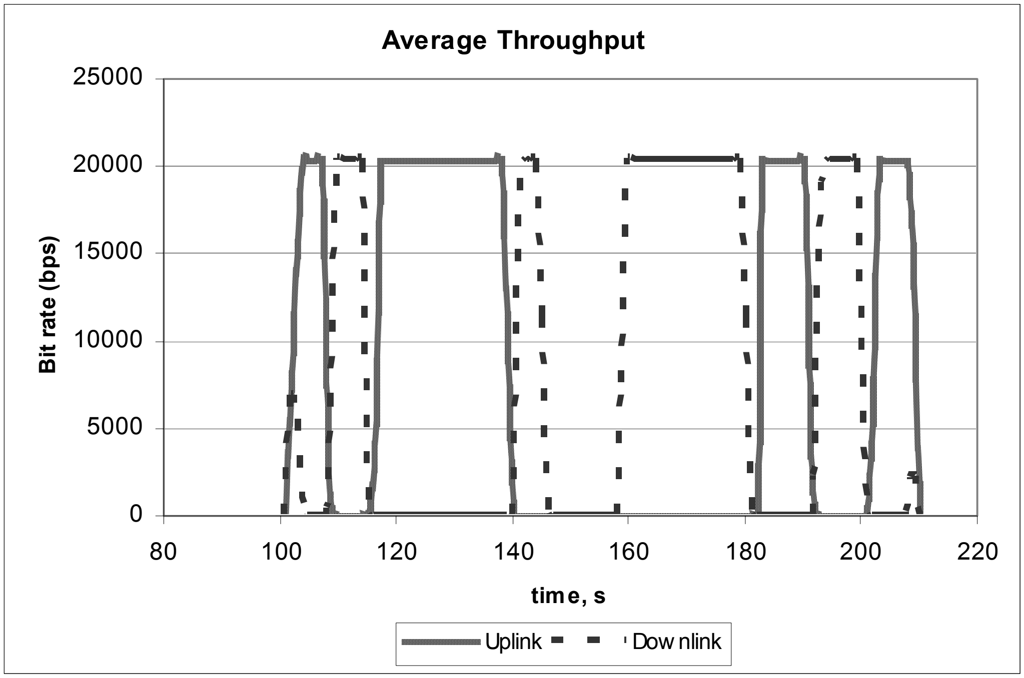

Figure 5.1: Throughput on access link when using an ATM WAN, 2_ATM_0.

5.1 Throughput and Link Utilization

5.1.1 Throughput

Throughput is used to characterize the rate at which traffic traverses the access link for the three different packet transport technologies considered. The throughput metric used includes the voice frames exchanged between LMR users, the packet overhead, and the signaling information required to setup and tear down the voice calls. To facilitate the comparison of the results and obtain a better understanding of the loading effects on the network, we first look at a straightforward scenario with two LMR users and 0% background utilization. The volume of data exchanged between the two users is identical, the only difference being the packet transport technology used for the WAN. Figures 5.1 through 5.3 show the throughput on the access link when using ATM, Frame Relay and IP WANs.

Using an ATM WAN results in the highest data rate on the access link at 20.3 Kbps

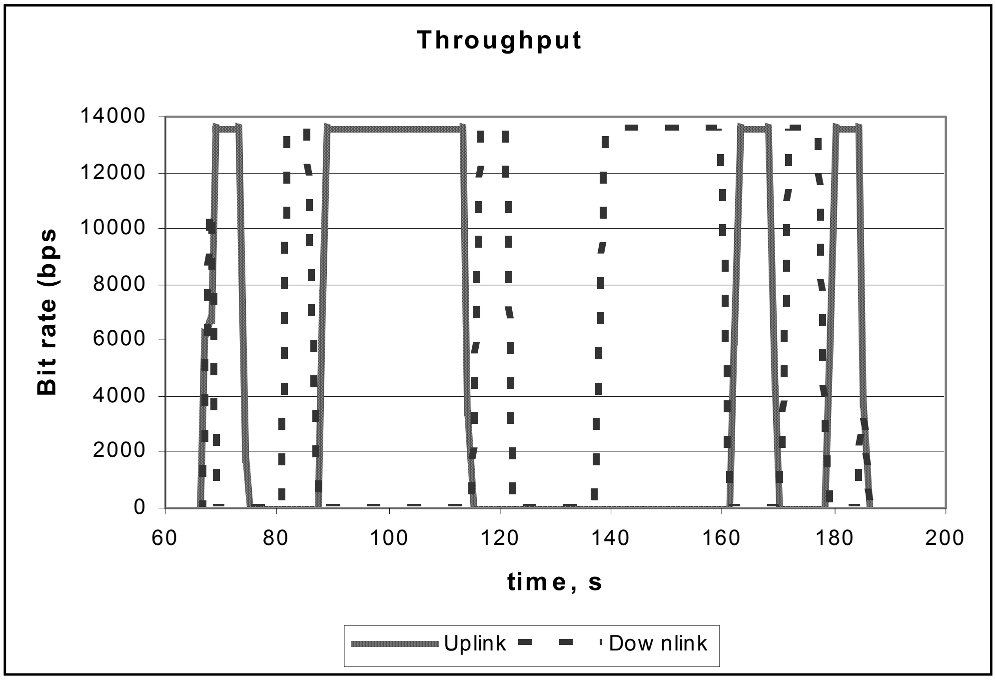

Figure 5.2: Throughput on access link when using a Frame Relay WAN, 2_FR_0.

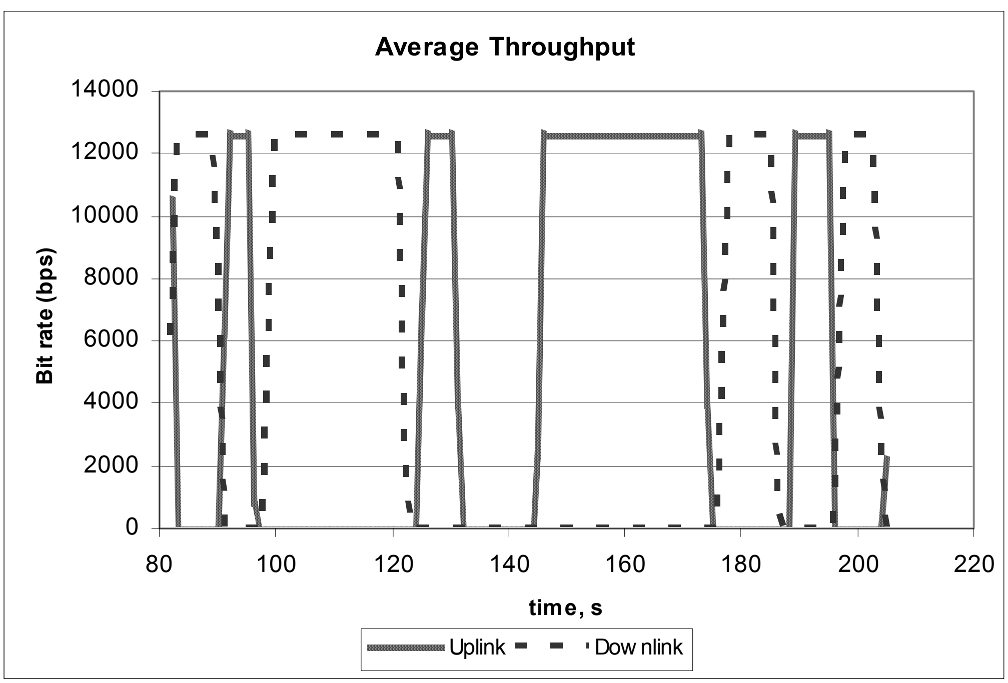

Figure 5.3: Throughput on access link when using an IP WAN, 2_IP_0.

(Figure 5.1). This is explained by the high percentage of transmission overhead associated with ATM. ATM Adaptation Layer Type 5 (AAL5) is used to segment IP protocol data units (PDUs) into ATM cells at the network edge. AAL5 is selected as it is simpler compared to other AALs and standardized inter-working functions (IWFs) for IP over ATM use AAL5 to provide multi-vendor interoperability [Wri01]. The size of each voice packet received by AAL5 from the network layer is 94 bytes (Chapter 4, Section 4.3). This requires three cells (159 bytes) for transmission across the ATM WAN resulting in a 59% bandwidth efficiency. This, of course, does not consider the RTP/UDP/IP overhead.

Packet transport over a Frame Relay WAN results in the second highest data rate on the access link at 13.6 Kbps (Figure 5.2). This data rate is significantly lower when compared to ATM. The maximum frame size that the Frame Relay network can process without requiring fragmentation is set to 504 bytes in order to protect the voice packets from experiencing high delay and jitter. As a result, most packets received from the network layer do not require fragmentation with the exception of the "INIVITE" and "200 OK" SIP messages. The Frame Relay overhead is 8 bytes (6 bytes fixed, 2 bytes variable) for each IP PDU received resulting in a 92% bandwidth efficiency.

Packet transport over an IP WAN does not introduce any additional overhead since the PDUs are already in native IP form. As shown in Figure 5.3, IP can support the unit-to-unit voice call between two LMR users at the lowest bit rate.

5.1.2 Average Link Utilization

The average link utilization is defined as the sum of the average data rate on the access link, in both[1] the uplink and downlink directions, as a percentage of the total bandwidth available. Data transferred between the ISI gateway and the WAN is defined as the uplink direction and

[1]Average link utilization in the uplink and downlink directions are not considered separately since the traffic load between LMR users in a unit-to-unit call is defined to be symmetrical on average (Section 4.3). However, this introduces the possibility of reaching 100% utilization in either the uplink or downlink direction of the access link and not being able to observe this on the average link utilization curve. To account for this limitation, the maximum link utilization is considered as a separate metric in the next subsection.

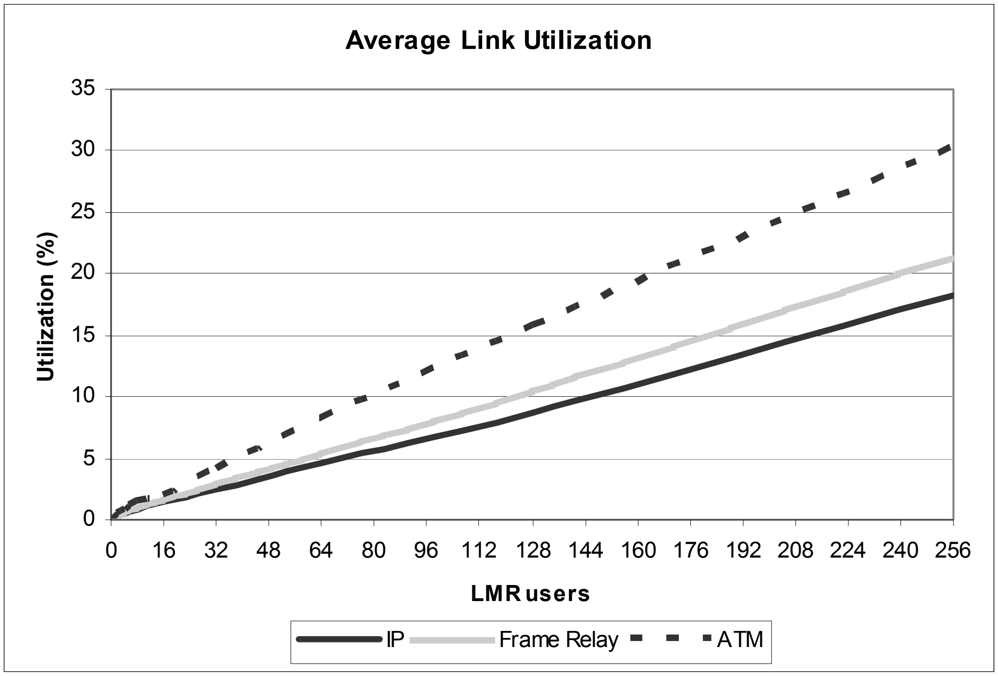

Figure 5.4: Average link utilization with 0% background utilization.

data transferred between the WAN and the ISI gateway is defined as the downlink direction. Since the access links provide T1 service at 1.544 Mbps, the total bandwidth available in both the uplink and downlink directions is 3.088 Mbps.

Figure 5.4 illustrates the average link utilization as a function of the number of LMR users active in the network when background utilization on the access link is 0%. Extending the observations made in Section 5.1.1 for two LMR users, we show that an ATM WAN requires the highest bandwidth to provide voice communications between LMR users. This is true for all numbers of LMR users considered. The average link utilization rises linearly from 0.5% for 2 users to 30.2% for 256 users. When using a Frame Relay WAN the average link utilization is consistently lower than ATM. The average link utilization rises linearly from 0.38% for 2 users to 21.3% for 256 users. IP imposes the lowest average link utilization ranging from 0.35% for 2 users to 18.16% for 256 users.

Notice that the slope of the average link utilization curve is steeper for ATM than it is

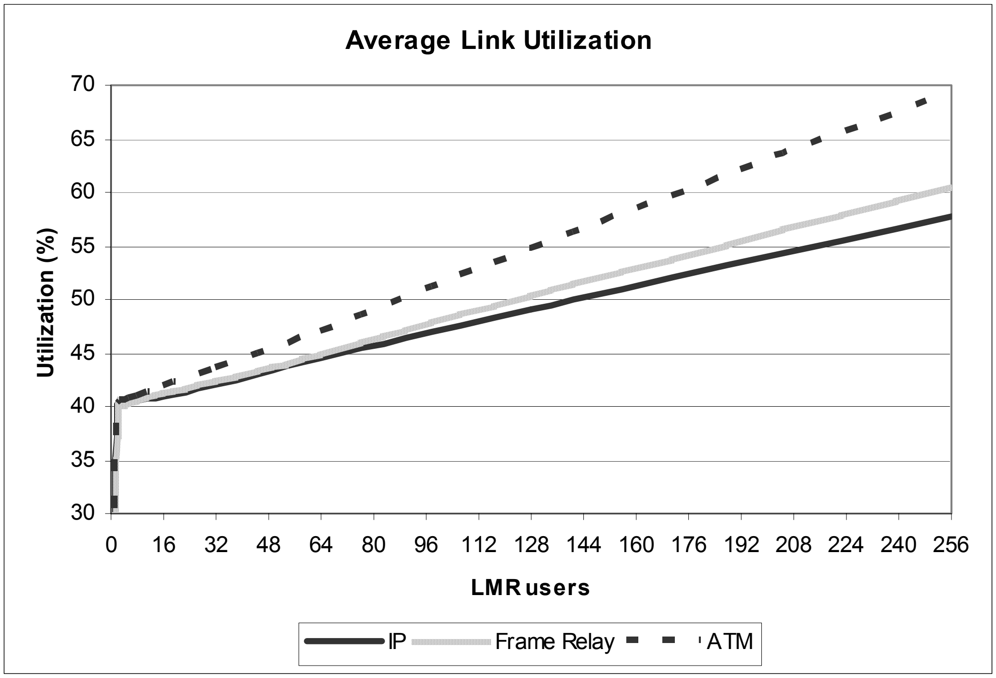

Figure 5.5: Average link utilization with 40% background utilization.

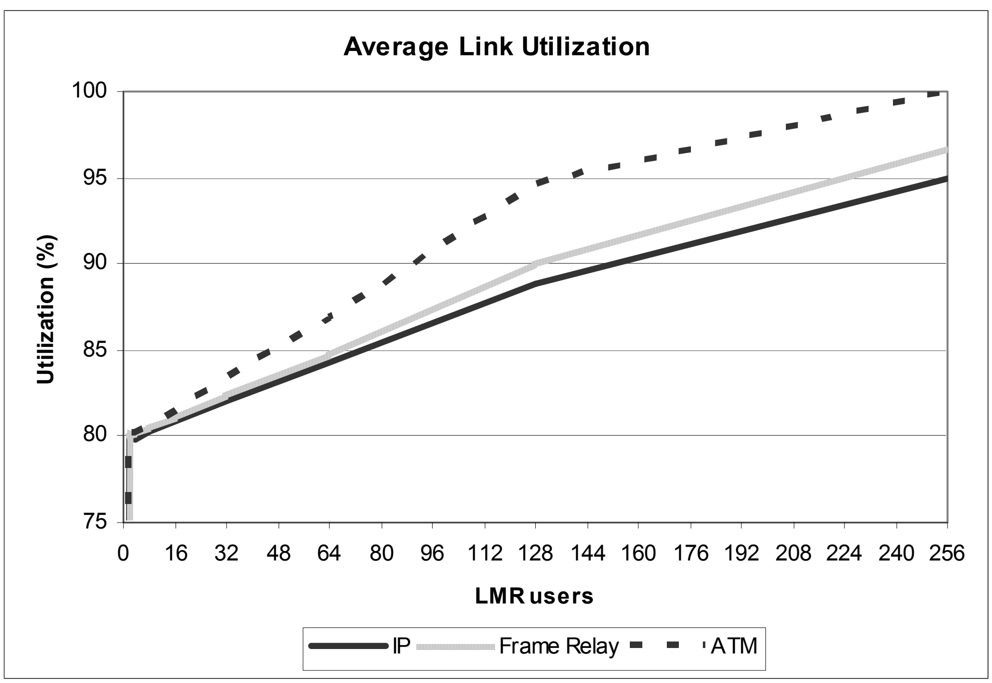

Figure 5.6: Average link utilization with 80% background utilization.

for Frame Relay and IP. The steepness of the slope, besides showing the average bandwidth required to support different numbers of LMR users, is also an indication of the bandwidth efficiency of the packet transport technology used. A steeper slope implies lower bandwidth efficiency. This is true as long as the maximum utilization is less than 100% in both uplink and downlink directions.

Figures 5.5 and 5.6 show the average link utilization when the background utilization on the access link is 40% and 80%, respectively. In the case of 80% background utilization (Figure 5.6), the average link utilization plots are not linear as they are in the case of 0% and 40% background utilization. This occurs because the link utilization in either the uplink or downlink direction reaches 100%, thus, causing network congestion. As a result, voice traffic between LMR users is delayed. Voice packets that would otherwise have access to the network are now denied access and buffered in queues. If the congestion is prolonged packets are eventually dropped. The change in the slope is most notable when the number of LMR users increases beyond 128[2].

The average link utilization will only reach 100% when both the uplink and downlink utilization is 100%. This occurs in the 256_ATM_80 scenario as illustrated in Figure 5.6.

5.1.3 Maximum Link Utilization

The maximum link utilization is defined as the highest utilization experienced by the access link in either the uplink or downlink direction during the length of the simulation experiment. Maximum utilization provides a strong indication of when a possible congestion may occur in the network. Due to the sensitivity of voice packets to delay and delay variation, maximum link utilization is, in many ways, a more useful indicator to characterize the network performance than the average link utilization. Moreover, it can be closely linked to the ETE delay experienced by the voice packets. Ideally, the instantaneous maximum link utilization

[2]Statistics are collected for specific numbers of LMR users in the range from 2 to 256 (Table 4.7). All performance decisions made are based on these particular numbers of LMR users.

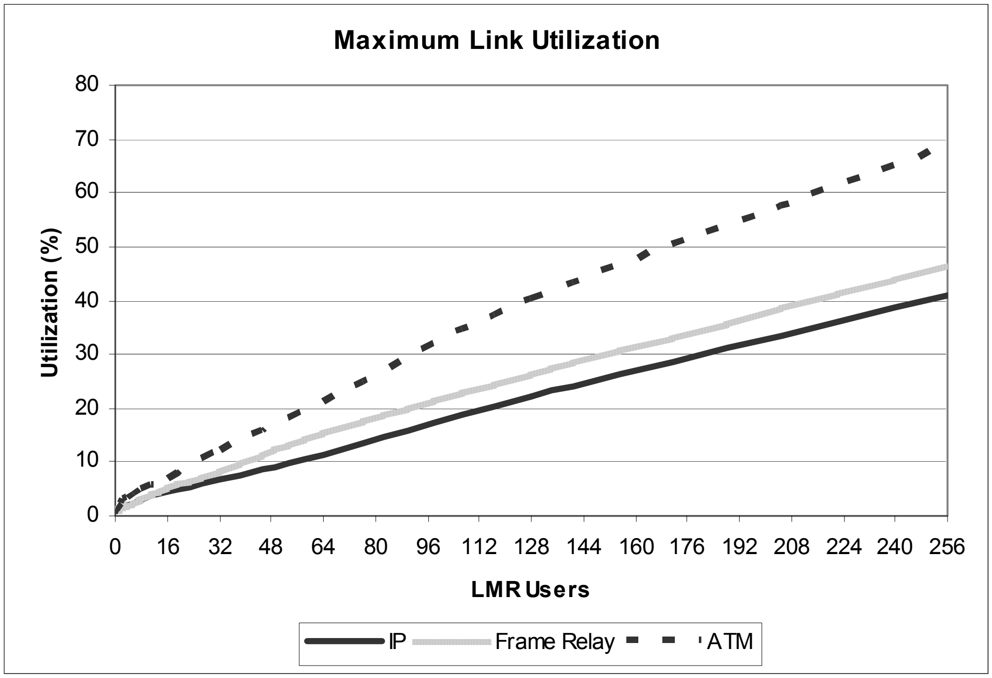

Figure 5.7: Maximum link utilization with 0% background utilization.

could be captured in the experiments. However, this requires extensive computing resources given the number of simulations experiments performed. Instead, a "bucket" mode is used to collect the data. The "bucket" size is defined to be 0.1 seconds. All samples collected during this interval are processed, averaged, and a single value is returned representing the sample average.

Figures 5.7 through 5.9 represent the maximum link utilization with 0%, 40%, and 80% background utilization, respectively. When using an IP or Frame Relay WAN, the access link reaches 100% utilization when the number of LMR users is 128 or more and background utilization is 80%, i.e., 128_IP_80, 256_IP_80 scenarios for IP, and 128_FR_80, 256_FR_80 scenarios for Frame Relay. This is illustrated in Figure 5.9.

When using an ATM WAN, the access link reaches 100% utilization in the following four scenarios: 256_ATM_40, 64_ATM_80, 128_ATM_80, and 256_ATM_80. In the 256_ATM_40 and 64_ATM_80 scenarios, the link reaches 100% utilization for a brief pe-

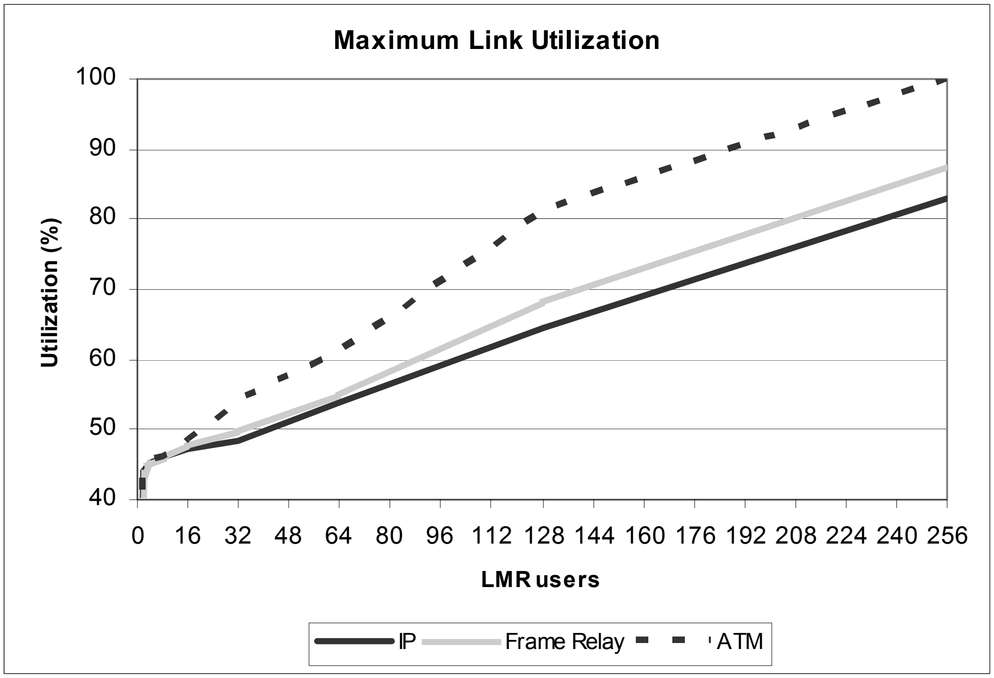

Figure 5.8: Maximum link utilization with 40% background utilization.

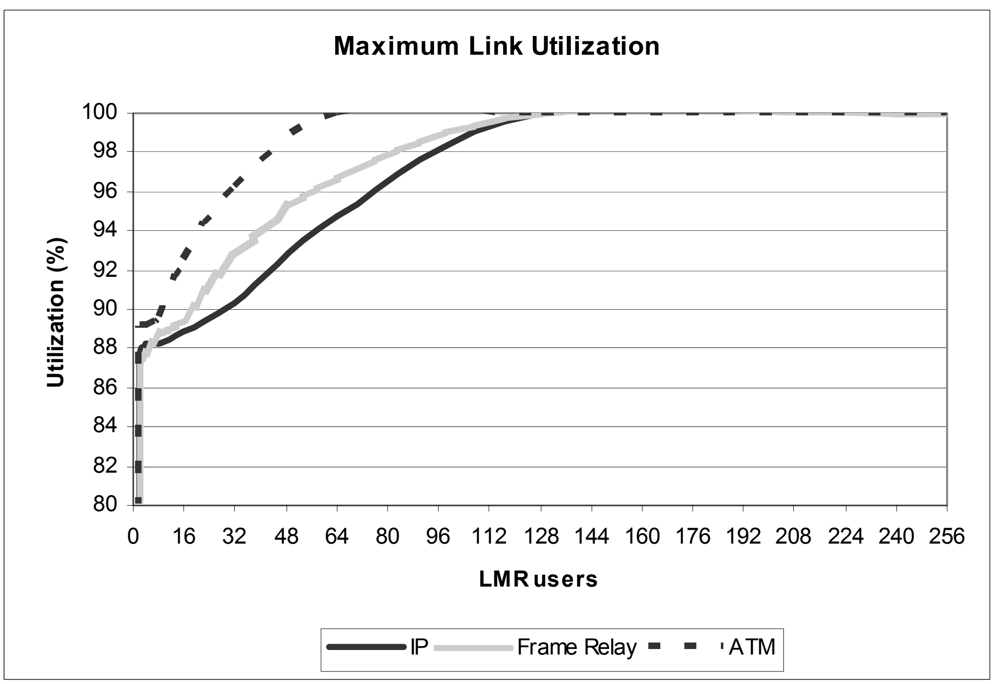

Figure 5.9: Maximum link utilization with 80% background utilization.

riod, while for the 128_ATM_80 and 256_ATM_80 scenarios the link is at 100% utilization for a prolonged period.

In some of the simulation experiments, higher maximum utilizations are obtained for a lower number of LMR users. For example, in 2_FR_80 the maximum link utilization is 87.71% where as in 4_FR_80 it is 86.22%. This can be explained as follows. The maximum utilization caused by the background traffic can be higher than the maximum utilization caused by the explicit traffic exchanged between LMR users. This is true when the number of LMR users in the network is small and the background utilization on the access link is high. Hence, the maximum utilization observed is in fact caused by the background traffic and not by the voice packets generated by the LMR users.

The maximum link utilization curve with 0% background utilization (Figure 5.7) can be used to predict the access link bandwidth required to support different numbers of LMR users. For example, if the access link available at a particular LMR base station is 64 Kbps this is equivalent to 4.145% of the total bandwidth of a T1 link. Drawing a horizontal line at the 4.145% level on Figure 5.7 and observing the intersection points with the maximum utilization curves yields that an ATM WAN can support up to 4 active LMR users, where as Frame Relay and IP WANs can each support up to 8 active LMR users. This observation only holds if the assumptions made in Section 4.4 regarding the duration and spacing of the voice calls between the LMR users are valid. Furthermore, there is no guarantee that the delay and delay variation performance will be satisfactory when using lower bandwidth access links even if the link can support the traffic load. We consider delay and delay variation next.

5.2 End-to-End Delay

Voice communications between LMR users is an interactive process and the quality of the communication is highly dependent on the ETE delay. High ETE delay restricts interactivity and results in echo (talker echo and listener echo) [Wri01]. ITU standard G.114 specifies

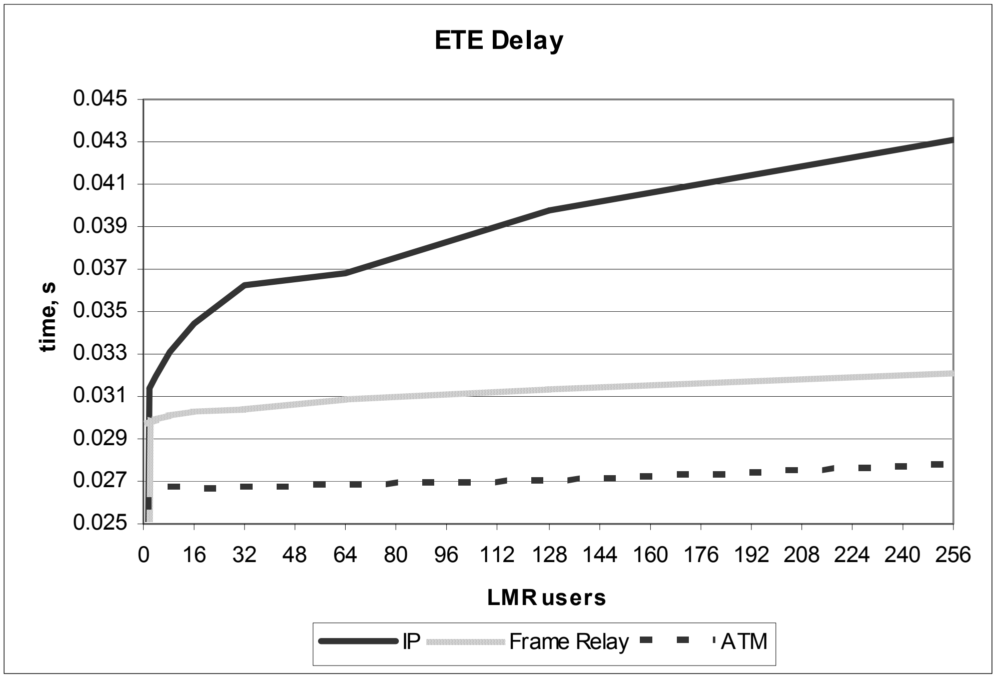

Figure 5.10: ETE delay with 0% background utilization.

that the ETE voice delay must be less than 150 ms [ITU00b] in order to maintain the flow in human conversation. This includes speech coder delay, packetization delay, queuing delay, transmit delay, network delay, air interface delay, and possible de-jitter buffer delay. The ETE delay metric used in this study considers the effects of queuing, transmit, propagation, and WAN switching delays.

ETE delay is measured from the time a voice packet is transmitted by an LMR user until the voice packet is received by the intended destination. Rather than measuring the ETE delay for every voice packet exchanged between LMR users, a sample of 100 voice packets is examined. Each packet is sent in one-second intervals. Figures 5.10 through 5.12 show the ETE delay for 0%, 40%, and 80% background utilization, respectively.

When the background utilization is 0%, ETE delay is kept below 45 ms on average for all three packet transport technologies considered and the maximum ETE delay observed is kept below 58 ms. The IP WAN offers the worst ETE delay performance, while the ATM

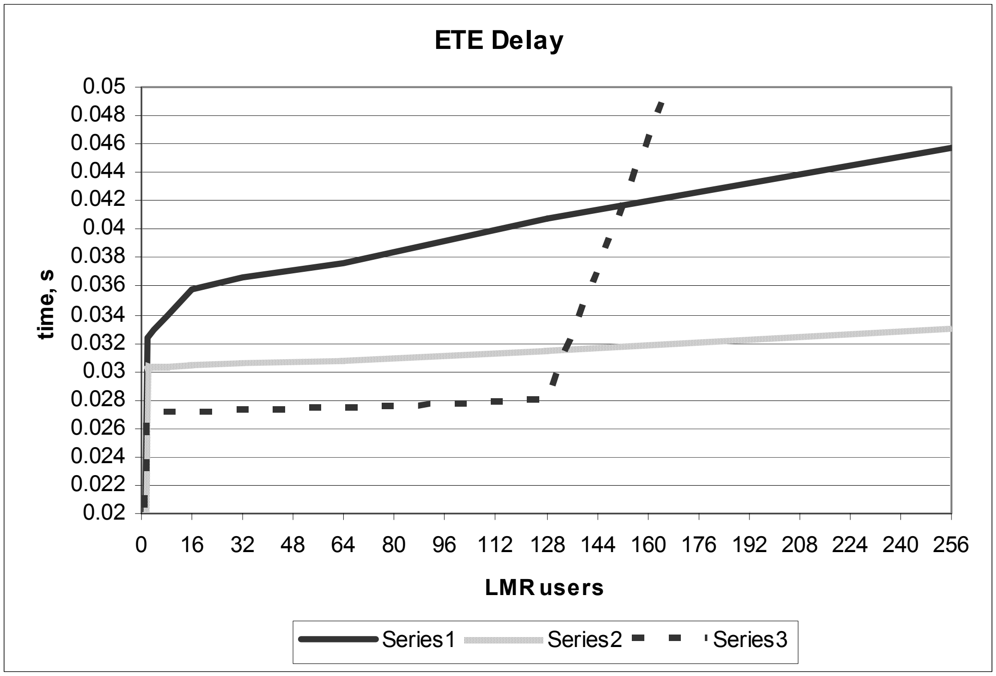

Figure 5.11: ETE delay with 40% background utilization.

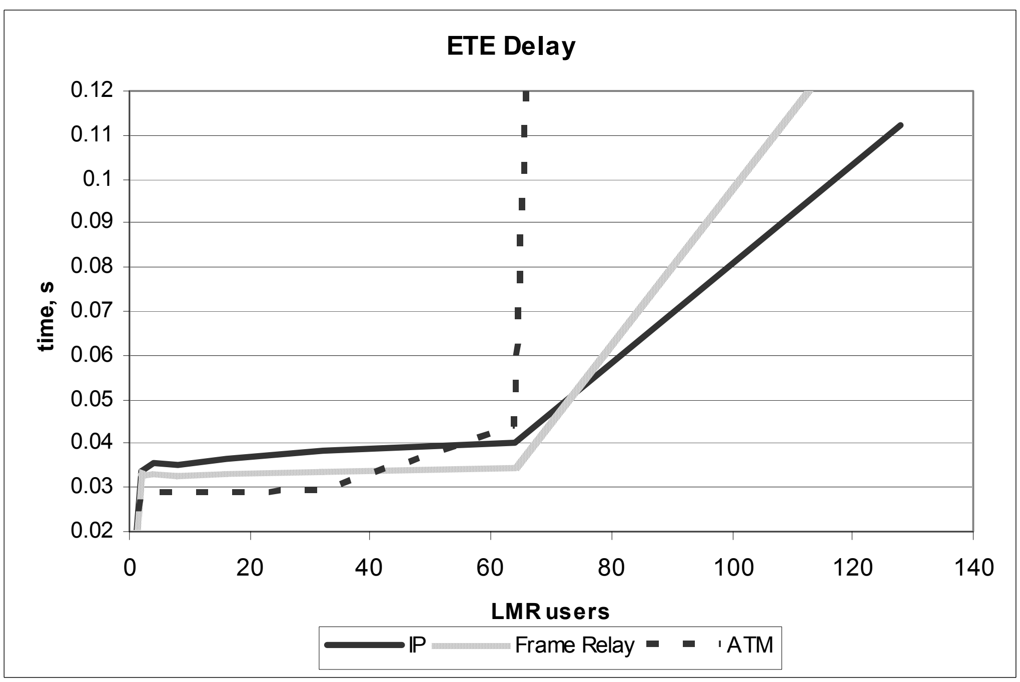

Figure 5.12: ETE delay with 80% background utilization.

WAN offers the best as illustrated in Figure 5.10. This is explained by the better SLAs specified for the ATM WAN. When using a Frame Relay WAN, ETE delay performance is improved compared to an IP WAN but is worse when compared to an ATM WAN. Table 5.1 summarizes the ETE delay performance when background utilization is 0%.

Table 5.1: ETE Delay with 0% Background Utilization

Number of LMR users	**IP**		**Frame Relay**		**ATM**	
	Mean (s)	**95% C.I.**	**Mean (s)**	**95% C.I.**	**Mean (s)**	**95% C.I.**
2	0.0314	±2.81E-05	0.0298	±3.72E-05	0.0264	±5.53E-05
4	0.0320	±6.51E-04	0.0300	±1.29E-04	0.0266	±1.01E-04
8	0.0331	±1.01E-03	0.0302	±4.46E-05	0.0267	±2.89E-05
16	0.0344	±1.91E-03	0.0303	±5.64E-05	0.0267	±4.97E-05
32	0.0362	±1.17E-04	0.0305	±4.15E-05	0.0267	±9.64E-05
64	0.0369	±3.87E-04	0.0309	±2.83E-04	0.0268	±1.56E-04
128	0.0398	±7.66E-05	0.0313	±2.18E-04	0.0270	±1.37E-04
256	0.0431	±7.03E-04	0.0321	±8.12E-05	0.0277	±2.26E-04

Notice that ETE delay reacts differently to an increase in the number of LMR users depending on the packet transport technology used. ETE delay when using an IP WAN is more sensitive to an increase in the number of LMR users compared to an ATM or Frame Relay WAN. This is explained as follows. First, recall that the ETE delay metric used represents queuing, serialization, propagation, and network switching delays. Propagation delay is fixed for all three transport technologies since it is distance-dependent and not technology-dependent. Transmit delay is small due to the relatively high bandwidth of the T1 access link, i.e., transmit delay for a 53-byte cell is 0.27 ms. Queuing delay at the ISI gateway is also small compared to the overall ETE delay even when 256 LMR users are active on the network. To quantify this, consider the following. When 256 LMR users are active, at most, 128 users can be transmitting voice packets across the ISI since LMR communications are half-duplex. Of these 128 users, we postulate that 64 users are transmitting voice packets in the uplink direction and 64 in the downlink direction. The inter-arrival time of packets generated by each user is 60 ms (Section 4.3), hence, even if 64 users are transmitting at the same time, the probability of having multiple packets received at a particular instant is small. In fact, the predominant source of delay is that imposed by the WAN.

For the Frame Relay and ATM WANs the simulation model specifies virtual circuits that provide QoS guarantees for the delay experienced by data traversing the WAN. As long as the data rate of the voice packets generated by the LMR users conforms to the bandwidth supported by the virtual circuit, data is sent across the WAN with minimal influence on the overall ETE delay as shown in Figure 5.10.

Figure 5.11 shows the maximum ETE delay with 40% background utilization. A significant increase in delay is experienced when using an ATM WAN and the number of users increases beyond 128. This occurs because the bandwidth utilization in the uplink direction reaches 100% for a brief period of time (Figure 5.8) causing an increase in the queuing delay at the ISI gateway. A maximum ETE delay of 584 ms is recorded. Such a delay value is well above all acceptable performance levels. Frame Relay and IP WANs can sustain an average ETE delay below 34 ms and 46 ms, respectively, for all numbers of LMR users considered.

When background utilization is increased to 80%, the maximum number of LMR users that can be supported at acceptable levels of ETE delay is significantly reduced as shown in Figure 5.12. With an IP or Frame Relay WAN at 80% utilization, up to 64 users can be supported at acceptable levels of ETE delay. With an ATM WAN, up to 32 users can be supported before the ETE delay performance begins to degrade.

As long as the utilization on the T1 access link is not allowed to reach 100%, the maximum ETE in all three packets transport technologies considered is less than 58 ms.

5.3 Delay Variation or Jitter

In our experiments, we define delay variation as the deviation from the mean ETE delay. This statistic is obtained from the ETE delay values recorded during the simulation experiments as follows. First, the average ETE delay is calculated for each simulation experiment. Then, the individual ETE delays are compared to this average. Delays that are higher than the average are recorded as positive delay variation and delays that are lower than the average are recorded as negative delay variation.

Figure 5.13 shows the delay variation experienced by the voice packets for the three packets transport technologies considered. Selected scenarios are illustrated that provide a good representation of the key differences observed. The main source of delay variation is the queuing delay experienced by each packet while traversing the WAN. Depending on the traffic conditions in the network and the QoS associated with each packet flow, the queuing delay at each switch or router can vary.

The worst case jitter occurs when an IP WAN is used. This is expected since there are no QoS guarantees for the IP packets traversing the WAN. Ultimately, each voice packet has to wait in the queue with all other packets and be serviced in the order received. An ATM WAN can provide the lowest delay variation as shown in Figure 5.13(e) since we define quantitative values for the delay variation experienced by ATM cells traversing the WAN. In fact, the ability to provide service guarantees for delay variation is one the major strengths of ATM that differentiates it from the other two packet transport technologies considered.

Parts (b), (d), and (f) of Figure 5.13 show the delay variation when network congestion occurs. This is indicated by the abrupt increase in delay variation. In such situations, delay variation can be misleading since most packets appear to experience delay variation below the mean ETE delay value (negative delay variation). This only occurs because the high delay variation recorded during periods of network congestion increases the mean ETE delay.

5.4 Call Setup Time

Call setup time[3] is measured as the time difference between when the SIP "INVITE" message is sent to initiate an LMR unit-to-unit voice call and when the "ACK" message is received

[3] The call setup time considered in this study does not include delays associated with the communications across the common air interface.

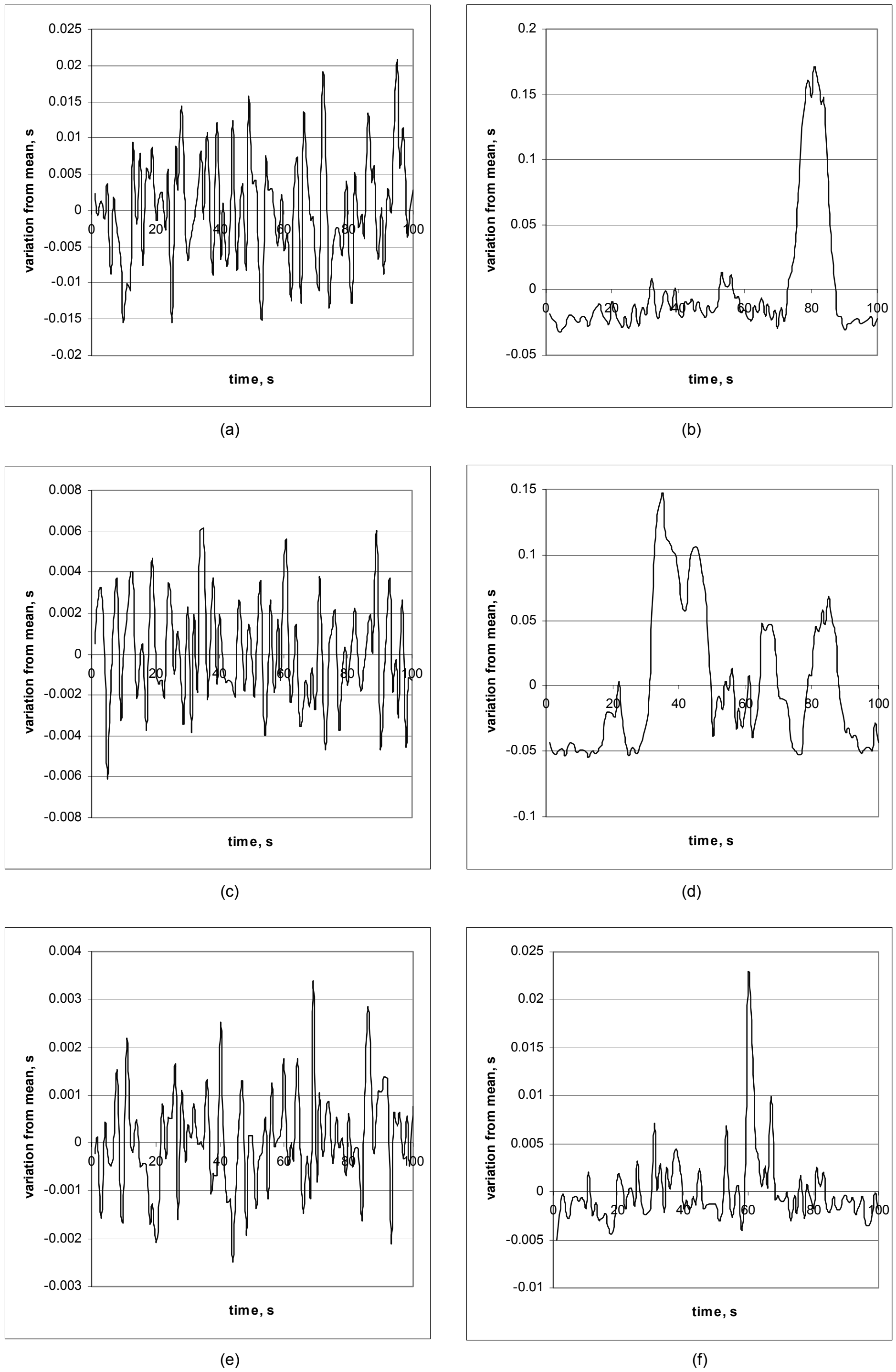

Figure 5.13: Delay variation: (a) 2_IP_80, (b) 128_IP_80, (c) 2_FR_80, (d) 128_FR_80, (e) 2_ATM_40, (f) 64_ATM_40.

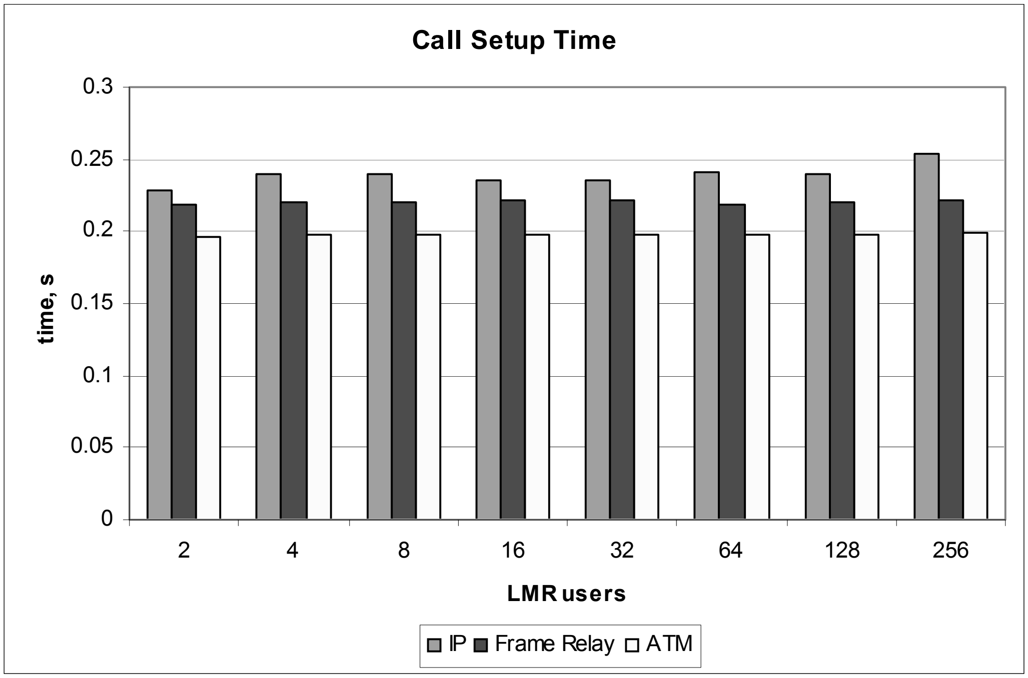

Figure 5.14: Call setup time with 0% background utilization.

(Figure 4.10).

The call setup time is recorded at twelve randomly selected instances during each simulation experiment. The results obtained provide an indication of the time required to setup a call if a user decides to establish voice communications across the ISI at any particular time. Results are averaged for each simulation experiment. Final results for all the simulation experiments are presented in Figures 5.14 through 5.16.

The Project 25 standards specify that throughput delay should be less than 350 ms for user-to-user communications through a single conventional repeater and less than 500 ms for unit-to-unit communications within an RF subsystem [TSB95]. The standards define throughput delay as the propagation delay of audio through the system [TSB95]. No particular specification is made for the delay requirements of the ISI. However, based on the information available, we postulate that a maximum call setup time of 300 ms is a feasible design objective.

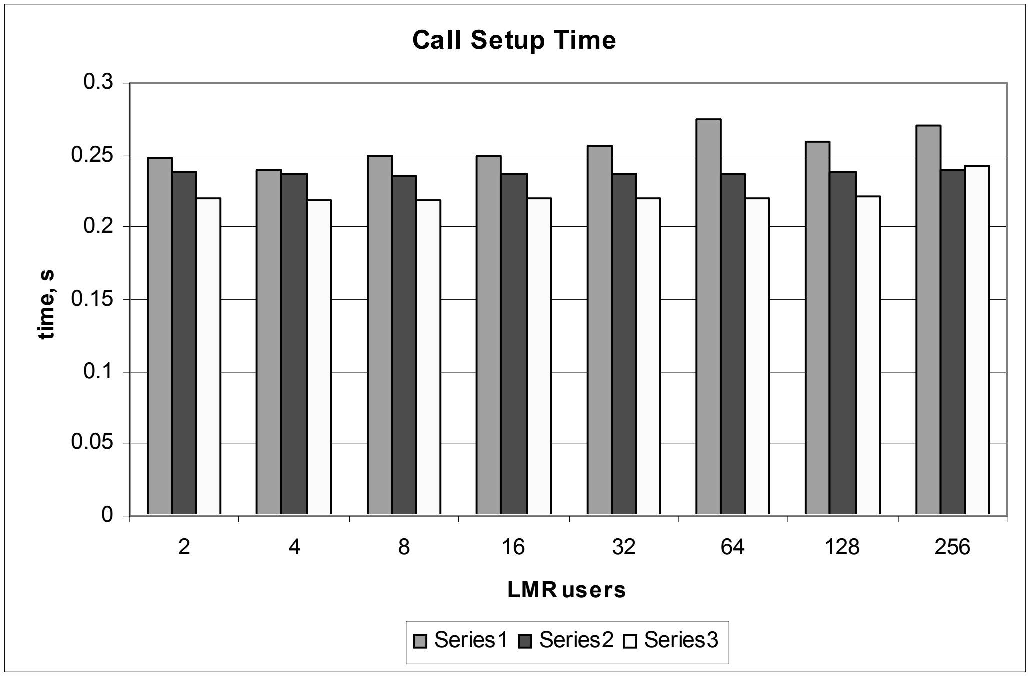

Figure 5.15: Call setup time with 40% background utilization.

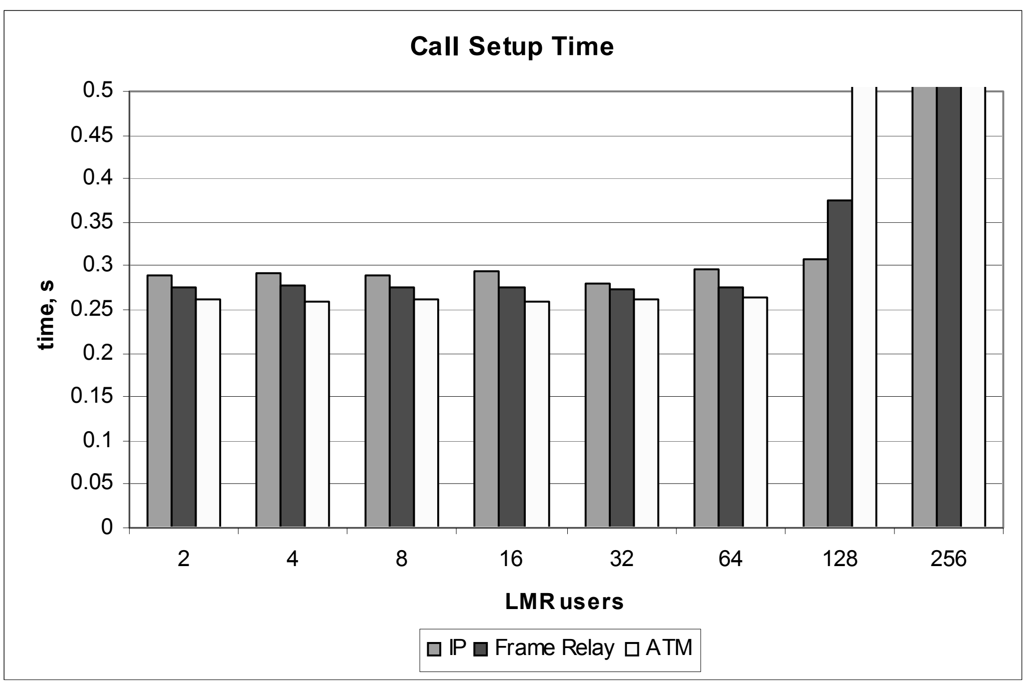

Figure 5.16: Call setup time with 80% background utilization.

Using an ATM WAN provides: (i) the fastest call setup time, and (ii) the most consistent call setup time. This is true as long as enough bandwidth is available on the access link to support the different numbers of LMR users considered. These results are expected given the fact an ATM WAN can provide the lowest ETE delay at a guaranteed QoS (Section 5.2). An IP WAN provides the worst call setup time. Moreover, there is no consistency in the call setup time for different numbers of LMR users. For example, in the 4_IP_0 scenario, the call setup time is 240 ms where as in the 32_IP_0 scenario the call setup time is 235 ms. This is because the delay variation in the best-effort IP WAN is significant. Typical SLAs for IP networks give guarantees for an average round trip delay of 55 ms, but also stipulate a maximum round trip delay of 100 ms [Wil02].

Increasing the background utilization of the access link also results in an increase in the call setup time as expected (Figures 5.15 and 5.16). To maintain the resolution of the graphs, call setup times that are equal or greater than 500 ms are labeled as 500 ms. In any case, call setup times exceeding 500 ms are unacceptable. Table 5.2 summarizes the highest average call setup times for the three different packet transport technologies considered. To get a more realistic representation of the call setup times expected to be observed in an LMR network, values obtained in scenarios where link utilization reaches 100% are not included in Table 5.2. A properly engineered design of the LMR network should provision enough bandwidth to ensure that maximum utilization of the access link is kept below 100%.

Table 5.2: Highest Call Setup Times when using IP, Frame Relay, and ATM WANs

Background Utilization	**Call Setup Time**		
	IP	**Frame Relay**	**ATM**
0%	254 ms	222 ms	199 ms
40%	274 ms	240 ms	221 ms
80%	296 ms	279 ms	263 ms

5.5 Summary

This chapter presented the results of the simulation experiments. Parameters that affect the performance of the packet-switched ISI proposed in Chapter 4 were analyzed. In particular, throughput, link utilization, ETE delay, delay variation, and call setup time were considered.

The results indicate that a packet-switched architecture for the ISI can take advantage of statistical multiplexing techniques to distribute network resources efficiently and, therefore, provide the capacity to support a large number of LMR users. Using an access link providing T1 service, we have shown that up to 256 active LMR users can be supported. The simulation model provides quantitative results for the network load required to support different numbers of LMR users. With this information available, it becomes possible to perform informed capacity planning to provision the appropriate network resources required to support different numbers of LMR users. As shown in the results, even brief periods of network congestion can cause serious repercussions on the performance and quality of voice communications in the LMR network.

Based on currently available SLAs and QoS guarantees, ETE delay and delay variation can be controlled at levels that are capable of supporting the timely delivery of real-time voice packets. As long as the utilization on the access link providing T1 service is maintained below 100%, the maximum ETE experienced by the voice packets in all three packets transport technologies considered is less than 58 ms. Under similar conditions, the call setup time using SIP signaling can be maintained below 300 ms.

The performance of a packet-switched ISI varies depending on the packet transport technology used for the WAN. Bandwidth efficiency is highest when using an IP WAN since voice packets are already in native IP form prior to transmission across the WAN. Bandwidth efficiency is lowest when using an ATM WAN. This is because of the high transmission overhead associated with ATM. We have shown that the bandwidth efficiency for ATM is 59%. Frame Relay has a bandwidth efficiency of 92%. When considering time-dependent metrics such as ETE delay, delay variation, and call setup time, an ATM WAN provides the best performance while an IP WAN provides the worst performance. This can be attributed to the fact that the ATM WAN supports QoS guarantees for delay and delay variation. However, the IP WAN provides best-effort service only. Selecting the appropriate packet transport technology for the WAN is a tradeoff between the delay that can be tolerated by the voice packets traversing the LMR network and the cost of bandwidth on the access link.

One apparent limitation of the background traffic model used is that it does not capture the burstiness of the data packets. Even though this may not influence the ETE delay results when using an ATM or Frame Relay WAN (given their ability to provide QoS guarantees), it can certainly influence the results observed when using an IP WAN. Depending on the size and burstiness of the IP packets we expect that the actual ETE delay observed in an IP WAN will be higher than that predicted by the simulation model.

The next chapter summarizes the research and outlines ideas for future work.

Chapter 6

Conclusions

6.1 Summary

Next generation LMR networks will make the transition from circuit-switched to packet-switched technology. Packet-switched networks utilize capacity efficiently, scale easily and seamlessly, and eliminate single points of failure by providing a distributed architecture. Furthermore, a packet-switched network will facilitate traffic integration. Voice, video, images, text messages, and OAM traffic can all be transported over a single data network.

SIP is gaining popularity as a signaling protocol due to its simplicity and efficiency. We argue that SIP is a suitable signaling protocol for a packet-switched ISI and define SIP messages for setting up and tearing down unit-to-unit voice calls across the ISI. SDP is used to describe how calls between LMR users are encoded. The session description is included in the body of the SIP "INVITE" and "200 OK" messages.

This is the first known published study that attempts to develop a framework and investigate the performance of a packet-switched ISI for LMR networks. Therefore, several assumptions are made. Justification for these assumptions is provided whenever warranted. OPNET Modeler is used to develop a simulation model for the packet-switched ISI. Traffic flows generated by LMR users engaged in unit-to-unit calls across the ISI are characterized. RTP is used for transporting the real-time voice packets between the LMR users. Traffic patterns for each LMR user are characterized by defining user profiles. Profiles consist of tasks and phases that model the behavior of each LMR user. Three packet transport technologies, ATM, IP, and Frame Relay, provide WAN connectivity for the LMR network. Each packet transport technology is modeled using a "cloud" object defined according to published SLAs.

Simulation experiments are carried out to characterize the performance of the packet-switched ISI. Throughput, link utilization, ETE delay, and delay variation are all considered as a function of the number of users active in the LMR network and the background utilization on the access link. Background utilization is used to represent additional data sources that can potentially share the available bandwidth with the LMR network.

6.2 Results

A packet-switched architecture for the ISI can take advantage of statistical multiplexing techniques to distribute network resources more efficiently compared to a circuit-switched network that explicitly allocates fixed resources for each LMR user. This makes it possible to support additional LMR users for a particular bandwidth.

Quantitative results for throughput and link utilization are obtained from the simulation model. With this information available, the bandwidth required to support different numbers of LMR users can be predicted. Accurate capacity planning for the packet-switched ISI is important. The simulation model indicates that even brief periods of network congestion can cause serious repercussions on the performance and quality of the voice communications in the LMR network.

We show that ETE delay and delay variation can be controlled at levels that are capable of supporting the timely delivery of real-time voice packets. The maximum ETE experienced by the voice packets in all three packets transport technologies considered is less than 58 ms assuming link utilization is maintained below 100%. Under similar conditions, the call setup time for establishing a unit-to-unit call across the ISI using SIP signaling can be maintained below 300 ms.

Additional traffic sources (represented as background utilization) can be supported on the same network while still maintaining acceptable performance for the voice traffic exchanged between the LMR users. This is true as long as QoS guarantees can be provisioned for the transport of the voice traffic and sufficient bandwidth is available on the access link to support the traffic load. When using ATM and Frame Relay WANs, that provide QoS guarantees for the voice packets traversing the LMR network, we find that delay and delay variation performance is better compared to the IP WAN that can only provide best-effort service. Delay and delay variation performance is best when using the ATM WAN. This is explained as follows: (i) packets traversing the ATM WAN have the lowest delay (as specified in the SLAs), and (ii) quantitative values are defined for delay variation in the ATM WAN.

One apparent limitation of the background traffic model used is that it does not capture the burstiness of the data packets. Even though this may not influence the ETE delay results recorded when using an ATM or Frame Relay WAN (given their ability to provide QoS guarantees) it can certainly influence the results observed when using an IP WAN. Depending on the size and burstiness of the IP packets, we expect that the actual ETE delay observed in an IP WAN will be higher than that predicted by the simulation model.

6.3 Future Work

The design of a packet-switched ISI for LMR systems is a new area of research. Even though some initiatives have been made to develop standards for a packet-switched ISI, no formal documentation has yet been published. To develop a feasible framework for a packet-switched ISI and obtain some initial performance results, several assumptions were made in our model. These assumptions form the basis for possible future work.

Before any protocol-specific implementations of the packet-switched ISI can be finalized, additional services provided by the ISI need to be considered. Ideally, the ISI should be able to support all the services defined under the common air interface, i.e., mobility management, group calls, unit-to-unit calls, text messaging, interface to PSTN, and other miscellaneous supplementary services. In this thesis we concentrated our efforts on unit-to-unit calls between LMR users. Additional research can investigate the performance of the packet-switched ISI for some of the other services identified above.

Bandwidth efficiency is an area in which improvements can be made. Due to the high overhead of RTP/UDP/IP headers coupled with the layer 2 overhead, a high percentage of the bandwidth available on the access link is utilized by overhead. Mechanisms for improving bandwidth efficiency can be considered in future work. Header compression, frame packing, RTP multiplexing, and the effect of using voice activity detectors (VADs) are some of the areas that can be investigated.

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Appendix A

Acronyms

ACELP	Algebraic Code-Excited Linear Prediction
ANF	Additional Network Features
APCO	Association of Public and Communications Officials
APDU	Application Packet Data Unit
ATM	Asynchronous Transfer Mode
BC	Basic Call
BTS	Base Transceiver Station
CAI	Common Air Interface
CDMA	Code Division Multiple Access
DMO	Direct Mode Operation
ETE	End-to-End
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission
FDMA	Frequency Division Multiple Access
NASTD	National Association of State Telecommunications Directors
FNE	Fixed Network Equipment
GFP	Generic Function Protocol
GSM	Global System for Mobile Communications

HLR	Home Location Register
HTTP	Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
IMBE	Improved Multi-Band Excitation
IP	Internet Protocol
ISI	Intersystem Interface
ISIGC	ISI Group Call
ISIIC	ISI Individual Call
ISIMM	ISI Mobility Management
ISISD	ISI Short Data
ISISS	ISI Supplementary Service
ISO	International Standardization Organization
ISSI	Inter-RF Subsystem Interface
ITU	International Telecommunication Union
LMR	Land Mobile Radio
MSC	Mobile Switching Center
NATF	North American TETRA Forum
NTIA	National Telecommunications Information Association
OAM	Operations Administration and Maintenance
OMC	Operations and Maintenance Center
OTAR	Over-The-Air Rekeying
PDO	Packet Data Optimized
PDU	Packet Data Unit
PINX	Private Integrated Network Exchange
PISN	Private Integrated Service Network
PMR	Private Mobile Radio
PSS1	Private Signalling System 1 (aka QSIG)
PSTN	Public Switched Telephone Network

QoS	Quality of Service
QSIG	Q-Signaling protocol
RF	Radio Frequency
RFC	Request for Comment
RFSS	Radio Frequency Subsystem
ROSE	Remote Operations Service Element
RTP	Real-Time Transport Protocol
SDP	Session Description Protocol
SDS	Short Data Service
SIP	Session Initiation Protocol
SMTP	Simple Mail Transfer Protocol
SNA	Systems Network Architecture
SOR	Statement Of Requirements
SUID	Subscriber Unit ID
SwMI	Switching and Management Infrastructure
TDMA	Time Division Multiple Access
TETRA	Terrestrial Trunked Radio
TIA	Telecommunications Industry Association
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UHF	Ultra High Frequency
V+D	Voice plus Data
VHF	Very High Frequency
VLR	Visitor Location Register
VPN	Virtual Private Networks

Appendix B

Tables of Results

Table B.1: IP WAN, 0% Background Utilization, Average

LMR Users	Average Throughput (bps)			Maximum Link Utilization (%)	ETE Delay (s)	Call Setup Time (s)
	Downlink	Uplink	Total			
2	5471.4	5408.8	10880	1.40	0.031385	0.229005
4	7281.6	7584.3	14866	1.73	0.031989	0.239734
8	12749.6	12311.7	25061	3.01	0.033107	0.240335
16	23270.1	23480.8	46751	4.45	0.034402	0.235567
32	36213.9	37499.7	73714	6.64	0.036227	0.235415
64	75010.8	75809.7	150821	12.99	0.036850	0.240609
128	127608.3	142024.3	269633	22.05	0.039770	0.239943
256	271246.7	289500.5	560747	41.08	0.043073	0.254244
Packet transport technology: IP Background utilization: 0% Simulation experiments average						

Table B.2: IP WAN, 0% Background Utilization, Simulation Experiment 1

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	5130.3	5726.9	10857	1.42	0.031371	0.224555
4	8223.3	7397.5	15621	1.77	0.031657	0.238685
8	11582.3	12154.3	23737	2.51	0.032591	0.243633
16	25200.9	24984.1	50185	4.57	0.035378	0.234156
32	33667.7	37937.9	71606	6.48	0.036287	0.237976
64	79012.9	80388.7	159402	13.13	0.037047	0.242272
128	127213.6	142760.8	269974	19.96	0.039731	0.241100
256	265936.8	286974.7	552912	42.18	0.043432	0.254542
Packet transport technology: IP Background utilization: 0% Simulation seed: 200						

Table B.3: IP WAN, 0% Background Utilization, Simulation Experiment 2

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	5647.4	5281.4	10929	1.35	0.031371	0.237907
4	6696.2	7330.5	14027	1.66	0.032654	0.241832
8	12791.1	10615.2	23406	3.27	0.034138	0.233738
16	24652.9	24929.7	49583	4.32	0.032448	0.238390
32	35611.4	35896.8	71508	6.29	0.036108	0.230293
64	74568.0	73974.1	148542	12.99	0.036455	0.237281
128	132139.3	139531.0	271670	26.43	0.039848	0.237629
256	271763.5	288377.9	560141	41.21	0.042356	0.253648
Packet transport technology: IP Background utilization: 0% Simulation seed: 222						

Table B.4: IP WAN, 0% Background Utilization, Simulation Experiment 3

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	5636.6	5218.2	10855	1.42	0.031414	0.224555
4	6925.4	8024.9	14950	1.77	0.031657	0.238685
8	13875.4	14165.6	28041	3.26	0.032591	0.243633
16	19956.6	20528.7	40485	4.47	0.035378	0.234156
32	39362.5	38664.3	78027	7.14	0.036287	0.237976
64	71451.5	73066.3	144518	12.85	0.037047	0.242272
128	123472.0	143781.0	267253	19.75	0.039731	0.241100
256	276039.7	293148.8	569189	39.84	0.043432	0.254542
Packet transport technology: IP Background utilization: 0% Simulation seed: 128						

Table B.5: IP WAN, 40% Background Utilization, Average

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	618671.9	621115.0	1239787	44.94	0.032410	0.247436
4	620921.8	619793.9	1240715	45.13	0.032874	0.239629
8	624439.1	625338.4	1249777	46.02	0.033953	0.249570
16	631780.5	632449.4	1261529	47.17	0.035799	0.249915
32	647972.9	651047.3	1299020	48.51	0.036592	0.256305
64	684445.8	689985.7	1374431	53.82	0.037628	0.274481
128	757832.4	755906.4	1513739	64.39	0.040700	0.258801
256	894454.5	885641.2	1780096	82.84	0.045684	0.270022
Packet transport technology: IP Background utilization: 40% Simulation experiments average						

Table B.6: IP WAN, 40% Background Utilization, Simulation Experiment 1

LMR Users	Average Throughput (bps)			Maximum Link Utilization (%)	ETE Delay (s)	Call Setup Time (s)
	Downlink	Uplink	Total			
2	617429.5	621151.6	1238581	45.68	0.031784	0.250232
4	622963.1	621964.4	1244927	45.08	0.032796	0.231168
8	625626.4	626831.0	1252457	45.97	0.035075	0.252952
16	629857.8	628848.9	1258707	46.84	0.037081	0.252952
32	646483.7	648304.9	1294789	48.21	0.037081	0.258441
64	678755.7	687339.0	1366095	54.79	0.036639	0.286791
128	761678.1	750196.1	1511874	60.98	0.039988	0.256100
256	901404.0	889384.4	1790788	80.99	0.046128	0.271603
Packet transport technology: IP Background utilization: 40% Simulation seed: 200						

Table B.7: IP WAN, 40% Background Utilization, Simulation Experiment 2

LMR Users	Average Throughput (bps)			Maximum Link Utilization (%)	ETE Delay (s)	Call Setup Time (s)
	Downlink	Uplink	Total			
2	618806.2	620837.2	1239643	44.58	0.033662	0.241844
4	619251.7	619322.7	1238574	45.55	0.033030	0.256549
8	623812.0	625652.9	1249465	45.79	0.031709	0.242805
16	634571.2	632255.3	1258723	48.04	0.033234	0.243842
32	648408.2	652025.2	1300433	48.72	0.035614	0.252031
64	685246.4	692666.9	1377913	53.38	0.037607	0.249861
128	762736.6	758248.9	1520985	66.24	0.042125	0.264202
256	886671.5	868119.1	1754791	82.26	0.044795	0.266859
Packet transport technology: IP Background utilization: 40% Simulation seed: 222						

Table B.8: IP WAN, 40% Background Utilization, Simulation Experiment 3

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	619779.9	621356.3	1241136	44.56	0.031784	0.250232
4	620550.7	618094.7	1238645	44.75	0.032796	0.231168
8	623878.8	623531.4	1247410	46.31	0.035075	0.252952
16	630912.4	636244.0	1267156	46.62	0.037081	0.252952
32	649026.7	652811.9	1301839	48.59	0.037081	0.258441
64	689335.2	689951.1	1379286	53.29	0.038639	0.286791
128	749082.4	759274.3	1508357	65.95	0.039988	0.256100
256	895287.9	899420.2	1794708	85.28	0.046128	0.271603
Packet transport technology: IP Background utilization: 40% Simulation seed: 128						

Table B.9: IP WAN, 80% Background Utilization, Average

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	1237496	1235964	2473460	87.42	0.033763	0.290505
4	1239748	1235399	2466147	88.25	0.035584	0.290963
8	1238342	1241448	2479789	88.29	0.035089	0.288927
16	1246777	1249644	2496421	88.86	0.036439	0.293996
32	1266277	1267286	2533563	90.30	0.038193	0.280677
64	1301489	1300518	2602007	94.72	0.040365	0.295967
128	1371408	1372752	2744160	100	0.112336	0.307601
256	1458346	1474242	2932588	100	3.304551	5.833282
Packet transport technology: IP Background utilization: 80% Simulation experiments average						

Table B.10: IP WAN, 80% Background Utilization, Simulation Experiment 1

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	1235871	1235171	2471042	88.38	0.033583	0.287467
4	1243935	1237107	2481042	87.96	0.034495	0.290141
8	1239800	1245244	2485045	88.59	0.034824	0.286693
16	1246170	1249152	2495321	88.66	0.036854	0.294946
32	1268905	1268403	2537308	89.18	0.038312	0.283143
64	1306498	1299199	2605696	96.04	0.040208	0.293007
128	1376295	1368300	2744595	100	0.069038	0.312419
256	1461184	1477353	2938537	100	3.379223	5.587644
Packet transport technology: IP Background utilization: 80% Simulation seed: 200						

Table B.11: IP WAN, 80% Background Utilization, Simulation Experiment 2

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	1234390	1238050	2472440	86.19	0.034122	0.296582
4	1236325	1233645	2469970	88.23	0.037762	0.292605
8	1237651	1236965	2474615	89.18	0.035618	0.293394
16	1245482	1246208	2491690	89.48	0.035610	0.292095
32	1268905	1264985	2533890	90.94	0.037954	0.275744
64	1304385	1298616	2603001	93.53	0.040680	0.301888
128	1363210	1377607	2740818	100	0.198932	0.297966
256	1454474	1471766	2926240	100	3.155207	6.324559
Packet transport technology: IP Background utilization: 80% Simulation seed: 222						

Table B.12: IP WAN, 80% Background Utilization, Simulation Experiment 3

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	1242228	1234671	2476898	87.69	0.033583	0.287467
4	1238983	1235446	2447428	88.55	0.034495	0.290141
8	1237574	1242135	2479708	87.10	0.034824	0.286693
16	1248679	1253573	2502252	88.43	0.036854	0.294946
32	1261020	1268470	2529490	90.77	0.038312	0.283143
64	1293585	1303740	2597325	94.58	0.040208	0.293007
128	1374719	1372348	2747066	100	0.069038	0.312419
256	1459379	1473608	2932988	100	3.379223	5.587644
Packet transport technology: IP Background utilization: 80% Simulation seed: 128						

Table B.13: Frame Relay WAN, 0% Background Utilization, Average

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	5643.5	6032.3	11676	1.41	0.029757	0.218433
4	9747.0	8895.5	18642	1.90	0.029998	0.219447
8	13471.2	14768.6	28240	2.88	0.030167	0.220144
16	24222.6	25184.9	49408	5.13	0.030309	0.221985
32	45148.1	46683.7	91832	7.65	0.030464	0.220830
64	84983.7	83926.3	168910	15.26	0.030892	0.217998
128	159321.1	163598.0	322919	26.11	0.031336	0.220369
256	328367.0	329070.7	657438	46.60	0.032108	0.221953
Packet transport technology: Frame Relay Background utilization: 0% Simulation experiments average						

Table B.14: Frame Relay WAN, 0% Background Utilization, Simulation Experiment 1

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	5174.2	6274.9	11449	1.56	0.029859	0.221198
4	9249.2	9414.8	18664	1.90	0.030061	0.220252
8	13783.5	13632.3	27416	3.57	0.030211	0.222524
16	24181.4	24960.4	49142	5.27	0.030479	0.221391
32	43308.8	48328.9	91638	7.19	0.030543	0.223249
64	87074.3	82826.4	169901	14.77	0.031027	0.218392
128	158788.7	163508.7	322297	26.42	0.031424	0.221912
256	324802.4	333385.3	658188	45.24	0.032088	0.223204
Packet transport technology: Frame Relay Background utilization: 0% Simulation seed: 200						

Table B.15: Frame Relay WAN, 0% Background Utilization, Simulation Experiment 2

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	6737.1	5427.7	12165	1.65	0.029692	0.217463
4	10181.1	8959.1	19140	1.90	0.030079	0.219594
8	12813.6	15598.4	28412	2.39	0.030183	0.220612
16	22032.2	25997.7	48030	5.38	0.030233	0.224450
32	46197.4	45619.7	91817	7.96	0.030453	0.219721
64	81348.2	86033.7	167382	15.07	0.030577	0.218116
128	156828.9	163476.2	320305	27.73	0.031483	0.220631
256	319319.5	337051.6	656371	47.49	0.032189	0.221986
Packet transport technology: Frame Relay Background utilization: 0% Simulation seed: 222						

Table B.16: Frame Relay WAN, 0% Background Utilization, Simulation Experiment 3

LMR	**Average Throughput (bps)**			**Maximum Link**	**ETE**	**Call Setup**
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	5019.1	6394.2	11413	1.01	0.029721	0.216637
4	9810.6	8312.7	18123	1.90	0.029854	0.218494
8	13816.6	15075.2	28892	2.69	0.030108	0.217295
16	26454.2	24596.7	51051	4.75	0.030215	0.220113
32	45938.2	46102.5	92041	7.80	0.030396	0.219520
64	86528.6	82918.7	169447	15.93	0.031071	0.217486
128	162345.7	163809.0	326155	24.19	0.031101	0.218563
256	340979.0	316775.1	657754	47.07	0.032046	0.220668
Packet transport technology: Frame Relay Background utilization: 0% Simulation seed: 128						

Table B.17: Frame Relay WAN, 40% Background Utilization, Average

LMR	**Average Throughput (bps)**			**Maximum Link**	**ETE**	**Call Setup**
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	619956.7	618312.8	1238269	43.45	0.030305	0.237881
4	622253.0	620658.8	1242912	44.81	0.030450	0.236930
8	627127.6	625926.3	1253054	45.92	0.030320	0.234911
16	637451.0	637206.5	1274658	47.85	0.030471	0.237106
32	654132.4	656653.6	1310786	49.81	0.030666	0.236490
64	691092.6	697319.4	1388412	55.10	0.030802	0.236433
128	775973.5	780627.5	1556601	68.14	0.031503	0.238098
256	931985.9	940802.2	1872788	87.67	0.033072	0.239800
Packet transport technology: Frame Relay Background utilization: 40% Simulation experiments average						

Table B.18: Frame Relay WAN, 40% Background Utilization, Simulation Experiment 1

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	621938.6	620644.4	1242583	44.48	0.030195	0.239278
4	624514.2	622098.0	1246612	45.30	0.030727	0.236445
8	627036.0	627199.9	1254236	46.43	0.030023	0.236204
16	636671.9	637869.6	1274542	47.29	0.030329	0.236539
32	652665.2	656683.4	1309349	50.15	0.030445	0.238163
64	694802.9	697253.2	1392056	55.89	0.030932	0.238312
128	774250.1	785851.1	1560101	67.47	0.031175	0.240054
256	929690.8	947508.0	1877199	88.35	0.032849	0.238612
Packet transport technology: Frame Relay Background utilization: 40% Simulation seed: 200						

Table B.19: Frame Relay WAN, 40% Background Utilization, Simulation Experiment 2

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	617822.2	617803.2	1235625	42.93	0.030292	0.236629
4	619840.0	619969.5	1239810	44.21	0.030276	0.237637
8	627560.4	625099.9	1252660	46.01	0.030802	0.236674
16	636563.0	639393.0	1275956	48.63	0.030807	0.237781
32	652557.4	656216.7	1308774	49.42	0.030510	0.234501
64	687046.4	694402.1	1381448	54.97	0.030371	0.232677
128	782400.7	770827.9	1553229	68.37	0.031857	0.240173
256	940237.9	923631.2	1863869	87.14	0.032964	0.241947
Packet transport technology: Frame Relay Background utilization: 40% Simulation seed: 222						

Table B.20: Frame Relay WAN, 40% Background Utilization, Simulation Experiment 3

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	620109.4	616490.9	1236600	42.93	0.030428	0.237737
4	622404.9	619909.0	1242314	44.91	0.030348	0.236707
8	626786.5	625479.1	1252266	45.31	0.030134	0.231854
16	639118.0	634357.0	1273475	47.62	0.030276	0.236997
32	657174.5	657060.7	1314235	49.87	0.031044	0.236806
64	691428.5	700302.8	1391731	54.45	0.031105	0.238312
128	771269.7	785203.6	1556473	68.57	0.031476	0.234066
256	926029.0	951267.4	1877296	87.51	0.033405	0.238842
Packet transport technology: Frame Relay Background utilization: 40% Simulation seed: 128						

Table B.21: Frame Relay WAN, 80% Background Utilization, Average

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	1235211	1237438	2472649	87.09	0.032704	0.275063
4	1236113	1239619	2475732	87.73	0.033123	0.278788
8	1243948	1242242	2486190	88.75	0.032818	0.275436
16	1249389	1253720	2503109	89.44	0.033100	0.275928
32	1272021	1270570	2542592	92.73	0.033629	0.272253
64	1305550	1311709	2617259	96.74	0.034533	0.275105
128	1387738	1391645	2779383	100.00	0.145642	0.373980
256	1493519	1494718	2988237	100.00	5.268378	8.757478
Packet transport technology: Frame Relay Background utilization: 80% Simulation experiments average						

Table B.22: Frame Relay WAN, 80% Background Utilization, Simulation Experiment 1

LMR	**Average Throughput (bps)**			**Maximum Link**	**ETE**	**Call Setup**
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	1234082	1241239	2475321	87.71	0.032705	0.269077
4	1238462	1238256	2476718	87.22	0.033216	0.281967
8	1246019	1240344	2486363	89.07	0.032216	0.270566
16	1248106	1247945	2496051	90.16	0.032822	0.272884
32	1268730	1270207	2538937	92.64	0.033472	0.271408
64	1308903	1315654	2624558	96.22	0.034738	0.271795
128	1390298	1383810	2774108	100	0.151376	0.293162
256	1501183	1488449	2989632	100	3.710520	8.828979
Packet transport technology: Frame Relay Background utilization: 80% Simulation seed: 200						

Table B.23: Frame Relay WAN, 80% Background Utilization, Simulation Experiment 2

LMR	**Average Throughput (bps)**			**Maximum Link**	**ETE**	**Call Setup**
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	1234941	1234441	2469382	86.85	0.032726	0.274172
4	1234313	1242945	2477258	88.00	0.032781	0.275736
8	1239882	1243746	2483629	88.87	0.032533	0.276112
16	1251624	1257525	2509149	89.03	0.032619	0.280292
32	1273995	1271305	2545300	91.72	0.033430	0.275432
64	1302131	1310083	2612214	96.88	0.034651	0.272483
128	1386581	1393517	2780098	100	0.086493	0.542518
256	1501098	1487311	2988409	100	6.248030	8.877963
Packet transport technology: Frame Relay Background utilization: 80% Simulation seed: 222						

Table B.24: Frame Relay WAN, 80% Background Utilization, Simulation Experiment 3

LMR	Average Throughput (bps)			Maximum Link	ETE	Call Setup
Users	Downlink	Uplink	Total	Utilization (%)	Delay (s)	Time (s)
2	1236609	1236634	2473243	86.70	0.032683	0.281939
4	1235565	1237657	2473221	87.96	0.033372	0.278660
8	1245942	1242636	2488579	88.31	0.033705	0.279630
16	1248437	1255691	2504128	89.13	0.033858	0.274608
32	1273339	1270199	2543538	93.82	0.033983	0.269919
64	1305616	1309389	2615005	97.11	0.034209	0.281038
128	1386334	1397609	2783944	100	0.199058	0.286260
256	1478277	1508393	2986670	100	5.846585	8.565494
Packet transport technology: Frame Relay Background utilization: 80% Simulation seed: 128						

Table B.25: ATM WAN, 0% Background Utilization, Average

LMR	Average Throughput (bps)			Maximum Link	ETE	Call Setup
Users	Downlink	Uplink	Total	Utilization (%)	Delay (s)	Time (s)
2	8001.7	7646.7	15648	2.13	0.026437	0.196681
4	13014.4	13851.7	26866	3.23	0.026668	0.198013
8	21533.1	23684.6	45218	4.79	0.026721	0.197761
16	31463.8	31458.8	62923	6.91	0.026621	0.197573
32	61715.3	64556.8	126272	12.16	0.026737	0.197681
64	124645.5	127606.9	252252	21.07	0.026809	0.197520
128	242779.4	244293.2	487073	40.15	0.027009	0.197810
256	460979.4	472561.4	933541	68.75	0.027732	0.198715
Packet transport technology: ATM Background utilization: 0% Simulation experiments average						

Table B.26: ATM WAN, 0% Background Utilization, Simulation Experiment 1

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	7567.6	7801.6	15369	2.03	0.026408	0.196309
4	11546.8	13539.3	25086	2.72	0.026756	0.197563
8	22088.5	25399.5	47488	4.86	0.026698	0.197131
16	28854.8	28542.5	57397	7.00	0.026641	0.197559
32	61245.2	67400.8	128646	12.82	0.026648	0.197382
64	125770.8	130114.2	255885	22.09	0.026674	0.197026
128	238574.1	245376.6	483951	46.11	0.027068	0.198282
256	457337.5	479312.2	936650	63.42	0.027749	0.199196
Packet transport technology: ATM Background utilization: 0% Simulation seed: 200						

Table B.27: ATM WAN, 0% Background Utilization, Simulation Experiment 2

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	7201.2	7438.7	14640	2.14	0.026410	0.196493
4	13711.5	14489.3	28201	3.54	0.026578	0.198627
8	20503.2	24105.9	44609	4.04	0.026748	0.198250
16	31960.9	33920.0	65881	6.89	0.026651	0.197433
32	64578.5	64889.4	129468	12.25	0.026743	0.197781
64	127423.4	125069.7	252493	20.84	0.026950	0.197829
128	248126.4	245703.9	493830	38.06	0.026870	0.197443
256	470810.4	456404.3	927215	70.85	0.027524	0.198664
Packet transport technology: ATM Background utilization: 0% Simulation seed: 222						

Table B.28: ATM WAN, 0% Background Utilization, Simulation Experiment 3

LMR Users	Average Throughput (bps) Downlink	Uplink	Total	Maximum Link Utilization (%)	ETE Delay (s)	Call Setup Time (s)
2	9236.4	7699.8	16936	2.22	0.026493	0.197241
4	13784.8	13526.6	27311	3.43	0.026668	0.197850
8	22007.5	21548.4	43556	5.46	0.026715	0.197902
16	33575.8	31914.0	65490	6.84	0.026571	0.197728
32	59322.3	61380.1	120702	11.42	0.026818	0.197880
64	120742.4	127636.8	248379	20.27	0.026801	0.197705
128	241637.6	241799.1	483437	36.28	0.027090	0.197704
256	454790.3	481967.7	936758	71.99	0.027922	0.198284

Packet transport technology: ATM
Background utilization: 0%
Simulation seed: 128

Table B.29: ATM WAN, 40% Background Utilization, Average

LMR Users	Average Throughput (bps) Downlink	Uplink	Total	Maximum Link Utilization (%)	ETE Delay (s)	Call Setup Time (s)
2	621835.1	622689.7	1244525	44.36	0.027030	0.219881
4	624778.7	625573.4	1250352	45.63	0.027070	0.219364
8	631712.3	633333.0	1265045	45.96	0.027137	0.218818
16	645158.2	647319.8	1292478	48.67	0.027124	0.220107
32	673673.5	673105.6	1346779	54.27	0.027257	0.220321
64	728860.6	721170.1	1450031	61.01	0.027344	0.220592
128	839084.7	849457.6	1688542	81.30	0.027940	0.221062
256	4241334.3	1079196.0	2136630	100	0.101028	0.242321

Packet transport technology: ATM
Background utilization: 40%
Simulation experiments average

Table B.30: ATM WAN, 40% Background Utilization, Simulation Experiment 1

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	621961.1	623592.6	1245554	44.58	0.026981	0.219308
4	624229.7	625317.9	1249548	46.86	0.027195	0.219439
8	631759.6	634269.0	1266029	46.18	0.027111	0.219278
16	647273.7	648219.5	1295493	48.93	0.027116	0.219918
32	674837.0	674038.1	1348875	56.05	0.027353	0.219922
64	736271.6	722994.8	1459266	61.43	0.027299	0.222813
128	829432.7	850519.7	1679952	84.56	0.027956	0.222358
256	1053649.0	1081060.0	2134709	100	0.101084	0.247175
Packet transport technology: ATM Background utilization: 40% Simulation seed: 200						

Table B.31: ATM WAN, 40% Background Utilization, Simulation Experiment 2

LMR Users	**Average Throughput (bps)**			**Maximum Link Utilization (%)**	**ETE Delay (s)**	**Call Setup Time (s)**
	Downlink	**Uplink**	**Total**			
2	621258.8	621021.1	1242280	44.88	0.026945	0.220181
4	624747.5	623254.2	1248002	45.38	0.026966	0.219153
8	633379.7	632489.4	1265869	45.62	0.027252	0.218465
16	643374.2	646848.6	1290223	49.34	0.027170	0.220503
32	675232.9	675468.2	1350701	54.30	0.027103	0.219463
64	728699.1	727929.5	1456629	59.62	0.027396	0.219692
128	845945.3	842717.8	1688663	81.55	0.027965	0.220991
256	1057304.0	1078828.0	2136132	100	0.117263	0.223840
Packet transport technology: ATM Background utilization: 40% Simulation seed: 222						

Table B.32: ATM WAN, 40% Background Utilization, Simulation Experiment 3

LMR	Average Throughput (bps)			Maximum Link	ETE	Call Setup
Users	Downlink	Uplink	Total	Utilization (%)	Delay (s)	Time (s)
2	622285.5	623455.4	1245741	43.62	0.027164	0.220153
4	625358.8	628148.1	1253507	44.64	0.027049	0.219502
8	629997.6	633240.7	1263238	46.09	0.027047	0.218711
16	644826.8	646891.3	1291718	47.75	0.027085	0.219901
32	670950.5	669810.6	1340761	52.45	0.027314	0.221579
64	721611.1	712586.0	1434197	61.97	0.027336	0.219272
128	841876.1	855135.3	1697011	77.80	0.027898	0.219838
256	10613050.0	1077700.0	2139050	100	0.084737	0.225947
Packet transport technology: ATM Background utilization: 40% Simulation seed: 128						

Table B.33: ATM WAN, 80% Background Utilization, Average

LMR	Average Throughput (bps)			Maximum Link	ETE	Call Setup
Users	Downlink	Uplink	Total	Utilization (%)	Delay (s)	Time (s)
2	1236913	1238080	2474993	89.08	0.028499	0.261347
4	1238291	1235824	2474115	89.21	0.028503	0.258583
8	1244017	1243695	2487713	89.57	0.028641	0.260712
16	1258758	1258315	2517072	92.6	0.028871	0.260253
32	1287861	1285254	2573115	96.31	0.029071	0.260461
64	1341416	1340349	2681765	100	0.044063	0.263729
128	1460831	1461020	2921851	100	2.431789	3.677352
256	1544000	1544000	3088000	100	6.341882	22.665598
Packet transport technology: ATM Background utilization: 80% Simulation experiments average						

Table B.34: ATM WAN, 80% Background Utilization, Simulation Experiment 1

LMR	Average Throughput (bps)			Maximum Link	ETE	Call Setup
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	1235424	1240377	2475801	88.19	0.028695	0.262734
4	1239017	1236365	2475383	88.05	0.028580	0.257591
8	1247745	1245180	2492926	90.28	0.028790	0.262052
16	1257535	1260589	2518124	93.54	0.028847	0.259285
32	1289666	1293468	2583133	97.86	0.029146	0.260438
64	1352445	1347474	2699919	100	0.031645	0.262749
128	1460634	1473234	2933867	100	2.418846	3.080320
256	1544000	1544000	3088000	100	6.196267	18.482612
Packet transport technology: ATM Background utilization: 80% Simulation seed: 200						

Table B.35: ATM WAN, 80% Background Utilization, Simulation Experiment 2

LMR	Average Throughput (bps)			Maximum Link	ETE	Call Setup
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	1239262	1236716	2475979	87.88	0.028364	0.259652
4	1238157	1233949	2472106	89.32	0.028603	0.258623
8	1240152	1242131	2482284	88.50	0.028549	0.259403
16	1258490	1260943	2519433	93.11	0.028893	0.261007
32	1290485	1280951	2571436	97.82	0.029090	0.260801
64	1340962	1342497	2683459	100	0.068343	0.264370
128	1461362	1447078	2908440	100	2.379424	3.834242
256	1544000	1544000	3088000	100	6.344201	25.045199
Packet transport technology: ATM Background utilization: 80% Simulation seed: 222						

Table B.36: ATM WAN, 80% Background Utilization, Simulation Experiment 3

LMR	**Average Throughput (bps)**			**Maximum Link**	**ETE**	**Call Setup**
Users	**Downlink**	**Uplink**	**Total**	**Utilization (%)**	**Delay (s)**	**Time (s)**
2	1236053	1237146	2473199	91.16	0.028437	0.261656
4	1237700	1237157	2474856	90.26	0.028327	0.259537
8	1244154	1243774	2487928	89.93	0.028584	0.260679
16	1260248	1253412	2513660	91.16	0.028872	0.260466
32	1283432	1281344	2564775	93.26	0.028976	0.260143
64	1330842	1331075	2661918	100	0.032201	0.264068
128	1460497	1462749	2923247	100	2.497095	4.117494
256	1544000	1544000	3088000	100	6.485179	24.468983
Packet transport technology: ATM						
Background utilization: 80%						
Simulation seed: 128						

Vita

Stavros A. Tsiakkouris was born on January 27, 1977, in Nicosia, Cyprus. He received a Fulbright scholarship that funded his undergraduate studies at Virginia Polytechnic Institute and State University (Virginia Tech). In May 2001 he received his Bachelor of Science in Electrical Engineering. He continued with graduate studies at Virginia Tech working as a Graduate Research Assistant, and in July 2002 he earned his Master of Science in Electrical Engineering. He is an active member of the Institute of Electrical and Electronic Engineers (IEEE), IEEE Communications Society, and IEEE Computer Society.

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