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Dependable and Extensible Next Generation Network Communication Services

by

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A thesis submitted to
the Faculty of Graduate Studies and Research
in partial fulfillment of
the requirements for the degree of
Master of Applied Science

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May 2001

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acceptance of the thesis

Dependable and Extensible Next Generation Network Communication Services

submitted by Michael Gawargy, B.Eng.
in partial fulfillment of the requirements for
the degree of Master of Applied Science

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May 2001

Abstract

This thesis investigates existing Next Generation Network (NGN) architectures and protocols, and proposes modifications and extensions to enable the creation of NGN services that are dependable and extensible. Extensibility is addressed at the application layer by building on the SIP protocol: first decoupling its underlying session establishment capability from its upper layer call setup functionality, then defining a new application protocol for extensible services on top of the decoupled base SIP. Dependability is addressed at the transport layer, by trading off strict sequencing for improved performance. This is achieved by extending the domain of a new partially ordered transport protocol, SCTP, into the services domain where it is used to carry service-related messaging between Communication and Application Servers. A simulation model is used to compare this SCTP approach with previous TCP schemes. The results achieved are promising, showing a 30% performance improvement in response times under typical traffic conditions.

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Dedication

This thesis is dedicated to the memories of two individuals who have made a profound impact on my life: to the memory of Aunt Fayza, my favourite aunt, who took care of us as if we were her own children, and to the memory of my close friend, colleague, and brother-in-law, Dang Minh Le, always enthusiastic, optimistic and fearless in life.

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Glossary and Definitions

AAL **ATM Adaptation Layer**

The portion of the ATM protocol that breaks up packets into 48-byte payloads. A 5-byte header is then added to form ATM cells. The AAL resides between the higher layer transport protocols and the ATM layer.

AIN **Advanced Intelligent Network**

The Telcordia (formerly Bellcore) defined version of the intelligent network, the SS7-based services component of the PSTN.

API **Application Programming Interface**

An abstraction, in the form of a programming language construct or message format, used by an application program to communicate with the operating system or with another system.

ASN **Abstract Syntax Notation**

An international standard for classifying data structures widely used in telecommunications applications.

ATM **Asynchronous Transfer Mode**

A network technology for LANs and WANs that supports realtime voice and video traffic in addition to data.

BCM Basic Call Model

A finite state machine modelling the call processing operations within telephony switch or call server.

BCSM Basic Call State Model

Term used interchangeably with BCM (see above).

CORBA Common Object Request Broker Architecture

A standard from the Object Management Group (OMG) for communicating between distributed objects (self-contained software modules). CORBA provides a way to execute programs (objects) written in different programming languages running on different platforms no matter where they reside in the network.

BICC Bearer Independent Call Control

An enhancement to ISUP, abstracting the protocol so that it is independent of the underlying bearer fabric (TDM, ATM or IP-based).

CCS7 Common Channel Signalling System Number 7

Same as SS7. The protocol and signalling network used in the public switched telephone system for setting up calls and providing services.

CTI Computer Telephony Integration

Term used to describe approaches that combine data with voice systems in order to enhance telephony services, typically in enterprise and call centre environments.

DP **Detection Point**

Intelligent Network term for the points during which the basic call model may be extended using off-board service logic.

DTMF **Dual-Tone Multi-Frequency**

The type of audio signals that are generated when you press the buttons on a touch-tone telephone.

EDP **Event Detection Point**

Intelligent Network term for a detection point that is dynamically armed within the context of an already-established dialogue between an SSP and an SCP.

H.GCP **H-series Gateway Control Protocol**

ITU-T standards temporary designation for the gateway control protocol later known as H.248.

IEEE **The Institute of Electrical and Electronics Engineers**

An organization of engineers, scientists and students in electronics and related fields. See www.ieee.org.

IETF **Internet Engineering Task Force**

An organization of working groups dedicated to identifying problems and providing technical solutions related to Internet technologies. See www.ietf.org.

IN Intelligent Network

The public switched telephone network services architecture of the 1990s, which was developed by Bellcore (now Telcordia) and the ITU to provide a variety of advanced telephony services.

INAP Intelligent Network Application Part

Intelligent Network TCAP-based protocol used in European SS7 networks that is used to query distributed service databases.

IP Internet Protocol

The network layer (layer 3) protocol of the Internet.

IPDC Internet Protocol Device Control

A protocol for media gateway control, preceding the standardization of the MEGACO / H.248 protocol.

ISDN Integrated Services Digital Network

International telecommunications standard for providing digital service on the access link from the customer's premises to the telephone network.

ISUP ISDN Signalling User Part

The SS7-based call signalling protocol used for call establishment and takedown.

ITU	International Telecommunications Union
	An international communications standards organization. See www.itu.ch .
ITU-T	ITU - Telecommunications Standards Section
	The arm of the ITU dealing with telecommunications standardization.
LAN	Local Area Network
	A communications network serving users within a confined geographical area
MDCP	Media Device Control Protocol
	A gateway control protocol that existed prior to the standardization of MEGACO / H.248.
MEGACO	Media Gateway Control Protocol
	The IETF protocol for gateway control in IP telephony.
MG	Media Gateway
	A device or network entity that converts media (or multimedia) into network payload.
MGC	Media Gateway Controller
	A network entity that controls media gateways using a gateway control protocol.
MGCP	Media Gateway Control Protocol

A gateway control protocol that existed prior to the standardization of MEGACO / H.248.

MIME Multipart Internet Mail Extension

A method for transmitting text and non-text files via electronic mail. This same approach is also used in other domains (such as SIP signalling).

MPLS Multi-Protocol Label Switching

An IETF specification for layer 3 switching to deliver the QoS required to support realtime voice and video.

MTP Message Transfer Part

The lower protocol layers of the SS7 protocol stack.

NAS Network Access Server

A node that provides remote users with dial-up access to the network.

NCS Network Control Signalling

The gateway control protocol used in packet cable telephony specifications.

NGN Next Generation Network

The evolution of the PSTN to support multimedia converged network services.

O-BCM Originating BCM

The finite state machine describing the handling of a call origination.

PIC Point In Call

IN term for the call processing operations that occur during a specific state in the BCSM (basic call state model).

PSTN **Public Switched Telephony Network**

The voice telephone network.

QoS **Quality of Service**

The ability to define a level of performance in communication networks.

RAS **Registration, Admission, and Status**

H.323 control channel used for registration, admission, and status.

RAS **Remote Access Server**

A node that provides remote users with dial-up access to the network.

RFC **Request For Comment**

IETF draft document that describes a technical proposes or specifies a recommended technology.

RSVP **Reservation Protocol**

A communications protocol that signals a router to reserve bandwidth for realtime transmission. RSVP is designed to clear a path for audio and video traffic eliminating annoying skips and hesitations. It has been sanctioned by the IETF, because audio and video traffic is expected to increase dramatically on the Internet.

SCCP **Signalling Connection Control Protocol**

SS7 protocol residing between the upper-layer TCAP and lower-layer MTP protocols.

SCP Service Control Point

A node in the SS7 network providing services to PSTN switches.

SCTP Stream Control Transmission Protocol

An IETF transport protocol initially defined to interconnect a Media Gateway Controller with its Signalling Gateways.

SG Signalling Gateway

A signalling gateway provides an interface between the IP Telephony domain and the PSTN signalling domain (typically SS7).

SG Study Group

Used to describe a working group within a standards forum.

SIP Session Initiation Protocol

An IETF protocol initially defined for IP telephony session establishment.

SIP-S SIP for Services

A term introduced in this thesis for a new services protocol based on SIP.

SIP-T SIP for Telephony

The use of SIP protocol for transit (inter call server) call setup.

SIP-U SIP for User Agents

A term introduced in this thesis for the original intended usage of SIP as a session establishment protocol between User Agents.

SS7 Signalling System Number 7

Same as CCS7. The protocol and signalling network used in the public switched telephone system for setting up calls and providing services.

SSP Service Switching Point

An IN term for a switch (or call server) capable of messaging with an SCP for advanced services.

T-BCM Terminating BCM

The finite state machine describing the process of handling an incoming call setup.

TCAP Transaction Capability Application Part

The SS7 protocol used by IN and AIN for communicating services-related messages.

TCP Transmission Control Protocol

IETF transport layer (layer 4) reliable protocol, that ensures that a given message is delivered reliably (accurately and in its entirety).

TDP Trigger Detection Point

Intelligent Network term for a detection point that is pre-provisioned on the SSP.

TTL Time To Live

A set maximum lifetime for a packet (before it gets discarded) as it travels through the network.

UAC User Agent Client

Term used in the SIP domain for the entity that initiates a session.

UAS User Agent Server

Term used in the SIP domain for the entity that handles an incoming request for session establishment.

UDP User Datagram Protocol

IETF transport layer (layer 4) protocol providing best effort (unreliable) data transmission.

VoATM Voice over ATM

The use of an ATM network to carry voice.

VoIP Voice over IP

IP telephony, or the transmission of realtime audio over an IP network.

WAN Wide Area Network

A communications network that covers a wide geographic area.

WG Working Group

Used within the context of standards forums to refer to a team of individuals working on addressing a specific problem domain.

Chapter 1

Introduction

NGN (Next Generation Network) systems are an evolution of, or in certain scenarios a revolutionary transformation of, the PSTN (Public Switched Telephony Network). NGN evolves the PSTN model towards packet-based converged multimedia communications. Figure 1 lists some of the key attributes of the NGN domain, as compared to the PSTN.

This thesis proposes modifications and extensions to existing NGN architectural models to address dependability and extensibility of NGN communication services. The NGN models, as previously defined, address basic call establishment quite well. The existing NGN architecture and protocols, however, do not address the services domain as well as they do the call setup domain, in that they do not provide a model to allow the creation of advanced NGN services in an extensible and reliable fashion. The types of services that are being referred to are enhancements and extensions to traditional telecom features that benefit from the convergence of voice and data, such as distributed call centre applications, network-based call logging, internet call waiting, and dynamic call redirection.

This thesis addresses extensibility of NGN services by providing a solution at the application layer, and addresses the dependability attribute with a solution at the transport layer. These two attributes (extensibility and dependability for the services domain) are addressed in separate chapters to come, followed by a chapter outlining a simulation model that was coded up and used to verify the proposed approach.

Figure 1 - NGN and PSTN Attributes Compared

1.1 Definitions and Assumptions

The term NGN (Next Generation Network) is used to refer to the transformation of the PSTN into a new converged packet-based multi-service network.

The terms MGC (Media Gateway Controller), Communication Server (CS), and Call Server, are used interchangeably to refer to the entity providing basic telephony and session establishment services. To use IN (Intelligent Network) terminology, this is the entity that runs the BCSM (basic call state machine).

The term Application Server (AS) is used to refer to the implementation of a functional entity that works in conjunction with the Communication Server to provide value-added services. Unlike the traditional PSTN Intelligent Network (IN) model, the use of the term Application Server here need not imply that it is always acting as a master to the Communication Servers.

This thesis enhances the domain of next generation communication systems. There are various protocols that come in to play in NGN architectures, some of which overlap in their applicability. As with many engineering problems, there are situations where multiple technologies exist to address the same problem space. One such example where multiple protocols exist is in the IP telephony signalling space, where both H.323 and SIP are seen as valid contenders. Although both are valid, only one is chosen to focus most of the discussions of this thesis. This is done to limit the scope of the thesis into a manageable size, and is not an indication that one technology is superior to the other.

Cases where assumptions are made to narrow the scope are listed below:

- Packet technology is assumed to be IP, although other technologies, particularly ATM also benefit from the guiding principles highlighted in this thesis.
- The domain of communications is assumed to be that of a public telecommunications carrier network as opposed to an enterprise network. Once again, the concepts should apply to enterprise environments equally well.
- The IP telephony signalling protocol between communication servers is assumed to be based on SIP. As mentioned earlier, this is by no means an indication of the superiority of SIP relative to H.323, but is a simplifying assumption for this thesis. It should be noted that there are other protocols with equal validity in this space, such as BICC, which extends classical ISUP protocols into the packet domain where the bearer path is decoupled from the communication servers.

- The gateway control protocol used between the communication server and the media gateways is assumed to be ITU-T H.248, which is identical to the IETF MEGACO protocol. Other gateway control protocols, such as MGCP (which is not currently on a standards track), the packet cable NCS protocol proposed by CableLabs, and many predecessor protocols to H.248, are not addressed in any detail.

1.2 Thesis Recommendations

There is a lot of research and industry energy being put forth to address various components of next generation communication systems, but not as much energy is being put forth to piece it all together. The research leading to this thesis has taken a broad look at the space of packet-based next generation communication networks, and made recommendations that are meant to enable the design of such networks that are:

- Scalable from thousands to millions of subscribers;
- Extensible to support the creation and modification of service logic;
- Simpler to design, maintain and manage.

The following guiding principles or recommendations are proposed to address enhancements to the NGN architecture to address the attributes just mentioned above:

- R01 - Limit transcodings to access gateways;
- R02 - Support bearer path multicasting;
- R03 - Decouple session control from media;
- R04 - Flatten the control architecture;
- R05 - Support standard communication state machines.

In addition to the above general principles (that will not be addressed in detail), two specific recommendations form the basis of the thesis contributions to come:

- R06 - Use a flexible and extensible application protocol for NGN services;
- R07 - Use partially ordered transport for dependable delivery of service messages.

The relationship between the proposed recommendations and the overall NGN functional domains is presented in Figure 2. The bolded portion of the diagram shows the services-related recommendations, which form the basis for the thesis contributions.

Figure 2 - Thesis Recommendations

Whereas much of the current and emerging research is addressing the bearer path (first audio then video streams) in NGN, less research has focussed on the requirements for advanced NGN services. The final two specific recommendations, R06 and R07, are in the space of service-related messaging between communication servers and application servers in the NGN architecture, an area believe to be of equal importance as the study of the control and bearer domains.

To demonstrate the importance of NGN service-related messaging, consider a scenario where a network is engineered to use a scheme such as MPLS to achieve reliable QoS for the bearer traffic over an IP packet network. If the service-related messaging information is not carried in a deterministically reliable fashion, a request to modify the bearer path as a result of an enhanced service would not be able to proceed, resulting in a reduced class of service to the end user. As well, if the service-related messaging information is not flexible or extensible, as new types of services are thought of, the network architecture would continually need new frameworks to support such services, potentially leading to a service interaction nightmare.

1.3 Thesis Objectives

This thesis addresses the problem of designing extensible and dependable NGN services. The objective is to make specific proposals to improve the NGN architecture and protocols as previously defined in the area of communication services and applications. This is achieved by providing contributions that specifically address the extensibility and dependability attributes in the NGN services domain.

1.4 Thesis Contributions

The contributions of this thesis relate to the final two recommendations (listed in section 1.2) in the area of NGN services: recommendation R06 that deals with extensibility at the application layer, and R07 that is related to dependability at the transport layer. The thesis contributions are expanded on in the chapters to follow, and are summarized below:

- Decouple the existing SIP protocol into a reusable base session protocol, separate from any upper-layer application layer protocol that handles communication sessions involving end-users.

- Define a new protocol, referred to as SIP-S (SIP for services), which sits at the application layer, on top of the decoupled base SIP protocol.
- Define a mapping between TCAP operations and this newly defined SIP-S protocol, thereby allowing traditional PSTN services (based on TCAP and AIN / IN) to be provided in NGN as a starting point, and then extended as required.
- In addition to being able to use TCP and UDP transport, enhance SIP to be able to use SCTP as an underlying reliable transport. SCTP, which is classified here as a partially ordered reliable transport protocol. This approach is shown to have better deterministic behaviour when compared to TCP. Service-related messaging between Communication and Application Servers benefits from this deterministic behaviour.

1.5 Thesis Organization

The structure of this thesis is as follows:

- Background on relevant technology and prior research approaches is presented in Chapter 2.

The contributions of the thesis are then detailed in the next two chapters:

- Chapter 3 details the proposal to address extensibility of services at the application layer.
- Chapter 4 gives details of the contribution to address dependability at the transport layer.
- Chapter 5 discusses a simulation designed to verify the approach proposed in this thesis.

- Chapter 6 summarizes the overall thesis recommendations mentioned in section 1.2, including the two specific recommendations in the services area which form the basis of the thesis contributions.
- Chapter 7 concludes the thesis, and presents material for further study.

Chapter 2

Background

2.1 Relevant Technology

This section highlights relevant standards and technologies that are referenced in the later sections. These technologies exemplify the current state of the industry and are used as a starting point for this thesis.

2.1.1 Emerging Packet Telephony Models

Today's telecommunication networks are being transformed into next generation networks, capable of handling multimedia traffic in addition to voice. The key standards forums where this transformation is being addressed are the IETF (Internet Engineering Task Force) and the ITU-T (Telecommunications Sector of the International Telecommunications Union).

In IETF, three specific working groups are addressing NGN protocols for use in the process of call establishment in this new network: MEGACO (Media Gateway Control), SIGTRAN (Signalling Transport) and SIP (Session Initiation Protocol). There are, of

course, various other working groups in the IETF that are addressing other important aspects of communication networks. However, the three mentioned here are closest to the topic addressed by this thesis, as they deal with the application and transport protocols being cited to have greatest impact from a network evolution standpoint. Other groups address security, quality of service, addressing schemes, mobility, and so on. These areas are of great importance but will not be discussed in great detail here.

In the ITU-T, three study groups are working in the space of next generation network services: SG11 (signalling requirements and protocols), SG13 (multi-protocol and IP-based networks and interworking), and SG16 (multimedia services, systems and terminals).

The standards bodies have proposed various functional models of the entities involved in next generation networks, to go beyond voice telephony into multimedia communications. These architectural models will be discussed as background to the thesis contributions.

The IETF working groups have identified a functional model of a gateway, decomposing it into the following: a Signalling Gateway (SG), a Media Gateway Controller (MGC), and a Media Gateway (MG), as shown in Figure 3. Specific working groups in the IETF have defined protocols that interconnect these functional entities in order to allow multi-vendor interoperability. The SIGTRAN working group (WG) addresses the SG-MGC protocol; the MEGACO working group addresses the MGC-MG protocol. This model is sufficient for representing what can be referred to as pure interconnect gateways. The word "pure" is used here in the sense that the gateway merely provides a bridge from the TDM domain with its PSTN protocols, to the IP (or ATM) domain with its data-centric packet-based protocols. RAS (remote access servers) / NAS (network access servers) and simple toll bypass VoIP / VoATM telephony servers are examples of pure gateways that are well served by the simple SG-MGC-MG model.

Figure 3 - IETF Packet Telephony Model

2.1.2 IP Telephony Signalling

There are currently three industry-accepted standards for packet telephony call signalling: H.323, SIP-T and BICC. Each will be described briefly in the subsections to follow. Call signalling is taken to mean the ability to establish (initiate and accept), tear down, and control a communications session or telephony call.

BICC Overview

BICC (Bearer Independent Call Control) is an evolution of ISUP (the SS7 call signalling protocol of the PSTN). It aims to isolate bearer related information from the rest of the information elements so that the same protocol may be used regardless of what network fabric is used to establish the bearer path (TDM, ATM or IP). The initial version of BICC

is focused on ATM (see ITU-T recommendations Q.1901 and Q.765), although research is underway in ITU-T SG11 to extend the work to IP networks.

SIP and SIP-T Overview

Submitted as IETF standards-track RFC 2543 mid-1999, SIP (Session Initiation Protocol) [17] has been widely talked about as an enabling protocol for a new suite of IP-based services. SIP-T (SIP for Telephony) is an application of SIP for telephony call signalling, and allows for the interworking of NGN Communication Servers with PSTN switching nodes. SIP-T defines an ISUP MIME type that allows ISUP messages to be carried within a SIP envelope without losing contextual information, thereby allowing PSTN-to-PSTN calls to transit an IP "tandem" network. There are two defined mechanisms for call signalling within SIP-T:

- The first allows (binary-encoded) ISUP message intact within SIP;
- The second defines interworking between ISUP and SIP parameters.

Both SIP-T and BICC are evolving to be agnostic of network fabric (IP or ATM), although they were initially designed with a specific fabric in mind (SIP-T on IP and BICC on ATM). Unlike BICC, which uses a carrier's SS7 network to carry signalling, SIP-T has no reliance on SS7 transport.

H.323 Overview

H.323 is an ITU-T specification for multimedia communication over a packet network without guaranteed QoS (quality of service) [24]. It includes a call signalling protocol (recommendation H.225.0), and an integrated framework making use of a variety of protocols (as shown in Figure 4). ITU-T SG16 initially standardized H.323 in 1996, and revisions are still under development (currently work is progressing on H.323v4). Recent ITU-T work has begun to also address supplementary services (in the H.450 series of recommendations).

The network intelligence node in the H.323 model is referred to as a Gatekeeper, and performs much of the same functionality as the MGC in recent IETF models of IP telephony. Whereas H.323 has achieved widespread deployment in enterprise networks, it has not been favoured for carrier service provider network configurations.

Audio Applications	Video Applications	Terminal Control and Management				Data Applications
G.711 G.722 G.723.1 G.728 G.729	H.261 H.263	RTCP	H.225.0 RAS Channel	H.225.0 Call Signaling Channel	H.245 Control Channel	T.124
RTP				X.224 Class 0		T.125
Unreliable Transport (UDP)				Reliable Transport (TCP)		T.123
Network Layer (IP)						
Link Layer (IEEE 802.3) or AAL5 (I.363.5) for VoATM						
Physical Layer (IEEE 802.3) or ATM (I.361)						

Figure 4 - H.323 Protocol Framework

2.1.3 Gateway Control

Whereas protocols like BICC, SIP-T and H.323 address call signalling between communication servers and end user agents, gateway control protocols address signalling between a communication server and media gateways. There are many variations of gateway control protocols in emerging carrier products. From a standards perspective, the IETF and ITU-T have both agreed to endorse the same standard: MEGACO / H.248. MEGACO is the IETF name for the protocol (as well as for the working group responsible for its definition); H.248 is the ITU-T designation for (literally) the same protocol [23]. MEGACO / H.248 can be used as a gateway control protocol alongside with all of the IP signalling protocols discussed in the previous section.

The MEGACO / H.248 protocol is arguably the most advanced of the gateway control protocols in existence today. Other gateway control protocols existed prior to the standardization of MEGACO / H.248, as shown in Figure 5.

Figure 5 - Evolution of Gateway Control Protocols

The basic functions of a gateway control protocol such as H.248 are as follows:

- Directing media flows across media gateways;
- Controlling media processing functions (such as transport adaptation, bearer encoding / decoding / transcoding, mixing of the bearer path);
- Relaying bearer-related events from the media gateway to the media gateway controller, and signals in the reverse direction.

2.1.4 SCTP Transport

The Stream Control Transmission Protocol (SCTP) is a new IP transport protocol, existing at the same level as UDP and TCP. The protocol is a by-product of the research and work of the IETF SIGTRAN working group [15], and has just recently received a standards-track designation as RFC 2960 [16]. This protocol is proposed in this thesis to be the ideal transport protocol for communications signalling needed for providing advanced NGN services.

SCTP shares some common characteristics with TCP:

- It provides a reliable transport service, ensuring that data transport across the network is accomplished error-free.
- It is connection-oriented, establishing a relationship between the endpoints of a session before the data is exchanged, and maintaining this relationship until completion of the transmission.

SCTP is classified by the author as a partially ordered reliable transport scheme, in that it only attempts to ensure the sequencing of related flows, but does not attempt to order all of the data. This partial ordering will later be shown to have beneficial impact on end-to-end performance and deterministic system behaviour. A key feature of SCTP known as multi-streaming enables partial ordering. This feature allows data to be partitioned into multiple streams that can be delivered independently of each other. Any message loss in one stream is independent of flows in other streams. By contrast, TCP utilizes a single stream of data, ensuring that delivery of the stream occurs with perfect sequence preservation. SCTP only concerns itself with order preservation within a stream.

It is worthwhile to note that SCTP has other attributes that make it suitable for messaging transport in transaction-oriented systems. Summarized below are the key attributes of this transport protocol.

- **Multi-streaming capability:** transmitting messages and processing received messages in one stream independent to the order and status of other streams. This allows communication applications to avoid head-of-line blocking by transmitting unrelated transactions using different streams.

- **Security:** an encryption cookie mechanism is built into SCTP to protect against the equivalent of TCP denial-of-service attacks between communicating entities using SCTP.
- **Network Congestion:** SCTP provides mechanisms to detect and manage network congestion.
- **Multi-homing:** SCTP supports multi-homed IP nodes, allowing load balancing, and faster fail-over / switch-over mechanisms.

2.1.5 Advanced Intelligent Network Concepts

The PSTN has undergone several generations of transformations to reach its current state:

- The migration from analog to digital interfaces;
- The separation of signalling from transport with the introduction of SS7;
- The separation of call setup from services with the introduction of the Advanced Intelligent Network (AIN) architecture.

AIN introduced various concepts and paradigms, some of which are important to carry forward into NGN networks, others of which are not as critical. Those believed to be important to carry forward are listed here:

- Separation in modelling of call processing from enhanced services;
- Definition of a standard call model for call processing;
- Separation of originating and terminating "half-call" models.

AIN introduced the concept of a standard call model, essentially a finite state machine, for describing the flow of call processing operations that occur within a switching centre. This model is referred to as the Basic Call Model (BCM), or the Basic

Call State Machine (BCSM). The BCSM is comprised of various states known as Points In Call (PIC), and Detection Points (DP) which are essentially transition points (between the states) at which the finite state machine may be extended. The definition of a standard state machine for call processing and session establishment is an important fundamental concept to be carried forward. The exact definition of the call processing states need not be identical to that of existing PSTN AIN / IN call models.

AIN separated, or decoupled the process of initiating a communications session from that of accepting (or rejecting) an incoming request to communicate. This manifested itself in the definition of two basic call models, essentially a pair of communicating state machines:

- An Originating BCM for specifying the process of initiating a call by a Calling Agent;
- A Terminating BCM for specifying the process by which a Called Agent accepts (or rejects) incoming requests.

Included in Appendix A are the widely accepted versions of the AIN / IN basic call models. The North American AIN and International IN variants of the originating basic call model are shown in Figure A1 and Figure A2 respectively; similarly, the accepted variants of the terminating basic call model are given in Figure A3 and Figure A4. As can be seen the diagrams in this appendix, the North American [21] and International [22] versions are quite similar in definitions of functionality.

The simulation model described in Chapter 5 makes use of an analogy between the PSTN AIN domain and the NGN model. This analogy allows us to use observed AIN message flows from the existing PSTN as a basis for the traffic patterns used in the simulation.

2.2 State of Related Research

2.2.1 Research on Applications of SIP

SIP was initially proposed for the setup of communication sessions directly between end user agents. While this model is appropriate where the endpoints involved in the session are intelligent, other models were required to handle thin clients. SIP was later enhanced with the introduction of proxy and redirect servers that may intervene to provide enhanced services on behalf of end users. More recently, SIP-T was introduced as a vehicle for allowing call setup messages to transit between communication servers that serve different domains.

As will be described in Chapter 3, this thesis proposes a new usage of SIP for services, extending its previous role for call signalling. Recent research has studied the way SIP-based networks can access PSTN SCP-based IN services [19], essentially via a signalling gateway in the SS7 network. However, this previous research did not propose a new SIP-based protocol for extensible services, as will be discussed later in this thesis.

Other studies, such as the work presented in [20], have looked at a specific service implementation using SIP technology. By contrast, this thesis provides a general framework for implementing a wide range of existing telecommunication services using a SIP-based protocol for application layer messaging between communication servers and application servers, enabling a whole host of services.

Figure 6 shows the existing SIP protocol domains, showing SIP signalling between peer network elements, and between network elements and end-user agents. This model will be enhanced as part of the thesis contribution presented in Chapter 3.

Figure 6 - SIP Protocol Domains

The network entities shown are explained below:

- UA is an end-user communication user agent endpoint that may physically be manifested as any one of the following: a software entity running on a PC or workstation such as a softphone application, SIP user agent software, or a java-based communication applet.
- CS is a network element acting as a communication server, call server, call agent, media gateway controller, or a SIP translation / registrar / proxy / redirect server.

2.2.2 Research on Transport Protocols and Signalling Domains

Partial Ordered Transport and Reliability

Previous research work presented in reference [9] looks at partially reliable transport services for the media (bearer) stream, and uses an analytic model to demonstrate the benefits of partial reliability over fully reliable transport (particularly TCP), in terms of reduced network delay and increased throughput for lossy networks. By contrast, the work in this thesis (detailed in Chapter 4) looks at a specific partially ordered reliable transport service, SCTP, not for media stream, but for application messaging required for

enhanced communication services. Instead of examining metrics like network delay and throughput, we examine parameters related to telecommunication applications such as transaction response time and the number of completed successful transactions. This work provides a complementary view to that presented in reference [9], looking at the applicability of partially ordered reliable transport mechanisms under lossless as opposed to lossy conditions (that is for cases where it is unacceptable to lose portions of an information flow). Whereas reference [9] presented an analytical model for comparing fully and partially reliable transport schemes for media (bearer) streams, this thesis presents a simulation framework for comparing (fully and partially ordered) reliable transport schemes for signalling.

Several research studies have concentrated on the effect of transport layer protocols on the QoS parameters associated with the streaming of audio and video content. The work in reference [7], and related work in reference [10], address the problem space from a point of view of comparing TCP and UDP to alternative transport schemes; other early studies such as reference [11] looked at the problem space from the perspective of using CORBA between heterogeneous computing environments. More recently, related work in reference [12] proposes a protocol for the setup of VoIP bearer streams. While the transport of voice and video bearer streams is clearly an important area in the development of reliable and deterministic next generation networks, there is an equally important area of concern: the reliability of the associated feature-related messages. This thesis focuses on this latter area, but also builds on research findings in the related area of partially reliable transport for multimedia bearer streams, particularly in the area of testing methodologies.

The schemes mentioned above that have been discussed in prior research deal with Partial Reliability, whereas this thesis focuses on Partial Ordered Reliability. Both of these approaches aim to increase determinism by loosening reliability. The spectrum of reliability for IP transport schemes can be summarized at a high level as follows:

Fully Reliable (such as TCP)

> Partial Ordered Reliability (such as SCTP)

> Partial Reliability (various)

> Unreliable (such as UDP).

Previous to the availability of SCTP, prior research referred to in reference [5] has examined the motivations behind partial ordered reliability, and had proposed a general specification model for a partial-ordered transport service. By contrast, the work here looks at a specific partially ordered transport service (SCTP), and its applicability in the NGN services protocol domain.

Improving Performance Relative to TCP

In this thesis, we propose using SCTP instead of TCP for transport of service-related messages. This alternative protocol (SCTP) is reliable, but only partially ordered, thereby achieving an increased level of predictability, at the expense of the relaxed ordering requirement.

Recent research documented in reference [2] supports the claim that a certain class of applications can benefit from a more predictable transport scheme relative to TCP. This prior research looks at the dynamics of TCP, and proposes modifications to the congestion control mechanisms of TCP, and extensions to existing routing algorithms, that lead to better predictability. By contrast, the work in this thesis focuses on the use of an alternative to TCP instead of modifications to it.

Other research mentioned in reference [4] has looked at characterizing HTTP traffic over IP networks using different transport protocols, looking for opportunities to improve performance. The analysis of HTTP traffic is of interest to the work here, simply because SIP, which is central to the services theme of this thesis, has similarities to HTTP. The work in reference [4] finds that there is room for HTTP performance improvements over

high-speed WAN connections, supporting the proposed usage of a new transport protocol for carrying SIP-based messages, to gain such improvements relative to using TCP.

Other recent research has looked at improving performance of packet technologies more generally. As an example, the work on IP/ATM switching systems cited in reference [3] proposes a hybrid approach utilizing both software and hardware-based routing techniques. This type of research is seen to be complementary to the transport approach proposed in this thesis for carrying service-related messages.

Testing Transport Protocols

In recent research, reference [8] presents a test environment to allow upper layer applications to use a single application programming interface (API) to test different underlying transport protocols. In this thesis, the simulation portion follows a similar approach. A uniform API is presented to the application layer entities regardless of underlying transport protocols being used in the simulation.

The work in this thesis examines the effect of transport protocols on application-layer service-related parameters; the work in reference [8] is more focussed on the effect of transport protocols on multimedia bearer streams.

2.2.3 SCTP Transport Applications in NGN

Figure 7 - NGN Protocol Domains showing Prior SCTP Usage

Figure 7 depicts the recently accepted model for decomposition of NGN network components, based on what is described in reference [14], with shading added to show the existing domains of SCTP usage. The MGC is the network element responsible for call processing. The MGC controls several media gateways that terminate physical line and trunk interfaces; the MGC also controls a signalling gateway for sending control messages towards the PSTN. For calls placed to or from resources outside of the control of a specific MGC, inter-MGC call signalling is used. Existing areas where SCTP is used are as follows:

- SCTP used to allow an MGC to communicate with its Signalling Gateway for call setup between the PSTN and NGN domains;

- SCTP used to allow an MGC to control its Media Gateways (based on recent research findings).

Prior research has already shown that the signalling protocol domain labelled "1" (call signalling to a signalling gateway) benefits from partially ordered reliable transport [16]. In fact, SCTP was designed specifically to satisfy the signalling transport requirements of this interface.

Recent IEEE papers and IETF research (as mentioned in reference [14]) are looking at the applicability of SCTP transport for the MEGACO / H.248 signalling protocol domain labelled "2" (media gateway control), as an alternative to using TCP and UDP with application layer framing for this interface. The use of TCP or UDP as the MEGACO / H.248 transport protocol is described in Annex D of H.248. ITU-T SG16 has recently added an Annex to explore the applicability of SCTP for interface "2", as a transport protocol for MEGACO / H.248. This Annex H to ITU-T H.248 was added with a note that this is "proposed text for decision"; there is also a related IETF informational RFC [18] issued summarizing research on the same topic.

By contrast to the existing and prior research just described, this thesis proposes that SCTP usage be expanded into the services messaging domain, as will be described in Chapter 4.

Impact of SCTP on Other Network Traffic

Recent research [6] has examined the impact of streams of SCTP traffic over an IP network carrying concurrent TCP flows, and has concluded favourably for SCTP. This recent study has shown that SCTP has the same incremental effect on other TCP traffic as TCP itself, meaning that SCTP does not cause extra burden on concurrent TCP traffic (no more so than TCP itself would do). By contrast to this finding, which indeed is important and relevant, the work presented in this thesis compares TCP and SCTP as underlying

transport mechanisms for carrying service-related messaging flows between communication servers and application servers.

Chapter 3

Extensibility at the Application Layer

This thesis proposes an enhanced architectural model relative to the existing NGN architecture (as previously defined in packet telephony standards forums and research), showing extensions to address extensible communication services. Missing in the IETF packet telephony model (discussed earlier) is the definition of application level entities within the packet network that enable communication services to be created in a scalable and extensible fashion. The term communication services, as used here, is not referring to the lower layer functions of network translations and routing. The communication services being referred to are the upper layer functions of session establishment, call processing, media resource usage, and supplementary services such as multiparty conferencing, call redirection, call logging, directory-based call establishment, and a host of yet-to-be-defined additional multimedia services. Such multimedia services may, in certain cases, start off being similar to related PSTN network services, but have extensions to make use of the enhanced network capabilities, flexibility and ease-of-use of an IP-based network.

In order to go beyond the simplistic packet telephony model of providing basis service, to one that can handle more complex forms of multimedia services and service interactions, this thesis proposes an extended architectural model to include an

Application Server in addition to the Communication Server and Gateways. The Application Server provides Communication Servers with access to enhanced service logic, using a new SIP-based protocol being proposed in this thesis.

3.1 SIP Modifications for Service Extensibility

A new proposed usage of SIP is presented in this thesis for message exchange between communication servers and enhanced application servers. The same generic SIP envelope is used for containing service-related requests and replies between these entities. A new method is also presented to map TCAP-style service-related messaging into SIP messages and parameters to allow parity with the existing PSTN network services (without inheriting the limitations of PSTN SS7 transport).

The following contribution is proposed in this thesis to address service extensibility:

- Begin with the existing SIP protocol, which is extensible by design, but not specifically in the service domain.
- Decouple the portion of SIP that deals with generic establishment of a session, from the upper-layer portion of the protocol that is application-specific (previously only defined for call setup).
- Rename the upper-layer call-control portion of SIP to SIP-U (SIP for User Agents) to distinguish it from the base SIP component that is reusable in different application domains.
- Define a new application layer protocol, SIP-S (SIP for Services), to ride on top of the decoupled base SIP protocol, and be used for extensible messaging between a Communication Server and an Application Server.

- Provide a mapping between existing PSTN TCAP service messages to this new SIP-S protocol, thereby allowing the existing PSTN service suite (such as AIN) to be available in NGN using SIP-S.

Figure 8 shows the previous SIP model and the extensions to the SIP protocol stack that is proposed in this contribution. Also shown is the new Application Server entity that interacts with Communication Servers using the new SIP-S protocol.

Figure 8 - Proposed Extensions to SIP Protocol Stack

Prior research has led to the specification of the two interfaces labelled SIP and SIP-T in the previous model; this thesis proposes the new interface labelled SIP-S between communication servers and application servers. Please note that this same interface may also be expanded to go between end-user agents and application servers, but this is not shown for simplicity.

The term "SIP-U" was introduced to refer to the portion of the protocol that had formerly been part of SIP, for communicating with user-agents (when an end user is initiating or accepting a communication session). The interface "SIP-T" is used between communication servers to set up transit sessions. The term "SIP-S", introduced here, is used to specify the SIP interface for services that are housed in an application server.

3.2 SIP-S and AIN Mapping

The SIP-S proposal is related to procedures that have been disclosed by the author of this thesis and a colleague under a recent patent filing [1]. This new usage of SIP to carry service-related information can be modelled after TCAP and AIN parameters.

SIP has been embraced as an NGN protocol with potential to enable a rich set of new services and applications. However, the SIP approaches used thus far (in both industry forums as well as research on the subject) have been somewhat ad-hoc in that they addressed specific services in isolation, lacking a general framework for extensible services, and not learning from decades of progress in PSTN network design.

Proposed here is an approach whereby existing TCAP service-related information elements may be mapped into the content of SIP-S messages, thereby enabling functionally equivalent versions of all existing TCAP-based service applications to operate in NGN architectures, without requiring the use of SS7 transport. Enhancements to the services (that were not foreseen when the service was initially designed) can then be made by adding additional service-related information to the SIP-S messages, as dictated by service needs. The extensibility of SIP as a protocol makes it well suited for service-related messaging exchange. This goes beyond the initial role of SIP as a user-to-user signalling protocol, and beyond its recent expanded role (SIP-T) as a protocol for signalling between transit communication servers.

Existing SIP request messages can be reused in this new context. Instead of initiating a communication session between end-user agents, or between communication servers, these same messages are used to initiate the equivalent of a TCAP transaction between a communication server and an application server. The following is a proposed mapping of SIP messages to the type of service-related information messages they can encapsulate:

- A SIP Invite is used to initiate a service-related transaction (similar to a TCAP Query package type);
- SIP Success (200 OK) messages are used to send back service-related response messages that terminate a transaction (analogous to TCAP Response package types);
- SIP Informational (1xx) response messages are used for intermediate service-related information exchange (analogous to TCAP Conversation package types);
- SIP Error Codes (4xx and 5xx) are used for failure cases, and a SIP BYE is used to abort a service-related transaction (similar to TCAP Abort package types).

This proposed mapping between SIP message types and TCAP package types allows the mapping of existing TCAP-based service implementations to quickly proceed in NGN.

In addition, a subset of existing already-defined SIP fields may be used in this context to refer to data typically housed in the SCCP headers for existing TCAP-based services:

- The "From:" field can be used to contain the originating network node (similar to an SCCP Originating Point-Code), but using more friendly and extensible addressing schemes (such as domain names);
- The "To:" field can be used to house the destination network node's address (similar to an SCCP Destination Point-Code);
- The "Call-ID" field can be used to house the equivalent of a TCAP transaction ID.

Additional fields can be housed in individual SIP-S MIME types corresponding to functional content of TCAP query and reply operations. For example, AIN TCAP query parameters representing the CallingPartyID, CalledPartyID and RedirectingPartyID may simply be mapped to functionally equivalent fields in a text-based MIME type corresponding to the query operation. The information in such fields need not be restricted to traditional telephone numbers, but can be expanded to allow carriage of additional user-friendly aliases. Such a mapping between PSTN AIN / TCAP parameters to SIP-S allows existing services to be supported as is (as a starting point), and to be readily extended by use of additional fields, or by expanding the allowable values that can be contained in existing fields.

To demonstrate the viability of the approach just discussed, a sample message flow is first presented based on a common existing AIN 0.2 service scenario (in Figure 9), followed by the corresponding SIP-based message sequence chart (in Figure 10) for the same service in an NGN architecture using an Application Server.

Figure 9 - AIN 0.2 Service Scenario

Figure 10 - Proposed SIP-S based Message Exchange

In Figure 9, the messages in square brackets correspond to AIN operation types (and corresponding AIN parameters contained therein, based on [21]). In the proposed SIP mapping shown in Figure 10, the corresponding messages map to SIP-S MIME types, containing tags and fields that are functionally equivalent to their AIN counterparts.

The use of SIP allows extensibility in the sense that new operation types and new service parameters may be readily added. The mapping from TCAP operations to SIP allows the existing well-known, deployed services to be taken as a starting point for implementation in the NGN architecture. This effectively expands the role of SIP in NGN networks, allowing it to serve a service-related role in addition to its previous call-related session-establishment function.

Chapter 4

Dependability at the Transport Layer

4.1 Service Dependability

In order to address service dependability at the transport layer, this thesis makes the following contribution: use a partially-ordered reliable transport protocol as the underlying transport mechanism to carry the previously-referenced service messages (SIP-S over SIP), trading off sequencing in exchange for deterministic transport behaviour. The thesis recommends the use of SCTP (an emerging protocol initially designed to handle signalling and control transport between a media gateway controller and its gateways) as the transport scheme for the service-related messaging between a Communication Server and Application Servers. SCTP, which can be referred to as having partial-ordered reliability courtesy of its multi-streaming capability, is proposed to be more appropriate for carrying service-related telecommunications messages than alternative fully reliable transport schemes (TCP). This will be validated using a simulation program to compare upper-layer telecommunications applications using alternative transport schemes, demonstrating the effectiveness of the SCTP approach.

4.2 Transport for Service-Related Messaging

The type of messaging under consideration for dependable transport is that between Communication Servers and Application Servers. The typical scenario is one in which a Communication Server needs to establish a dialogue with an Application Server in order to exchange session-related, service-affecting information. Prior to the availability of SCTP, this could be done using the following lower layer protocol alternatives:

- SS7 - using the MTP layers for reliability, but appropriate in the PSTN not NGN;
- TCP - inherently reliable but lacking in determinism;
- UDP - burdens the applications with addressing reliability (if required).

With the availability of the IETF SIGTRAN working group's SCTP protocol, a more appropriate mechanism is available for providing carrier-grade reliable signalling between Communication Servers and Application Servers in the NGN:

- SCTP - reliable but partially ordered and more deterministic than TCP.

This thesis contribution proposes the use of SCTP instead of TCP for service messaging. The SCTP and TCP approaches are compared from an upper layer application perspective in Chapter 5, using a model of communication server messaging to an application server, showing the value and appropriateness of SCTP for messaging between communication servers and peer application servers as depicted in Figure 11, label "A". This proposed usage of SCTP is in addition to its initial application for MGC-to-SG communication (Figure 12, label "C"), and the recent proposals for MGC-to-MG communication (label "D"), which can loosely be classified as master-slave configurations (the MGC is master of its gateways). Also shown for completeness in Figure 11 (label "B") is an area proposed for further study where SCTP may also be appropriate.

Figure 11 - New SCTP Application Being Explored

Figure 12 - Existing Applications of SCTP

SS7 lower layer protocols (MTP) will not be considered in the analysis as the focus is on emerging NGN architecture, not on the existing PSTN. Within the context of a packet based converged network, while it is appropriate to use SS7 for pure interworking with

existing PSTN switching fabrics, it is more appropriate to use the packet network itself (not the SS7 network) for transport of signalling information between Communication Servers and Application Servers within NGN.

UDP was not considered in the analysis since it is not a reliable transport protocol. Reliability is a mandatory requirement for the transport of communication service messages.

4.3 Impact of SCTP

As mentioned earlier, recent research [6] has demonstrated that the impact of SCTP streams on other TCP traffic over the same underlying network is in fact quite comparable to the impact of similar TCP connections on other TCP traffic over the same network. This effectively shows that SCTP does not negatively affect concurrent reliable traffic on the same network. This thesis contribution complements this earlier finding, by proposing that SCTP be used instead of TCP for transporting communication service-related messages, as the multi-streaming capability of SCTP allows it to yield more deterministic response characteristics for this class of applications. This, taken with the fact that SCTP is also more secure in some regards than TCP (SCTP has an encrypted cookie mechanism that guards against equivalent denial of service attacks on TCP), makes SCTP more suitable than TCP for such services messaging applications. The upcoming chapter compares SCTP and TCP for service messaging using a simulation framework.

4.4 SCTP Interfaces Summarized

Figure 13 shows the NGN end-to-end architecture model, enhanced with the proposal of this thesis contribution, and showing the domains where SCTP transport may apply:

- Interface A is the focus of this contribution, to use SCTP as a transport protocol for service-related messaging between Communication Servers and Application Servers.
- Interface B shows call setup signalling between Communication Servers. The use of SCTP transport for this interface is proposed for further study.
- Interface C shows signalling between an MGC and its Signalling Gateway for call setup. This is the interface for which SCTP was initially defined.
- Interface D shows control of Media Gateways by an MGC. Recent research, as described earlier, has proposed that SCTP be used for this interface.
- Interface E shows enhanced application messaging between Application Servers. The use of SCTP transport for this interface is proposed for further study.

To summarize, the use of SCTP for Interfaces C and D existed prior to this thesis. Interface A is the focus of this contribution, while the use of SCTP transport for Interfaces B and E is proposed for further study. Note that in the diagram to follow, the PSTN portion shown is strictly used for interworking of call control between the PSTN and NGN domains.

Figure 13 - Expanded NGN Protocol Domains Expanded and SCTP

Chapter 5

Verification using Simulation

This chapter details the simulation framework used to verify the proposal made to use SCTP as the transport vehicle for the SIP-based service messages described earlier.

5.1 Experimental Process

In order to compare the SCTP-based approach with the TCP-based one, a series of PC-based (Windows 32-bit) applications were written in C/C++ to alternatively make use of the following stacks for messaging between simulated models of the peer level Communication and Application Server entities:

- A native Windows NT 4.0 TCP/IP stack (Windows Sockets 2.0);
- A new SCTP software stack.

Figure 14 shows the testbed configuration used to simulate application-layer messaging using alternative underlying transport schemes.

Figure 14 - Testbed Configuration

The actual model of these entities will be explained further in the section to follow, but the parameters of the experiment are as follows.

Measurements:

- ΔT_{txn} , the response time as seen by the communication server (delta between initiation of a transaction to an application server, and receipt of the final message in that transaction).
- The percentage of successful transactions for which all of the messages sent by the application server are received by the communication server within a predetermined application layer timer, T_1 ($\forall$ messages in the transaction, $\Delta T_{\text{msg}} \leq T_1$, where ΔT_{msg} is the delta between the time an outgoing request message is sent from a Call Server and the time the incoming reply message is received).

Variables:

- Transport type (SCTP or TCP) is the primary variable examined.
- The available network bandwidth is varied.
- The reliability of the underlying network is varied.

Constants:

- The call arrival rate, and percentage of calls requiring advanced service logic, are kept constant.
- The statistical mix of application layer transaction traffic is kept consistent (please see the next section).

5.2 Messaging Model

In order to perform the analysis from an application perspective, a profile of typical messages exchanged between Communication and Application Servers is needed. By drawing analogies to the PSTN, we can look at existing profiles for messages exchanged between a Service Switching Point (SSP) and a Service Control Point (SCP). Such profiles, based on existing PSTN services, provide a realistic basis for the types of data exchanged between Communication Servers providing session establishment services, and Application Servers providing value-added enhanced services.

This proposed approach to model messaging exchanges between an MGC and an application server based on AIN operations, is consistent with recent proposed approaches for implementing NGN enhanced services and applications, and with the proposal to map AIN operations into SIP messages.

Figure 15 draws parallels between the PSTN architecture and NGN IP-based architecture. In the NGN architecture, the Call Server and Media Gateways are distributed over an IP fabric; in the PSTN architecture, the SSP and its peripherals are tightly coupled and communicate using proprietary protocols. However, from an enhanced services perspective, both architectures allow for application servers to provide value-added communication services.

In the NGN architecture, Call Servers can communicate with Application Servers using IP-based transport, to gain access to enhanced applications and services; in the PSTN, SS7-based transport is used between the SSP and SCP to gain access to enhanced telephony services. It is this analogy that allows us to model Communication Server to Application Server messaging, based on patterns observed in the existing PSTN network for SSP-SCP messaging.

Figure 15 - Analogy between PSTN and NGN Architecture

5.3 Simulation Design

A framework was developed to allow different transport protocols to be compared under simulated upper layer traffic loads. This framework was implemented on Windows NT 4.0 using C/C++, and executed on Intel-based Pentium processor workstations.

The aim of the simulation runs is to compare the SCTP and TCP approaches, from an application layer perspective of response time.

As part of the simulation framework implemented, both "client" and "server" portions of the simulated nodes (Communication and Application Servers) were written, for the different underlying transport schemes. The client portions are representative of a communication server that processes calls and user-initiated communications sessions based on a Poisson (statistically independent and memoryless) distribution. The server portions are representative of an application server that is queried by a communication server for the subset of the communication sessions that require additional service-related functionality.

The following outlines the program logic for the simulation driver and the components that model the communication and application servers:

- A mainline driver emulates call processing by initiating "calls" using an exponential distribution for inter-arrival time.
- Once a call is initiated, a decision is made whether or not the call requires advanced service processing functionality from an application server. A uniform distribution is used to make this decision.
- For calls that require such advanced service processing functionality, a service-related query message needs to be built and sent to the application server. The byte-length of this message is then determined.
- The number and byte-length of the parameters to be sent in the query are determined based on a deterministic and a probabilistic component. The deterministic part is based on mandatory parameters and message headers that are always present; the probabilistic part is based on optional parameters. AIN 0.1 (subset of AIN 0.2 [21])

service profiles observed in existing telecommunication networks were used as a basis for the probabilistic values (details are given in section 5.4).

- Based on the query operation type that had been determined, the corresponding response operation type is also determined using a probability distribution (again, based on observed AIN 0.1 patterns in today's network).
- Once the response operation type is determined, a similar process is followed to determine the size of the corresponding reply message that is to be sent back from the application server to the communication server "client", as was done for the query message (deterministic and probabilistic components).
- At this point, the simulation driver knows the size of the query message to be sent to the application server, and has also determined the size of the corresponding reply message to be sent back by the application server.
- The driver proceeds to send an appropriately sized message through the underlying transport mechanism in place. The size of the reply message to be sent back is included within the query sent (simply to avoid having the server components of the simulation need to worry about such decisions as response message and parameter determination).
- The appropriately modelled application server, running on a different network node, receives the incoming query, and sends back an appropriately sized reply message using the same signalling transport protocol.
- The driver receives this reply, and records the measured response time for the entire transaction (in milliseconds).

The network nodes are interconnected using a direct 100BaseT Ethernet connection, and a software-based WAN emulator, "The Cloud" by Shunra Software, is used to emulate network characteristics (throughput and delay) between the communication and

application servers. A summary of the key capabilities of this network emulator is included in Section 5.5, along with images of representative screen captures.

The communication server is modelled to be multithreaded. As such, the simulation loop creates a new thread to take care of sending the query message and awaiting a reply. This is a realistic modelling of communication servers, but does require that the implementation of the transport protocol stack (the equivalent of `wsock32.lib` for the Windows Sockets 2.0 implementation of TCP/IP and UDP/IP) be thread-safe (or have a framework wrapped around it to make it so).

The same simulation is run using TCP and SCTP as the underlying transport vehicle. For each run, initial measured response time values are discarded to account for any instability at the start of the experiments. Multiple simulation runs are invoked, and the statistical mean and standard deviation values are calculated for the different scenarios considered.

The average call arrival rate was set to 100ms (equivalent to 360K BHCA), and the percentage of calls requiring advanced service functionality was set to 100% (this is higher than the existing PSTN service scenarios, but is believed to be more indicative of NGN service scenarios). The techniques presented in reference [25] were used as guidelines in designing the simulation, and in measuring and analyzing the results.

5.4 Traffic Model Details

Included in this section are tables listing the probability of sending a specific query message type, and the probability of receiving a specific reply message type (for a given query message type). The message distributions are shown graphically in Figure 16. Also included in this section are the detailed lists of parameters for each query and reply message used in the simulation, showing parameter size and probability of inclusion in a

specific message type. The mandatory headers along with a subset of the message parameters are always present in the message, and therefore lead to the deterministic component of the message size. Other parameters have a non-zero inclusion probability that is less than unity. These parameters lead to the probabilistic component of the size of a given message.

Figure 16 - Query and Reply Message Distributions

The tables that follow give the traffic model parameters used in the simulation (based on the existing PSTN AIN 0.1 messages). Table 1 shows the query message distribution; Table 2 shows the distribution of reply messages for a given query message type. These query and reply messages represent messages sent from the Application Server and Communication Server, respectively. Each query and reply message contains a series of parameters, some of which are always present, whereas the presence of others is based on a uniform probability distribution. Listed in Table 3 and Table 4, respectively, are the query and reply parameters, with their byte size and probability of inclusion in each message type. Blank entries in the tables indicate that a parameter is invalid (never included) in a specific message; other entries with zero probability indicate that a parameter is valid in a message, but is assumed never to be present (based on data gathered from existing networks).

Q_M = Query Message		Probability ($Q = Q_M$)
Q_1	Orig_Attempt	0.15
Q_2	Info_Collected	0.10
Q_3	Info_Analyzed	0.35
Q_4	Network_Busy	0.05
Q_5	Term_Attempt	0.35

Table 1 - Query Message Distribution

R_N = Reply Message		Probability ($R = R_N \mid Q = Q_M$)				
$\downarrow$	Q_M = Query Message $\rightarrow$	Q_1	Q_2	Q_3	Q_4	Q_5
R_1	Analyze_Route	0.55	0.35	0.55	0.85	
R_2	Continue			0.20		
R_3	Disconnect	0.15	0.10	0.05	0.05	0.05
R_4	Authorize_Term					0.20
R_5	Forward_Call					0.55
R_6	Send_to_Resource Final	0.30	0.55	0.20	0.10	0.20

Table 2 - Reply Message Distribution (for a given query message)

Query Parameter	id	byte size	Probability of Inclusion in Query Message				
			Q ₁	Q ₂	Q ₃	Q ₄	Q ₅
Mandatory Header		46	1.000	1.000	1.000	1.000	1.000
Q_P_AccessCode	1	70		0.020	0.020		
Q_P_AcgEncountered	2	38	0.002	0.002	0.002	0.002	0.002
Q_P_Amp1	3	46	0.002	0.002	0.002	0.002	0.002
Q_P_BearerCapability	4	38	1.000	1.000	1.000	1.000	1.000
Q_P_CalledPartyId	5	72			1.000	1.000	1.000
Q_P_CalledPartyStationType	6	38					0.700
Q_P_CallingPartyBgid	7	72			0.020		
Q_P_CallingPartyId	8	72	1.000	1.000	1.000	0.500	
Q_P_Carrier	9	44	0.500	1.000	0.020	0.020	0.400
Q_P_ChargeNumber	10	72	0.400	0.300	0.100	1.000	0.400
Q_P_ChargePartyStationType	11	38	0.350	0.300	0.100	0.200	0.350
Q_P_ClearCause	12	38					
Q_P_CollectedAddressInfo	13	72		0.900	0.020		
Q_P_CollectedDigits	14	70		0.080	0.020		
Q_P_FailureCause	15	38					
Q_P_GenericName	16	36					0.080
Q_P_Lata	17	70	0.200	0.200	0.200	0.800	0.400
Q_P_OriginalCalledPartyId	18	72		0.020	0.020	0.020	0.020
Q_P_RedirectingPartyId	19	72		0.020	0.020	0.020	0.020
Q_P_RedirectionInformation	20	40		0.020	0.020	0.020	0.020
Q_P_Tcm	21	70		0.010	0.010	0.010	0.010
Q_P_TriggerCriteriaType	22	38	1.000	1.000	1.000	1.000	1.000
Q_P_UserId	23	90	1.000	1.000	1.000	1.000	1.000

Table 3 - Query Parameters (size and probability of inclusion in specific messages)

Reply Parameter	id	byte size	Probability of Inclusion in Reply Message					
			R1	R2	R3	R4	R5	R6
Mandatory Header		46	1.000	1.000	1.000	1.000	1.000	1.000
R_P_AlternateBillingInd	1	44						
R_P_AlternateCarrier	2	44	0.040				0.040	
R_P_AlternateTrunkGroup	3	50	0.040				0.040	
R_P_AmaAltBillNumber	4	72	0.001					
R_P_AmaBusinessCustomerId	5	72	0.005	0.005	0.005	0.005	0.005	
R_P_AmaDigitsDialedWc	6	70	0.010	0.010	0.010	0.010	0.010	
R_P_AmaDigitsDialedWc2	7	70						
R_P_AmaDigitsDialedWc3	8	70						
R_P_AmaDigitsDialedWc4	9	70						
R_P_AmaDigitsDialedWc5	10	70						
R_P_AmaLineNumber	11	72						
R_P_AmaLineNumber2	12	72						
R_P_AmaSlpId	13	48	0.040	0.040	0.040	0.040	0.040	
R_P_Amp1	14	46						
R_P_AnswerIndicator	15	36						
R_P_CalledPartyId	16	72	0.600				0.050	
R_P_CallingPartyId	17	72	0.030				0.050	
R_P_Carrier	18	44	0.200				0.200	
R_P_ChargeNumber	19	72	0.020				0.030	
R_P_ChargePartyStationType	20	38	0.020				0.020	
R_P_CollectedDigits	21	70						
R_P_ControllingLegTreatment	22	38				0.050		
R_P_DisconnectFlag	23	36						1.000
R_P_DisplayText	24	552				0.050		
R_P_OutpulseNumber	25	70	0.100				0.100	
R_P_OverflowBillingInd	26	44						
R_P_PassiveLegTreatment	27	38	0.010				0.010	
R_P_PrimaryBillingInd	28	44	0.005	0.005	0.005	0.005	0.005	
R_P_PrimaryTrunkGroup	29	50	0.200				0.200	
R_P_ReducingPartyId	30	72	0.020					
R_P_RedirectionInformation	31	40						
R_P_ResourceType	32	38						1.000
R_P_SecondAlternateBillingInd	33	44						
R_P_SecondAlternateCarrier	34	44	0.008				0.008	
R_P_SecondAlternateTrunkGroup	35	50	0.008				0.008	
R_P_StrParameterBlock	36	2856						1.000
R_P_Tcm	37	70	0.010			0.010	0.010	

Table 4 - Reply Parameters (size and probability of inclusion in specific messages)

5.5 Network Emulator

A software-based network emulator known as The Cloud¹ was used to model network conditions during the simulation. This software-based emulator runs on Windows NT 4.0, and has low host CPU utilization.

The software allows emulation of the following parameters:

- Bandwidth (symmetrical or asymmetrical);
- Latency (fixed or probabilistic);
- Packet Loss, Out of Order and Duplicated Packets;
- Bit Error Rate, Link Disconnect;
- IP Fragmentation;
- Network Congestion.

More information about this product can be obtained from the vendor's website at www.shunra.com. Representative screen shots of the software are included in the figures to follow.

The alternative to using a software-based emulator such as the Shunra product would be to use a hardware-based network emulation system. The hardware approaches are more complex to configure, but are appropriate for use when a high degree of precision is required (100 μ s instead of 10-100ms accuracy). For the simulation model used in this thesis, the software-based approach was adequate.

¹ "The Cloud" is a WAN emulation software package produced by Shunra Software Ltd.

Figure 17 - WAN Emulation Program "The Cloud" by Shunra Software

Figure 18 - WAN Emulation Program showing Network Traffic

5.6 Simulation Runs

Representative simulation runs were executed using the WAN emulator to model different signalling channel sizes, and to inject delay into the signalling path. The bandwidth of the signalling channel was set to 64kbps (DS0), 128kbps (2xDS0) and 1.544Mbps (T1 capacity). A normal distribution was used to model delay on the channel,

using mean delays of 200ms and 100ms for the different runs, and a standard deviation of 40ms and 20ms, respectively.

For each of the representative network conditions emulated, 10 separate traffic runs were completed for both TCP and SCTP transport. The random number generator used in the simulation program was seeded with different values for the different runs. The results of the TCP and SCTP runs are shown in the tables that follow.

BW, Delay	DS0, 200ms		2xDS0, 200ms		T1, 200ms		2xDS0, 100ms	
RUN	sample mean	std dev	sample mean	std dev	sample mean	std dev	sample mean	std dev
01	1234.28	418.12	1093.53	307.76	969.65	206.56	618.10	216.28
02	1214.55	404.14	1083.78	303.21	956.21	203.29	623.31	213.83
03	1213.54	399.96	1080.24	315.12	954.72	206.82	615.74	213.53
04	1299.49	447.13	1138.27	333.24	989.16	224.37	654.42	231.33
05	1222.17	408.79	1081.96	313.53	952.29	207.25	612.49	212.50
06	1249.03	419.63	1094.30	315.86	959.47	200.70	625.39	221.36
07	1205.94	388.89	1068.27	301.88	934.83	202.00	601.77	211.13
08	1211.94	399.12	1077.20	299.85	951.50	208.99	608.03	212.55
09	1222.84	412.36	1067.82	304.31	954.27	207.99	607.15	212.85
10	1230.40	398.08	1077.49	299.47	960.85	203.88	613.55	213.46

Table 5 - Response Time Results for Individual TCP Runs

BW, Delay	DS0, 200ms		2xDS0, 200ms		T1, 200ms		2xDS0, 100ms	
RUN	sample mean	std dev	sample mean	std dev	sample mean	std dev	sample mean	std dev
01	755.58	223.39	637.57	123.85	495.85	53.84	398.27	114.16
02	731.46	224.49	599.46	119.75	495.94	51.89	382.86	109.99
03	734.77	220.97	594.68	118.72	479.60	52.99	376.00	110.79
04	775.70	246.71	643.68	136.26	481.11	50.25	402.26	121.46
05	733.89	218.74	591.70	124.37	484.33	56.09	380.14	113.23
06	744.45	225.88	599.89	129.81	479.28	53.26	394.80	116.15
07	715.70	223.47	588.33	119.80	479.10	54.85	375.96	109.69
08	724.41	218.04	595.00	122.76	478.14	57.16	381.71	108.59
09	725.19	221.25	588.81	127.76	477.76	53.51	383.90	113.56
10	730.32	220.27	596.54	121.79	473.30	51.11	394.75	111.09

Table 6 - Response Time Results for Individual SCTP Runs

5.7 Simulation Results and Analysis

The statistical parameters used to aggregate the results from individual runs into an average response time, for each of the modelled network conditions, are as follows:

- A series of 10 sample mean response times from the individual runs is used to calculate the average response time.
- A 95% confidence interval was used in calculating the error associated with the average response time using the sample means from the individual runs.

The overall results are summarized in Table 7, showing the response time as measured from the communication server's perspective. All response times for the simulation runs were well within the typical acceptable values from a services perspective (typically 2 to 3 seconds based on existing PSTN specified values). As such, the percentage of successful transactions was effectively 100% for both TCP and SCTP,

and was therefore not treated as a meaningful metric for this analysis. The response time metric, however, was found to be useful in the comparison.

Network Conditions	TCP average response time	SCTP average response time
64 kbps (DS0) channel; 200ms avg delay	1230 $\pm$ 20	737 $\pm$ 12
128 kbps channel; 200ms avg delay	1086 $\pm$ 15	604 $\pm$ 14
1544 kbps (T1) channel; 200ms avg delay	958 $\pm$ 10	482 $\pm$ 5
128 kbps channel; 100ms avg delay	618 $\pm$ 11	387 $\pm$ 7

Table 7 - Summarized Response Time Results

The results show SCTP having lower response times than TCP for equivalent scenarios. It is also interesting to see less variance (more predictability or determinism) for the SCTP results in Table 6 than the TCP results in Table 5.

A scatter plot of the response times relative to the combined size of the query and reply payload messages is generated for a representative scenario. This helps to quickly ensure that any outliers are removed from the observed results. The scatter plots corresponding to using a 64 kbps channel between the simulated communication server and application server are shown in Figure 19 and Figure 20 for TCP and SCTP, respectively.

Figure 19 - Representative Scatter Plot for TCP Run

Figure 20 - Representative Scatter Plot for SCTP Run

Looking at the scatter plots the larger message payloads resulted in greater response times observed from the application layer (as expected). Discarding initial data points until the simulation reaches a steady state allows network parameters being emulated to stabilize. As a result of this approach, no outliers were expected in the scatter plots.

In all plots observed, there were two distinct clusters present in both the TCP-based and SCTP-based measurement results. To explain this phenomenon, the actual service messaging profiles were examined closely. These profiles were based on existing observed messaging exchanges in the PSTN network based on AIN 0.1 services. The root cause of the distribution having two distinct clusters is related to one specific AIN reply operation having a corresponding message size larger than many of the other operations. As such, it is expected that simulated calls that encounter this type of reply operation will be part of the cluster having larger message payloads.

To verify that we have correctly attributed the cause of the observed clustering, the percentage of expected observations in each cluster is calculated based on the reply mix used. The expected percentage of calls is calculated based on the following formula (please see traffic parameters in section 5.4 for more detail and note that R_6 corresponds to the large reply message):

$$\sum_i prob(Q = Q_i) \times prob(R = R_6 | Q = Q_i)$$

By plugging in the figures from Table 1 and Table 2 (on page 48), we obtain an expected percentage of 24.50%. Using a large sample size, the percentage of the samples having messages larger than 3000 bytes is calculated, yielding a result of 25.59%. This value is quite close to expectations, thereby validating the clustering explanation.

Another interesting observation can be made by measuring the ranges in response time values for the observations in the two regions of the graphs presented in Figure 19 and Figure 20. It is worth noting that these ranges are more than 1.5 times as large for the

TCP-based runs, relative to the SCTP-based ones. This is another confirmation of the deterministic behaviour of SCTP relative to TCP.

As the available bandwidth of the channel between the communication server and application server increases, the difference in observed response times between SCTP and TCP gets smaller. Similarly, as the network delay decreases, the gap between SCTP and TCP decreases.

Figure 21 - Summarized Results

The summarized results are shown graphically in Figure 21. Please note that the error margins in the data being plotted are too small to show on the graph. The above results clearly show favouritism towards SCTP. SCTP was expected to have better results than TCP, but the delta between the two was observed to be larger than initially anticipated. Further investigation into the root cause revealed some simplifications in the error-path handling of the SCTP underlying stack implementation. This simplification led to improved stack performance (since the SCTP protocol has just been standardized, early protocol implementations did not fully implement error scenarios). Please note, however, that the increased overhead of error-path implementation is not expected to be more than

10%, and that the results observed would still show favouritism towards SCTP (by approximately 30%), even when such overhead is taxed onto to the SCTP results. These results definitely show promise for the SCTP approach. It is also important to note that, not only was SCTP observed to lead to better (lower) response times, but it also led to more deterministic response time characteristics (less variance) as compared to TCP.

Chapter 6

Recommendations

This section summarizes the overall thesis recommendations, beginning with the ones that relate to NGN architectures in general, followed by the two specific recommendations that address the area of dependable and extensible NGN services architecture (central to the thesis contributions discussed in the previous chapter). The first set of recommendations that will be discussed is complementary, but not specific to the main contributions of this thesis in the area of NGN services. The thesis contributions are based on the final two recommendations (R06 and R07) related to NGN services: one is related to extensibility at the application layer, and the other is related to dependability at the transport layer.

6.1 Limit Transcodings To Access Gateways (R01)

This recommendation suggests limiting the number of transcodings to two (or less) for any given audio call: one at the ingress access gateway, and one at the egress gateway. This leads to significant improvements in speech quality and a simplified network

architecture, and can be engineered to do so without limiting the flexibility of end-user devices.

Related to this, it is recommended that an uncompressed voice scheme (such as RTP streams using the ITU-T G.711 codec over IP, or AAL1 traffic over ATM) be used in the core of a carrier's network (transit, tandem, long-haul), and for interconnection between carrier networks (local, inter-exchange, national, international). This simplifies the network architecture, leads to improved speech quality, and has tolerable implications on bandwidth utilization when considering the shift towards more data-centric network traffic. Note that this recommendation is intended for toll-quality conversational speech; for wideband audio, CD quality and high fidelity audio communications, a different coding scheme is to be used. Although this recommendation is in the context of narrowband voice, the same concepts could be made to apply to wideband audio with some adaptation.

Note that both H.323 and SIP allow the end-users to negotiate that choice of codecs and associated parameters between gateways and edge devices. It is suggested that the ingress and egress gateways handle a variety of voice transcoding schemes on the access side, but convert to a common uncompressed voice scheme on the network side.

Some would argue that using compression of the audio packets in the core of the network is important to decrease the bandwidth requirements of the network. This argument, however, becomes less relevant as the amount of non-voice traffic being carried in the core network exceeds that of voice.

6.2 Support Bearer Path Multicasting (R02)

This recommendation suggests expanding the scope of existing gateway control protocols to explicitly support multicasting, without allowing the multicast operation to be detected

by end-user devices. Features such as lawful interception require such multicasting of the bearer path to lawful interception data access points. As well, services that need mid-call in-band detection of DTMF tones, or detection of speech commands, can do so without disrupting the speech path if multicasting is used. Handover between wireless gateways is another application where multicasting simplifies the network configuration. As well, any service that needs to revector the bearer path is simplified by having multicast operations supported by the gateway control protocol. Revectoring of the bearer path refers to the breaking and re-establishment of the bearer path. The design of bridging and conferencing applications is also simplified by support of multicasting at the gateway control level.

6.3 Decouple Session Control From Media (R03)

This guiding principle, to decouple session control from media transport, is not entirely new, but is important enough to mention explicitly. Signalling and control protocols are exchanged between communication servers, whereas media streams flow between the media gateways (being controlled by the communication servers). There are scenarios where the signalling and control information needs to traverse through a particular "tandem" communication server, whereas the associated media streams need not traverse through the media gateway under the control of that communication server. Decoupling signalling from media resources allows this scenario to be realized efficiently. This decoupling also allows enhancements to service logic to become applicable to different media types.

6.4 Flatten The Control Architecture (R04)

It is recommended that services be migrated to the edge of the network, and a flattening of the communication server architecture be pursued. The classical PSTN telecom model contained End Offices for end user line access, and a hierarchy of Toll and Tandem Offices to aggregate traffic and route it between major centres. The model proposed here preserves the communication services at the End Office, and migrates service logic from the Tandem Offices towards the End Office. The network is simplified at the Communication Server layer due to the reduction in tandem nodes. Please note that a hierarchical model may still be employed in the core of the packet network over which multiple different traffic types are carried.

6.5 Standard Communication State Machines (R05)

The Communication Servers that provide basic voice service logic and supplementary services to end-users need to maintain state information related to session establishment. It is recommended that standard finite state machines be used to model the progress of call setup in communication servers: one state machine to model the initiation of a session, and an independent state machine to model the processing of an incoming request for communication. In the SIP context, these two state machines would relate to the User Agent Client and User Agent Server respectively. In an AIN / IN context, these state machines are similar to the Originating and Terminating Basic Call Models. In fact, it is recommended that the AIN call models be used as a starting point on which to base the design of these standard state machines. Some of the points in call defined in the AIN state machines may be optimized out, but the core of the call model is still believed to be applicable. Furthermore, just as the AIN call models may be extended using what are

referred to as detection points in the AIN context, the finite state machines resident in the communication servers need to extensible state machines.

Another way to look at this recommendation is that the Call Model design pattern, presented in reference [13], should be applied consistently to the finite state machines that operate within NGN Communication Servers.

6.6 Extensibility for Access to NGN Services (R06)

Traditional PSTN services interfaces have been built on SS7 infrastructure using application layer protocols such as TCAP and AIN; other PSTN and enterprise telephony approaches use primarily proprietary CTI protocols for niche applications. Unfortunately, both the traditional AIN and CTI approaches have presented frameworks that lack extensibility.

In AIN, the standard-setting bodies are typically slow to react to new market and service needs. Some AIN and IN variants allow for extension parameters to be included in TCAP and INAP messages. In reality, however, the size of AIN messages is bounded by the underlying SS7 transport mechanisms that prohibit the inclusion of any complex exchange of data. Furthermore, the extension parameters are defined using binary encoded ASN.1 whose rigid definition does not naturally lead to extensibility.

CTI approaches in existence today typically start with a view of the network that is based on first party call control, then aim to extend the paradigm to third party call control with only limited success in niche applications such as call centres. CTI protocols have yet to extend their applicability to general network communication services.

Recognizing the deficiencies of existing approaches, a novel proposal is made herein, to use emerging NGN extensible protocols to achieve greater flexibility and extensibility in the domain of next generation services and applications. SIP (Session Initiation

Protocol), which is previously only defined for call setup (initially for setting up communication sessions between user agents, and recently expanded for communication sessions between call servers), is proposed as the basis for upper layer messaging exchange between communication servers and application servers. This extensible approach was believed to be novel enough that a colleague and myself have a patent pending on a method of implementing this recommendation [1].

6.7 Partial Ordered Dependable Transport (R07)

The communication exchange between Communication Servers and Application Servers should be carried over a dependable (reliable and deterministic) transport mechanism. By maintaining reliability at the transport layer, but trading off some sequencing requirements, we are able to achieve greater deterministic behaviour, as observed from the perspective of upper layer applications.

Reliability is needed to maintain attributes of carrier grade systems, but determinism is also needed for the same reason. Carrier grade systems need to be characterized to handle a certain traffic load. When that load is exceeded, service will be denied to additional requests for session establishment or the grade of service provided will be decreased based on a pre-determined policy.

The requirement for reliable transport rules out the use of UDP (without an upper-layer scheme for reliability). The need for deterministic behaviour in carrier-grade systems rules out the use of TCP except for certain scenarios where a managed network is used and QoS policies are administered appropriately.

A new peer-level transport protocol to TCP and UDP, known as SCTP, is recommended for use. SCTP was initially defined to allow information exchange between a call server or media gateway controller, and an SS7 signalling gateway. As

shown in this thesis, SCTP, which is referred to here as a partially ordered reliable transport protocol, has wider applicability than its original intent.

Note that the use of QoS schemes such as RSVP with TCP, and MPLS, help address determinism of the bearer and control path (where administered properly), but have not been examined as part of this research on enhanced services. Instead the focus has been on the use of partial ordered transport to achieve better deterministic behaviour for enhanced IP-based communication services.

Other research has looked at partially reliable transport of the audio media streams. In that context, partial reliability is taken to mean allowing data loss within a pre-defined loss pattern. In this thesis, by contrast, partial ordered reliability is proposed. In the context of signalling transport, partial ordered reliability is taken to mean that messages and information streams are delivered to the application layer free of error, but not necessarily in the same order as they were sent. Messages that need to be ordered are sent within a context that preserves ordering; however, no attempt is made to order information between unrelated contexts. This relaxation of sequencing results in better deterministic behaviour, as was demonstrated using the simulation framework in Chapter 5.

Chapter 7

Conclusions and Further Works

This research has started with an investigation of enhanced communication services in next generation networks, making recommendations for the architecture of such networks so that they are scalable, extensible and simpler in design, and detailing specific contributions related to extensible and dependable services.

One contribution is made at the transport layer to address dependability (reliability with increased determinism relative to prior approaches). The term "partially ordered" reliable transport is used to describe SCTP, an emerging protocol initially intended for transport between a communication server and its signalling gateway. This same protocol is shown to have applicability between communication servers and application servers in an NGN model. The relaxing of sequencing in SCTP allows it to have increased deterministic behaviour. A simulation methodology is described that allows the comparison of different transport schemes from an upper layer application perspective. An implementation of the simulator is written and exercised to compare SCTP to TCP, showing that SCTP is more appropriate for service messaging in that it has more favourable results in terms of the number of completed fully-featured calls, and service-related response time deterministic behaviour.

Another contribution is made at the application layer, proposing the use of an extensible protocol for service-related information exchange between communication servers and application servers. It is proposed that a new SIP-based protocol be used for such a role. The SIP protocol, as previously defined, was extensible but was also closely coupled to call initiation. In order to make use of SIP in this new services context, a decoupling of the base components of SIP is proposed. A new protocol, termed SIP-S (for services) is then defined, based on the decoupled SIP base, to allow communication servers access to the services of an application server. A high-level mapping is shown between TCAP-based messaging to SIP-S. Such a mapping allows the creation of NGN services that are extensions of (or are a superset of) their TCAP-based service counterparts (such as AIN). The mapping presented shows how TCAP package types and operations can map to SIP-based messages and MIME types, and how TCAP parameters can map to tags and fields within the MIME types and headers.

The contributions in this thesis enable the design of a dependable and extensible NGN communication services architecture. The proposed usage of SCTP is presented to address dependability for transport of the messages between nodes in an NGN network. A new protocol, SIP-S, based on a decoupled version of the existing SIP protocol, is presented to address extensibility of service-related messaging.

The recommendations and contributions of this thesis outline an end-to-end solution to address extensibility and dependability for NGN services:

- At the Application Layer, the SIP-S contribution addresses service extensibility.
- At the Transport Layer, the contribution on the expanded domain of SCTP addresses service dependability.
- Additional recommendations are also proposed to suggest improvements in NGN control and bearer domains, but are not the primary focus of the services theme being addressed in the main thesis contributions.

The thesis contributions in the areas of service extensibility and dependability, when taken together, lead to the protocol stack shown in Table 8, proposed in this thesis for use between communication servers and application servers.

OSI Layer	Protocol
Application (7)	SIP-S
Presentation & Session (6/5)	SIP
Transport (4)	SCTP
Network (3)	IP

Table 8 - Proposed Protocol Stack

The new proposed application protocol, SIP-S, can carry AIN-like or TCAP-like information elements. SIP-S is encapsulated within a SIP envelope as an extensible services protocol, and then carried reliably using SCTP transport over IP.

7.1 Future Works

This section highlights some areas that are for further study.

- This thesis has focussed on the carrier service provider network configuration. Applicability to corporate enterprise communications systems can be examined further, particularly within the context of networks supporting H.323.
- This study has focussed on a network where the control, signalling and bearer are carried over an IP fabric; ATM-based networks can be examined further. Much of the same messaging for service-related functionality is applicable to both ATM and IP-based packet environments, so the results are expected to be similar for ATM.
- This thesis has looked at the usage of SCTP as a reliable transport vehicle between Communications Servers and Application Servers. The same type of study can be

done for messaging between Communication Servers and each other (referred to as Inter Call Server signalling). A similar simulation framework to the one presented in this thesis could be used as a basis for modelling Inter Call Server signalling.

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Appendix A - IN Call Models

The IN / AIN call models commonly used in the existing PSTN network are given in this section. Both the North American and International widely accepted variants are shown. The originating call models are presented first (model the behaviour of outbound call initiation), followed by the terminating call models (that model the behaviour of an inbound call request).

Figure A1 - AIN Originating Basic Call Model (North America)

Figure A2 - CS-2 Originating Basic Call Model (International)

Figure A3 - AIN Terminating Basic Call Model (North America)

Figure A4 - CS-2 Terminating Basic Call Model (International)