Advanced Intelligent Network Services

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Abstract

The telecommunications industry is undergoing a fundamental revolution with the introduction of intelligent networks. In this thesis, we first give an overview of the intelligent network, which includes the review of the motivation, technology, evolution, standards, components and the open issues of the intelligent network. The emphasis of the thesis is put on the intelligent network service creation. Three AIN (Advanced Intelligent Network) services, namely, the call by name, the automatic telephone directory service and the information query service, are designed and implemented in the AIN Service Creation Environment (SCE). An AIN multimedia fax service which is a novel application of the AIN technology is proposed. With a gateway between the telephone network and the Internet, we use AIN to provide automatic addressing, dynamic routing and flexible selecting of document type for the multimedia fax service. The idea and the technique used in this AIN multimedia fax service bring up a new AIN service opportunity which allows telephone subscribers to access abundant Internet resources. The performance of this service is evaluated based on the simulation model of priority queuing system.

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I am truly grateful to my beloved husband Jian Wang for his consistent support, without which this work would not have been possible.

List of Abbreviations

ABS	Alternate Billing Services
ADJ	Adjunct
AIN	Advanced Intelligent Network
ANI	Automatic Number Identification
ATDS	Automatic Telephone Directory Service
B-ISDN	Broadband Integrated Service Digital Network
BCP	Basic Call Process
CCAF	Call Control Agent Function
CCF	Call Control Function
CCIS	Common Channel Interoffice Signaling
CPR	Call Processing Record
CS	Capacity Set
CSO	Customer Service Office
CV	Call Variable
DFP	Distributed Functional Plane
DLN	Dialed Line Number
FC	Functional Component
FE	Functional Entity
FEA	Functional Entity Action
GFP	Global Functional Plane
GSL	Global Service Logic

GSS	General Service Specification
IN	Intelligent Network
IP	Intelligent Peripheral
ISCP	Integrated Service Control Point
ISDN	Integrated Service Digital Network
LAN	Local Area Network
MIME	Multipurpose Internet Mail Extensions
MM	Multimedia (MM fax call: Multimedia fax call)
MOC	Maintenance and Operation Console
MSAP	Multi-Service Application Platform
MVI	Multivendor Interaction
NCP	Network Control Point
PE	Physical Entity
PIC	Points in Call
PMCN	Personal Mobile Communication Network
PSTN	Public Switch Telephone Network
SCE	Service Creation Environment
SCEF	Service Creation Environment Function
SCF	Service Control Function
SCP	Service Control Point
SDF	Service Data Function
SF	Service Feature
SIB	Service Independent Building Block
SL	Service Logic
SMAF	Service Management Agent Function
SMF	Service Management Function

SMS	Service Management System
SN	Service Node
SPC	Stored Programming Control
SRF	Service Resource Function
SS7	Signaling System Number 7
SSF	Service Switching Function
SSP	Service Switching Point
STP	Signal Transfer Point

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Chapter 1

Introduction

The telecommunication industry is undergoing unprecedented changes, as it tries to keep pace with the fast emergence of new technologies and the diverse market demands. The changes involve: i) the increased use of network elements that are controlled by or interface with software; ii) a desire to share data and distribute application processing among network elements; iii) the need for standard interfaces between network elements; and iv) the user demand for more sophisticated telecommunications services and rapid delivery of services. This evolution is particularly evident in the quick adoption of the intelligent network technology by network providers all over the world.

1.1 Motivation of Intelligent Networks

Today, many telecommunication services are developed by means of switch software. Creation of a new service is achieved almost exclusively by the switch vendors. Traditionally, new services in the telephone network are implemented by the switch manufacturers on the local switching systems; service specific data need to be developed in every switch. With this approach, teams of expert developers are required to write thousands of lines of code that will enable a switching system to support a new service. This calls for a lengthy and costly endeavor: up to five years and millions of dollars are needed for providing a new service [DMK]! In addition, the services created in this manner can not be customized. Even the smallest modifications to existing service offerings are dependent on the switch manufacturers.

Intelligent networks is the solution for the above problem. The main idea of intelligent networks is to meet the needs of telecommunication service providers by providing the ability to introduce new services in a rapid and cost effective way. From the network point of view, the term "intelligent network" is used to describe an architectural concept for the provision of telecommunications services which are characterized by extensive use of central information processing techniques, modulation of the network functions, reusable standard functional entities, and efficient use of network resources [JaTh94].

1.2 Introduction to Intelligent Networks[1]

The intelligent network can be viewed in three dimensions [CDLS94], as shown in Figure 1.1. They are the following.

- the standards that define it;
- the technology that creates it;
- the services that result from it.

[1]Intelligent Network (IN) and Advanced Intelligent Network (AIN) represent the same concept. The difference only lies in the capacity and technology enhancement of the networks (see chapter 2). As a general concept, IN and AIN are used interchangeably in this thesis.

The first two points will be discussed in chapter 2. The last point is the focus of this thesis, which is presented in chapters 3 and 4. The rest of this chapter will introduce the components of an intelligent network, and then present a service example to show how the intelligent network works.

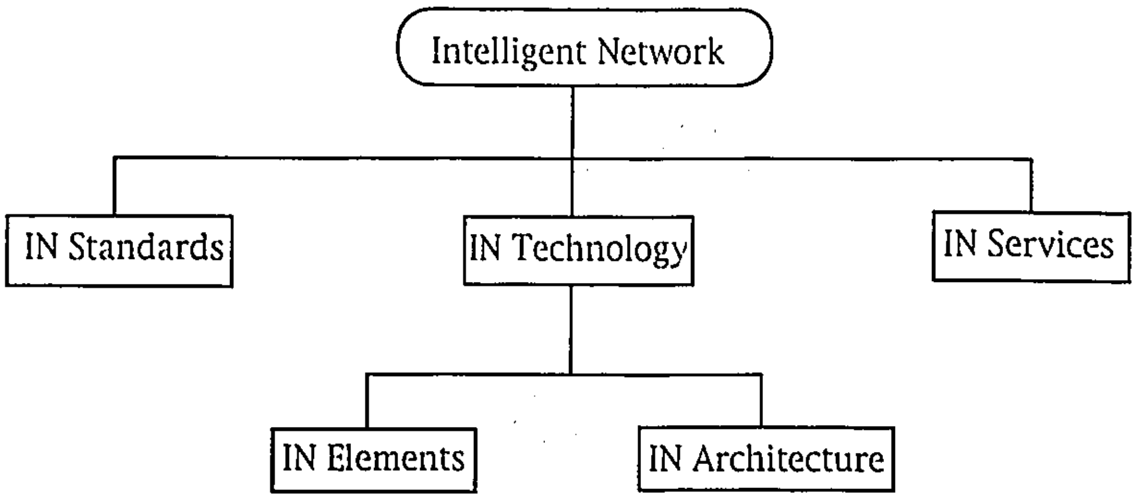

Figure 1.1 The Aspects of the Intelligent Network

1.2.1 Components of the Intelligent Network

The components of intelligent network are shown in Figure 1.2. A brief description of the functionality of each component follows.

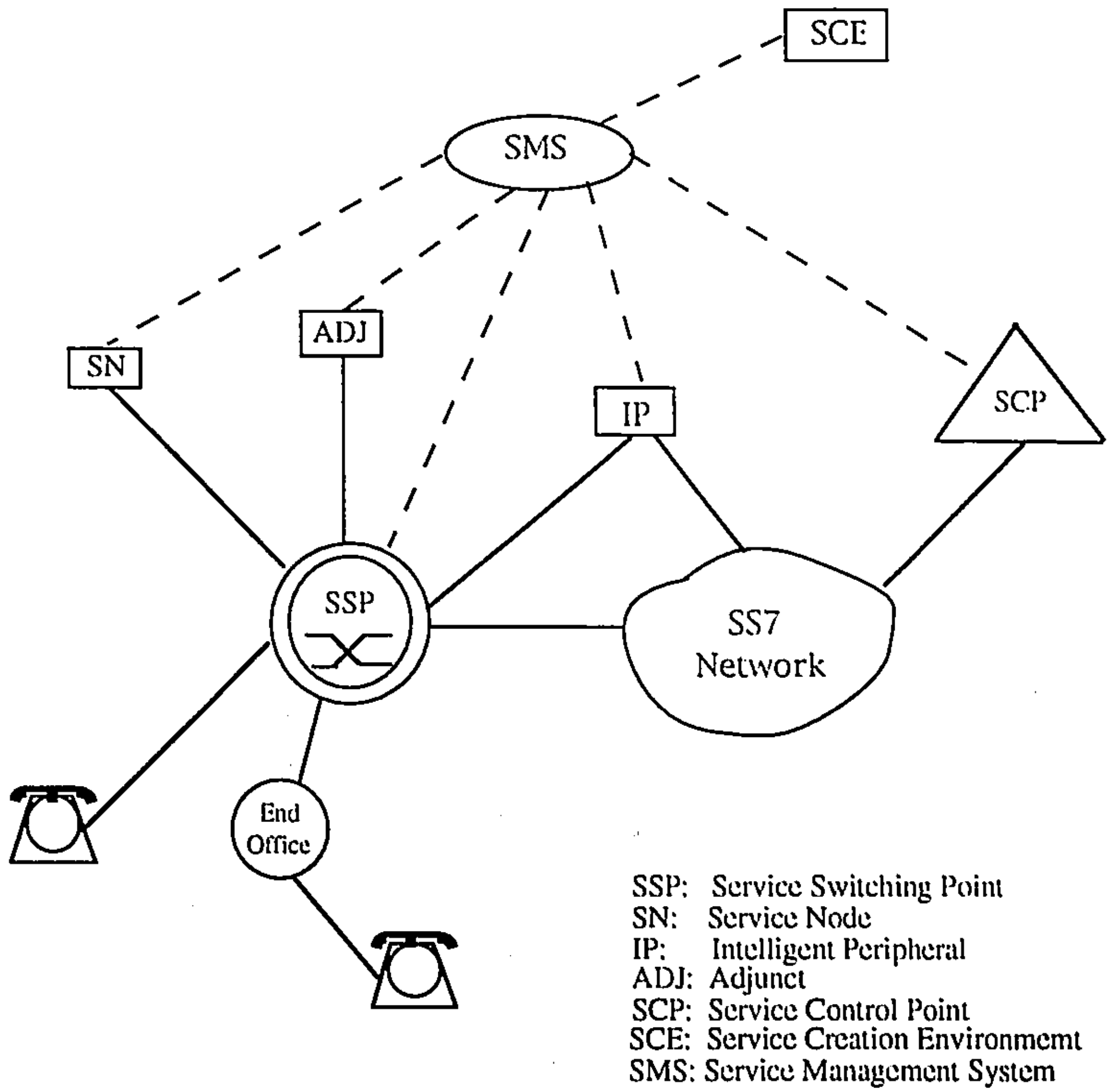

Figure 1.2 Components of the Intelligent Network

Service Creation Environment (SCE)

The SCE is a software tool set that allows the telephone company personnel to define, create, test, and activate services for customers. This tool set provides a friendly graphical user interface, so that users can quickly design, modify, validate and test new services.

Service Control Point (SCP)

The SCP is an entity which executes various service logic functions. It includes a database containing service logic, customer and system information. Instead of working on every switch, a new service is created by inserting a new call process record (CPR[1]) in a centralized database. A change in service is made by disconnecting, removing, or modifying an existing CPR in the database.

Service Switching Point (SSP)

The SSP is a switching system. Its primary new capability is to recognize attributes, called triggers[2] , of both outgoing and incoming calls. These triggers indicate a need for the intelligent network special treatment. When a trigger is detected, the SSP will initiate a query to the service control point for instructions on how to handle the call.

Intelligent Peripheral (IP)

The IP provides a new generic interface for end-user interactions. It hosts special resources, such as speech recognition, voice identification, text-to-voice facilities for customization of services, and supports flexible information interaction between a subscriber and the network. The functionality of IP is to play announcements and prompt to the caller, to collect the caller's responses and to return its collection to the SCP.

Service Management System (SMS)

The SMS is a centralized operations system for service provision, maintenance and administration. It's responsible for inputting customer records (specific CPRs), verifying the records, and distributing them to a proper SCP. SMS's maintenance and administration functions include database backup, recovery and updating, system reporting, and user access security.

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[1] CPR is a graphic representation of the service (see details in Chapter 3).
[2] A trigger is a event indicator in the call processing model (see detail in Chapter 2).

Common Channel Signaling System No. 7 (SS7)

The SS7 network provides a transport function for the intelligent network service control messages. The SSP and SCP communicate with each other through SS7 Transaction Capabilities Application Part (TCAP) messages.

Adjunct (ADJ)

The ADJ is functionally equivalent to an SCP, but is directly connected to a SSP.

Service Node (SN)

The SN contains functionality equivalent to the ADJ and IP. It connects directly with one or more SSPs and can engage in flexible information interactions with callers.

Some intelligent network services have local characteristics in nature. Instead of implementing this kind of service logic in the SCP, using ADJ or SN to store the service logic can provide better performance for the services, and can reduce SS7 network traffic, which is important because the SS7 network is a potential bottleneck when intelligent network services increase in volume.

Not all above components need to be used to provide intelligent network service capabilities. The decision concerning which network elements are to be used depends on a variety of factors, including:

- existing network characteristics and deployment plans;
- types of services tc be deployed;
- ubiquity of services to be deployed.

1.2.2 The Power of the Intelligent Network

An example of IN services, namely the "pizza order" service, is presented to show the power of the intelligent network. Using the intelligent network, we do not need to know where the nearest pizza store is located, what its open hours are, and how busy it is. We simply dial a unique "pizza number" (e.g., 737-1111), no matter where we are in the country. The network will route the AIN call to the nearest pizza store which is open and not in overload of pizza orders . The result is a hot pizza within minutes.

The flow of events and actions that might take place in the intelligent network is shown in Figure 1.3. Note that the SS7 is not shown in the figure for simplicity.

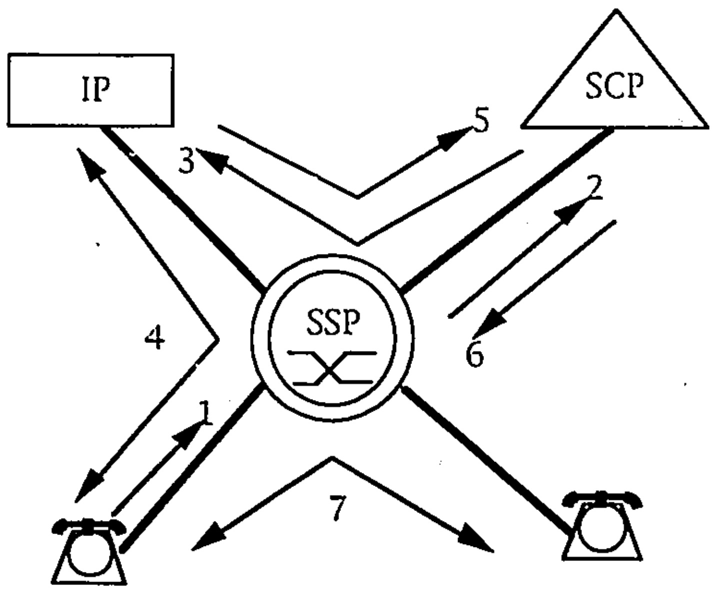


Figure 1.3 Illustration of Events and Actions Involved for the "Pizza Service"

(1) A user dials the "pizza number", the SSP detects the trigger which indicates the request for "pizza service";

(2) the SSP initiates a TCAP query message based on the trigger detected, then sends it to the SCP;

(3) Upon receiving the TCAP message from the SSP, the SCP invokes the service logic of "pizza service" which indicates the SCP to inform the IP to play a pre-defined announcement;

(4) the IP connects the user through the SSP, prompts the user to identify user's specific need, such as if user has preference of pizza store, and if yes, which store. Assume the user likes "pizza hut";

(5) the IP returns user's response to the SCP;

(6) the SCP determines the telephone number of a proper pizza store based on some criteria, such as the pizza store user wants, the nearest store to the user, the store's open hours, busy situation of the store, and then sends this number to the SSP;

(7) the SSP connects the caller to a pizza store which can provide the satisfactory service to the caller.

It can be seen that not only can the intelligent network provide value added services, but also it can be easily customized to fit user special needs due to the SCP centralized control and the IP flexible user interaction. For example, when store open hours or busy status changes, or a new store opens, etc., these information changes can be quickly reflected in the SCP. Furthermore, this service can be modified to suit other customers, e.g. retailers. Other intelligent network services could be free phone calls (800 calls), premium rate calls (900 calls), credit card calls, and so on.

1.3 Applicability of Intelligent Networks

As illustrated in section 1.1, the intelligent network is an architectural concept which results in service, switch and equipment independent networks. Compared to an intelligent network, the networks, such as the Public Switched Telephone Network (PSTN), Broadband or Narrow band Integrated Services Digital Network (B-ISDN or ISDN), cellular networks, and so on, are reduced to access forms. Figure 1.4 shows that the intelligent network is a general concept above access networks. Consequently, in the future it will be able to set up free phone calls, credit calls, etc., regardless of the type of network used to access the intelligent network level. The intelligent network level will be able to offer the same capabilities (services) to all access networks in a uniform way, for example, making number translation depending on time and origin [JaTh94].

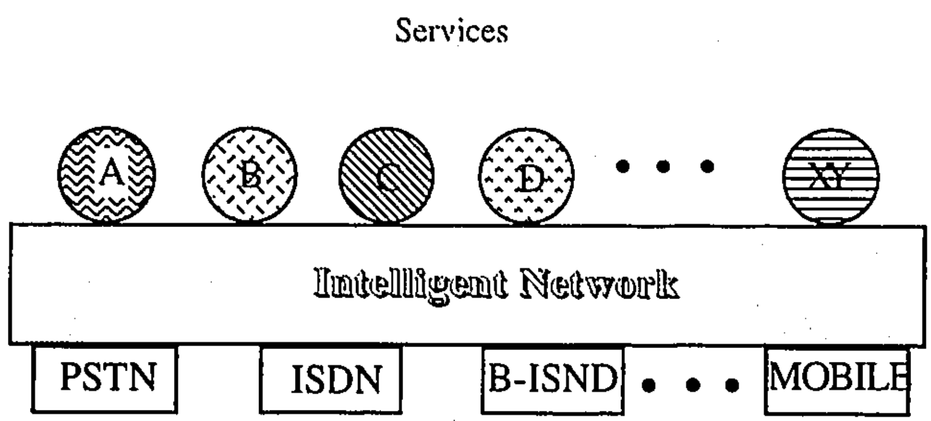


Figure 1.4 Intelligent Network as a General Concept

Moreover, it will be possible for users with different means of access to call one another, depending on the condition of the weakest part and on the relevant services. The intelligent network will control the call set up and might also determine whether the connection between two users with different means of access is allowed for the call or services (see Figure 1.5).

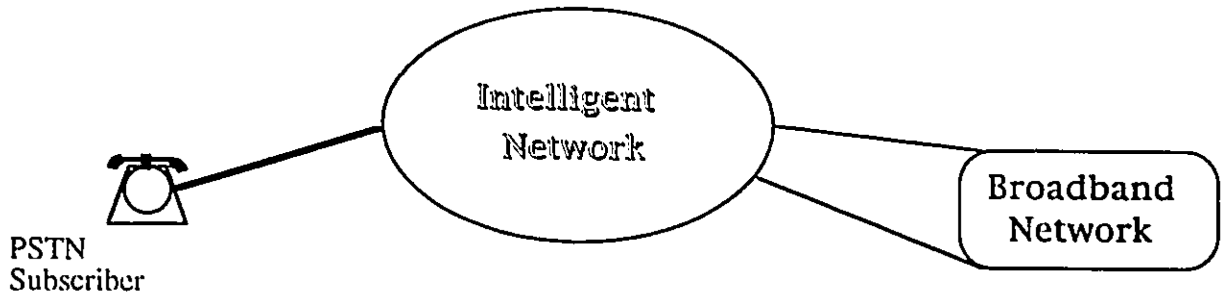


Figure 1.5 The Weakest Part Determines the Connection

It has been claimed that the term "intelligent network" will not be used beyond the 1990s because the AIN approach will be the only way to build various kinds of networks [JaTh94]. In the 1970s, the "Stored Programming Control" (SPC) was a new concept for building telephone switches. With the SPC, computers were used to control centralized intelligence (control logic) in telephone exchanges. The SPC is seldom motioned today because it is quite commonplace.

Much similar to the revolution in building switches (nodes) with the SPC technique, the intelligent network implementation signals the start of a revolution in building networks. The first generation of SCPs on intelligent networks is simply components that control and centralize intelligence in the network. In the future, the AIN concept will be firmly integrated and commonplace in all networks. So AIN is a principle goal of the telecommunication and data communication evolution and will sooner or later become a reality in all networks.

1.4 Structure of this Thesis

In this chapter, we have introduced the driving force, the principle and the components of the intelligent network. In Chapter 2, we will give an overview of intelligent networks, which includes the evolution, standards, some terminology and open issues of intelligent networks. In Chapter 3, the platform for service creation and service creation process are described, and the services we created are presented. In Chapter 4, an AIN multimedia fax service, which is a novel application of the AIN, is designed and simulated. The simulation model and the simulation results are presented and discussed. Chapter 5, the last chapter, contains the summary and conclusion of this thesis.

1.5 Thesis Contributions

The main contributions of this thesis are summarized below:

- Design of some new AIN services and implementation of some of them with the Bellcore SPACE system (**section 3.3.1, 3.3.2,** and **3.3.3**);
- Design of an AIN multimedia fax service, which is a novel application of AIN technology (**section 4.1**);
- Building a simulation model for the performance evaluation of the above AIN multimedia fax service (**section 4.2.3, 4.2.4, 2.4.5, 4.2.6, 4.2.7, 4.2.8** and **4.3.2**);
- Evaluation of the performance of the designed AIN multimedia fax service (**section 4.3.4**).

Chapter 2

Overview of the Intelligent Network

In this chapter, we give an overview of the intelligent network, which includes the evolution, standards, some terminology and open issues of the intelligent network.

2.1 Evolution of Intelligent Networks

During the past 30 years, telephone networks have greatly expanded in terms of the number of subscribers served and the volume of traffic carried. The expansion of services has been driven by the needs of the increasingly sophisticated business and residential subscribers, and has been enabled by the widespread deployment of stored program control (SPC) technology in the telephone networks. During the 1970s, the SPC intelligence was beginning to be introduced in the systems whose functions were to support the network's management and maintenance.

Beginning in 1981, the network intelligence reached a new plateau when AT & T introduced the use of centralized databases to support calling card and 800 services. These databases are located at the network control point (NCP) system and are accessed by SPC switches via the Common Channel Interoffice Signaling (CCIS) network. This centralized approach allows the introduction of some services that would otherwise be impractical due to the complexity of managing large amount of volatile data at every SPC switch.

In 1984, Bellcore introduced the term "intelligent network" with the first standard, IN/1, which was based on a centralized service control point (SCP). The SCP was used to support alternate billing services (ABS) and 800 calling. ABS provides calling card validation and other line information functions for collecting the third-party billing. Access to SCPs from switches is provided via the Signaling System 7 (SS7) networks.

IN/1 was a very simple design, but when a new service was introduced, both the SSP and SCP had to be updated. That is, services were implemented as switching system features that interacted with network databases on service-by-service basis. It didn't support a common, service-independent call model. It was recognized that a more generic approach was needed in order to satisfy a potentially large number of services. The effort resulted in the IN/2 architecture in 1987.

IN/2 included a greatly expanded set of switching and SCP capabilities, known as functional components (FCs), and a new system, called the Intelligent Peripheral (IP). The FCs defined the atomic functionality elements of the architecture and the IP provided a platform for deploying service-assistance capabilities. In addition, IN/2 placed more emphasis on the concept of standardized service creation environment, which would allow services to be defined and implemented without reference to specific IN elements. The basic idea of IN/2 was that only the SCP would be updated for a new service.

However, implementing IN/2 would require a major updating of many local switches, which would cause unacceptably high risks and could not be realized in a sufficiently short time frame. As an interim stage solution, IN/1+ was defined in 1988.

IN/1+ can be viewed as a subset of IN/2. It was intended to introduce service-independent capabilities to the network within a few years. Similar to IN/2, IN/1+ used FCs and IPs, but they were limited to a small set of functions needed to support voice-band services. With IN/1+, SSPs react to pre-defined events (or triggers, explained in section 2.3.2) in the switching process by requesting assistance from SCPs, which execute the service logic. The SSP then receives orders in the form of standardized functional messages through SS7, which are independent of the service. With this architecture, it was hoped that the need for major modifications to a large number of local switches would be minimized.

IN/1+ abandoned later based on the realization that it would be advantageous to achieve better alignment between the IN architecture and the architecture of network switches. In 1989, Bellcore established the Multivendor Interaction (MVI) forum with the purpose of collecting the best ideas from Telecommunication companies and switch manufacturers to define an architecture that would meet the long-term objectives of Telecommunication companies and have the support of the switch vendors.

The IN evolution that started by the MVI forum led to the concept of Advanced Intelligent Network (AIN). An evolutionary path was defined by a series of AIN releases. Each release contains additional architectural attributes and capabilities for supporting services.

AIN/1 (or AIN release 1) was defined in 1991. It is the first complete intelligent network architecture that provides a service and switch independent platform for execution of service logic. More importantly, for the first time, a majority of operating companies and manufactures have agreed on a common goal: an AIN with the capabilities outlined in AIN/1.

AIN/1 was targeted for later deployment. It was recognized that there was a huge gap between the existing network and the envisioned AIN/1 network. AIN/0.1 and AIN/0.2, released in 1992 and 1993 respectively, provide a gradual progress towards the direction of the AIN/1 target.

As the US telecommunication industry struggled with the best evolution path to the AIN, the CCITT (now ITU-T) began working on recommendations for the intelligent network. As result, the Capacity Set 1 (CS-1) and CS-2 were defined in 1992 and 1994 respectively. The term "capacity set" refers to the set of services and service features that can be constructed by using service independent building blocks (SIBs). The CS-1 enables the low-risk introduction of a wide range of advanced services together with rapid service delivery and customization capabilities [JDJV92]. The set of target CS-1 services and service features, with an emphasis on capabilities that provide flexible routing, flexible charging, and flexible user interaction, are presented in Appendix 1

The conceptual model of the IN has been defined by CCITT to provide a platform for developing the intelligent network, which is the topic of the next section.

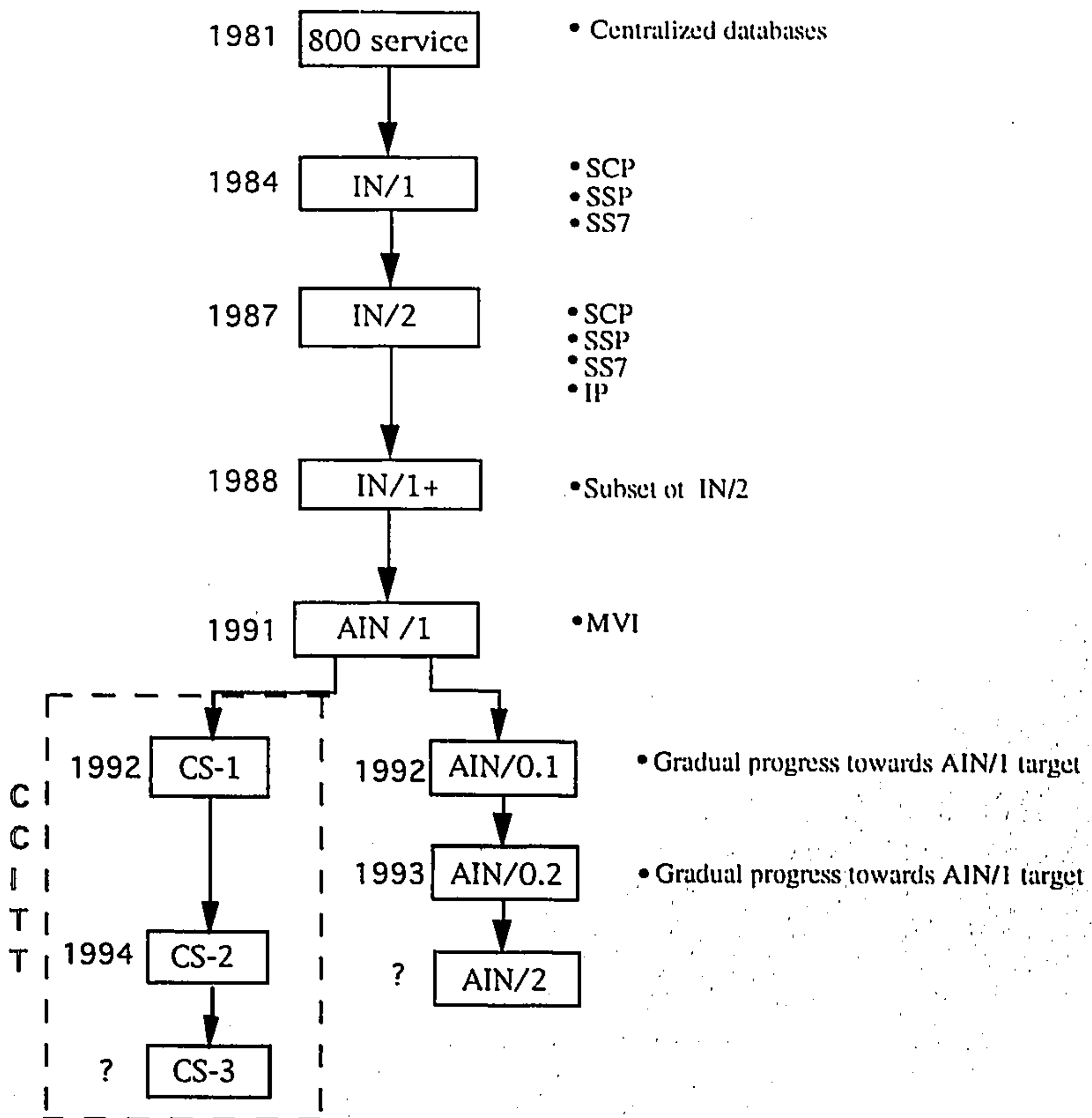


Figure 2.1 Evolution of the Intelligent Network [DMK]

2.2 CCITT (ITU-T) IN Conceptual Model

The ITU-T intelligent network concept model is an architectural concept that meets the needs of telecommunication service providers to rapidly, cost effectively, and differentially satisfy their existing and potential market needs for services, and to improve the quality and reduce the cost of network service operations and management [GRKK93]. This intelligent network architectural concept has the following characteristics:

- it is applicable to all telecommunications networks, e.g., PSTNs, ISDN, B-ISDN, packet switched public networks, mobile networks, etc.;
- it enables service providers to define services independent of service specific developments of equipment vendors;
- it enables network operators to allocate functionality and resources efficiently, independent of network specific developments of equipment vendors.

The intelligent network concept model is illustrated in Figure 2.2. It consists of four planes representing different levels of the IN abstraction:

- a service plane;
- a global functional plane;
- a distributed functional plane;
- a physical plane.

This conceptual model shows the IN as an integrated framework, within which all other IN concepts are identified, characterized, and related to each other.

It is essential to distinguish between intelligent network architecture and the intelligent network conceptual model. The architecture will evolve with increasing service requirements and emerging technologies, but the conceptual model is intended to remain consistent; and it provides better understanding of the intelligent network concept. Table 2.1 illustrates the relationship between the implementation and the conceptual model.

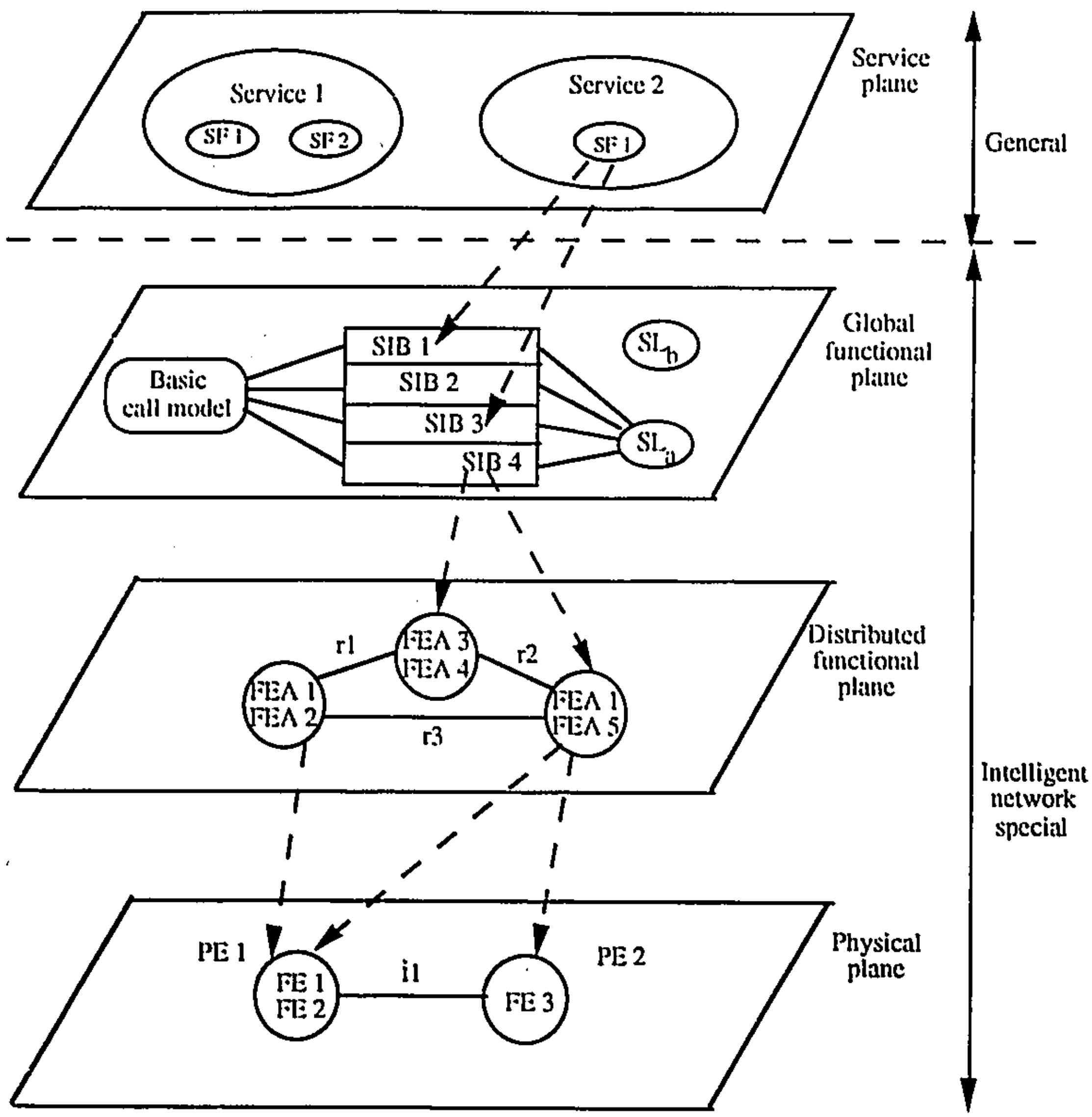
Conceptual Model	Implementation
Service plane	Services (the plane users see)
Global functional plane	Service-independent building blocks (SIBs)
Distributed functional plane	Functional entities (e.g., service switching functions, etc.)
Physical plane	Physical products (e.g., SCP, SSP)

Table 2.1 IN Conceptual Model vs. IN Architecture

2.2.1 Service Plane

The service plane provides a view that is exclusively service-oriented. It describes services from a user's perspective without indicating how the services are implemented. The service plane consists of the smallest functions called service features (SFs). A SF is larger than a service-independent building block (SIB) but smaller than a service. A service is realized by grouping SFs.

An important issue of the service plane is the service or the service feature interaction. The service interaction means that undesired effects occur when two or more services or SFs are used together. The SFs on the service plane are mapped to the global functional plane (GFP) by combining SIBs on the GFP using global service logic (GSL).



SF: Service feature
SIB: Service independent building block
SL: Service logic
FE: Functional entity
FEA: FE action
rN: Relationship N
PE: Physical entity
iN: Interface N

Figure 2.2 ITU-T Intelligent Network Conceptual Model

2.2.2 Global Functional Plane

The global functional (GFP) describes units of service functionality, referred to as service-independent building blocks (SIBs). SIBs are the smallest building block that can be found in an intelligent network. A SIB is defined as "a standard reusable network wide capability residing in the global functional plane used to create service features" [JaTh94]. The basic call process (BCP) is a special SIB that handles all activities necessary for a normal call. SIBs are combined by the GSL to provide a service feature. SIBs are service-independent. The GSL is service-dependent. The description of how the SIBs are linked together (or the GSL) is often referred to as a service script.

The first set of SIBs that will be commonly used, the Capability Set 1 (CS-1), is defined by the ITU-T (CCITT). The CS-1 is not a standard, but only a recommendation. The successors CS-2, CS-3, etc., are intended to be standards. The subsequent capability sets will place emphasis on the mobility and inter networking between intelligent networks.

A SIB on the global functional plane must be found in one or more functional entities on the distributed functional plane (DFP).

2.2.3 Distributed Functional Plane

The distributed functional plane (DFP) describes the functional architecture of an intelligent network in terms of units of network functionality, referred to as functional entities (FEs), and the information that flows between functional entities, referred to as relationships. The DFP architecture is vendor and implementation independent because the functional entities and information flow describe independently how the functionality is physically implemented or deployed in the network. The intelligent network CS-1 and the DFP architecture are shown in Figure 2.3.

Call Control Agent Function (CCAF)

This function handles user access to the services. It resides in local exchanges.

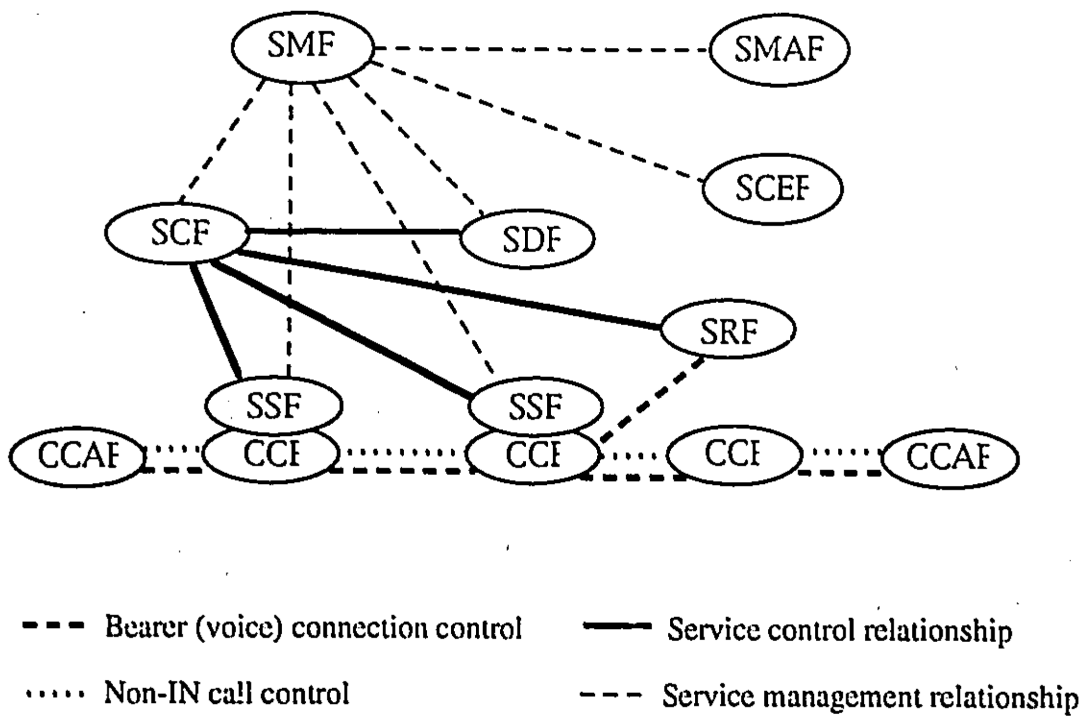


Figure 2.3 The Distributed Functional Plane

Call Control Function (CCF)

The CCF or call processing provides the means for establishing and controlling the bearer service on behalf of the network users. The CCF can detect service triggers of intelligent networks.

Service Switching Function (SSF)

This function determines when the IN service logic should be invoked and how the switching system interacts with IN service logic. The SSF is responsible for switching a call or service to a particular location based on the instructions of the service control function.

Service Control Function (SCF)

This is the core of the intelligent network. This function controls the whole process of handling calls or services by giving instructions to the SSF/CCF, the SDF (service data function), and the SRF (specialized resource function). Service logic and associated data are found only in the SCF.

Service Data Function (SDF)

This function assists the SCF with data about the customers and the networks.

Specialized Resource Function (SRF)

This function plays the role of interface between users input data and the SCF. It prompts queries to a user, and collects the user response.

Service Creation Environment Function (SCEF)

The task of this function is to define, develop, and test IN based services.

Service Management Function (SMF)

This function virtually interfaces with all other function entities. It is used to load and manage the service logic.

Service Management Agent Function (SMAF)

This function operates as the interface between operation/maintenance staff and the SMF.

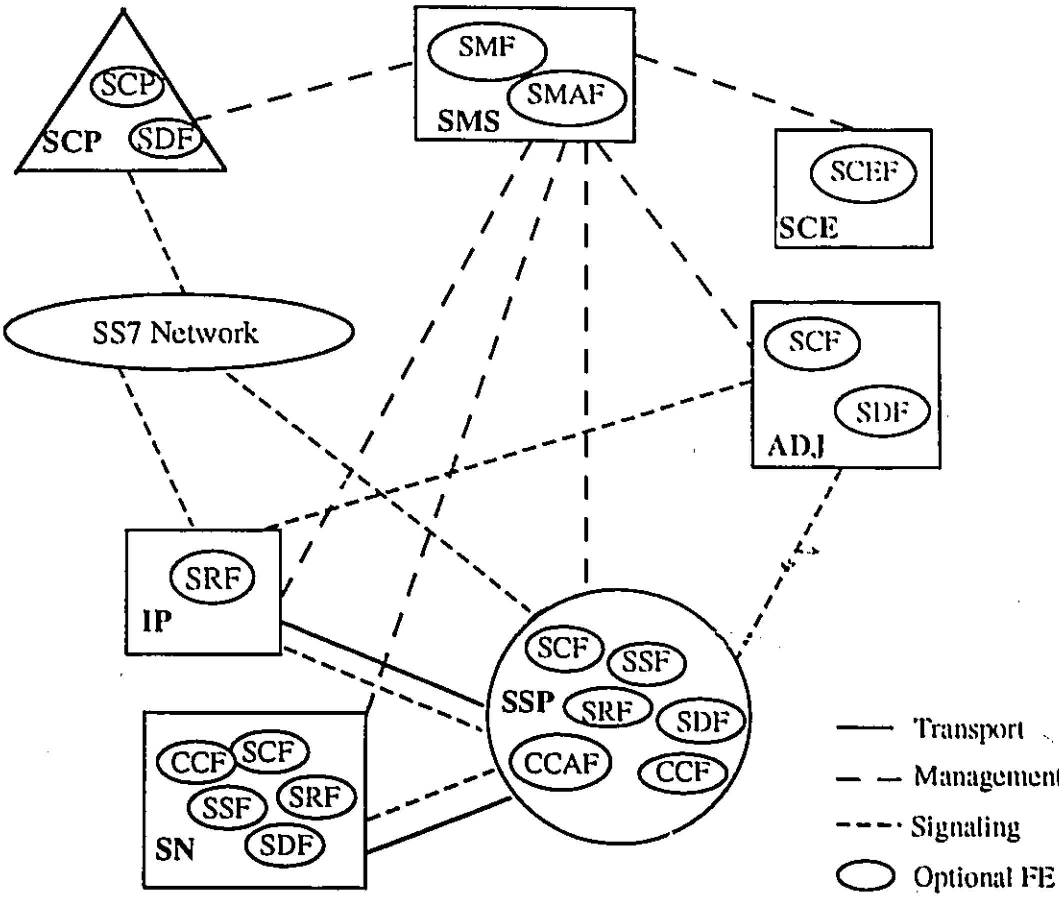
2.2.4 Physical Plane

The physical plane shows how the DFP functions can be mapped into physical products on the networks. In chapter 1, the physical components of intelligent networks, namely SSP, SCP, IP, SN, ADJ, SMS, SCE, are explained. These physical nodes are the concern of the physical plane, that is, how an FE on the DFP is allocated to a physical entity (or physical node). Figure 2.4 shows a physical plane architecture.

Each FE must be found in at least one PE, but a PE can consist of more than one FEs of different types. If for some reason, for example, it's needed to load two SCFs in the same place, these two SCFs must be regarded as two different PEs, that is, two SCPs. Note that not all PEs would be needed for a particular intelligent network implementation. The PEs required are dependent on many factors such as existing network characteristics, deployment plans, types of services being deployed, and so on.

2.3 Technology behind the Intelligent Network

As described in previous sections, the primary objective of the intelligent network is to reduce the time (and of course, the cost) required to introduce a new service in the network. Conventional techniques and conventional software programming require somewhere between two to five years from service conception to service deployment and operation (at cost up to $5 millions)[DMK92]. With intelligent networks, this time can be reduced to the maximum of six months. How the intelligent network makes this possible? The answer is given in the following.



PEs (physical entities):
ADJ adjunct
IP intelligent peripheral
SCP service control point
SSP service switching point
SN service node
SCE service creation environment
SMS service management system

FEs (functional entities):
CCF call control function
CCAF call control agent function
SCF service control function
SDF service data function
SRF special resource function
SSF service switching function

Figure 2.4 IN Physical Plane Architecture

2.3.1 Separation of Functionality and Using Centralized Control

The approach that the intelligent network uses is "separation" and "centralization": separate the functional architecture from the physical architecture, or separate the service functionality from physical devices (separate connection control from service control); distribute limited software control; provide centralized access to the shared data. The communication among function entities is achieved via the common channel signaling networks (SS7). With the above method, the work required for providing new services can be reduced to create or modify the service logic and relevant databases in one or several SCPs, rather than to "squeeze" software to a large number of switches. The work is further speeded up by using the reusable SIBs which are provided in the intelligent network platform.

Call control and call switching functions have historically been tied up with each other in the telephone exchanges, with no possibility of separating them. A call is set up according to a step-by-step procedure, with each node deciding to which node to proceed next, making it impossible to obtain an overview of the call set up from one location. The service logic is embedded in every switch that the service might invoke.

When the service control, such as a number translation service for 800 calls, is moved from switches to a SCP, control of the whole service is centralized at a single point in the network. This point, which is the SCP, has a total overview of the execution of the service, including logic and data. If the service is changed or withdrawn from the network or a new service is introduced, it is mainly the SCP that will be affected. The switches (SSPs) become "dumb" switches, knowing nothing about the service logic, but only detecting the triggers (explained in the next subsection) of services, and switching according to the order of the SCP.

2.3.2 Call Processing Model and Triggers

The capabilities of the intelligent network are based on its call processing model (or call model) which is built on the call processing infrastructure of existing digital switches. The call processing model is the set of states (or Points In Call, PIC), transitions, and trigger check points (or detection points), which are used to illustrate the different states that a call can go through, from the origination (the user picks up the phone) to the termination (the user hangs up). Figure 2.5 [FS89] illustrates a generalized set of states through which a call will progress during the call processing.

A trigger refers to an event in the call processing that is made visible to the IN service logic. When a trigger is encountered, the call processing will be suspended at SSP, to allow a query to be sent to the SCP and a subsequent set of instructions to be returned by the SCP. The call processing will then be resumed after the SSP receives the instructions of the SCP. The trigger check points are the places where the SCP can affect the way of call processing, that is, manipulate and control the call processing to provide services.

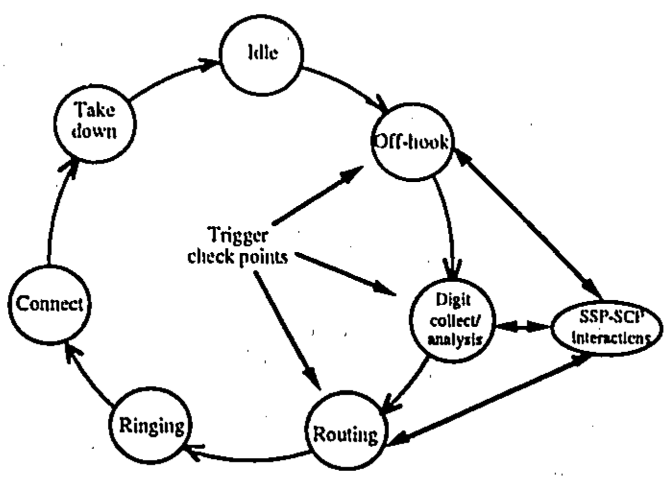


Figure 2.5 Generalized Call Processing Model and Triggers

For example, the call processing model in Figure 2.5 provides three trigger check points, namely the off-hook, digit collect/analysis, and routing trigger check points. As soon as a user picks up the phone, the off-hook trigger is encountered. The SSP checks the number of that telephone in a table to see if an off-hook trigger is set for that phone. If not, the call processing proceeds. Otherwise, the call processing is suspended; and the SSP sends a query to the SCP. The off-hook trigger can support hot line services that automatically route calls to particular destinations. For example, the emergency phones on campus will be directly connected to a police office, or a hot line phone in an airport will be automatically routed to a taxi company. The digit collect/analysis trigger provides a place for the SCP to carry out a broad range of service logic, e.g., the service of "pizza order", described in chapter 1. When the unique "pizza number" (e.g., 737-1111) is dialed, the SSP checks whether the digit collect / analysis trigger is set for that telephone number. If so, the SSP sends a query to the SCP. The "pizza order" service will be invoked in the SCP. The SSP then will process the call according to the service logic returned by the SCP.

It can be seen that the more trigger check points a call model has, the more services can be provided. For instance, if the ring and busy trigger check points are added in the call processing model shown in Figure 2.5, the services such as call forward, call ring back can be provided. Each trigger in a call model is service independent. The evolution of intelligent networks will expand on the variety of triggers. The AIN/1 call model is presented in Appendix 2. A flexible, generalized trigger table is a key SSP capability that enables quicker service provisioning.

2.3.3 Supporting Network Intelligence via Common Channel Signaling System Number 7 (SS7)

As described in above subsections, the magic of the intelligent network is to separate function entities resulting in the SSP and the SCP. The communication between the SSP and the SCP is achieved through an SS7 network which uses a separate channel from the voice connection and uses packet information transfer. The SS7 network delivers SSP's query message (TCAP message) to the SCP, and SCP's response message (TCAP message) to the SSP. So the SS7 network is an integral part of the intelligent network architecture.

2.4 Issues in AIN Paradigm

Since AIN technology is fairly new, a number of significant technique details need to be carefully addressed and formulated. The following are some of AIN issues which are discussed in the AIN literature.

2.4.1 Service Creation Environment (SCE) and Service Creation Process

Service creation is the cornerstone of intelligent networks. The SCE is a part of the IN architecture, which facilitates the fast service creation process. The success of the intelligent network depends heavily upon building a comprehensive SCE [EFH92]. A reuse-driven SCE, which employs the service independent building block (SIB) concept, is described in [BSK94]. Based on a reusable and pre-tested set of SIBs, service providers can respond to the market in a timely manner, and the service productivity, quality, and reliability can be assured. Through international standardization efforts and further research studies, the SIB concept will become a reality in the near future to enhance the service creation speed and quality. Paper [MOYN94] describes designing of the SCE with the service software verification method. The verification method avoids actual machine tests as much as possible, and provides the verification through a service software automatic generation system and simulation using an IN service simulator. Use of this SCE shows that the manpower necessary for verifying service logic programs can be decreased.

In [DSDG92], the requirements for service creation platform, which can progressively improve flexibility and accelerate the service creation process, are presented. The service creation process involves seven steps: conceptualization, definition, specification, development, verification, deployment and monitoring. To enable the faster creation of new services, the inter-operable, compatible and flexible tools are required to support the service creation life cycle. Services must be specified in unambiguous detail without limiting flexibility for modification and customization. Papers [YNNU92], [DaH92] and [HCW94] addressed the issues of methods for the customer service requirements collection, the opportunity to personalize service for individual customers, the service specification process, and the verification contents of service specifications.

2.4.2 Service (or Feature) Interaction

The service/feature interaction occurs when services interact with each other and consequently cause adverse behaviors which were not intended by the designer. Take the automatic alarm service and the call forward service as an example. The automatic alarm service is used to alarm people at a specified time; the call forward service is used to forward an incoming call to other people when the incoming call is not answered within a certain period of time. The design of these two services is completed, posing no technical problems when used together; however, it might cause unexpected results: a morning wake-up call is unexpectedly forwarded. So the service interaction must be carefully considered before introducing a new service to the network. Testing all possible combinations of services is too complicated to accomplish practically. Although a good service (feature) specification with complete service (feature) assumptions is important, a general solution is needed for dealing with the bulk of interactions without feature-to-feature, interaction-specific logic.

In [YW92], knowledge expressing the relationship between service functions and service interactions is used. The knowledge is compared with service features extracted from a new service specification by experts; the experts must first extract the service features from the new service specification for automatic detection. Since this approach depends on the extracted service features, it is not always good. In [RB92], service specifications are specified by an executable language and the designers detect service interactions by using an interpreter. [ITO94] intends to automatically detect adverse behaviors of feature interactions by analyzing the new service specification along with existing service specifications that are specified in accordance with the state transition model.

2.4.3 Common Channel Signaling Network (SS7)

As described previously, the common channel signaling system plays an important role in the intelligent network. At present, the traffic load applied on the SS7 network is relatively low, thus there are no performance problems in terms of delays and blocking of packets traveling through the SS7 networks. However, as the number and complexity of AIN services increase, the load applied on the SS7 network will increase. As a result, the performance of the signaling network might degrade, prohibiting the application of new AIN services, or even penalizing the existing services. For the realization of global intelligent services, the planning and management of the SS7 network need to be addressed carefully.

In paper [FLM94], the authors discuss the issues related to the definition of end-to-end performance objectives to SS7 network elements. Two categories of performance objectives are defined for SS7 network planners: availability objective and utilization objective. Paper [MFW94] identifies various scenarios where inter network signaling interactions would take place in the framework of the IN, and then identifies various requirements to cope with the scenarios.

2.4.4 Congestion Control

Due to the centralized control architecture of the intelligent network, the SCP handles all the service requests from SSPs. To prevent the SCP from overloading, congestion control is required. The call gapping and windowing techniques can be used in congestion control for intelligent networks. The call gapping refers to a technique which only allows a pre-determined number of calls to make requests to a SCP per second. If the traffic demand is higher than that number, only the pre-determined number of calls are sent per second; the rest of calls are blocked or temporarily stored. When call gapping is introduced on all connections from SSPs to the SCP, the traffic from the SSP to the SCP can be controlled. However, this doesn't guarantee that the SCP is not overloaded, since one call might involve several queries which need to be processed by the SCP.

The window technique takes into account the traffic load on the SCP. The window size determines how many SSP call requests can be accepted by the SCP based on the number of outstanding calls, that is, the number of calls that are being processed and haven't been finished. When windowing is used from all SSPs to the SCP, the actual load on the SCP decides the traffic flow from each SSP, so the risk of overloading the SCP is less than when call gapping is used. Call gapping and windowing are studied by simulation in [PB94].

2.4.5 Application in Mobile Communications

One of the most promising areas for the application of the intelligent network concept is in the Personal and Mobile Communication Networks (PMCNs). Current IN architecture lacks the ability to support terminal mobility. In [BJ92], the intelligent network concepts in mobile communications is described. The author presents an architecture for the mobile intelligent network along with various implementations and discusses major IN aspects in mobile networks. The functional architecture which enhances the existing IN architecture to provide both wire line and wireless networks services is also proposed in [SHJP93]. In [JHSH92], the authors discuss the IN requirements for implementing a personal communication network. They discuss the trigger check points of the SSP and functional entity actions of the SCP and the requirements to implement IN personal communication services.

Chapter 3

Design and Implementation of AIN Services

In this chapter, the platform of service creation and the service creation process will be first described. Then, the services we created will be presented.

3.1 Platform for Service Creation

Three services, namely, the automatic directory service, the call by name service, and the information query service, are designed and implemented at the AIN platform which consists of the Integrated Service Control Point (ISCP) and the associated Bellcore SPACE software.

3.1.1 Integrated Service Control Point (ISCP)

The ISCP is the service control point of the intelligent network developed by Bellcore, and it functions as network service database. This ISCP is an on-line, real-time, faulttolerant database system that provides call treatment information in response to the query from the SSP. The application software with service-defining logic to create individual call-handling instructions are stored in the ISCP. The ISCP determines what actions should be taken, then issues special call handling instructions to the SSP.

The ISCP node achieves very high reliability through hardware and software redundancy coupled with automated error detection and recovery logic. To achieve an even higher degree of reliability, application traffic may be load-shared by two distinct ISCP nodes. In this mated configuration, application traffic is distributed between the two ISCPs by the STPs (Signal Transfer Points in SS7 networks). If one of the ISCPs fails, the remaining ISCP takes the full traffic load until the failed ISCP is brought back into service. For mated operation, each of the ISCP mates must contain a full copy of application customer data and must have sufficient processing capability to handle the full traffic load.

The hardware architecture of the ISCP is shown in Figure 3.1. The hardware platform for the ISCP is the call processor. For hardware redundancy purposes, there are two call processors for one ISCP. Each call processor is connected to each other by duplex Local Area Network (LAN). The maintenance and operation console (MOC) operates and maintains the ISCP. A local MOC, located at the ISCP site, accesses the call processors through duplex LANs. An optional remote MOC accesses the local ISCP over a dedicated line. An optional dial-in MOC accesses the local ISCP over a dial-up line.

What network services are supported by an ISCP depends on which applications are installed. In future versions, the ISCP will be capable of providing simultaneous services for multiple applications, depending on the amount of traffic and data required per application. In version 1.0 of the ISCP, one application is available: the Multi-Service Application Platform (MSAP).

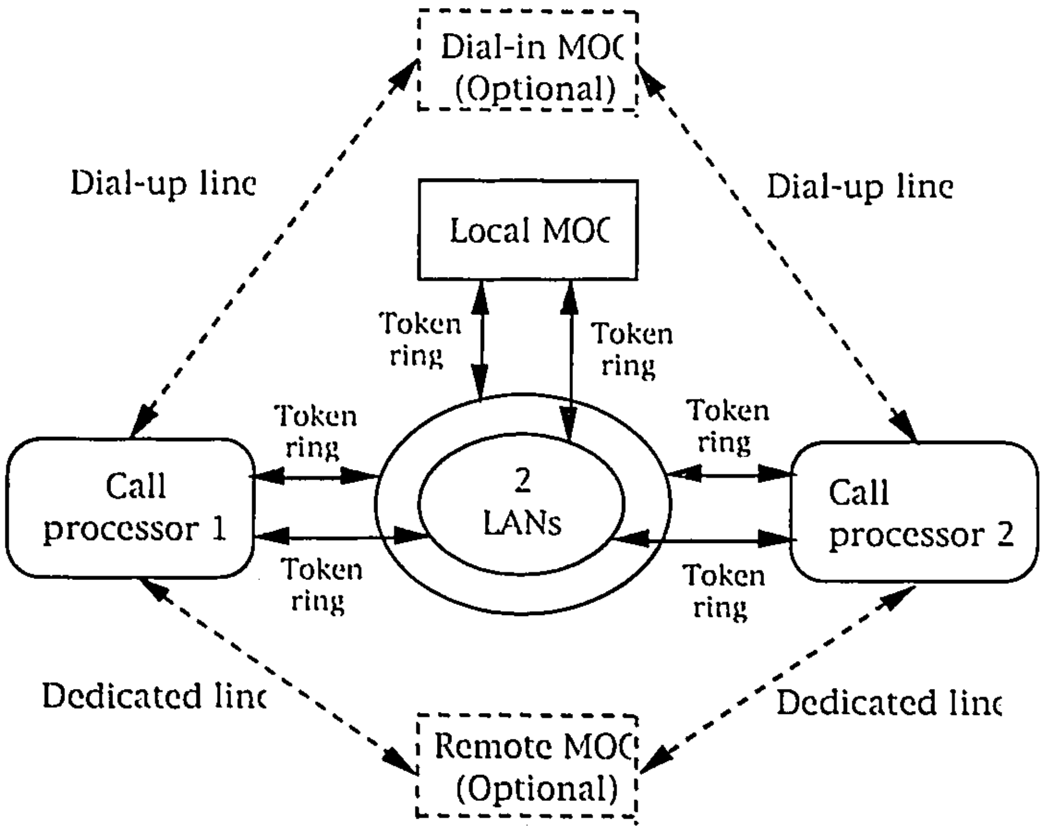


Figure 3.1 ISCP Hardware Architecture

3.1.2 Multi-Service Application Platform (MSAP)

The MSAP provides real-time execution of the services defined by the SPACE system which is the operation support system for the ISCP. After a subscriber record is created, validated, and tested by the service designer on the SPACE system, it is sent to the MSAP for live call processing. As queries arrive at the ISCP from the SS7 network, they are routed to the MSAP for processing. There are two categories of MSAP processing:

- call processing;
- administrative processing.

The call processing includes the functions required to process a query and to format a response, such as:

- parsing the query;
- determining what service logic to invoke;
- interpreting the service logic to determine the call treatment
- formatting a response.

When the MSAP receives a query, it derives a database access key through service key analysis and retrieves a Call Processing Record (CPR) from the database. The CPR references feature logic that the MSAP retrieves. The retrieved feature logic contains logic operations and data used to direct call processing. Based on the results of interpreting the feature logic, the MSAP uses directives in the CPRs to determine the call treatment (e.g. route the call, block the call, or play an announcement). The call treatment is encoded in the response and returned.

The administrative processing consists of subscriber-specific administrative functions. The need for administrative processing is determined during call processing. The administrative functions include:

- sampling at application level;
- sampling at customer level;
- collecting selected call data;
- sampling the number of attempts;
- sampling the number of completion.

3.1.3 Service Creation Tool and Terminology

The ISCP provides a workstation facility that allows telephone company personnel to define, create, test, and activate IN services. These tasks are accomplished using the

SPACE software of Bellcore. With the SPACE, the service designer and provider can perform the following tasks in a fast and cost-effective way:

- Create new services;
- Customize previously designed services using service templates;
- Perform service validation and testing to confirm that the service logic performs correctly;
- Activate, or deploy, services into the live network environment;
- Administer and maintain services.

Services are created using Call Processing Records (CPRs), Call Variables (CVs), tables, templates, and General Service Specifications (GSSs) that are stored in the SPACE database. After a service is created, it needs to be validated, tested by local red-line and remote red-line traces, activated, and enabled. Each of these service creation components or phases is described below.

CPR

The CPR is a graphic representation of a service. Using one or more nodes (provided by the SPACE, and each represents a particular action or decision) in one or more graphs, a CPR displays the flow of decisions and actions that are made during the call processing. There are three types of CPRs; each is denoted by a trigger. The CPR trigger is based upon whether the call processing logic will apply to incoming calls, outgoing calls, or whether the CPR is accessed during call processing to obtain the supplementary call logic. The incoming call trigger is called the Dialed Line Number (DLN) trigger; the outgoing call trigger is the called Automatic Number Identification (ANI) trigger; the last type of trigger is called Hand-over trigger. When the trigger is the ANI or DLN trigger, the CPR key is the customer's telephone number. The CPR key can be any alphanumeric string (up to 13 characters) when the trigger is the Hand-over trigger. A

CPR is built by connecting the graphic nodes (called CPR nodes), provided by the SPACE system

CVs

The CVs identify the data elements whose values are processed by service logic execution routines. CVs are defined by a tag name, scope of variable, data type, and an optional initial value. There are two types of CVs: pre-defined and user-defined. Pre-defined CVs are those CVs whose name, scope, and data type are defined in the service definition interface. User-defined CVs are defined by service designers.

Table

The table is a single or multi-columns lists of data that are used when processing a CPR. Not all CPRs have tables associated with them. The table key may be any alphanumeric string that the user assigns to it.

Template

The template is a master for a service CPR logic from which copies (or instances) can be made and customized to satisfy different customer service requirements. A template is made from a CPR. Not all of the call processing logic in the CPR is made available for customization when the CPR becomes a template. In this way, nodes that are critical to the implementation of the service are kept the same for each customer that uses the service.

GSS

The GSS is a written description of a service, followed by lists of required and optional nodes that are to be used when creating the CPRs that graphically represent the service. The GSS key may be any alphanumeric string that the user assigns to it. One GSS is delivered with the SPACE software: "generic". This GSS is used when no other GSS can be associated with a CPR. The generic GSS is useful for service creators who are experimenting with new services because it is defined with no nodes in the required nodes list and all nodes in the optional nodes list, which makes any combination of nodes allowable to a CPR.

Validation

The validation is the means for determining whether the CPR is true to its associated GSS and whether the CPR logic is acceptable by the call processing software. Any CPR that is associated with an active GSS may be validated. In order to pass validation, any CPR must be designed to use the nodes indicated by the GSS with which it is associated.

Local red-line testing

The local red-line testing is used to test the call processing logic for CPR records that have not yet been sent to the MSAP. However, when any change is made to a CPR (even to the one that has already been sent to MSAP), the local red-line testing should be performed again.

Remote red-line testing

The remote red-line testing is used to test the call processing logic for a CPR that has been sent to the MSAP. The remote red-line testing mechanism stores trace information for a CPR each time a call is made to that CPR from the MSAP environment.

Activation

The activation causes CPRs and tables to be sent to the MSAP and ultimately to be used in the live telephone environment.

Enabling

The enabling freezes a GSS or template record so that no further changes can be made. when a GSS record is enabled, it can be associated with a CPR. When a template is enabled, it is available for customization.

3.3 Service Creation Process

Figure 3.2 illustrates the procedures for creating an AIN telephone service, using the SPACE software.

A GSS captures service-specific information that is relevant to each CPR associated with it. The GSS associated with a CPR defines what nodes are required and what are optional for that CPR so that it represents the service the GSS describes. A GSS is created prior to and independently of a CPR. The GSS defines the scope for the service that the CPR represent. Once a GSS is complete, it should be enabled. The enabling process prohibits any further modifications to the GSS so that records associated with it can be validated against the static information.

After a CPR is created by combining the CPR nodes based on the service logic and by setting up proper call variables and tables, the CPR should be validated. The validation facility determines if the CPR has met the criteria stated in the GSS with which it is associated. It also checks whether the CPR logic is appropriate or not. The validation software provides ERROR and WARNING messages to the service designer. The error must be corrected before the CPR can be activated.

Once the CPR has been successfully validated (no error messages associated with the CPR), it should be tested using the local red-line trace facility. Local red-line trace simulates the placement of a call using the CPR that has been just created and validated, and allows the designer to see whether the CPR works correctly. If the local red-line trace facility encounters any problems in processing the simulated call, it displays messages to inform the nature of the problem. If the tested CPR causes any red-line trace error messages, the problem need to be corrected with modifying the CPR and to be re-tested by red-line trace.

The next step in the service creation process is to create a template from the service CPR. A template preserves the service logic in the CPR, but allows certain values defined in the CPR to be customizable. When the template is made, adding nodes to the graph is not permitted. The customization is achieved by declaring values for nodes or branches based on the customer specific needs, for example, changing the value of the Carrier node to use different carrier, or changing a time value of a decision node to route a call based on different time requirement.

After a template is defined from a CPR, it is ready to be enabled. The enabling process prohibits further modifications to the template, so that instance CPRs associated with the template implement exactly the same service.

Once the template has been enabled, instance CPRs can be created from it. Typically one instance CPR is created for each customer using the service. Instance CPRs must be validated and tested in the same way in which initial CPRs are validated and tested.

When the initial CPR or the instance CPR has passed validation and has been successfully tested using local red-line trace, the CPR is ready to be activated. Activation makes the CPR available for live traffic to use.

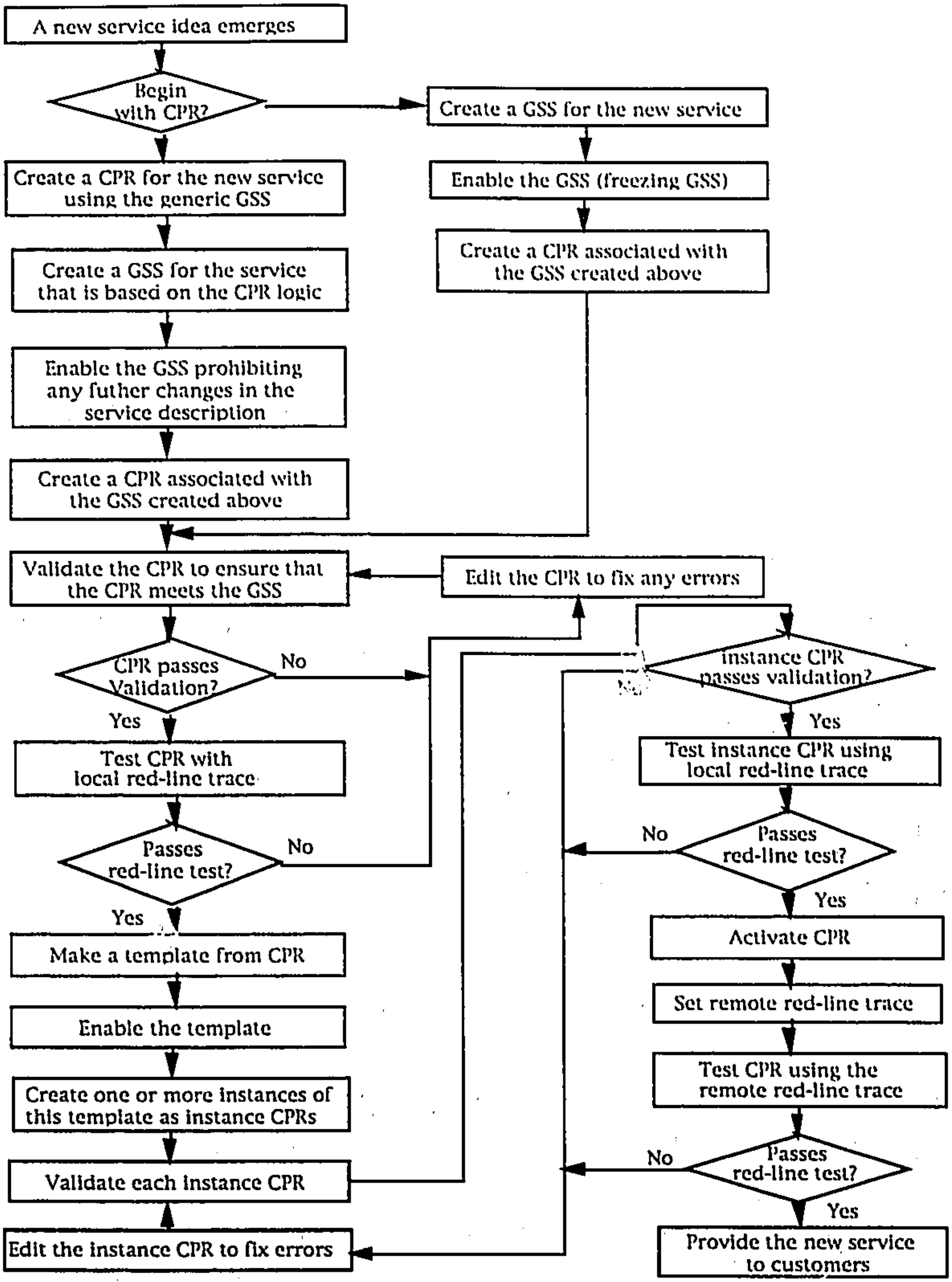


Figure 3.2 Service Creation Procedure

To perform a more thorough set of tests on the newly-activated CPR, the remote red-line trace flag for the CPR is set after passing the local red-line test. An expiration date and time is established for the flag that limits the duration of the call trace messages that are received. Once the trace flag is set, the call processing software sends red-line trace messages back to the SPACE software for each live traffic call which uses the CPR. These remotely generated red-line trace messages are viewed from the SPACE system's user interface in the same manner that locally generated red-line trace messages are viewed. When the live traffic tests are successful, the service is ready to be used by customers.

3.3 AIN Services

We have proposed several services in our project report [MYBG95]. Below we describe three of them.

3.3.1 Call by Name Service

We might have the experience that we want to call a person or organization but could not remember the telephone number. The Motivation of Call by Name Service is to allow the caller to key in an alphanumeric string. The alphanumeric string can be a person's or organization's name, or the alias for the person or organization which can be easily remembered. For the simplicity of the description, this alphanumeric string is referred as the name of a telephone. The call by name service is responsible for translating the name to a telephone number which is associated with that name. The service procedure is described as follows.

Step 1

A user picks up a phone, and presses a special key (e.g., #) or a sequence of keys to indicate string input rather than digit input. When a SSP receives the signal of the special key, it knows that the string input function is required, and prepares to collect the caller's input at string input mode.

Step 2

The user inputs each characters of the name with a proper sequence. Figure 3.3 shows the layout of a typical telephone key pad.

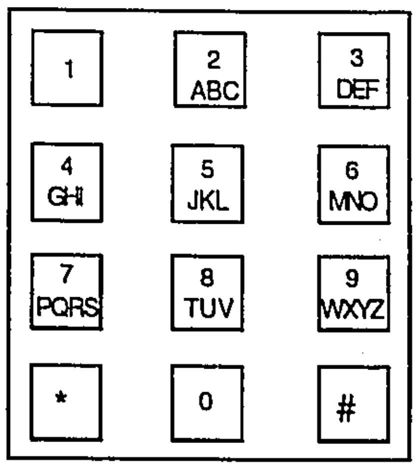


Figure 3.3 Layout of Telephone Key Pad

Each character can be represented by two digits. A simple way to achieve the character representation can be like the following: first pressing the digit where the character resides, then pressing the digit which is the position the character occupied. For example, "K" is input by pressing 52, "Y" is input by pressing 93. The digit "1" can be used to separate the words in a name. To indicate the end of the user input, the special key "#" is pressed.

Step 3

After the system receives the user input, it plays the announcement repeating the name of the user input, and asks the user to verify the input by pressing a pre-defined key, e.g., pressing 1 for the right name, or pressing 2 for re-input. If 2 is pressed, the system goes to step 2 again, otherwise, it goes to step 4.

Step 4

In this step, the SCP searches the "name to number" database, and looks for the telephone number associated with the name (assuming the name is a unique search key). When the SCP successfully finds the telephone, it sends the number to the SSP, and the SSP makes connection based on the number received from the SCP. In case the number is not found by the SCP, the system informs the caller of the problem , and provides an optional way to solve the problem by allowing the user either to use the service again (going to step 2) or to use the automatic telephone directory service (which is described next).

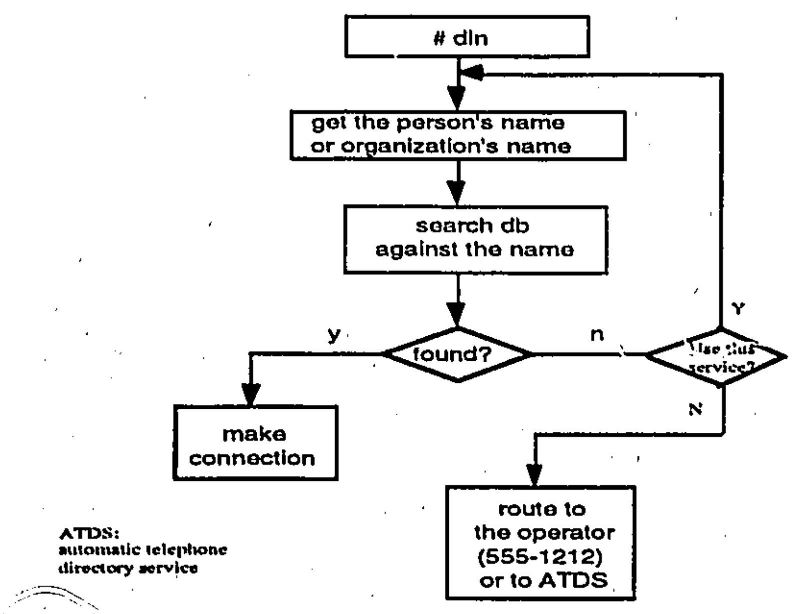


Figure 3.4 Block Diagram of Call by Name Service (implemented)

3.3.2 Automatic Telephone Directory Service

Every year, telephone companies in the U.S. spend over \$1.5 billion providing directory assistance service [MLGB94]. It usually takes an operator about 25 seconds to complete a directory assistance call. Reducing one second in this average processing time means savings of over \$60 million a year. The purpose of automatic telephone directory service is to speed the directory assistance call processing by automation, and to reduce the work of human operators.

The service is invoked by dialing a universal number, e.g., 411-1111. When the SSP gets these digits, the DLN trigger is detected. The SSP initiates a query and sends to the SCP, where the automatic telephone directory service logic are stored. The service can be provided by using only the dial-key telephone, or the dial-key telephone and the speech recognition facility. The service procedure is as follows.

<u>Step 1</u>

When the service is activated, first it prompts the user for the preferred language of interaction. For instance, pressing 1 represents English; pressing 2 represents French (assume that only English and French are used).

<u>Step 2</u>

If the speech recognition system is used, the user might be asked to repeat some words or expressions so that the system can get used to the voice and the accent of the user. If for some reason the system is unable to process the user's speech, it will inform the user to use the dial-key pad of telephone to response the prompt of the system.

<u>Step 3</u>

The system asks if the person or the organization that the user wants to research is in local or remote. The indication of local and remote can be defined as: pressing 1 for local, pressing 2 for non-local (remote). If the number of person (or organization) is local, the system goes directly to step 7.

Step 4

When the number is long distance, the system asks if the person or organization is within North America (Canada, U.S.A) or international (Europe, Asia, Africa, Australia, Central or South America). If the call is within North America, the system goes directly to step 6.

Step 5

When the number is long distance and is outside North America number, the system asks the user to give the name of the country (through the dial-key pad or speech recognition facility) or the country code (through the dial-key pad of the telephone). The system validates the user's input and allows the user to confirm or re-enter it, so that the wrong input can be corrected. However the user can be allowed to enter and re-enter the input for the pre-defined number of times. If the number is exceeded without having a correct input, or the next step decision (routing of the call to the directory service system which serves the country the user desired) can not be made based on the user's input, the system switches the user's call to a human operator.

Step 6

The system asks for the area code (or the state, province, city, town). The procedure is similar to the one in step 5. After collecting this information, the system tries to find the area code; if successful, then it routes the user call to the directory database server of that area; if failed, it informs the user and connects the user to a human operator.

Step 7

After the system successfully identifies the directory database of the required area, it tries to find the number for the user. This task can be carried out by a service logic similar to the call-by-name service logic described earlier (see section 3.3.1). If the name is not unique in the database, the system informs the user about the number of items on the list and asks if i) the user can provide additional information, or ii) the user prefers to hear the list with additional information and allows the user to make a choice, or iii) asks the user to provide additional information. The possible additional information that the system could ask are the following:

- the area which the called party is located;
- the possible address of the called party (if it is known to the caller);
- part of the postal code (if the caller remembers);
- a few of the digits in the telephone number (if the caller remembers);
- the type of profession (e.g., education, engineering, automobile sales, etc.).

What information is contained in the system depends on the database design of the system. After one or several telephone numbers (allowing user to try dialing the number one by one), the service is finished. If one number is presented to the user, the service might go to step 8, if the user wishes so. In case the system is unable to find a telephone number for the user, it informs the user, and then directs the call to a human operator, if the user wants to do so, otherwise it terminates this automatic directory service.

Step 8

After finding a number based on the searching criteria provided by the user, the system asks if the user wants the system to dial the number automatically, and acts according to the response of the user. This feature is quite useful to some users who have difficulty in recording the number (e.g., when users call from a pay-phone or through the cellular while they are driving). If automatic dialing is not required, the service is finished.

The complete service logic is presented in Figure 3.5. Figure 3.6 shows the implementation of the service using the SPACE software. Due to the limitation of the available functions of SPACE, not all the features of the service designed have been implemented.

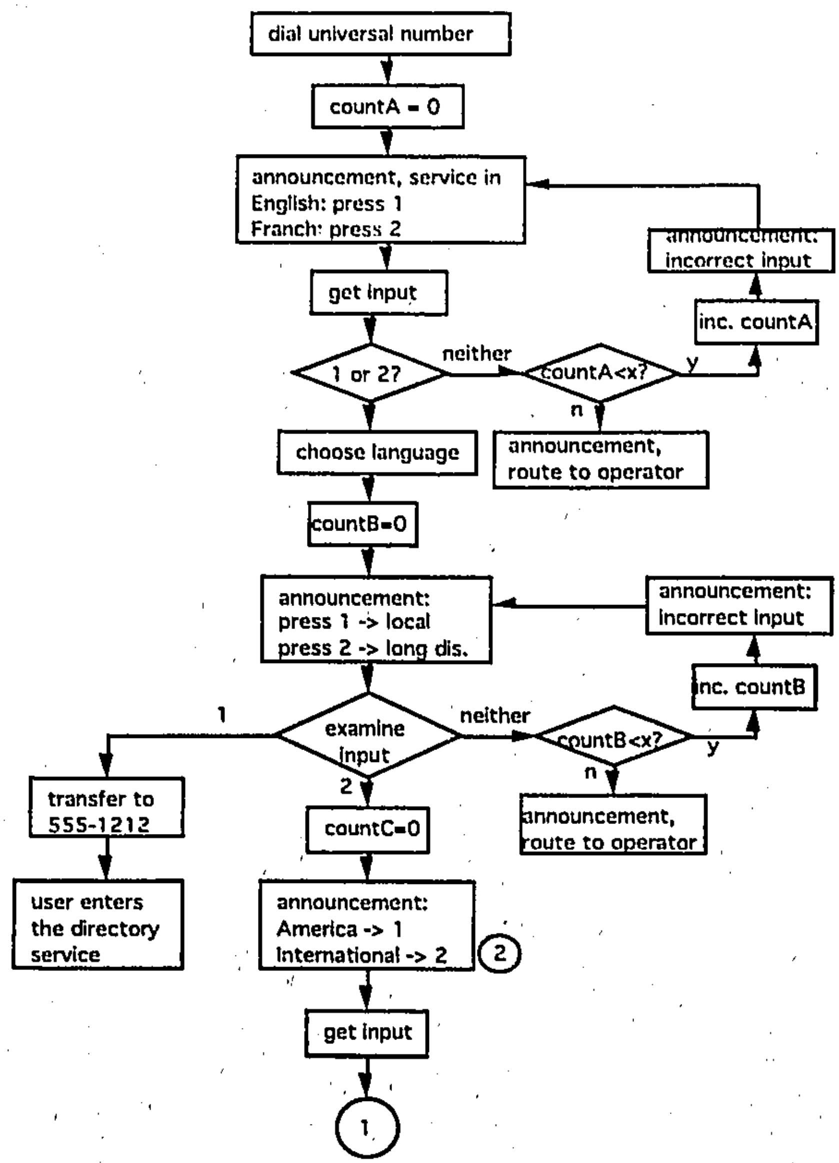


Figure 3.5 Automatic Telephone Directory Service (cont'd)

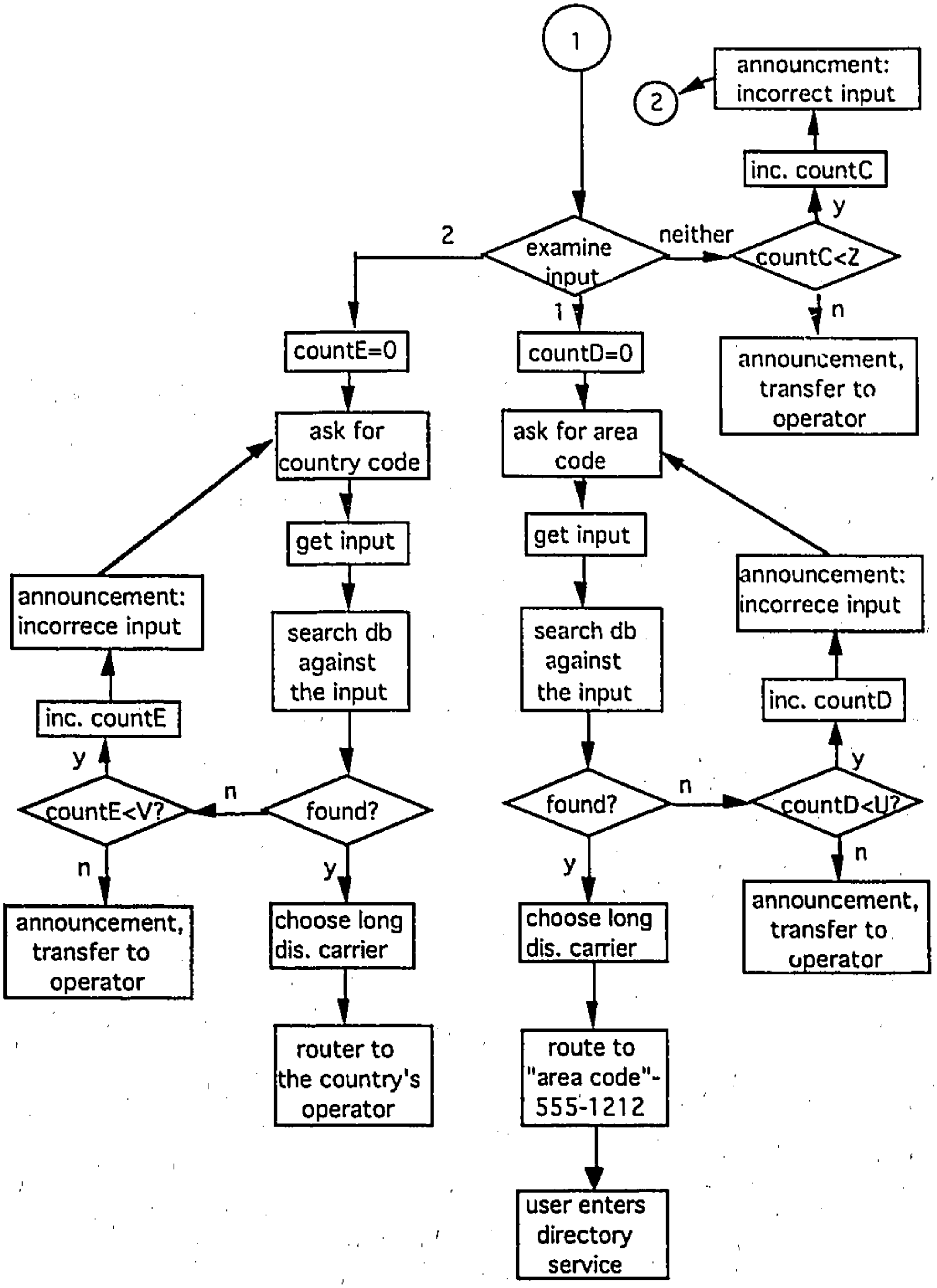


Figure 3.5 Automatic Telephone Directory Service (cont'd)

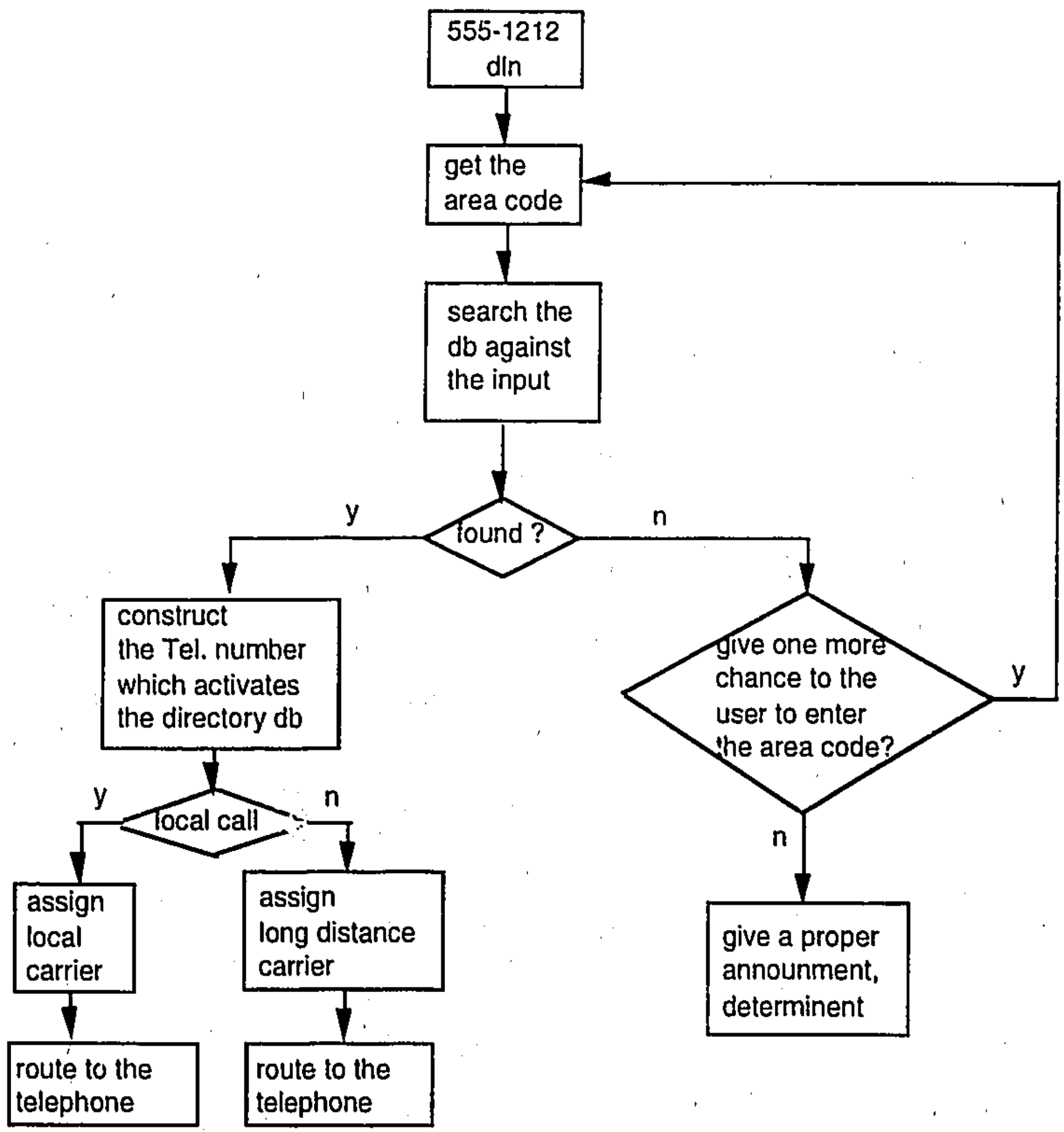


Figure 3.6 Automatic Telephone Directory Service (implemented)

3.3.3 Automatic Information Query Service

The objective of this service is to allow people to query information in a fast and cost effective way. This service handles the user's various types of information requests by routing the user's call to the nearest database server which can answer the requests through mail, fax, voice, or e-mail. Potential customers of this service could be government, department stores, libraries, banks, and various other medium and large size organizations. For the purpose of simplicity in the description of this service procedure, a government information query, e.g., information about "capital gain", is assumed.

<u>Step 1</u>

The user dials the number which represents the automatic information query service. This number can be a 1-800 number (the cost of service is charged to the called party) or an 1-900 number (the cost of the service is charged to the calling party).

<u>Step 2</u>

The local SSP recognizes that this call requires special handling, then passes the request to the SCP along with the calling number and the called number. The SCP consults the appropriate database to determine the area from where this call is originated and finds out which Customer Service Office (CSO) should be involved (assuming that the SCP is supported by several geographically distributed information databases referred as CSOs). In order to reduce the cost (making the call as local call as possible), the service always tries to connect the user to the nearest CSO.

<u>Step 3</u>

The SCP connects the calling party to an appropriate CSO. At this point, there are two alternatives. The CSO might use speech recognition or telephone dial-key pad for interaction with users. The system asks the caller if he/she prefers service in English (indicated by pressing 1) or French (indicated by pressing 2). If speech recognition is used, the CSO asks the caller to repeat several words or sentences in order to let the speech recognition system get used to the user voice and accent. In case the speech recognition system is unable to proceed, the CSO informs the caller that his/her only option is to use the dial-key pad. Otherwise, the system goes to step 4.

<u>Step 4</u>

The CSO asks the user what government department is of his/her interest. The user answers the request. The CSO system processes it. When the CSO figures out the department, it informs the user that he/she will be connected. This helps the system to identify which branch of the search tree should follow. If the CSO can not identify the department, it informs the user of its failure, and announces that it will list a number of options, asking the user either to press a button (when the user identifies one of the listed items as the department he/she might be interested in), or to say a specific word of the department he/she is looking for. The CSO allows the user to request the system to back-track on the list so that a correction of error answer or reconsideration of an answer can be performed. If the system is unable to identify the department, the CSO informs the caller and transfers the call to a human operator.

<u>Step 5</u>

The CSO asks the caller what information he/she wants to know, for example, information about taxation, the deadline for filling an income tax form, and other information which can be passed to the caller either by the record of an answer machine, or by delivering hard copies (fax, mail, e-mail). The caller responds either by using the dial-key pad to make the choice from a provided list (when the list is not long) or by using a speech recognition system to let the service know his/her inquiry. If the system is unable to answer the user's query for some reason (e.g., the information is not available in the databases, or the user's input does not quite fit the search key of the databases, etc.), it allows the caller to re-input up to a certain number of times. If the system fails to answer the user's information query, it informs the caller about the problem and then connects the user to a human operator.

Step 6

This step might be omitted depending on the type of information the caller requires. For example, if the information queried is very short (like the deadline for filling an income tax form), the answer is given by an answer machine, and this step is omitted. When the answer is long and needs to be given in the form of a document (e.g., a specific government regulation), this step is taken place. Assume the caller is interested in the information on "capital gains". The system provides a brief summary of the regulation through an answer machine or by providing a hard copy of the summary, depending on the length of the summary (the system might ask the caller to indicate specific items in the regulation). Afterwards, the system informs the caller of the existence of a certain bulletin that contains the information, and asks the caller if he/she wants to receive this bulletin. The caller responds by pressing the appropriate button. If the caller wants the hard copy of the information, the system asks the caller to indicate the way for delivering the information, e.g., delivering the information by mail, fax or e-mail, and asks for some additional information which might be needed, e.g., the fax number, the postal address, the e-mail address, the suitable time for the delivery of the information (this option is needed to cover the case that the caller needs to switch the fax or home computer on), the telephone number at which the caller can be contacted in case something goes wrong. When the service involves a charge for the information delivery, the CSO arranges the method of payment (assuming the cost is announced to the caller before arranging the information delivery). Different options are available. One possibility is to pay through the telephone company. Another way is to use a card which the caller gives and authorizes through a PIN (Personal Identification Number) number. During the verification process, the CSO connects the caller to the database of the caller's bank. The bank authorizes the transaction, and makes an immediate transfer of the funds.

<u>Step 7</u>

The CSO terminates the call, keeps in memory the information that is required in order to complete the remaining tasks, and carries them on.

The above is the detailed description of the service procedure. Figure 3.7 is the flow chart of this service procedure. Figure 3.8 displays the flow chart of a limited scale implementation of the service with the SPACE software.

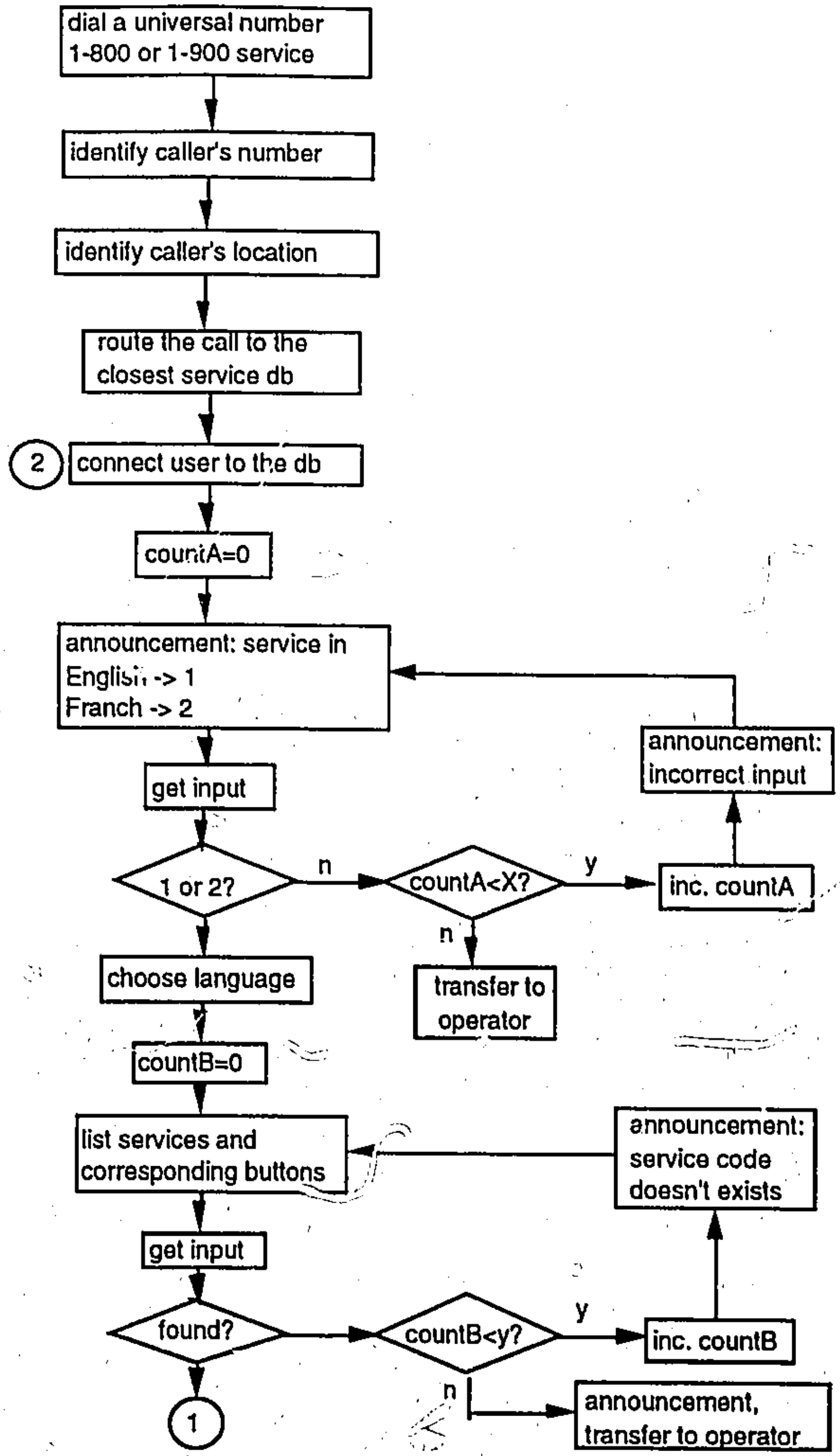


Figure 3.7a Automatic Information Query Service (cont'd)

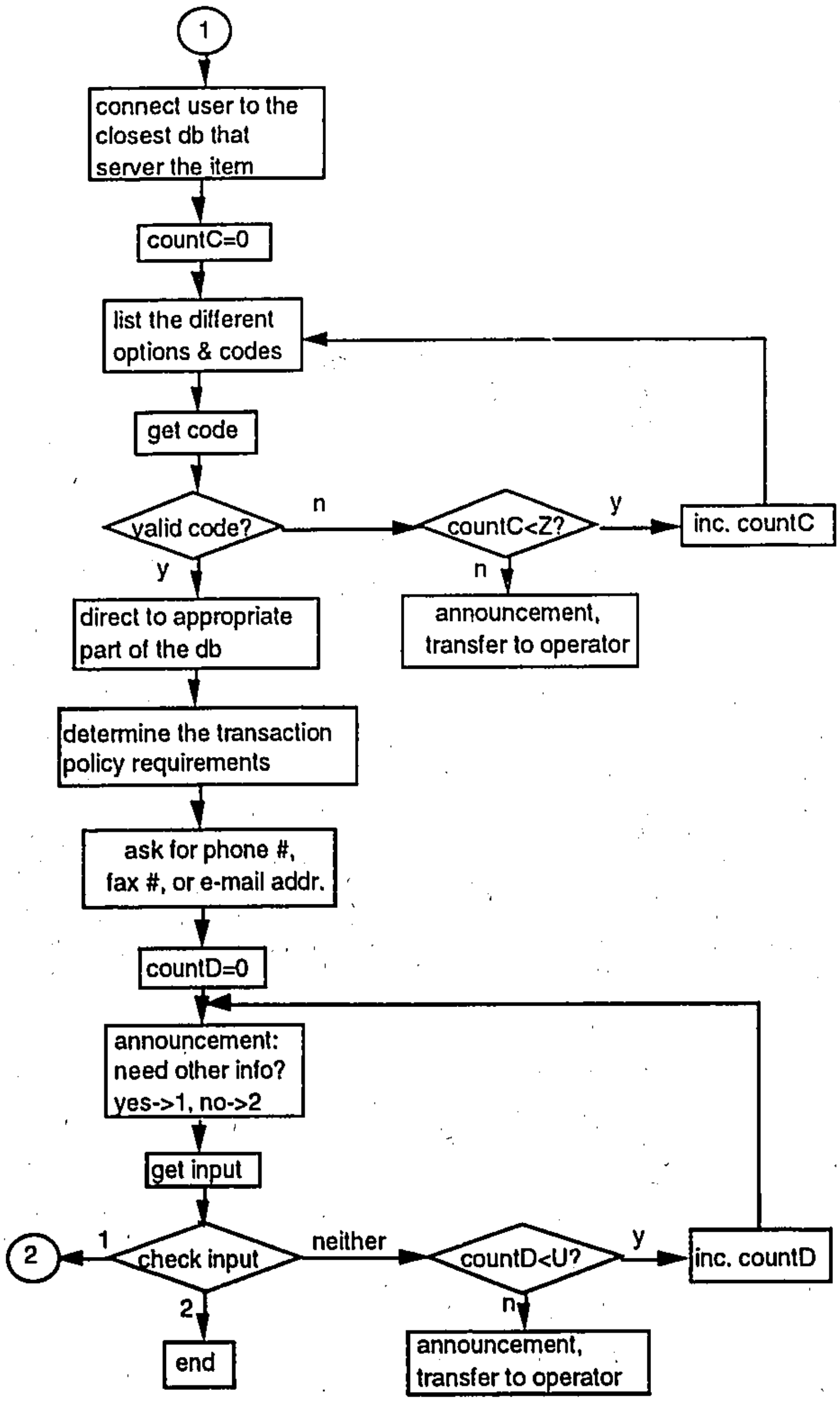
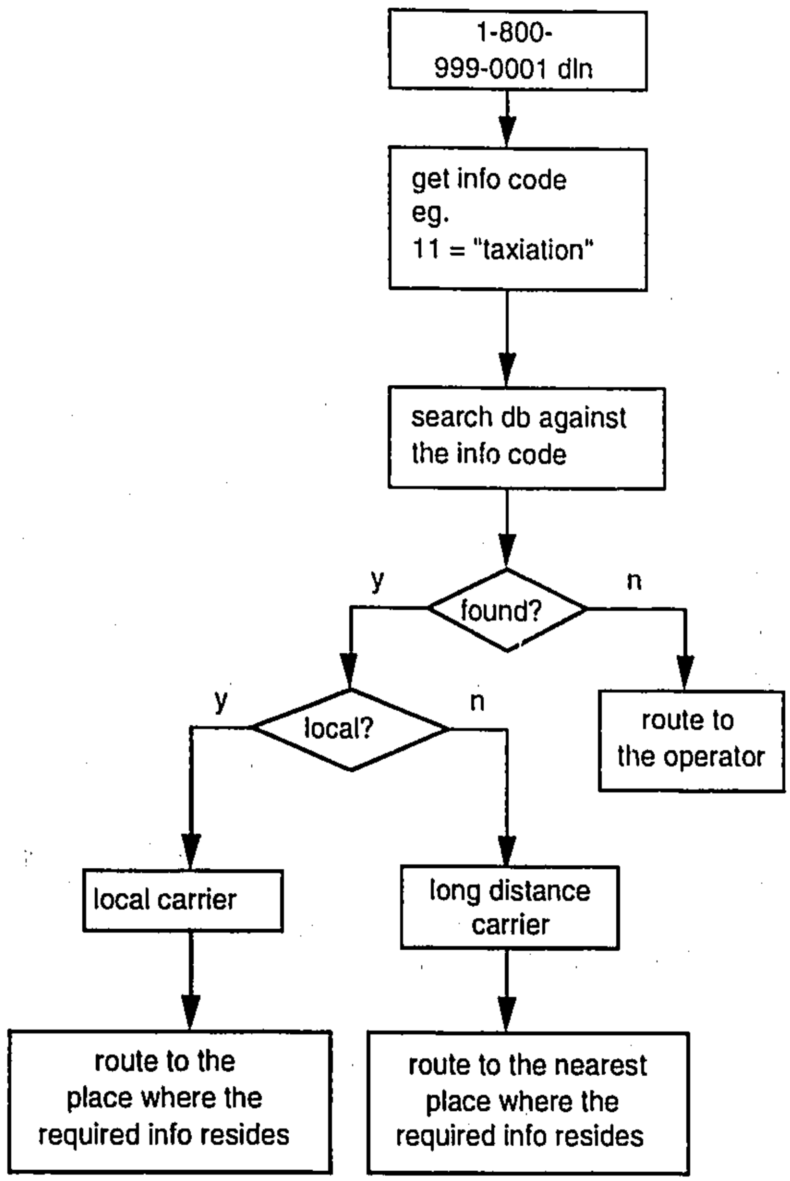


Figure 3.7b Automatic Information Query Service (cont'd)



Database: info. vs nearest place where the infomation resides

	TELEPHONE
11	613-111-1111
12	604-222-2222

Figure 3.8 Automatic Information Query Service (implemented)

Chapter 4

Design and Performance Evaluation of an AIN Multimedia Fax Service

Although originally resulting from the needs for various voice services over the Public Switched Telephone Network (PSTN), the oldest network in the world, the concept and technology of the AIN have become the principal guideline for building services in all kinds of networks. Introducing network intelligence in the Broadband Network (B-ISDN) will require the design of a broadband call model and a broadband IN architecture. The future broadband intelligent network will provide various sophisticated multimedia services and support user mobility in a fast and cost efficient manner. Some examples of broadband IN multimedia services could be automatic remote multimedia database access, video on demand, network interconnection, multimedia calls, and so on. In this chapter, we describe the design of a multimedia fax service over the existing AIN architecture, instead of over the broadband intelligent network which is being under investigation and development.

4.1 Description of the AIN Multimedia Fax Service

The growth and ubiquity of the fax machine has been quite extraordinary since the early 80's. It is predicted that by the year 2000, there will be 100 million fax machines in the US alone and 70% of those fax machines will be PC-based and able to support the transfer of multimedia documents [BSA93].

On the other hand, the Internet is the fastest growing network in the world. Services such as electronic mail, developed by the Internet community since 1982 allows users worldwide to exchange information. These e-mail standards, which are very effective for textual messages, have begun to show their age with the emergence of multimedia communications. The Multimedia Fax-MIME gateway [PHG94] provides the means of offering fax services to Internet users and Internet services to multimedia fax users. It performs the protocol translations between Binary File Transfer fax protocol (which is used by the fax machine to transfer compressed scanned images of documents across the telephone network) and the Multipurpose Internet Mail Extensions (MIME), a standard that is used to support the transfer of multimedia messages over the Internet. Figure 4.1 illustrates the operation of Multimedia Fax-MIME gateway. The Multimedia Fax-MIME gateway offers bi-directional multimedia messaging. The MIME messages (text, image, audio, video) can be routed across the Internet to a local Fax-MIME gateway. Over the telephone network, the Fax-MIME gateway would use BFT (Binary File Transfer) as a tunneling mechanism to deliver the MIME message to a multimedia fax machine or use Group3 to transfer regular fax messages to traditional fax machines. To send a fax to an Internet user, the multimedia fax machine would use BFT to ship MIME-encoded messages to the Fax-MIME gateway. The gateway would strip off the BFT header and mail the message to the destination e-mail address. Thus, using MIME and BFT, we can facilitate bi-directional multimedia messaging through the Fax-MIME gateway. Similarly, the BFT header can be used to send requests to the gateway for FTP or new services from a fax-enabled PC. Traditional fax machines, however, would not be able to send messages to the Internet because they transmit only scanned images of documents, not ASCII text.

The AIN technology can enhance the multimedia fax service by dynamically addressing communication entities based on the callers needs, callers' equipment, time-of-day, services, and so on. The service presented here takes the advantages of the AIN technology, and provides the value added service to the fax and Internet users.

Potential users of this type service are a lot of industries such as traveling, hotel and resorts, and restaurants, etc.. The idea of this service is the following:

- assign a unique telephone number to the service. The number serves as the identifier of the service (not a fixed terminal), e.g., providing hotel reservations through the Internet.
- use the IP to perform user interaction, which provides assistance to the SCP in dynamically routing the call and determining the Internet address of the remote database servers. For instance, a caller planning to travel to New York city may use this service. The caller may need to provide to the IP various with parameters describing the requirements of his/her desired hotel (the location, preferred hotels, price range, etc.). The SCP will then route the call to a particular database server (containing the hotel information) located in New York city. The IP can also get the

information on the caller's equipment the caller uses, e.g., multimedia fax machine, or traditional fax machine, or personal computer with fax card.

- use the SCP to determine the call treatment based on the specific user's information (obtained through the IP), e.g., route the call to the proper location, associate the call with a particular Internet address, activate different protocols for information retrieval (more explanation on this will follow in the next paragraph), deliver the retrieved information to the destination which is specified by the user, and allocate the fax gateway closest to the user.

For the hotel reservation case, the service could provide some useful information (which depends on the databases in the Internet) such as video clips of the hotel rooms along with the text providing information on the price, the map of surrounding streets and transportation, etc. It also provides facilities which allow the caller to make the reservation through a computer or a fax machine.

The detailed procedures of the service are presented in the following flow charts, Figure 4.2a to Figure 4.2f.

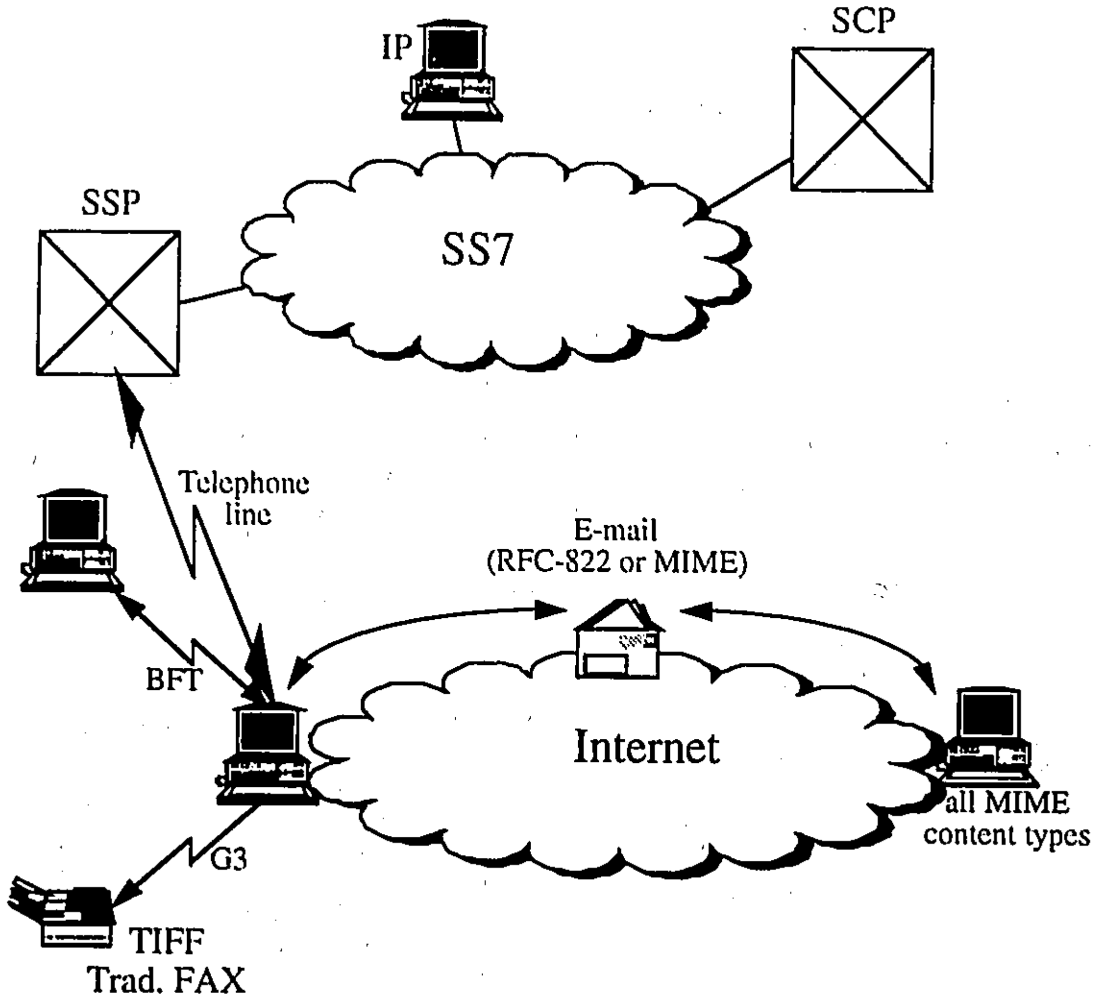


Figure 4.1 Multimedia Fax-MIME Interworking

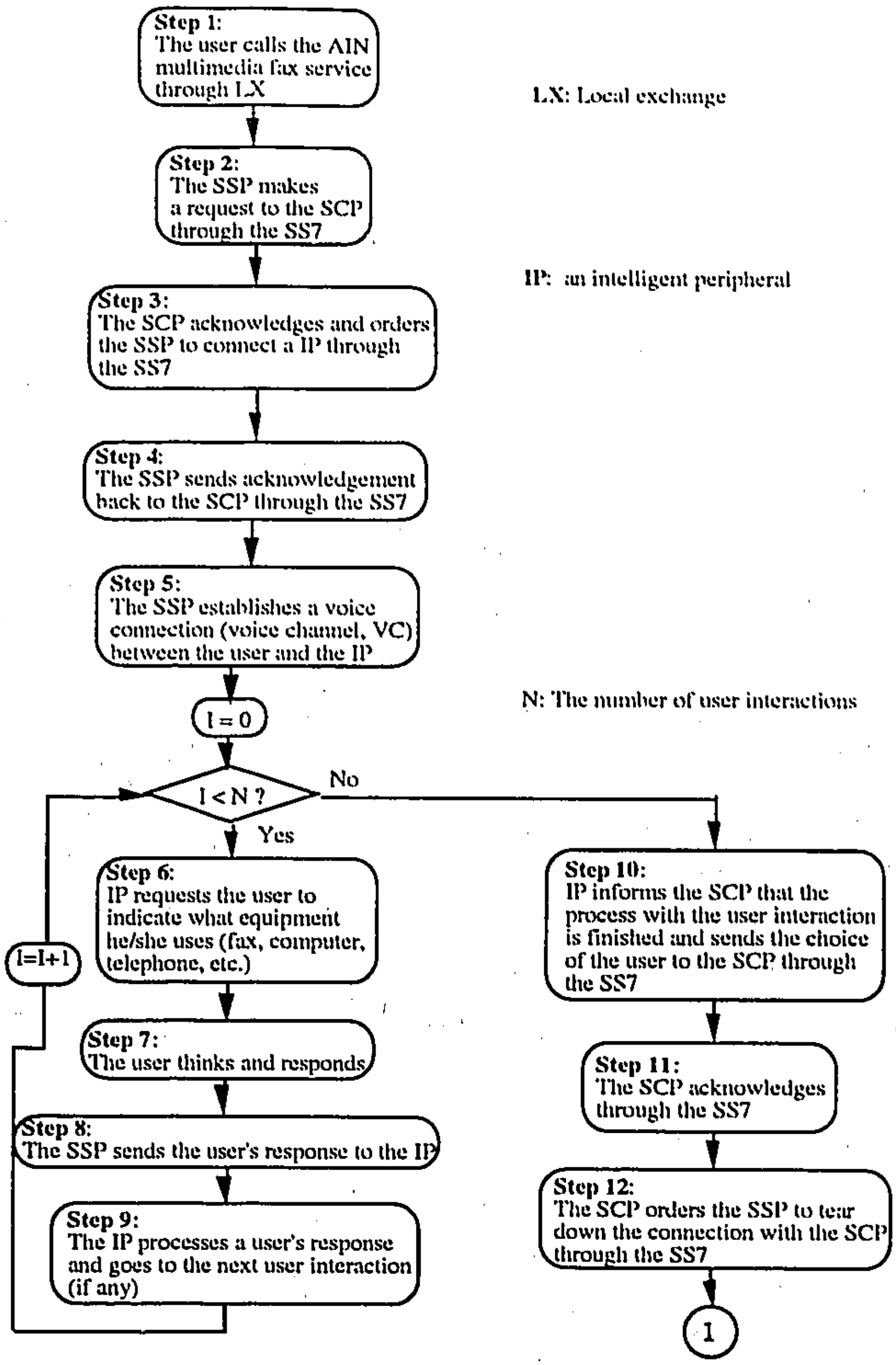


Figure 4.2a AIN Multimedia Fax Service Procedures (cont'd)

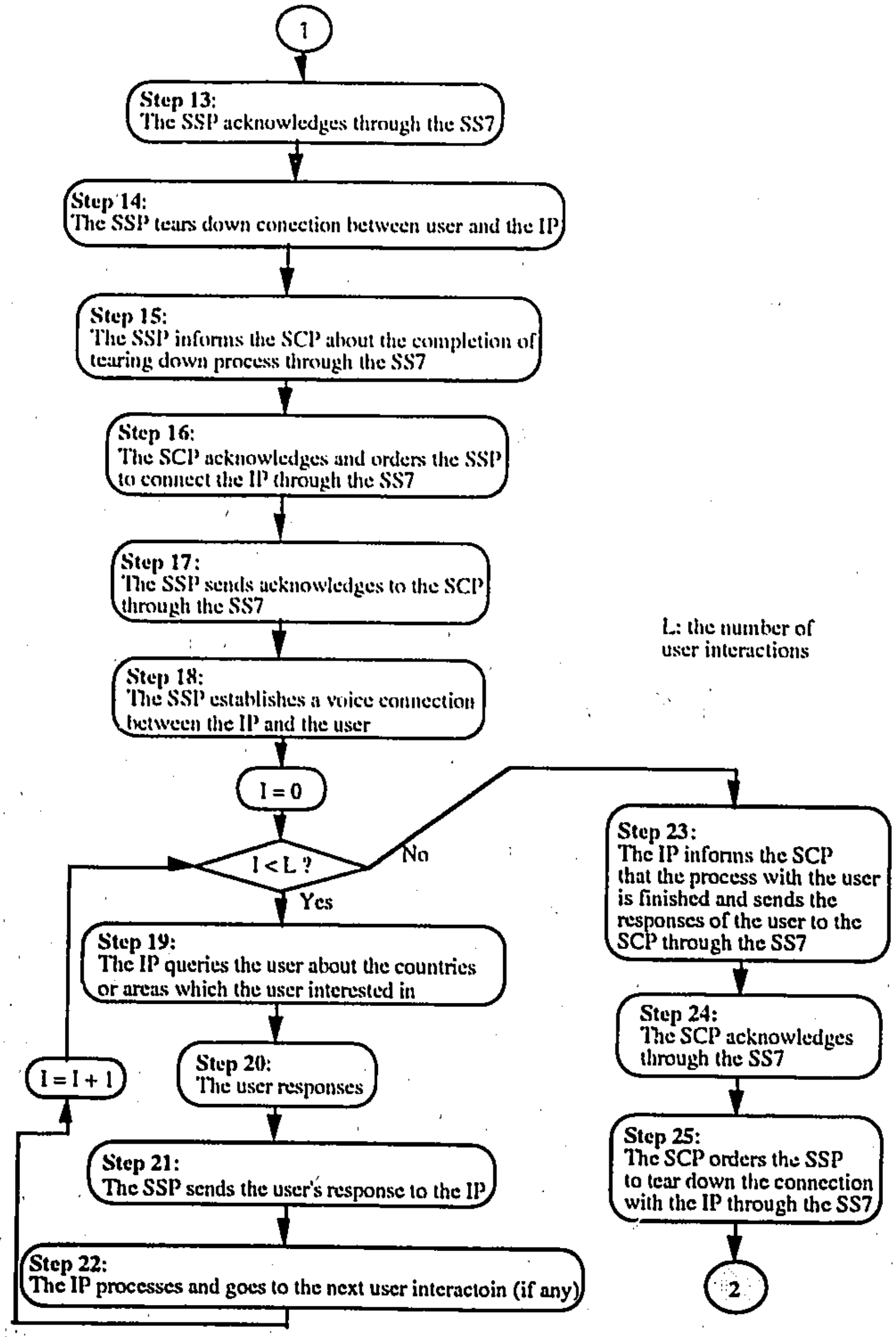


Figure 4.2b AIN Multimedia Fax Service Procedures (cont'd)

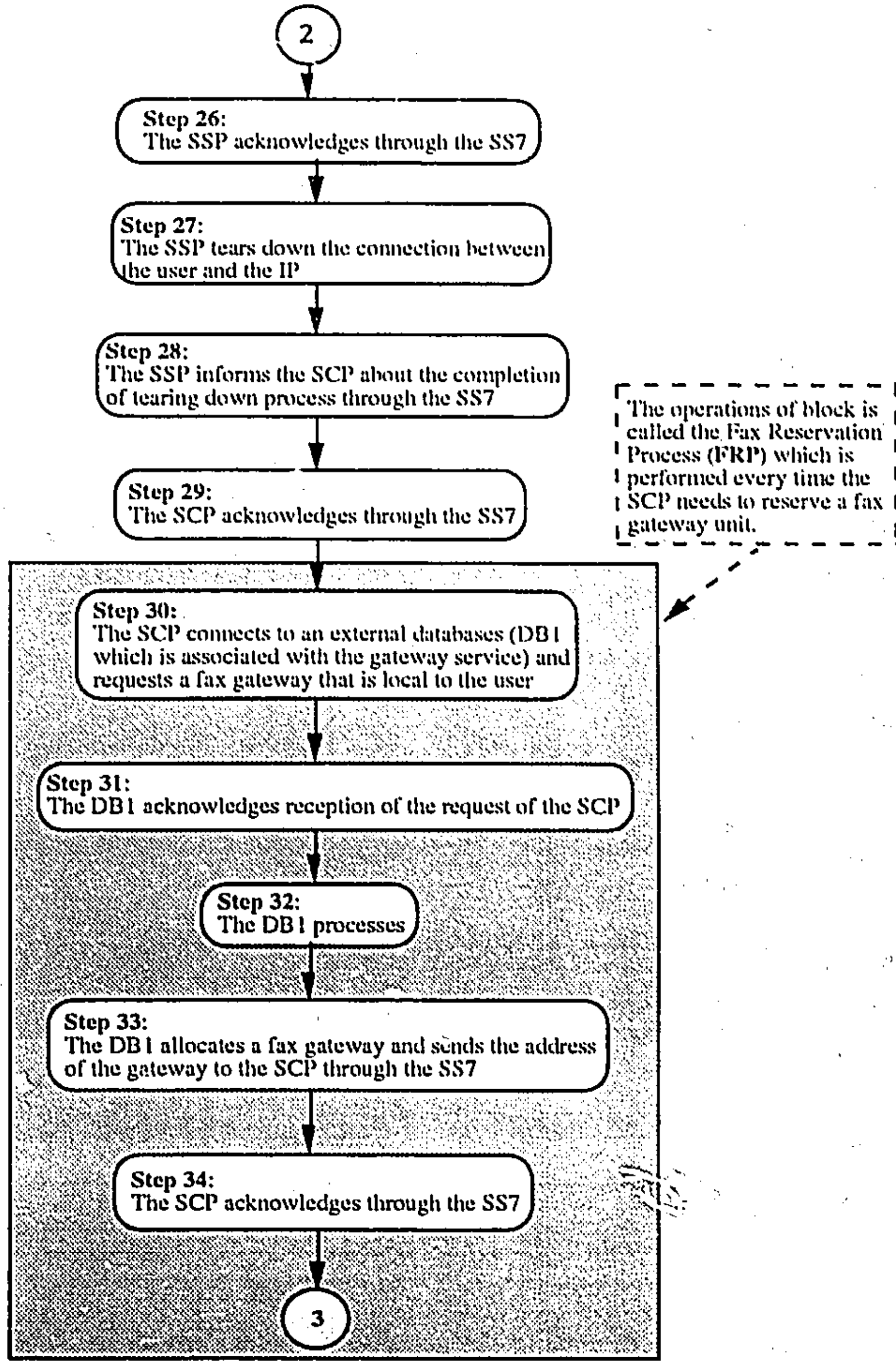


Figure 4.2c AIN Multimedia Fax Service Procedures (cont'd)

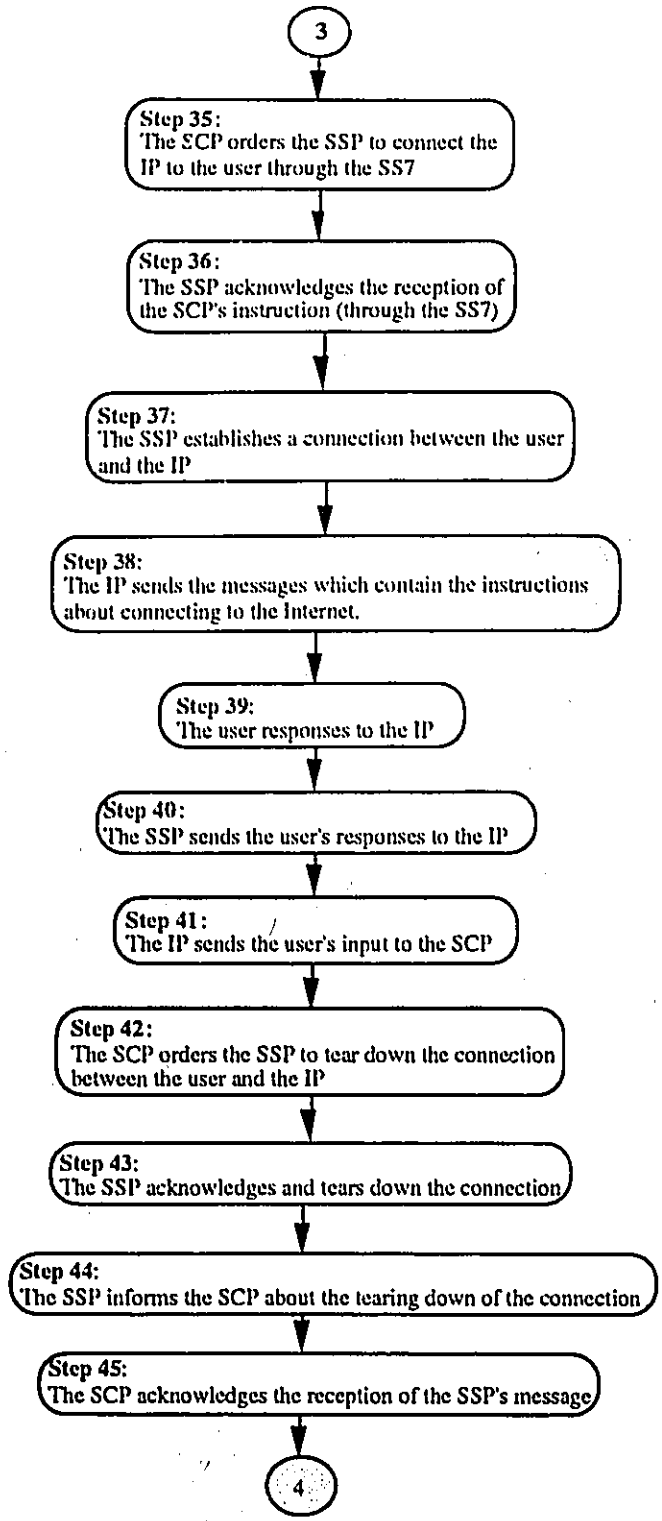


Figure 4.2d AIN Multimedia Fax Service Procedures (cont'd)

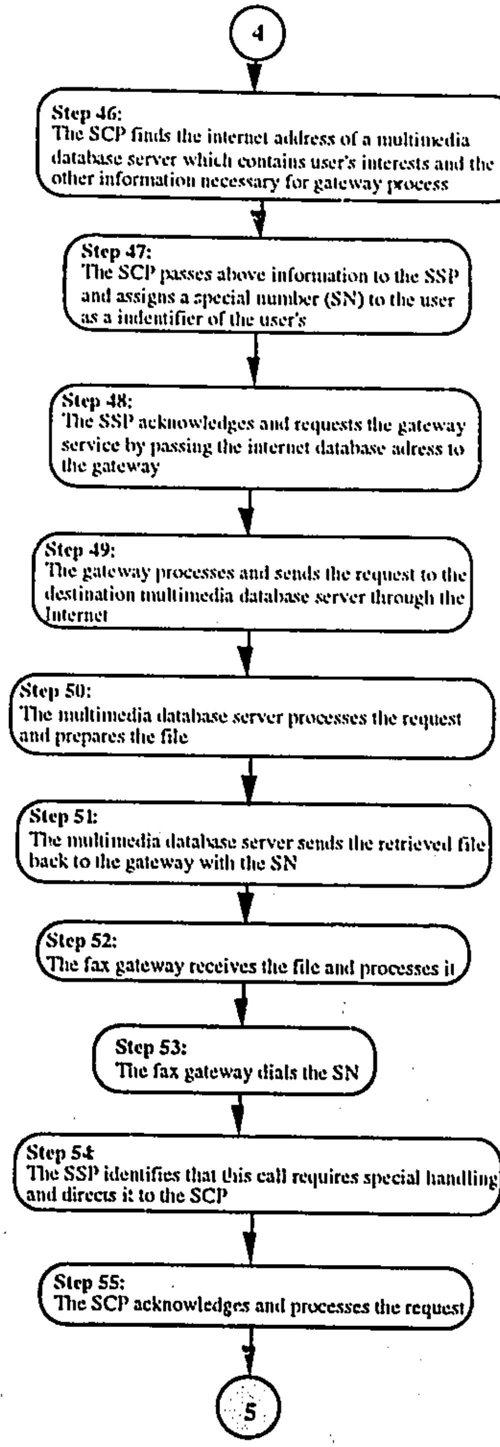


Figure 4.2e AIN Multimedia Fax Service Procedures (cont'd)

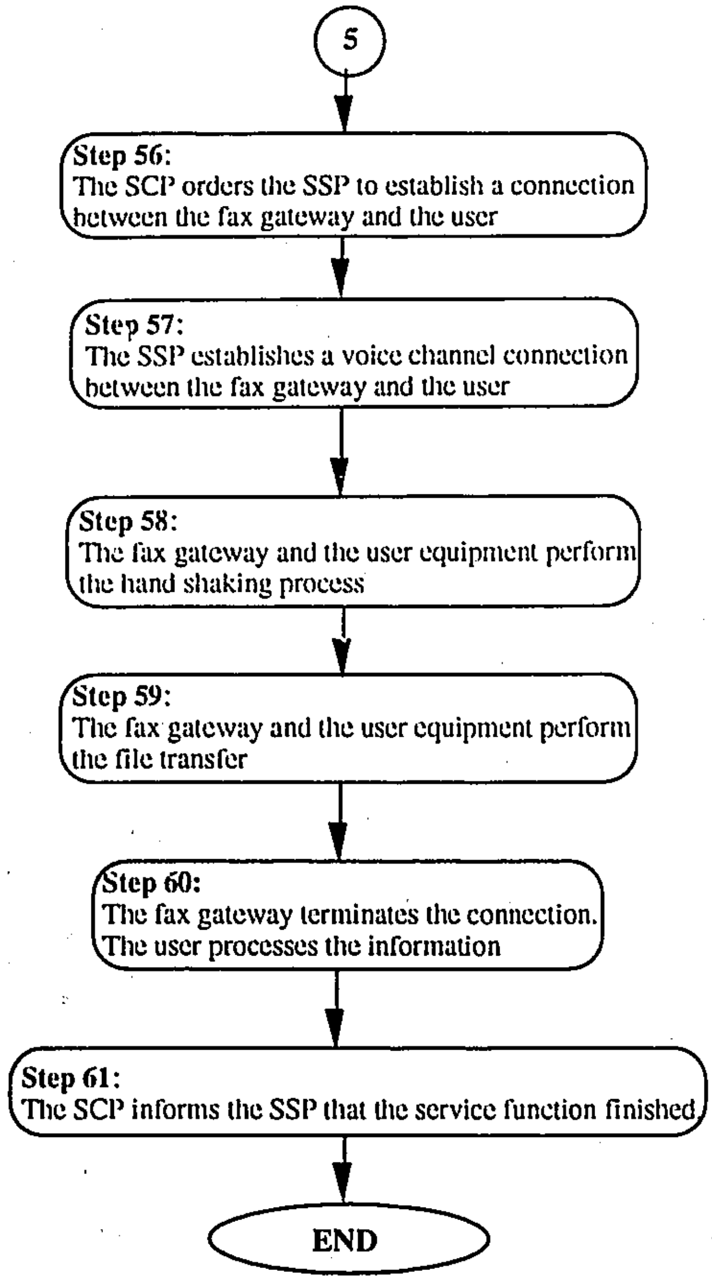


Figure 4.2f AIN Multimedia Fax Service Procedures

4.2 Modeling of an AIN Multimedia Fax Service

In this section, the queuing model for each network component of the AIN multimedia fax service is described in detail. The objective of the modeling is to evaluate the performance of the service in terms of the average call set-up time and the call blocking probability. The modeling is based on the M/G/1 non-preemptive priority queuing system.

4.2.1 Priority Queuing System

In this section, we review the non-preemptive priority queuing theory. The treatment provided here will be brief. Details can be found in [BG77]. Consider the M/G/1 queuing system in which the arriving calls (or packets) are divided into n different priority classes. Class 1 has the highest priority, class 2 has the second highest priority, and so on. Some symbols are defined as the following.

k: the priority class of a call

λ_k: the arrival rate of class k

μ_k: the mean service rate for class k

$\overline{X_k}$: the mean service time of class k, $\overline{X_k} = 1/\mu_k$

T_k: the average delay for class k

W_k: the average queuing waiting time for class k

ρ_k: the system utilization for class k, $\rho_k = \lambda_k / \mu_k$

R: the mean residual service time

Assume that the overall M/G/1 system utilization is less than one:

$$\rho_1 + \rho_2 + \cdots + \rho_n < 1$$

When this assumption is not satisfied, there is some priority class k for which all calls that have priorities less or equal to k experience infinite average delay. The average delay of calls with higher priority than k will be finite.

The waiting time in a M/G/1 non-preemptive priority queuing system is determined by the following formula:

$$W_k = \frac{R}{(1-\rho_1-\cdots-\rho_{k-1})(1-\rho_1-\cdots-\rho_k)} \qquad (1)$$

The average delay (defined by the time elapsing between the moment the call enters the queuing system and the moment the call exits the system) for each call of class k is given by:

$$T_k = \frac{1}{\mu_k} + W_k \qquad (2)$$

and R is obtained by

$$R = \frac{1}{2}\left(\sum_{i=1}^{n}\lambda_i\right)\overline{X^2} \qquad (3)$$

where $\overline{X^2}$ is the second moment of the service time averaged over all priority classes, namely,

$$\overline{X^2} = \frac{\lambda_1}{\sum_{i=1}^{n}\lambda_i}\overline{X_1^2} + \cdots + \frac{\lambda_n}{\sum_{i=1}^{n}\lambda_i}\overline{X_n^2}$$

Substituting in equation (3) yields

$$R = \frac{1}{2}\sum_{1}^{n}\lambda_i\overline{X_i^2} \qquad (4)$$

The average waiting time in the queue and the average delay for each class can be obtained from (1), (2) and (4). The final expressions are as the following:

$$W_k = \frac{\sum_{i=1}^{n} \lambda_i \overline{X_i^2}}{2(1-\rho_1-\cdots-\rho_{k-1})(1-\rho_1-\cdots-\rho_k)} \tag{5}$$

$$T_k = \frac{1}{\mu_k} + W_k$$

4.2.2 Model of Service Switching Point (SSP)

The main functions of the SSP are:

- to accept (or reject) call requests
- to suspend the call processing in order to wait for instructions from the SCP (or ADJ/SN)
- to suspend the processing of a call when the call processing requires interaction between the users and the IP
- to resume the call processing of stopped calls as a result of acting on instructions from the SCP (or ADJ/SN)
- to connect the caller to the final destination phone numbers

All these functions can be modeled by a non-preemptive priority queuing system as shown in Figure 4.3. The model consists of four different priority queues. The first queue (referred as emergency_queue, or emg_queue) collects emergency calls (e.g., 911) which can never be rejected. Emergency calls are given the highest priority. The second queue (referred as wait_scp_queue) handles the on-going calls which are waiting for further call processing based on the SCP instructions. This queue has the second highest priority since it is associated with on-going calls which can not be blocked. In addition to that, since the SCP expects the response of the SSP to its request before continuing processing, it is important that the SSP executes the request as soon as possible so that the SCP does not remain idle. The third queue (referred as wait_ip_queue) contains the on-going calls which are involving the user interactions through the IP. This queue has the third highest priority. The fourth queue (refereed as new_call_queue) contains the requests of new calls since it is preferable to block new calls rather than interrupt calls that are already in service. So this queue is assigned the lowest priority.

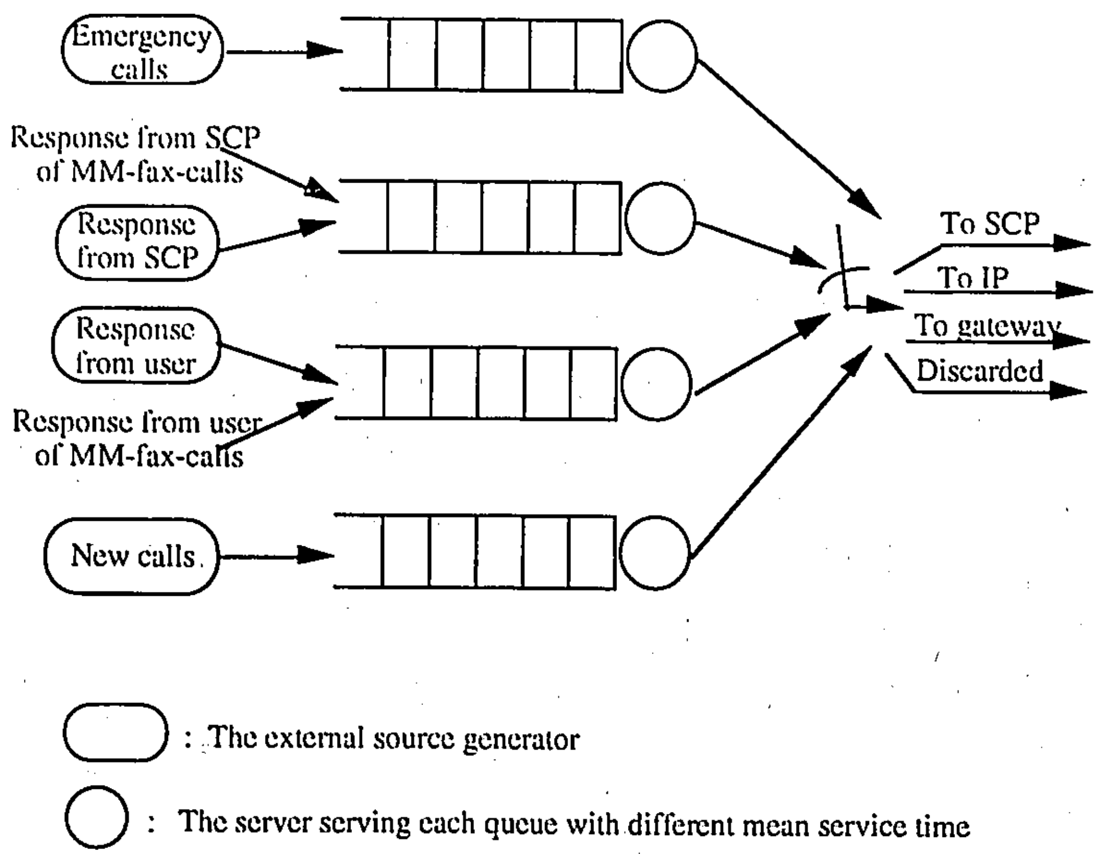


Figure 4.3 Model of Service Switching Point

The first and the last queues receive the traffic generated by the Local Exchanges (LXs) which are connected to the SSP. The traffic to the second and the third queues is determined by the nature of the accepted calls, e.g., some of calls are ordinary calls (non AIN calls) which do not generate any traffic input to the second and the third queues; whereas some other types of calls (AIN calls) require several interactions between the SSP and SCP, IP and the SSP, and consequently they generate traffic to the second and the third queues. The traffic load depends on the service characteristics. For example, if the service involves many user interactions with the IP, there will be more traffic load to the wait_ip_queue. The parameters of traffic input of all the queues and the service time for each class of traffic are defined as follows.

- $\lambda_{SSP\text{-}E}$: the mean arrival rate of emergency calls
- $\lambda_{SSP\text{-}SCP}$: the mean arrival rate of calls waiting SCP instructions
- $\lambda_{SSP\text{-}IP}$: the mean arrival rate of calls waiting user interactions
- $\lambda_{SSP\text{-}NEW}$: the mean arrival rate of new calls
- $\mu_{SSP\text{-}E}$: the mean service rate for emergency calls
- $\mu_{SSP\text{-}SCP}$: the mean service rate for the calls waiting in the wait_scp_queue
- $\mu_{SSP\text{-}IP}$: the mean service rate for the calls waiting in the wait_ip_queue
- $\mu_{SSP\text{-}NEW}$: the mean service rate for accepting new calls

In the simulation model, we assume that all the traffic arrival rate follows a Poisson distribution, and all the service times are exponentially distributed. P_{MM} is the probability that a call will request the multimedia fax service. So among all the calls, only a percentage of $P_{MM} \times 100$ calls are the calls marked as "MM-fax-calls" (Multimedia-fax-calls). When the server of the SSP completes the service of a call, it will direct the call to the SCP, IP, or gateway if the call is "MM-fax-call", and discards all calls of other types. Whether an AIN fax call is sent to SCP, IP, or gateway depends on the stage of the call processing. The information of the call processing stage is stored in a pre-defined packet format (described in the next paragraph).

It is convenient to use packets to represent calls which may be in any call processing stage (in any queues). From now on we will use packets and calls interchangeably. Packets are generated by external traffic source generators. There are four types of traffic generators, defined as $I_{SSP\text{-}E}$, $I_{SSP\text{-}SCP}$, $I_{SSP\text{-}IP}$ and $I_{SSP\text{-}NEW}$. Each generator is associated with an arrival rate for a particular queue. In our simulation model, packets generated by $I_{SSP\text{-}E}$, $I_{SSP\text{-}SCP}$ and $I_{SSP\text{-}IP}$ have only one field, and packets generated by $I_{SSP\text{-}NEW}$ have three fields, as shown below:

packet format 1:

type_of_call

packet format 2:

type_of_call	num_of_ip_interactions	time_stamp

For the packets generated by $I_{SSP\text{-}E}$, $I_{SSP\text{-}SCP}$ and $I_{SSP\text{-}IP}$, the value of "type_of_call" field is set to 0. For packets generated by $I_{SSP\text{-}NEW}$, the value of "type_of_call" field is set to 1 with probability P_{MM} (indicating the call with the request of our AIN multimedia fax service), and 0 with probability ($1-P_{MM}$), indicating calls of any other types of service. The field "num_of_ip_interactions" is set to an integer indicating the number of user interactions involved in the service. The IP node decrements the value of "num_of_ip_interactions" by 1 each time the packet visits the IP. When this field reaches zero, all user interactions are finished. The IP will then send user's response(s) to the SCP. At this point, the SCP has all the user information required for the service, and sends the call processing information to the SSP. The SSP resumes the processing of the call and sends the packet to the gateway. For all the other types of calls (non MM-fax-calls), the field of "num_of_ip_interactions" is ignored since when the SSP figures out that the call is not MM-fax-call, it does not check the field of "num_of_ip_interactions", but discards the packet after finishing serving it. The field "time_stamp" is used to calculate the call set-up time. Each packet generated by the $I_{SSP\text{-}NEW}$ and marked as "MM-fax-call" is stamped the time at which the packet enters the node. Each network node checks the "time_stamp" field of the packet, compares it with the current time, and calculates the delay of the packet (delay = current time - value of "time_stamp" field). If the delay exceeds a certain threshold, the packet is blocked (then discarded). For example, if the delay threshold is equal to 20 seconds, then the packet with a delay value greater than 20 seconds is considered as being blocked. In the simulation, the call blocking probability due to the lack of the buffer space in the SSP, SCP and IP has not been considered since all the queues have infinite buffers.

When a call process is initiated, it goes first to the LE, and from there it is forwarded to the SSP. There is some time elapsing between the moment the call is initiated and the moment where the SSP receives the call request. This is the combined delay due to the propagation time through the lines, as well as the waiting and processing times at the LE. In order to avoid complication of our model, we ignore this time and assume that users are directly connected to the SSP.

As described previously, the traffic entering the various queues of the SSP is generated by a variety of services (including the multimedia fax service). As we are interested in evaluating the response time of the AIN service, we only examine the behavior of the MM-fax service as it progresses through the network. We do this by allowing only packets marked as "MM-fax-call" to go through all the network elements that the service involves during its execution. The total traffic generated by services of other types is considered as the aggregated traffic which is simulated by as a Poisson traffic. This type of traffic (generated by other services) has only local meaning, that is, it doesn't propagate through the network, consequently, it does not generate traffic to other network nodes. Its traffic is absorbed in the local node (the node in which the traffic source generators reside). The effect of the other services on the network is considered by using external traffic generators at each network node .

In order to keep the mean arrival rate of aggregated traffic unchanged, we do the following. Whenever a packet from the SCP or the IP arrives at the SSP and is inserted in the proper queue, the next packet generated by the external traffic source (associated with that queue) is discarded. For example, if the "wait_scp_queue" (the second queue) receives a packet of "MM-fax-call" (from the SCP), the next packet generated by the external source generator $I_{ssp\text{-}scp}$ is ignored.

4.2.3 Model of Service Control Point (SCP)

The main functions of the SCP (or AJD/SN) are:

- to accept SSP requests and to process them
- to activate the IP to perform the function which is responsible to interact with users
- to receive user information collected by the IP
- to send the call processing information to the SSP

The SCP model is built in a similar way to the SSP model, presented in the previous section. The SCP model is shown in Figure 4.4. The model consists of three queues. The highest priority is given to the first queue which is responsible for handling the emergency calls. The second queue which collects SSP requests has the second highest priority since the request from the SSP is virtually the first step of invoking the service logic stored in the SCP. The last priority queue is allocated to handle the packets from the IP, that is, the user responses.

We define the first queue as SCPemg_queue, the second as SCPuser_resp_queue, and the third as SCPssp_req_queue. The parameters of traffic load to each queue and the mean service rate for different queues are defined as follows:

- $\lambda_{SCP\text{-}E}$: the mean arrival rate of emergency calls which need SCP processing
- $\lambda_{SCP\text{-}SSP}$: the mean arrival rate of SSP requests
- $\lambda_{SCP\text{-}IP}$: the mean arrival rate of incoming user responses from IP
- $\mu_{SCP\text{-}E}$: the mean service rate for processing emergency calls
- μ_{SSP}: the mean service rate for accepting SSP queries
- $\mu_{SCP\text{-}IP}$: the mean service rate for processing the input of IP

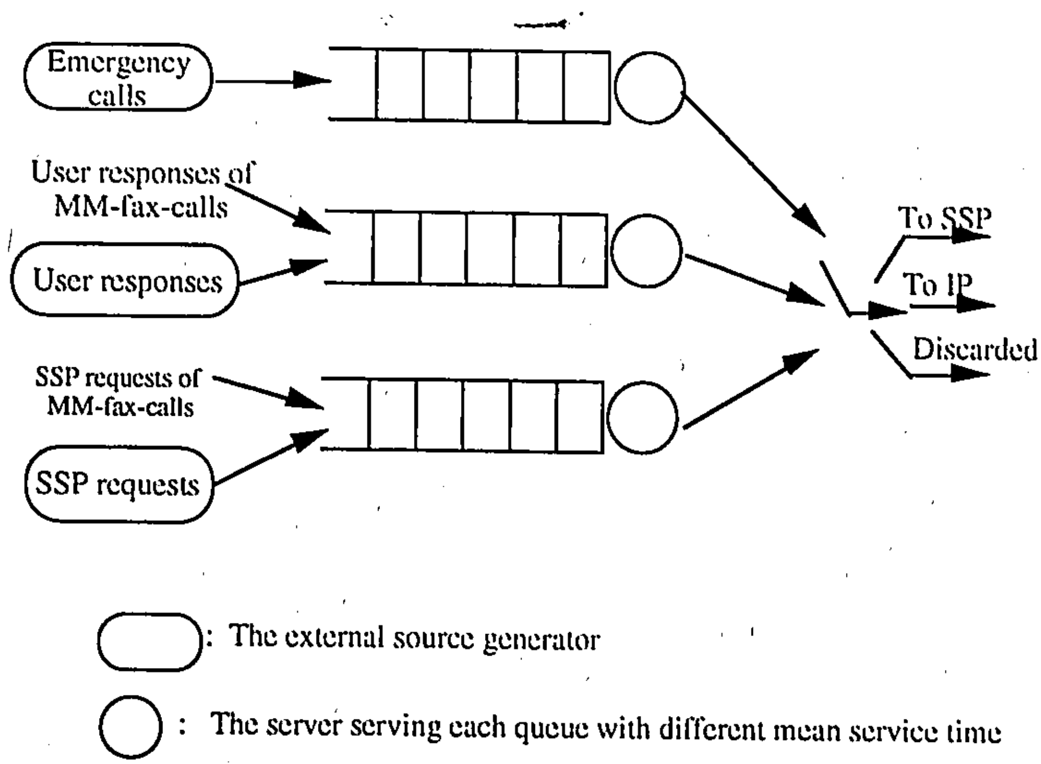


Figure 4.4 The Model of Service Control Point

The three arrival rates are modeled by three external traffic source generators (I_{SCP-E}, I_{SSP}, I_{SCP-IP}) with different mean values of Poison distributions. Each service time is exponentially distributed. Packets generated by all external traffic sources take the packet format 1(described in 4.2.3 subsection) which contains only one field, "type_of_call". All these packets are marked as "calls of other types" (the value of "type_of_call" is set to 0) since the MM-fax-call is generated only at the SSP node (see Figure 4.3). After processing a packet of any queue, the SCP checks the packet type. If the packet represents an other types of calls, it is discarded, otherwise, the "num_of_user_interactions" of the packet is checked. If the value of "num_of_user_intractions" is equal to 0 (meaning that no user input is required), the SCP formulates the response for the SSP, and then sends this information to the SSP. In order to keep the average mean arrival rate of aggregated traffic unchanged, the same approach as the one used in the SSP model is used.

4.2.4 Model of Intelligent Peripheral (IP)

The IP, as introduced in chapter 1, is a system which controls and manages functions such as voice synthesis, announcements, speech recognition, and digit collection. Its main simulated functions are:

- to accept messages from the SCP (these messages convey the information required to inform the IP about the resources or procedures to be used on the call
- to connect the caller through the SSP
- to prompt users and collecting user responses
- to return the user input to the SCP

The above IP functions are also modeled by the priority queuing system shown in Figure 4.5. The model consists of three queues. We assume that emergency calls are directly connected to human operators rather than to intelligent peripherals. Consequently, there is no emergency call processing in the IP node. The higher priority is allocated to the queue which collects the instructions of the SCP. This forces the IP to first execute the commands of the SCP, thus reduces the time that the SCP will keep the call idle. Two external traffic generators ($I_{IP\text{-}SCP}$ and $I_{IP\text{-}SSP}$) are used to model the overall traffic load of two different types in the IP node. Each external traffic generator randomly generates traffic with Poison distribution. The service time for each queue has exponential distribution. The parameters of this queuing model are defined as follows:

- $\lambda_{IP\text{-}SSP}$: the mean arrival rate of user responses (through the SSP)
- $\lambda_{IP\text{-}SCP}$: the mean arrival rate of SCP messages
- $\mu_{IP\text{-}SSP}$: the mean service rate for interaction with users (through the SSP)
- $\mu_{IP\text{-}SC}$: the mean service rate for processing user responses and sending them to the SCP

The external traffic generators generate packets with packet format 1, and set the field to value of 0 since the external traffic is only used to the model effect of other services. After serving a packet, the IP checks the "type_of_call" field. If a packet represents a call of other types, this packet is discarded, otherwise, the IP checks the "num_of_user_interactions" field. If the value of this field is not equal to 0, which indicates that the call (represented by this packet) has more user interactions to be processed, the IP will decrement this value by 1, and send the packet to the SSP (representing that one more prompt is sent to the user through the SSP). If the value of field is equal to 0, meaning that all the required user interactions are completed, the IP will send this packet to the SCP (In our model, the IP sends all the user responses to the SCP).

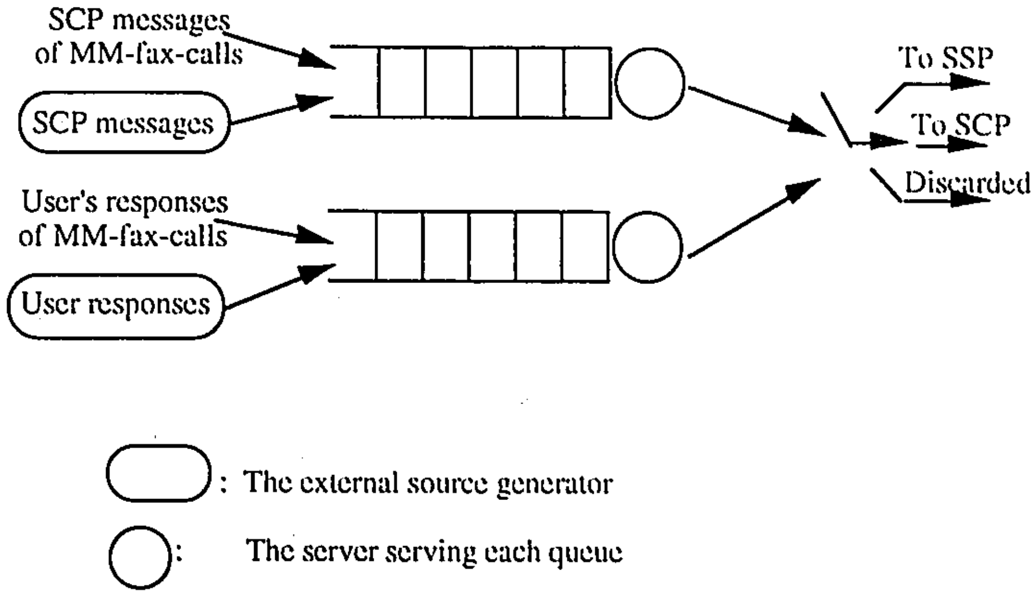


Figure 4.5 Model of the Intelligent Peripheral

4.2.5 Model of the SS7 Network

The SS7 network is used to interconnect the SCP to SSP, the SCP to the IP, and the SSP to the IP. From a view at the application level, the SS7 network receives a message from a network node (the SSP, SCP or IP), and delivers this message to another network node with some delay. The amount of the delay depends on the traffic load of the SS7 network. In our simulation, we model the delay, $t_{ss7\text{-}d}$, that is experienced by a packet traveling through the SS7 network as the sum of one fixed part (T_c) and a negative exponentially distributed variable part (t_v). This model is accurate when the signaling links are lightly loaded [PB94], so that messages usually do not have to wait for being queued before transmission. To study the performance under a heavy load of signaling traffic, a more detailed model is required. Figure 4.6 shows the model of the SS7 network used in our simulation.

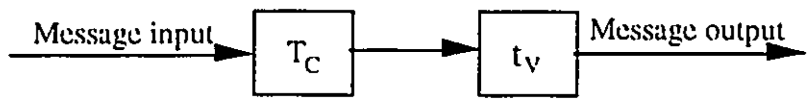


Figure 4.6 Delay Model of the SS7 Network

4.2.6 Model of the Gateway between the AIN and Internet

The gateway model used in our simulation is illustrated in Figure 4.7. Upon connection to the SSP, the gateway will perform proper Internet access, based on the messages of the SSP. The traffic load of the gateway is modeled by using traffic generators for each type of traffic (signal, text , image and video traffic), on both sides (PSTN, INTERNET). Each traffic arrival pattern follows the Poisson distribution. The text, image and video file sizes are conformed to the exponential distribution, but we impose minimum and maximum boundaries on the file size, that is, we choose a minimum and a maximum file size for each type of the files (text, image and video). If the file size generated by the exponential distribution generator is smaller (or greater) than the minimum (or maximum) value, the file size takes the minimum (or maximum) value. In this way, we avoid having a zero or infinite file size.

The gateway is connected to the AIN PSTN through an ISDN line, and is connected to the INTERNET through a T1 line. The ISDN line is modeled as 24 servers, each having the service rate of 64K bits per second. The T1 line is modeled as one server with the service rate of 1.544M bites per second.

The gateway receives files, or calls, from both the PSTN and the INTERNET, stores them in the memory for protocol translation and file transmission to the INTERNET or the PSTN. In figure 4.7, the buffer sizes of buffer 1, buffer 2 and buffer 3 are not fixed, but the sum of these buffer sizes is fixed to the capacity of the gateway. The gateway processing rate is also fixed.

In Figure 4.7, the call blocking occurs in two situations: i) all transmission servers are busy when the AIN MM call arrivals; or ii) there is not enough memory in the gateway to hold the AIN MM call.

4.2.7 Models of the Internet and the Database Server

In our simulation environment, we model Internet and the database server as two delay nodes. The delay experienced by a message while traveling through the Internet or processed by a database server residing in the Internet is modeled by a random delay which statistically follows a constant delay (T_{INT} for the Internet, T_{DB} for the database server) and an exponential distribution with a mean value (t_{INT} for the Internet, t_{DB} for the database server). The model of Internet is illustrated in Figure 4.8. The model of the database server in the Internet is shown in Figure 4.9.

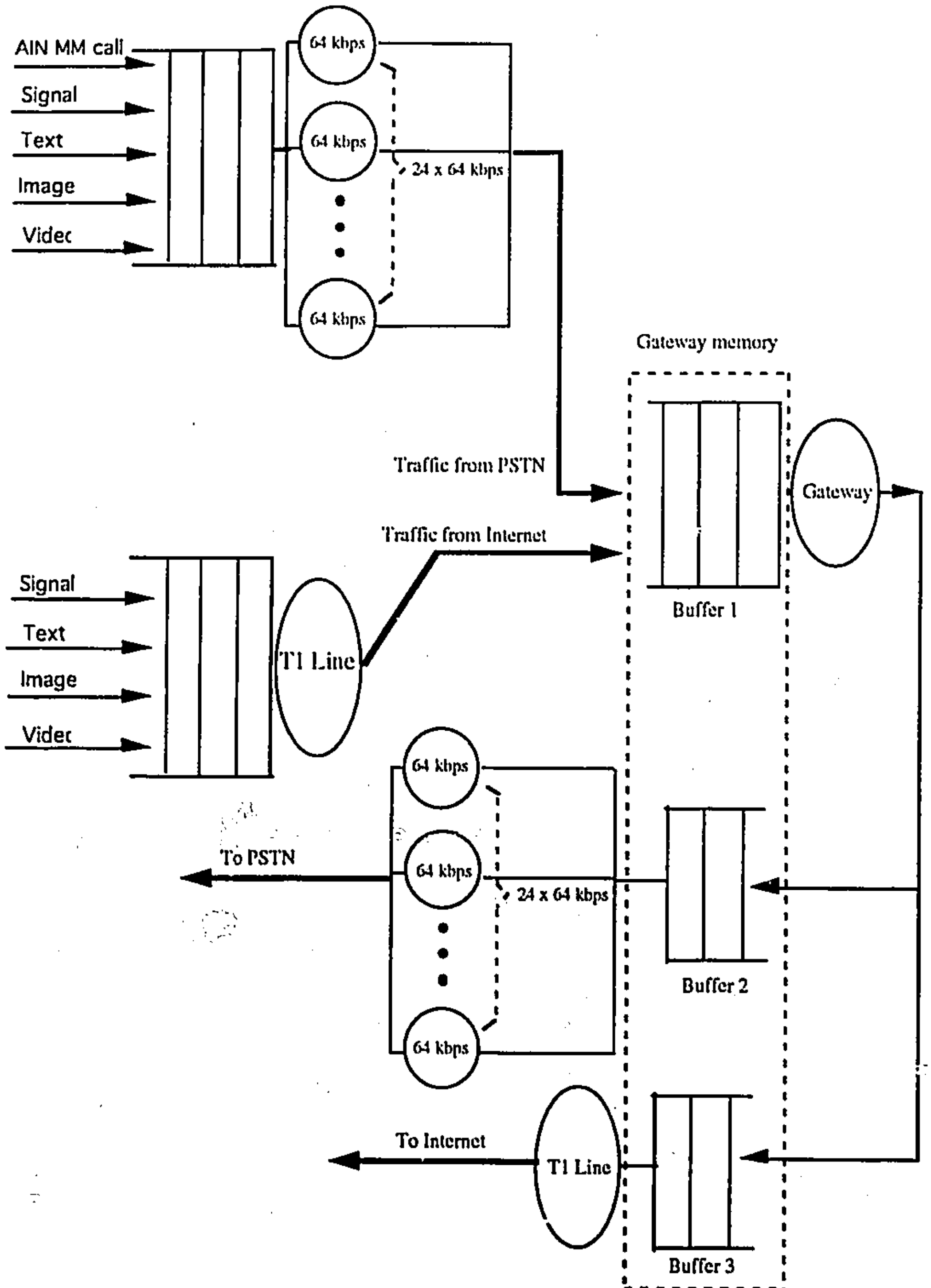


Figure 4.7 Model of the Gateway

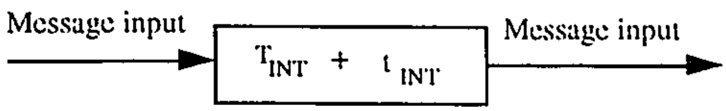


Figure 4.8 Model of the Internet

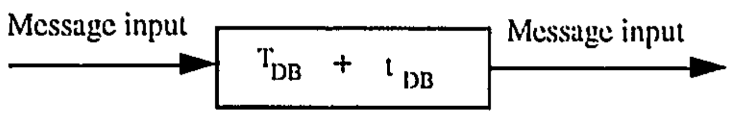


Figure 4.9 Model of the Database Server in Internet

4.3 Simulation of the Multimedia Fax Service

Based on the queuing models presented in the previous section, this section presents the simulation of the multimedia fax. The simulation is built using the OPNET software which is a relatively new and powerful simulation tool for telecommunications [OP93]. A brief description of the OPNET is presented in Appendix 3.

4.3.1 Simulation Model

Figure 4.10 gives the simulation model at the Network level (the highest level of models in the OPNET). The Network level's model consists of eight nodes, namely, the SSP, SCP, IP, Gateway, SS7, Internet and database server in the Internet. Figure 4.11 presents the detailed context of each node.

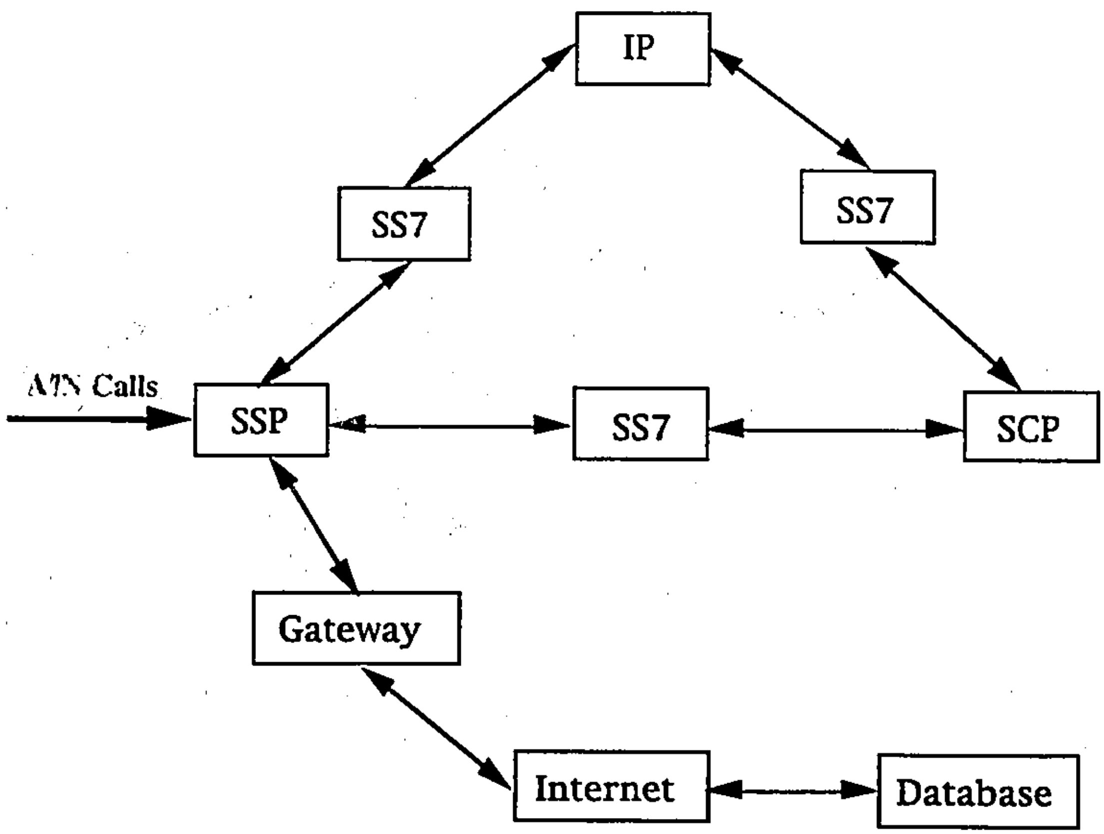


Figure 4.10 Simulation model at the Network level of OPNET

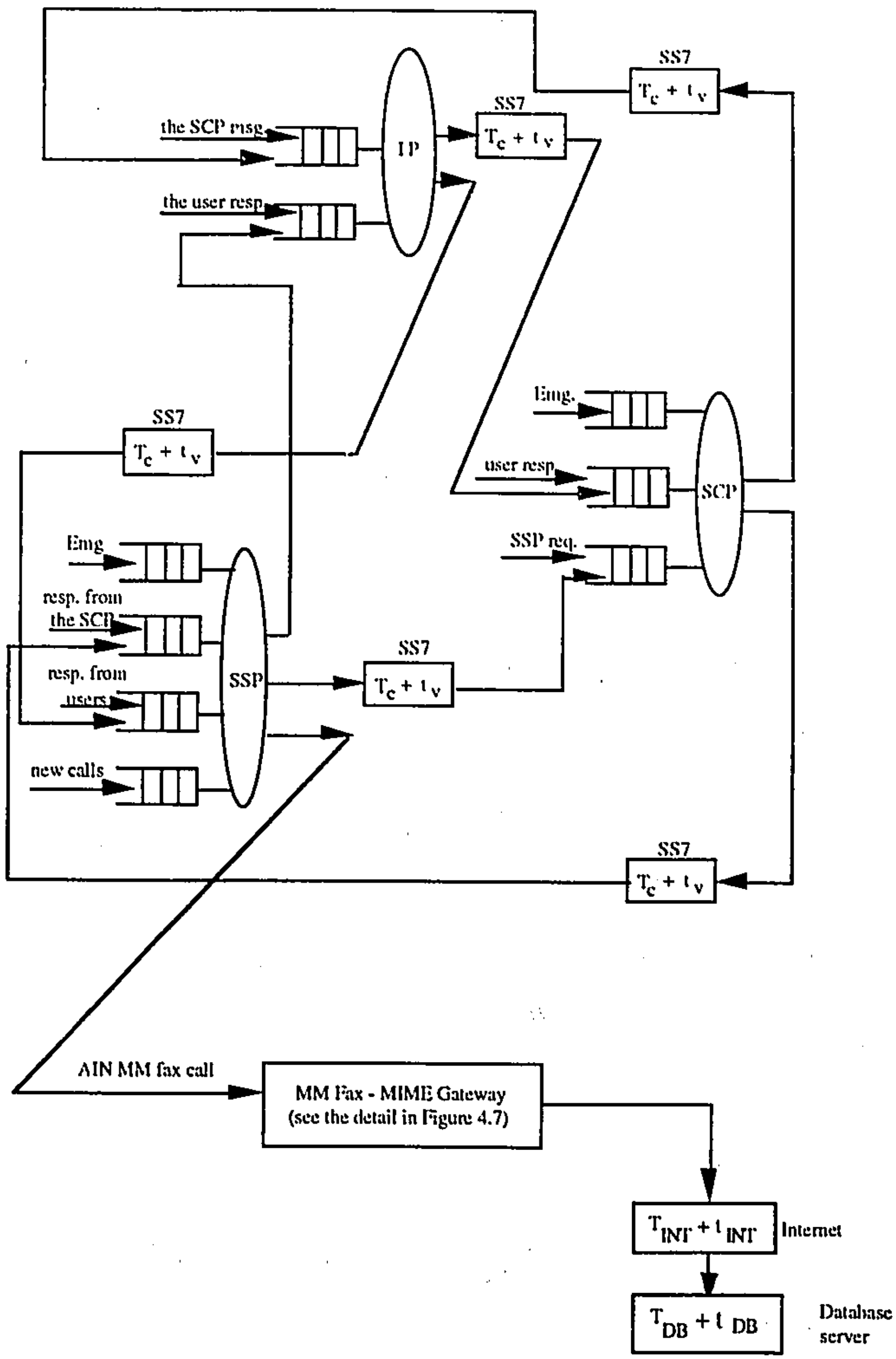


Figure 4.11 The Entire Simulation Model

4.3.2 Validation of Simulation Models

The entire system simulation model consists of several nodes. In turn, each mode consists either a non-preemptive priority queuing system or a delay center as described section 4.2.2 to 4.2.7. To assess the correctness of the simulation model, we validate the operation of each network elements. This validation is carried out by isolating each element and running individual simulations. Obviously, this operation is only performed for each network element modeled by a non-preemptive queuing system. The results obtained from these simulations are then compared with the numerical results of the non-preemptive queuing system described in section 4.2.1. Table 4.1 shows: i) the values, obtained from the simulation (under a confidence level of 95%), of mean delay for the queuing system which is used to model the SSP, SCP, IP, and Gateway (the number of sub queues might be changed, but they are all of the non-preemptive priority queuing systems); ii) the values obtained using the formula (2) in the section 4.2.1. The time of simulation was set to 5000 seconds. The parameters used in the simulation and the calculation are as follows:

$\lambda_1 = 0.2$ packets/sec

$\lambda_2 = 10$ packets/sec

$\lambda_3 = 5$ packets/sec

$\lambda_4 = 100$ packets/sec

$\mu = 200$ packets/sec

As seen from the table, the results show close agreement. Therefore, we can conclude safely that the model is valid. We run the delay model which is used in our simulation with one second mean delay for exponential distribution. The value of mean delay obtained from the simulation is equal to one second. Since both the priority queuing model and the delay model are valid, we conclude that the entire system's simulation model is valid.

	Simulation Value (sec)	Calculated Value (sec)	Discrepancy (%)
Queue 1	0.007809	0.007883	0.939
Queue 2	0.007841	0.007884	0.0054
Queue 3	0.008214	0.00828	0.008
Queue 4	0.01244	0.01235	0.0073

Table 4.1 Simulation Values vs. Calculated Values

4.3.4 Simulation Results

In our simulations, we make some assumptions:

- 50% of the total calls are AIN calls which make requests of the SCP services;
- 50% of the total number of AIN calls involve the IP services;
- on the average, each AIN call, with user interactions, requires two interactions with the IP.

One set of parameter values used in the simulation is listed below. All the following parameter values, except for the values of x_{ssp} (SSP mean service time), P_{MM} (the percentage of our multimedia fax service call) and the packet field "num_of_ip_queries" (indicating the number of user interactions), will be referred later as the **basic value set.**

For the SSP node:

$\lambda_{SSP-NEW}$ = 100 calls/second

$\lambda_{SSP-SCP}$ = 50 calls/second

λ_{SSP-IP} = 50 calls/second

λ_{SSP-E} = 0.1 calls/second

Service time $x_{ssp} = 1/\mu_{SSP}$ = 3ms, 3.5ms, 4ms[PB94]

where $\mu_{SSP} = \mu_E = \mu_{SCP} = \mu_{IP} = \mu_{NEW}$

$P_M = 1\%$

For the SCP node:

λ_{SCP-E} = 0.1 calls/second

λ_{SSP} = 50 calls/second

λ_{SCP-IP} = 25 calls/second

Service time $x_{scp} = 1/\mu_{SCP}$ = 10ms[PB94], where $\mu_{SCP} = \mu_{SCP-E} = \mu_{SCP-SSP} = \mu_{SCP-IP}$

For the IP node:

λ_{IP-SSP} = 50 calls/second

λ_{IP-SCP} = 25 calls/second

Service time $x_{ip} = 1/\mu_{IP}$ = 10ms, where $\mu_{IP} = \mu_{IP-SSP} = \mu_{IP-SCP}$

For the fax gateway node[1]:

Service rate $\mu_{Gateway}$ = 25 M bits/sec

Buffer size = 2 G bytes

For the Internet node:

T_{INT} = 50ms, t_{INT} = 50ms

For the database server in the Internet:

T_{DB} = 50ms, t_{DB} = 50ms

[1]. The traffic load in the gateway is specified in each simulation case.

For the SS7 node:

$T_C = 10ms$, $t_V = 1ms$ [PB94]

For the packet formats:

The value for the field "num_of_ip_interaction" is chosen among 2 to 6.

For the delay threshold:

Assume that the time that each packet stays in a network node (see Figure 4.9) is not greater than 20 seconds; otherwise, the call is considered blocked.

For an SSP service time equals to 3ms and the number of user interactions equals to 2, Figure 4.12 gives the instantaneous behavior of the connection time against the simulation time, which illustrates the stability of the simulation system.

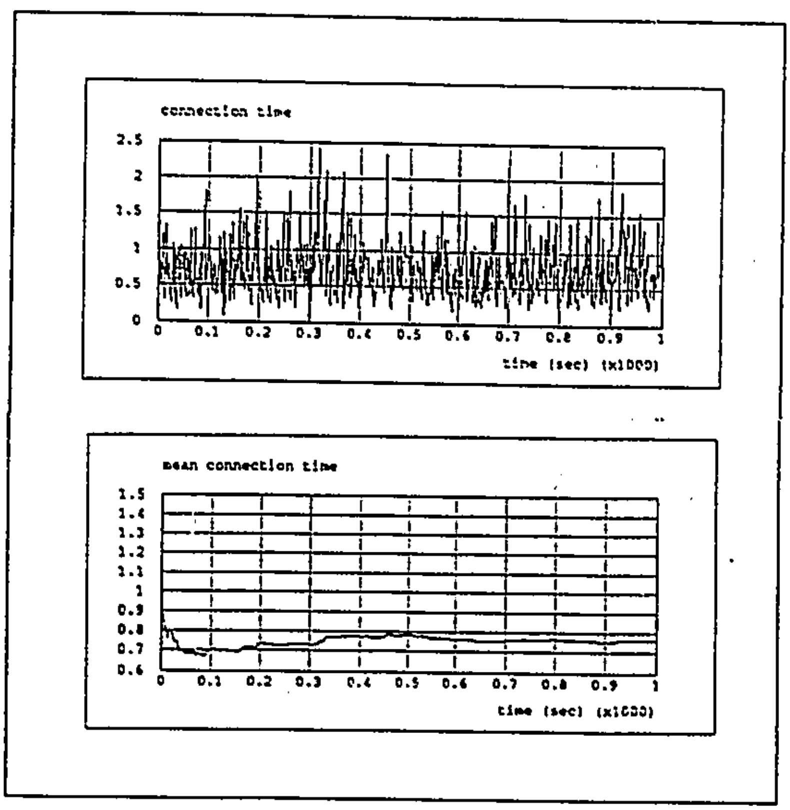


Figure 4.12 Instantaneous Behavior of the Connection Time

In following, the mean call set-up time and the blocking probability for the call of the multimedia fax machine service are evaluated at different aspects. If the value for a parameter is not specified, it takes the value indicated in the basic value set.

Part 1

In this part, we look at the performance of the proposed AIN MM fax service by varying the number of user interactions with the IP. Five different cases are considered. The parameters for case 1 are as follows.

- the SSP mean service time = 3.5ms
- the inter arrival time of signaling = 0.1sec.
- the inter arrival time of text files = 1 sec.
- the inter arrival time of image files = 30 sec.
- the inter arrival time of video files = 120 sec.[1]
- signal file size = 400 K bytes
- text file size: min. = 100 bytes, max. = 10 K bytes, mean = 500 bytes
- image file size: min. = 50 K bytes, Max. = 150 K bytes, mean = 120 K bytes
- video file size: min. = 10 M bytes, Max. = 120 M bytes, mean = 50 M bytes
- gateway service rate = 25 M bits / sec.
- gateway memory = 2 G bytes
- the number of user interactions with the IP = 2, 3, 4, 5, or 6.

In case 2, we only change the SSP mean service time to 3ms. The parameters for case 3 are the same as the ones in case 2 except that the video mean file size is increased to 80M bytes. The parameters for case 4 are the same as the ones in case 2 except that the

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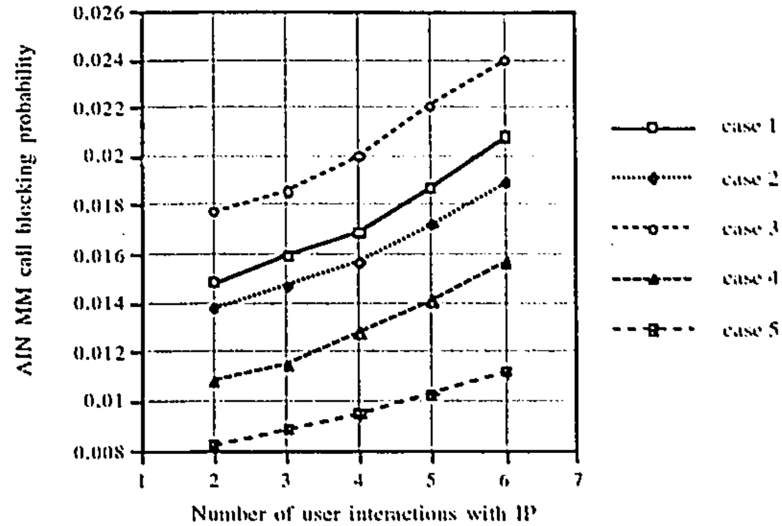
[1] In part 1, we concern that video traffic are only in one direction, from INTERNET to PSTN. Other traffic (signal, text and image) are in both directions, that is, from INTERNET to PSTN, and from PSTN to INNEERNET.

gateway service rate is changed to 30M bits / sec. The parameters for case 5 are the same as the ones in case 2 except that the video mean file size is reduced to 25 M bytes.



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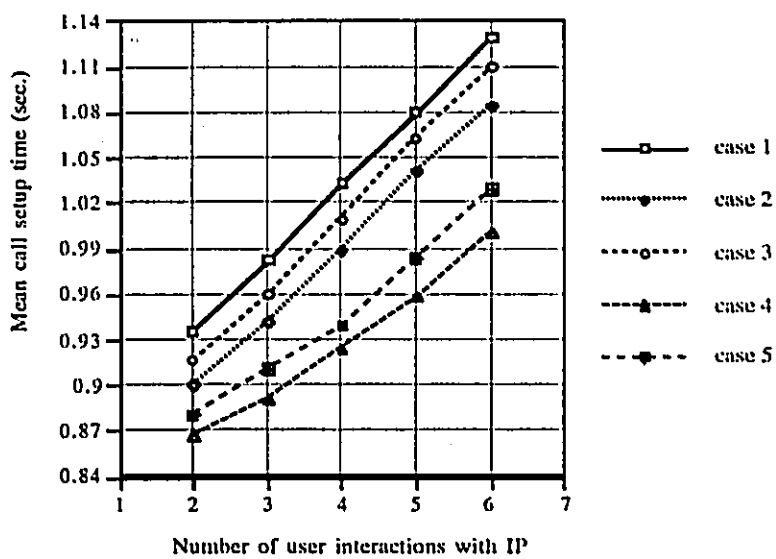


b

Figure 4.13 Call blocking probability vs. number of user interactions

Figure 4.13a shows the call blocking probability at different cases. When the SSP mean service time decreases, the call blocking probability decreases. This is because we define that the call is considered blocked when the call stays in a queuing system for a time[1] longer than 20 seconds. Decreasing the SSP mean service time reduces the time that the call stays in the queuing system, so reduces the call blocking probability. For the same reason, when the gateway service rate increases, the call blocking probability decreases.

The traffic on the gateway also has impact on the call blocking probability. Take the video file as an example. Larger mean video files will put more load on the transmission line and the gateway memory. Hence as the video mean file size increases (decreases), the call blocking probability increases (decreases).

The mean call setup time is affected, in a similar way, by the number of user interactions with the IP, the SSP mean service time, the gateway service time and the video mean file size. Figure 4.13b illustrates the performance of the mean call setup time for all five cases under consideration and as a function of user interactions with the IP.

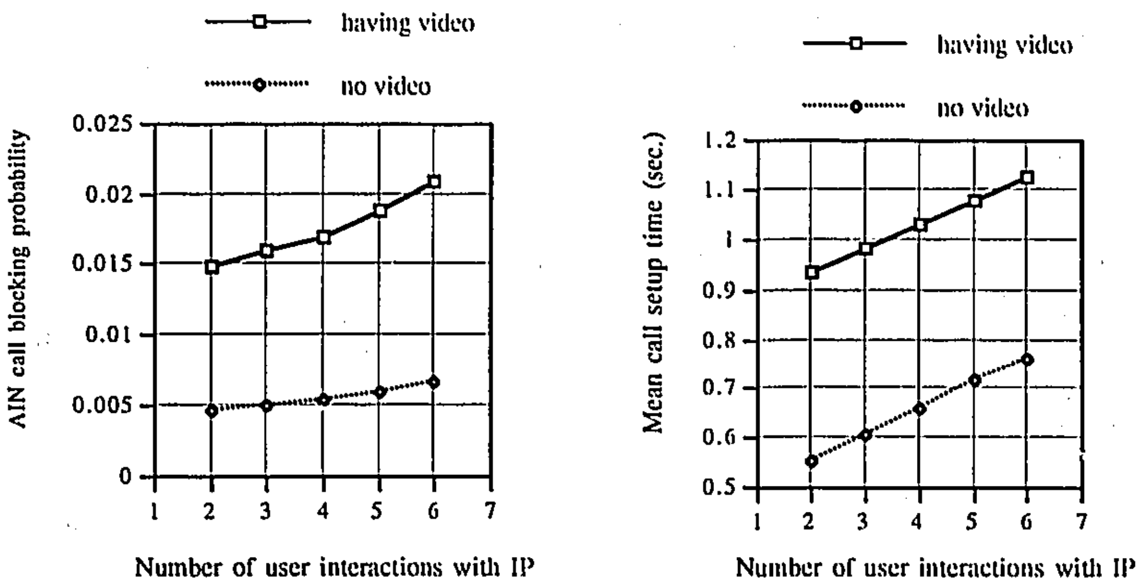
Part 2

In this part, simulations are run to show the impact of video traffic on the performance of the AIN service. For this part, we take case 1, 2 and 4 used in the previous part of our studies. For all these three cases, we further consider the use of video and compare to the case when no video is included in the AIN service. The mean video file size used in this part is set to 50 M bytes.

Figure 4.14a to 4.14c depict the results of this second part. From these figures, the results show that the use of video has a big impact on the performance of the service. For

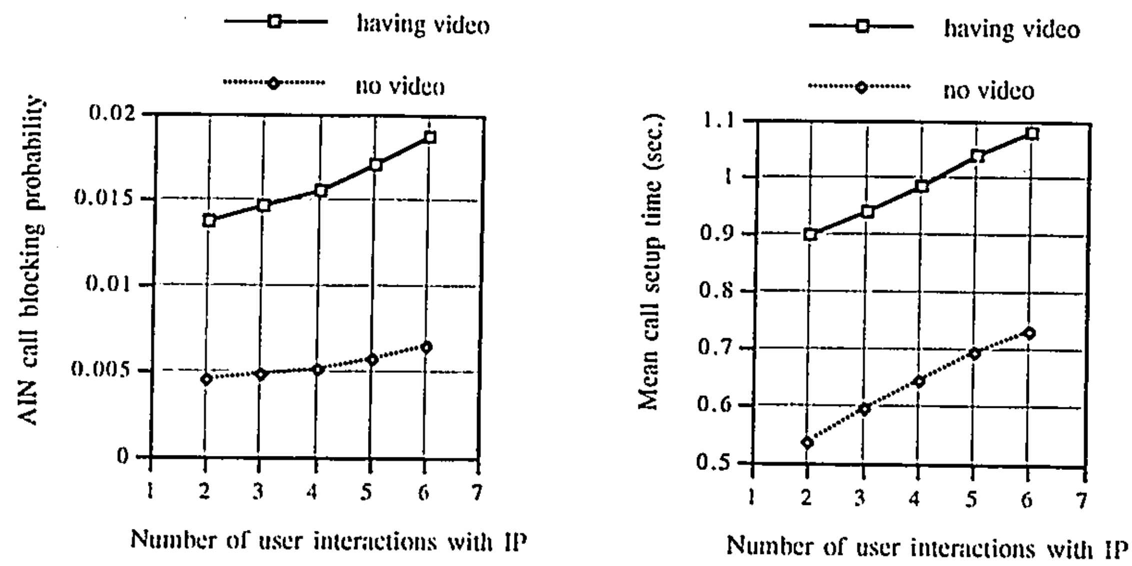
[1] The time the call stays in a queue plus the service time of this call.

all cases, the call blocking probability increases by at least 1%. However, we observe that as the number of user interactions increases, the blocking probability starts to increase at a faster rate. This is mainly due to the fact that as the already established calls become more demanding, new calls are blocked. On the other hand, the mean call setup time is increased half a second when video is used, for all the three cases under consideration. However, this difference between the results for both cases is maintained constant as a function of the number of user interactions. This confirms once again that as the number of user interactions increase when video is being used, new users are stopped from entering the system. The call setup delay accounts for the users who were successful in getting connected to the AIN service.



a

Figure 4.14 Video Traffic Impact on the Performance(Cont'd)



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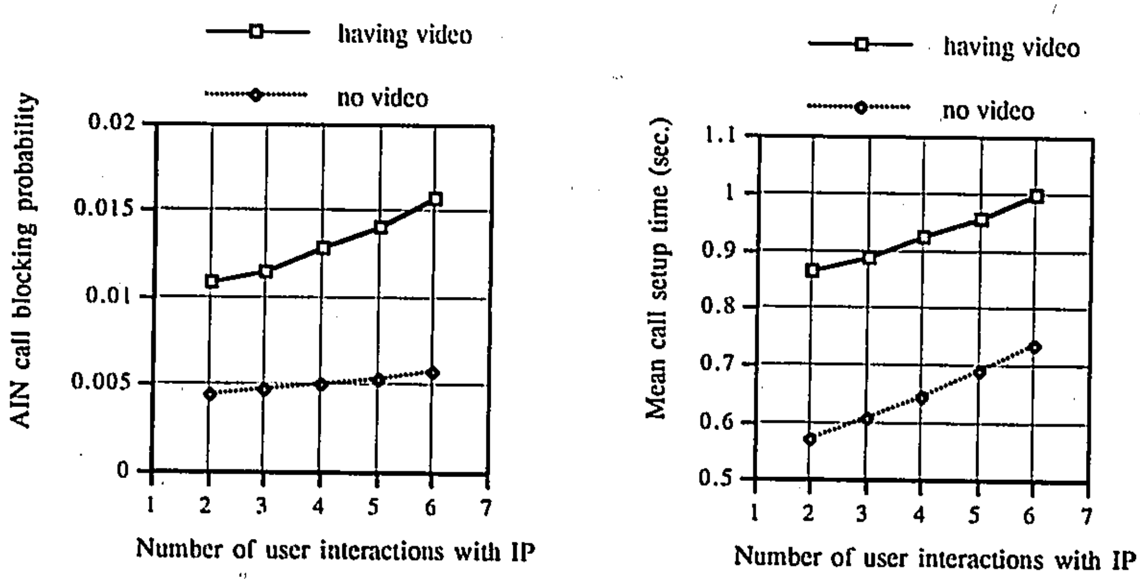

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Figure 4.14 Video Traffic Impact on the Performance

Part 3

In this part, we fix the number of user interactions with the IP to 4, and evaluate the performance against several video traffic situations. Both the INTERNET and the PSTN have video traffic to the gateway. The minimum and maximum video file sizes are set to 1 and 50 M bytes, respectively while various mean video file sizes are set to 2, 3, 5, 10 M bytes. Other parameters are the same as ones in case 2 of part 1.

We can see, from Figure 4.15 and 4.16, that if the video file size is increased, the mean inter arrival rate of the video files must be decreased in order to achieve acceptable performance. For example, if the acceptable call blocking probability is 5%, then for the mean video file size 2, 3, 5 and 10 M bytes, the inter arrival time of the video traffic must be not less than 2, 4, 8 and 10 minutes, respectively. Similarly, if the acceptable mean call setup time is 1 second, for mean video file size 2, 3, 5 and 10 M bytes, the inter arrival time of the video traffic must be not less than 3, 4, 5, and 8 seconds, respectively.

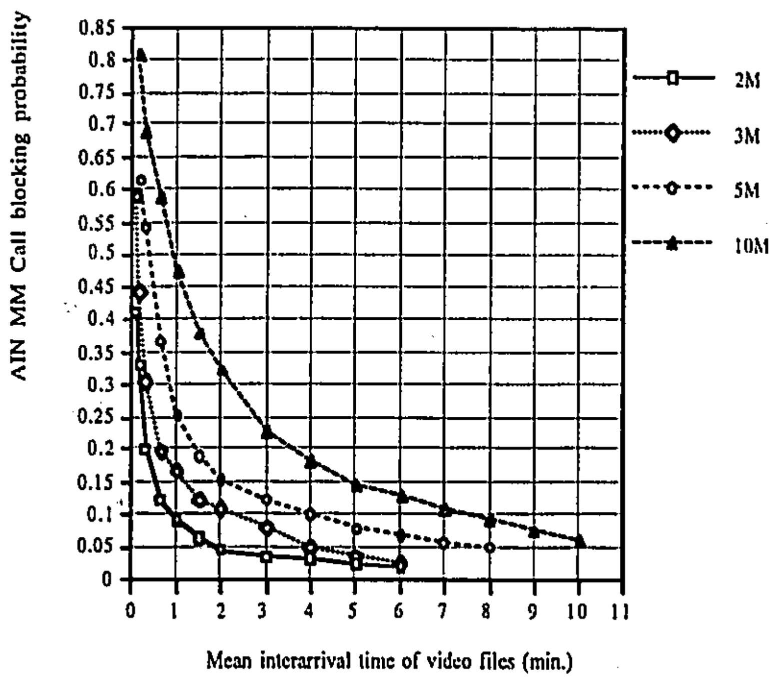


Figure 4.15 Call Blocking Probability vs. Video Traffic Load

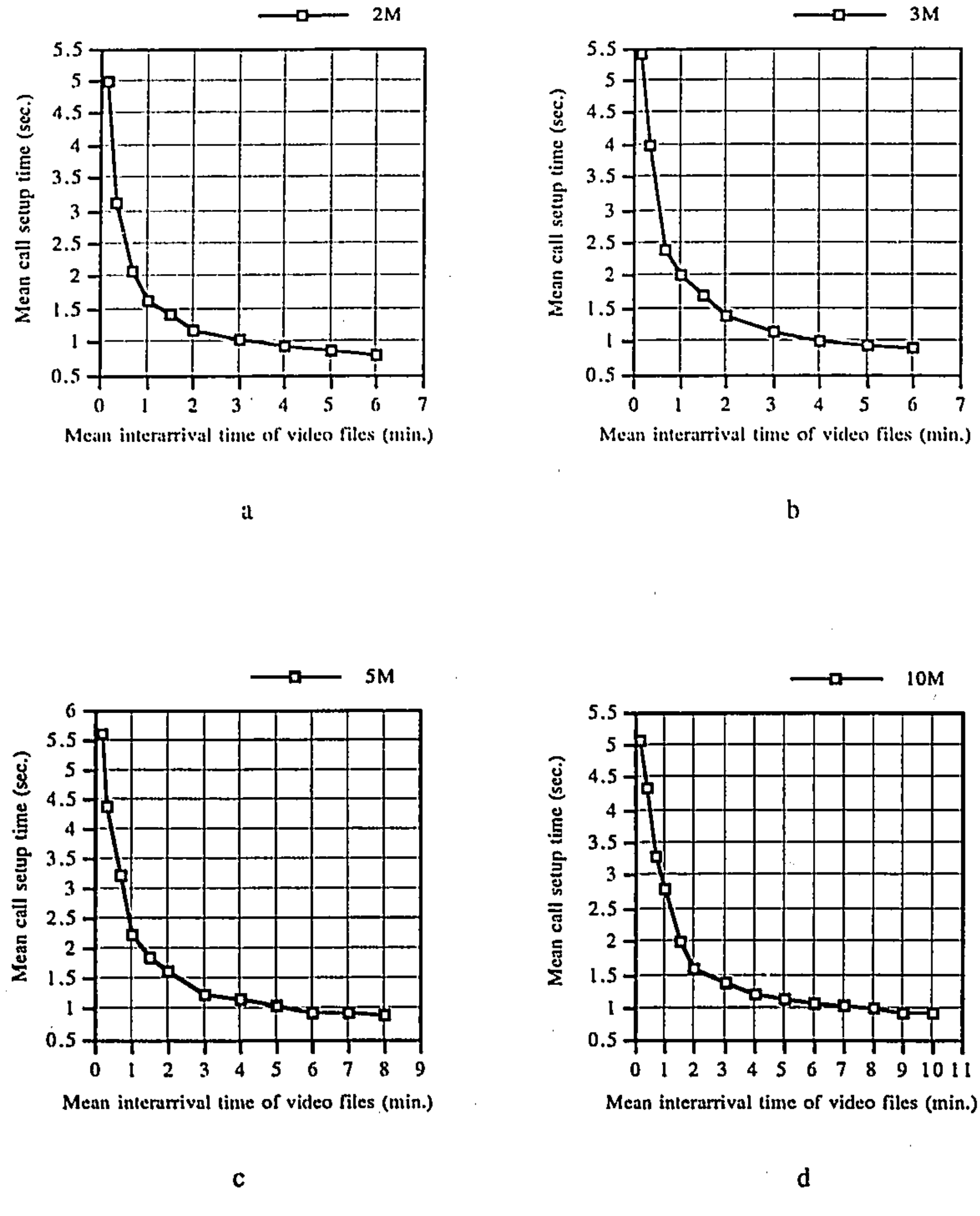
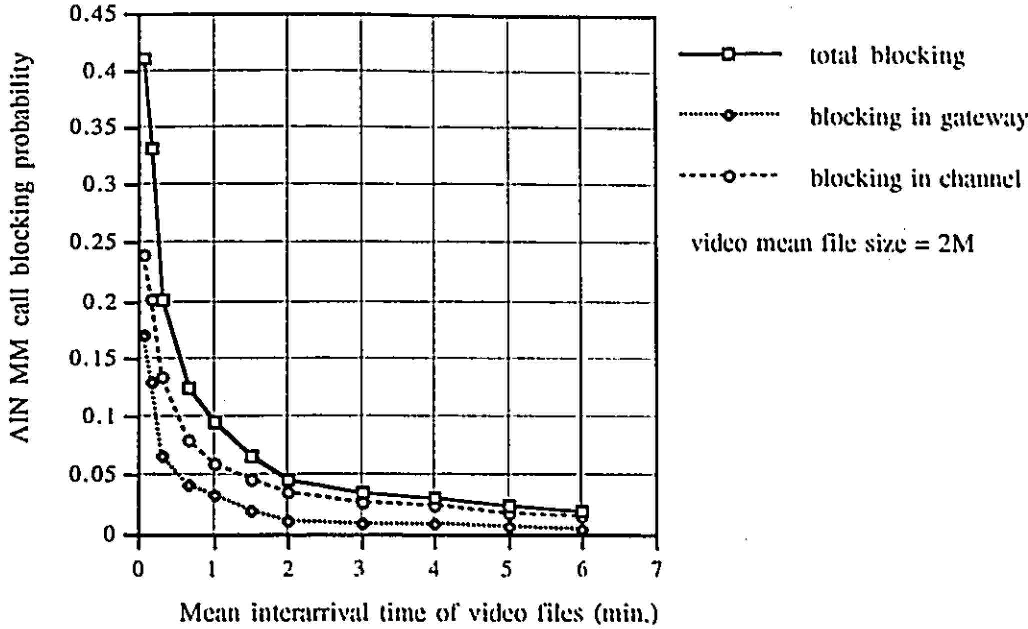


Figure 4.16 Mean Connection Time vs. Video Traffic Load



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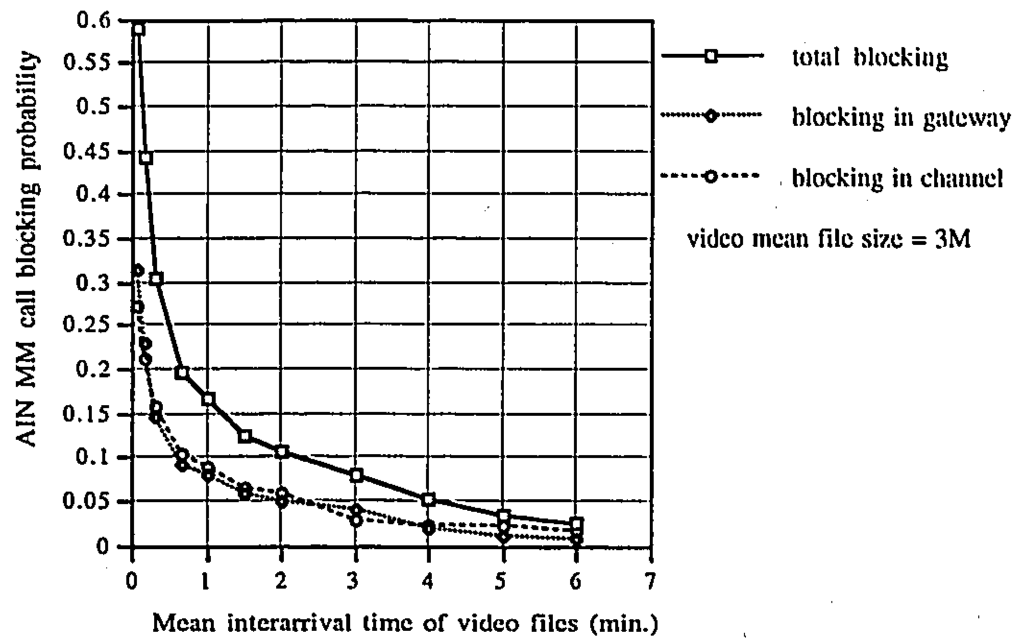

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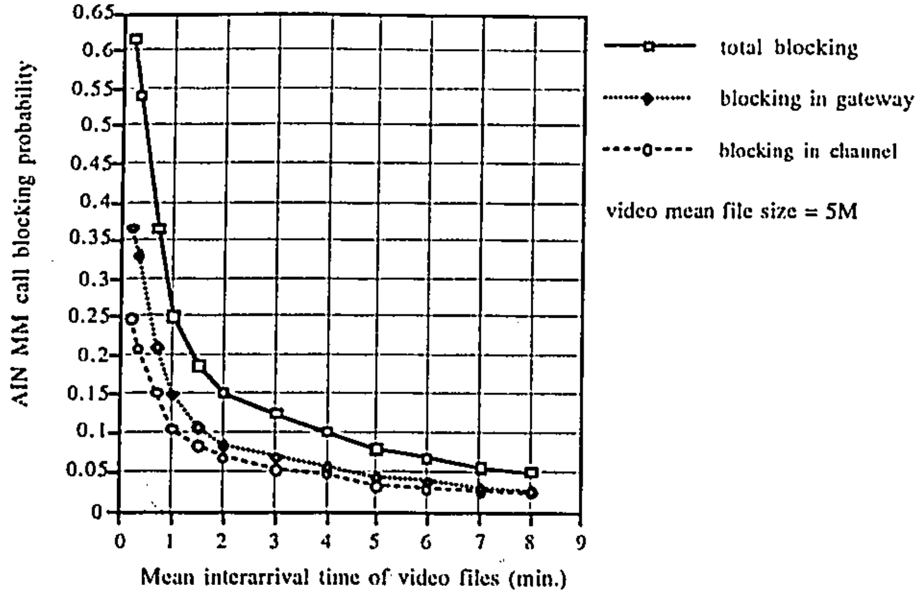
Figure 4.17 Call Blocking Probability vs. Video Traffic Load (cont'd)



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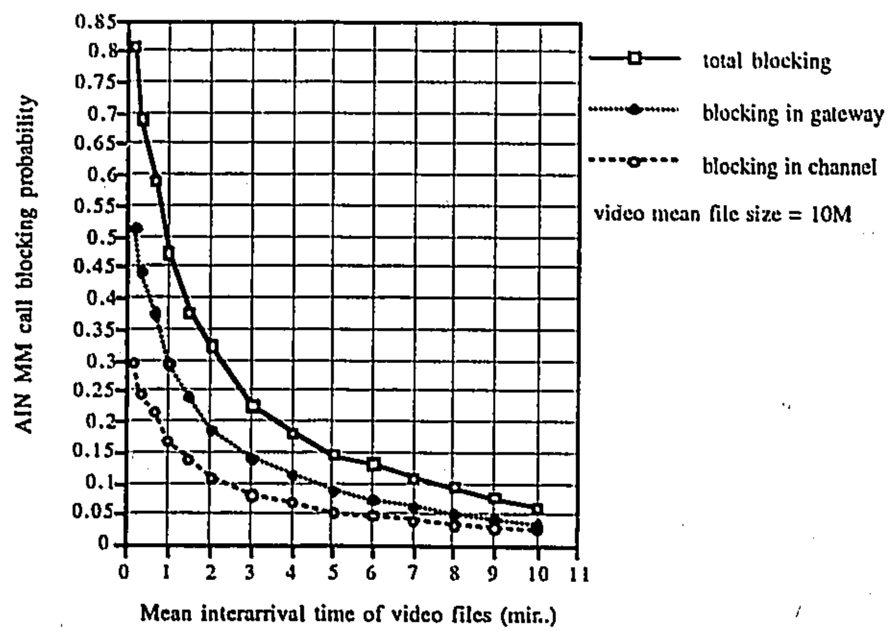

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Figure 4.17 Call Blocking Probability vs. Video Traffic Load

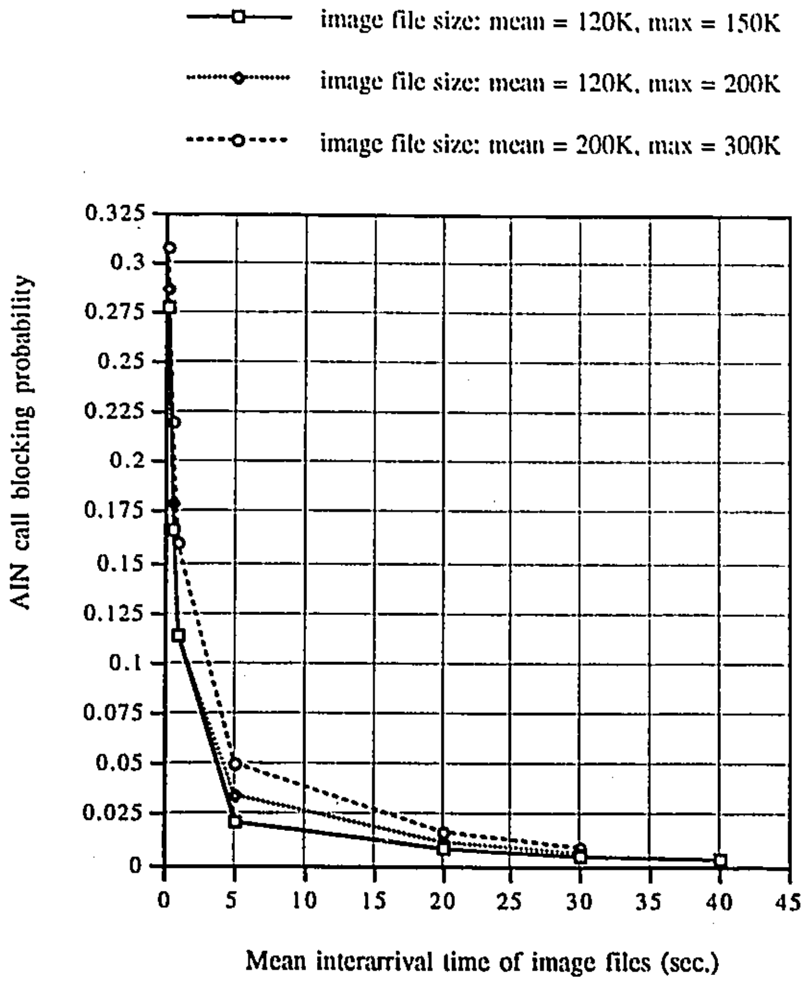

Figure 4.18a Call Blocking Probability vs. Image Traffic Load

Part 4

Figure 4.17 illustrates two major components which contribute to call blocking. With the selected parameters, the call blocking occurs mainly due to the blocking in the transmission channel and the gateway. As the video mean file size increases, more calls are blocked in the gateway . The reason is that bigger video mean file size means that both PSTN and INTERNET put more traffic load to the gateway. This increases the chance that the AIN call is blocked in the gateway.

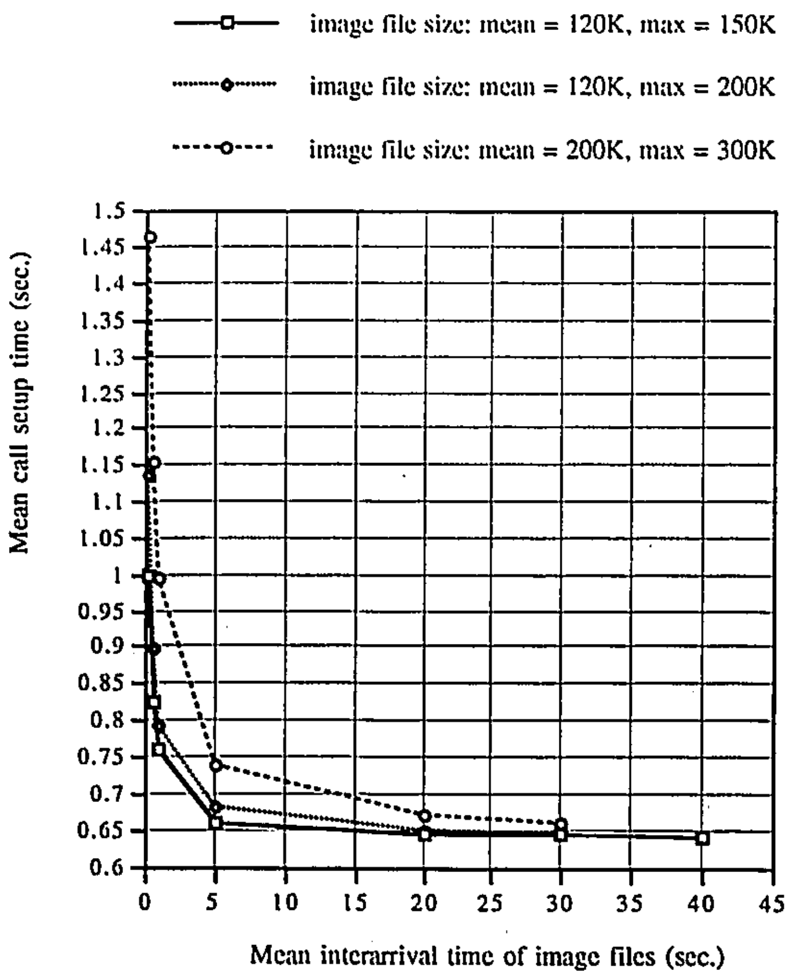

Figure 4.18b Mean Connection Time vs. Image Traffic Load

Part 5

In this part, we assume that there is no video traffic and investigate the image traffic's impact on the call blocking probability and on the mean call setup time. Figure 4.18a and 4.18b shows that since the image file size is much smaller than the video file size and there is no video file, the inter arrival time of image files can go as small as to 5 seconds (for the call blocking probability is not greater than 5%) and the mean call setup time is less than one second (under the parameters we have chosen).

4.3.5 Summary

The performance of the AIN multimedia fax service is evaluated under different conditions. The simulation results illustrate how various parameters could affect the performance. The number of user interactions is one of the call blocking probability factors. When designing the database (for this multimedia fax service), we should, in order to achieve low call blocking probability, minimize the number of queries between the SSP and the IP (see Figure 4.13, Figure 4.14). Video traffic has a great impact on the performance. The heavy load of the video traffic (more frequent video file arrivals or larger video file size) would degrade the performance. The bottleneck to the call blocking probability might be on the gateway when the video traffic increases (see Figure 4.14, Figure 4.15, Figure 4.16, and Figure 4.17). The image traffic has similar impact on the performance (see Figure 4.18). However, since the image file size is smaller, its impact is smaller. Increasing the mean service rate of the gateway and the SSP and the memory of the gateway will improve the performance of our proposed AIN multimedia fax service.

Chapter 5

Conclusions

5.1 Summary of the Thesis

The telecommunications industry is undergoing a fundamental revolution with the introduction of intelligent networks. In this thesis, we investigate this fairly new concept and technology in different aspects:

- the standards that define the intelligent network;
- the technology that implements the intelligent network; and
- the services that can be provided by the intelligent network.

Chapter 1 gives the introductory remarks on the intelligent network, which includes the motivation, technology, components, and the applicability of the intelligent network. The main driving force of the intelligent network is to dramatically reduce the time and cost in the creation of new telephone services, which is achieved by separating the control of calls from the call itself. The services are implemented by service logic programs residing at Service Control Points (SCPs), Service Node (SN) or Adjunct (ADJ), and the call switching is performed, according to the instruction of the SCPs (or SN/ADJ), by the Service Switching Points (SSPs). The Intelligent Peripheral (IP) adds the flexibility to services through the dynamic user interactions. The Common Channel Signaling System No. 7 (SS7) network provides a transport function for the communication between the SCPs and the SSPs or IPs. The AIN service creation environment (SCE) provides a platform for the fast service creation, and the Service Management System (SMS) is responsible for the service provision, maintenance and administration. The operation of the intelligent network is built on distributing the application processing among network elements and using the centralized control. The concept of the intelligent network represents a principle goal of the telecommunication and data communication evolution, and the AIN approach will become the way to build any kinds of networks.

The evolution, standards and the main open issues of the intelligent network are presented in Chapter 2. The evolutionary path of the intelligent network can be depicted through a series of standards (recommendations), from Bellcore's IN/1 to AIN/0.2, and from CCITT's CS-1 to CS-2. Both Bellcore and ITU-T (CCITT) are working towards the new standards. The goal of this path is

- to reduce the complexity of the switches;
- to use cheaper general-purpose computers;
- to provide standardized protocols and interfaces between intelligent network components, which are independent of network equipment and services; and
- to smooth the evolution.

The primary objective is to reduce the time and cost in the introduction of new services. The CCITT (ITU-T) IN conceptual model is an architectural concept, which provides the guideline for implementing the intelligent network. The call processing model is the key of intelligent networks. Different networks, such as the telephone network, the ATM (Asynchronous Transfer Mode) network, the mobile network, etc., have different call processing models. Since AIN technology is fairly new, a number of significant technical details, for example, the service interactions, congestion control, service independent building blocks, etc., need to be carefully addressed and formulated.

Service creation is the emphasis of this thesis. Three services, namely, the "call by name" service, the "automatic telephone directory" service and the "information query" service, are presented in Chapter 3. All these services are designed and implemented (or partly implemented) in the AIN Service Creation Environment (SCE) which uses the Bellcore SPACE system. The services take the AIN advantages of flexible user interaction and automatic telephone number translation. The "call by name" service translates the callee's "name", which is easy for remembering, to a telephone number. The "automatic telephone directory" service makes use of the IP to achieve fast and reliable telephone number seeking. The "information query" service allows people to query information in a fast and cost effective way. It handles the user's various types of information requests by routing the calls to the nearest database server which can answer the requests through mail, fax, voice, or e-mail.

In Chapter 4, we use existing AIN technology to provide the multimedia fax service. With a Multimedia Fax-MIME Gateway between the telephone network and the Internet, the intelligent network can provide automatic addressing, dynamic routing and flexible selecting of document type for the multimedia fax service. The approach of this AIN multimedia fax service brings up a new AIN service opportunity which allows telephone subscribers to access abundant Internet resources. The service is simulated with OPNET (a powerful simulation tool). The components of the system, such as the SSP, SCP, IP and Gateway, are modeled by the priority queuing system. The SS7 network, Internet and the database server in the Internet are modeled by delay nodes since, at the application level, the effect of these components is adding some delay to the call processing. The service performance is evaluated in terms of the mean call setup time and the call blocking probability, and at different scenarios. The simulation results illustrate the performance functions of traffic load, the number of user interactions and the mean service time of the SSP and the gateway.

5.2 Suggestions for Future Research

From the literature of the intelligent network, we notice that only a little work has been done on the performance evaluation of AIN operations. Part of this thesis presents preliminary studies on the performance issues of the intelligent network. As an extension of this thesis work, future research could address the following aspects:

- models of traffic sources, and the performance under different traffic source models;
- the model of the SS7 node which considers the heavy load of signaling traffic;
- the congestion control in the SCP node and the SS7 node;
- the model which involves more than one SSPs, SCPs, and IPs.

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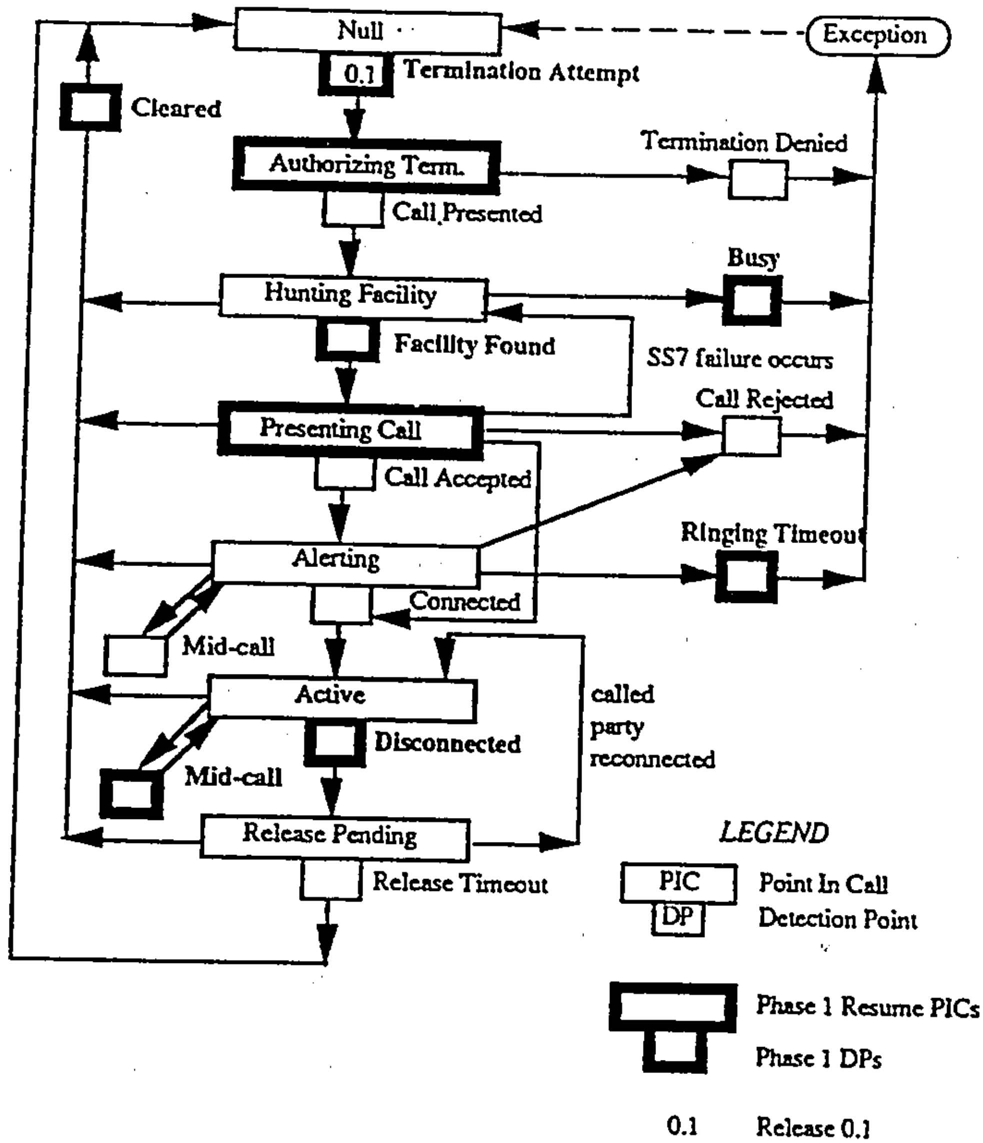
Appendix

Appendix 1: CS-1 Services and Service Features

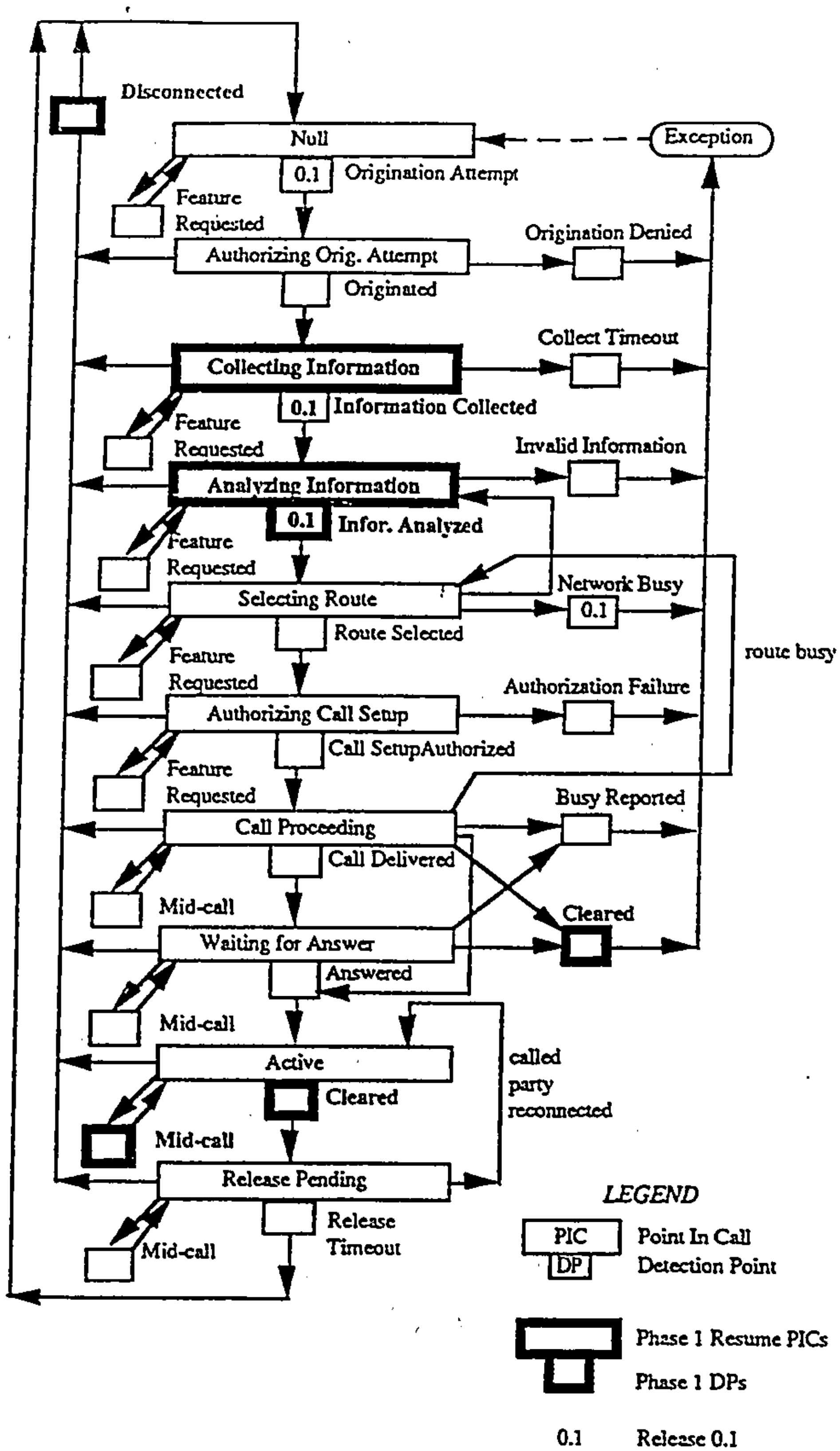
Reverse Charging
Call Gapping
Call Queueing
T_Call Screening
Customer Profile Management
Origin Dependent Routing
Time Dependent Routing
Abbreviated Dialing
Authorization Code
Off-Net Calling
Mass Calling
Premium Charging
One Number
Call Logging
Call Forwarding
*Call Waiting
*Meet-Me Conference
*Call Hold with Announcement
*Automatic Call Back
Call Distribution
Call Limiter
O_Call Screening
Closed User Group
Follow-Me Diversion
Customized Recorded Announcement
User Prompter
Authentication
Off-Net Access
Attendant
Split Charging
Private Numbering Plan
Customized Ringing
Call Forwarding in BY/DA
Personal Numbering
*Multi-way Calling
*Call Transfer
*Consultation Calling

Note: The service features indicated with a * may be only partially supported in CS1, because they require capabilities beyond those of Type A services.

Appendix 2: AIN Release 1 Call Model



Terminating Basic Call Model



Originating Basic Call Model

Appendix 3: A Brief Description of OPNET

OPNET is designed to provide a comprehensive development environment supporting the modeling and performance-evaluation of communication networks and distributed systems. The system behavior and performance are analyzed by performing discrete event simulations. The key features of the OPNET package are:

• Object oriented

OPNET models are specified in terms of objects, each with configurable sets of attributes. The package supports flexible definitions of new objects with programmable characteristics and behavior in order to address as wide a scope of systems as possible.

• Specialized in communication networks and information systems.

Model building blocks focus primarily on communications and information processing to accelerate development efforts for networks and distributed systems.

• Hierarchical models

The OPNET models are hierarchical to naturally parallel the structure of actual communication networks. There are four levels in the hierarchical models: the Network level, the Node level, the Process level and the Parameter level. This feature, plus the object oriented feature, makes the OPNET models easily to be designed and understood.

• Graphical specification

Wherever possible, models are entered via graphical editors which provide an intuitive mapping from the modeled system to the OPNET model specification.

• **Flexibility to develop detailed custom models**

The OPNET provides a flexible, high-level programming language with extensive support for communications and distributed systems. This environment allows the rapid modeling of communication protocols and algorithms.

• **Automatic generation of simulations**

Model specifications are automatically compiled into executable, efficient, discrete-event simulations implemented in the C programming language.

• **Integrated analysis tools**

Performance evaluation, and trade-off analysis require large volumes of simulation results to be interpreted. The OPNET includes a tool for graphical presentation and processing of simulation output.

• **Advanced debugging**

All OPNET simulations automatically incorporate support for program testing via a sophisticated interactive debugger.