ADDIS ABABA UNIVERSITY
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DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING

ASSESMENT OF NEXT GENERATION NETWORK, SERVICES AND MIGRATION STRATEGIES

The Case of Ethiopian Telecommunication Corporation

A thesis submitted to the School of Graduate Studies of Addis Ababa University in partial fulfillment of the requirements for the Degree of Masters of Science in Electrical Engineering

By: Bewket Abrha

Advisor: Dr.-Ing Hailu Ayele

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LIST OF ACRONYMS

ATM	Asynchronous Transfer Mode
BGP	Border Gateway Protocol
BT	British Telecom
CFS	Cash Flow Statement
CO	Central Office
CPE	Customer Premise Equipment
DiffServ	Differentiated Services
DSLAM	Digital Subscriber Line Access Multiplexer
DWDM	Dense Wavelength Division Multiplexing
ETC	Ethiopian Telecommunications Corporation
GW	Gateway
IETF	Internet Engineering Task Force
Intserv	Integrated Services
IP	Internet Protocol
IRR	Internal Rate of Return
ITU	International Telecommunications Union
LAN	Local Area Network
LE	Local Exchange
LSP	Label Switched Paths
MG	Media Gateway
MGC	Media Gateway Controller
MGCP	Media Gateway Control Protocl
MOS	Mean Opinion Score
MPLS	Multi-Protocol Label Switching
MUX	Multiplexing
NGN	Next Generation Network
NPV	Net Present Value
PBX	Private Branch Exchange
PC	Personal Computer
PDH	Pleio Synchronous Hierarchy
POTS	Plain Old Telephony Service
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RSS	Remote Subscriber Stage
RSVP	Resource Reservation Setup Protocol
SDH	Synchronous Digital Hierarchy
SIP	Session Initiation Protocol
SME	Small and Medium Enterprise
SOHO	Small Office/Home Office
STB	Set Top Box
STM	Synchronous Transport Module
TDM	Time Division Multiplexing
TV	Terminal Value
TVM	Terminal Value Multiple
VoB	Voice over Broad
VoD	Video on Demand

VOIP	Voice over Inter Protocol
VPN	Virtual Private Network
WAN	Wide Area Network
WDM	Wave Division Multplexing

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Abstract

The development of telecommunication systems during the past two decades has been from the circuit switched towards the packet switched paradigm. Initially a distinction was made between telecommunications and computer networks which is hardly applicable in today's reality. Incumbent network operators are running legacy circuit-switched telephony networks which inhibit them from enjoying the benefits of packet switching. Many operators have now started the paradigm shift in the telephony network by moving to packet based technologies. This new approach is often called Next Generation Networks (NGN). According to ITU *Next Generation Network (NGN) is* a packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It offers unrestricted access by users to different service providers. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users NGN enables the network operators to run all services on one network, i.e. voice, data and video.

In this thesis the transformation of the legacy network to NGN in general and the Ethiopian Telecom migration in particular will be examined. This work reviews the evolution of voice telecommunications and attempts to analyze the network and service point of view as the industry migrates from circuit switched networks to packet switched networks. For the network part, the work tries to asses the legacy PSTN network elements with the NGN softswitched network. In the service part, a service modeling tool from Alcatel have been used to analyze and model next generation network services for the case of Ethiopian Telecommunication Corporation.

CHAPTER I
GENERAL INTRODUCTION

1.1 Motivation of the Report

From the introduction of telephony in the late 19th century the basic concept of setting up a dedicated switched tunnel capable of transmitting the human voice in a more or less recognizable form between two end users has not changed. Connections were established dynamically through switching offices and transmission throughout the telephone network was analogue, with the actual voice signal being transmitted as an electrical voltage from source to destination.

With the advent of digital electronics and computers, digital transmission became possible. Since the 1970s telecommunications networks have evolved towards digitally switched networks where each conversation is transmitted as a statically reserved 64 kb/s stream. Today the Public Switched Telecommunications Network (PSTN) stands at the brim of a new evolution, the transformation towards packet based, Next Generation Networks (NGN). While most people agree that the future of telecommunication service provider networks is based on NGN concepts, less is know about the financial impact it will have. The turning point that NGN marks represents a unique chance to build up a new network with increased efficiency. The choice faced by incumbent operators now is how to manage this transition from a voice centric circuit switched network to a data centric packet switched network taking into consideration that the main revenue provider in the short term is still simple voice switching but that to ensure long term survival it will be essential to support a platform for more advanced data and multimedia services

The author in his role in as an Engineer in different areas of Ethiopian Telecommunications Corporation has been involving in various circumstances to the next generation network technology. The major one for the initiation of this work is a Workshop held in Nairobi, Kenya on “Transitional Scenarios towards NGN”, from August 30-September 03, 2004, sponsored by ITU, IEE, and Alcatel.

1.2 Overview

After a brief description of overview and motivation of the thesis, chapter two will describe the general problem description. The goal of the research with the proposed thesis structure will be presented and the scope of the work will be defined. The basic limitation in relation to the whole work in general and to the modeling tool in particular will be presented.

In the third chapter the Current ETC networks which will be affected with the evolution will be presented. Here the technology standards of the existing transmission, switching and data networks are discussed.

In chapter 4 the architecture and services of the next generation networks based on softswitch technologies will be described in detail. The softswitch architecture like the legacy PSTN can be described as having three elements: Access, Switching, and Transport. Then the basic reason to evolve for most carriers, which is new NGN Services, will be presented.

Chapter 5 is all about next generation network assessment, the technology part. First the performance metrics of a class 4/5 switch like Reliability, Scalability, QOS and Features will be described in relation to Softswitch. Then detail case studies will be seen for the major Incumbents NGN migration plan. Then the case of ETC will be seen in brief. In ETC, most of the local exchanges located in Addis Ababa are from Ericsson, AXE 10. The next generation network architecture of Ericsson is called Engine, and the basic elements to change from PSTN point of view will be addressed here.

Chapter 6 starts with brief introduction on Investment Analysis: Time Value of Money. Then follows a description of the Alcatel Modeling tool. Here some NGN services are selected and modeled in line with ETC and Ethiopian market. Then the results of the modeling tools in terms of Cash Flow Statement and Service analysis are presented.

And the last chapter will be all about the conclusion and possible future works to be done on some of the important issues relating to NGN network platforms and services.

CHAPTER II
PROBLEM DESCRIPTION

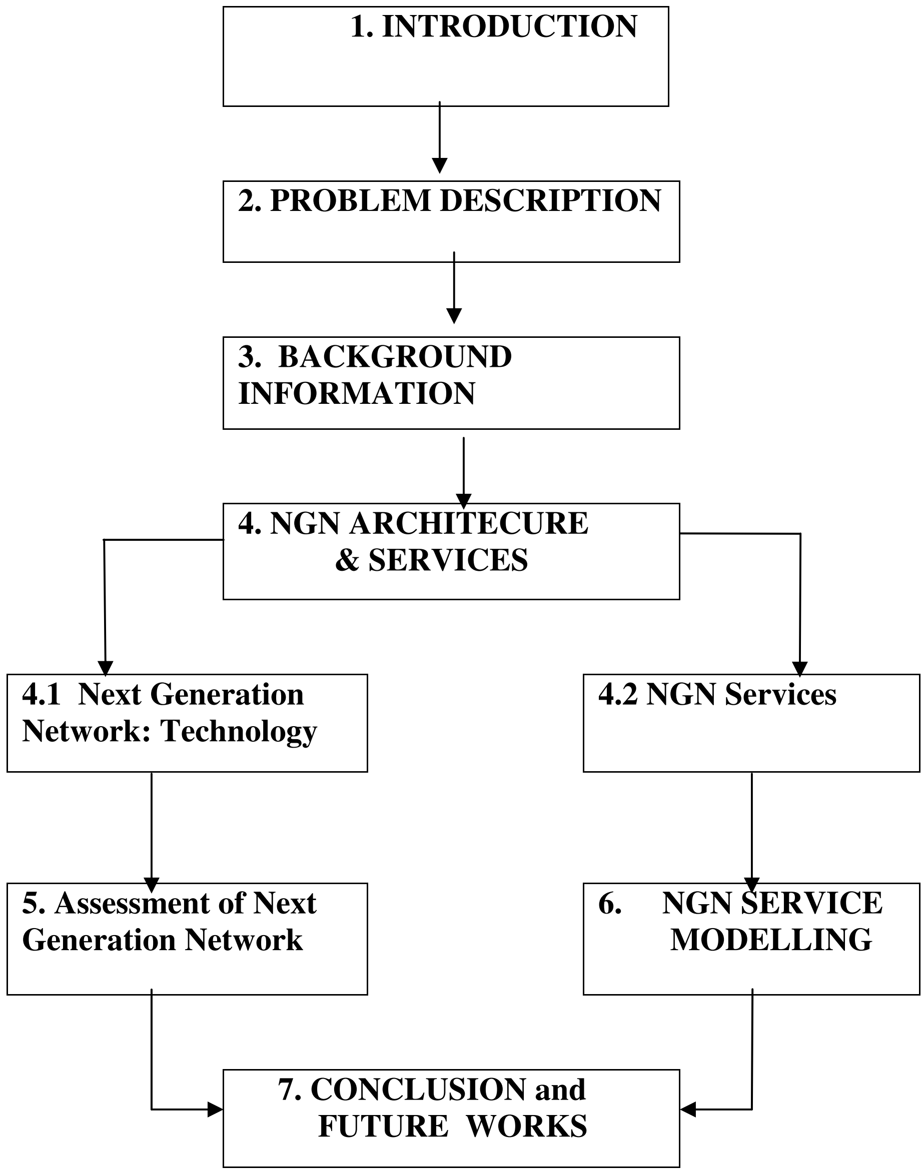
2.1 Research Goal, NGN Economics

Today operators are being faced by several and not always exclusive issues concerning their market and their network. Some of the issues are: Network Expansion, Introduction of new services, Competition aspects and regulation, and CAPEX/OPEX savings. These are taken to be the main drivers for NGN. Other questions to be raised are the choice of the NGN strategy that would guarantee profitability, and to select the available solutions best for the business.

NGN economics is there to answer operator's questions about the economical issues on the evolution of their own network. It may answer them by comparing the business value of the solutions and the related strategic choices. There are two main areas of NGN economics which shall be addressed separately. The first is to determine which new ***services*** are to be addressed using NGN, and the second is to know how to evolve the existing Public Switched Telephone Network towards NGN. The separation comes because of the need to better handle the complexity, and on their nature these are two different strategic questions that need to be addressed separately. New services may require quick deployment of small/medium size network elements to address focused end users segments while PSTN evolution concern a big network with an evolution process which can be extended along the coming 10 years. Nevertheless, consistency between both approaches is needed to ensure future convergence.

This work addresses the evolution of the PSTN by dividing the problem in two major parts: Network and Services. First is the Network part. The main network elements behind the legacy network are the TDM Class 4/ Class 5 switches, and in the NGN networks the carrier grade softswitches. Now that time may have come to deploy NGN, and the work tries to assess the problem by measuring the performance metrics of the elements from both worlds, studying the major technology supplier's position and the best practices of the migration strategies of some incumbent operators. The second part of the paper is about services. This part tries to evaluate the economic feasibility of next generation network services in the Ethiopian Telecommunication Corporation market, and identifies the new services business characteristics in terms of economic parameters.

2.2. Thesis Structure



2.3. Scope and Limitations

In general, the scope of this thesis work focuses on *modeling* some Next Generation Network Services and *technical assessment* of the carrier grade softswitches in relation to the Class 4/ 5 switches of the wire line network.

The major limitations on this paper include

- Focus exclusively on the PSTN
- On the Network part, the approach was made by assessing the Softswitch and Class 4/5 Switches based on their major performance metrics. Assessment on the whole ETC network migration was not made due lack of the Network Modeling Tool, and also due the proposals made by different suppliers on the migration of the existing network.
- There are also some limitations on the NGN Alcatel Service Modeling tool as suggested by Alcatel. The model doesn't support strong volume reduction for CAPEX and it doesn't also consider specific network architecture.

CHAPTER III
BACKGROUND INFORMATION

3.1 History Roundup of ETC

The introduction of telecommunication in Ethiopia dates back to 1894. In those early years, the new technological scheme contributed to the integration of the Ethiopian society when the extensive open-wire line system was laid out linking the capital with all the important administrative centers of the country. Most of the telecommunication network, however, was completely destroyed during the Italian Fascist aggression and later on Ethiopia had to start the development of its telecommunication facilities all over again. When the Imperial Telecommunications Board of Ethiopia was established in 1953, it was granted full administrative and financial autonomy in carrying out its mandate. The major objectives of the Board were: to undertake the expansion of telecom services through out the nation, to represent Ethiopia at all International fora regarding telecom, to allocate and control all communication frequencies, and to train the required personnel in a way expedient to its operation. In order to achieve its objectives, the organization had undergone through series of development programs. One of the major activities undertaken was the establishment of a satellite communication earth station to facilitate international communication services in 1979. By the year 1988, the first digital exchanges went operational in Addis Ababa and other major towns for the first time. Later on in 1996 it was established as a state owned corporation. New services were introduced then; Internet in 1996, Mobile in 1997, and the Digital Data Network in 2001. Just before the implementation of the seventh Telecom development program (1998-2001), in the transitional years, several projects have been executed to rehabilitate the network damaged by the 17 years of war during the Military Regime.

According to the statistics released recently, the current country's telecom penetration goes to around 0.9 %, and this figure increases if mobile subscription is included. [1]

3.2 The Transmission Network

The current ETC transmission network comprises of a hybrid combination of PDH/SDH on microwave and fiber communication lines in the capital and to regional centers. Currently ETC is deploying an optical fiber backbone infrastructure emanating from Addis Ababa along to the major national backbone routes and international connectivity via Djibouti through three consecutive phases. The first phase on completion will serve a total number of around 50 cities and towns in the country.

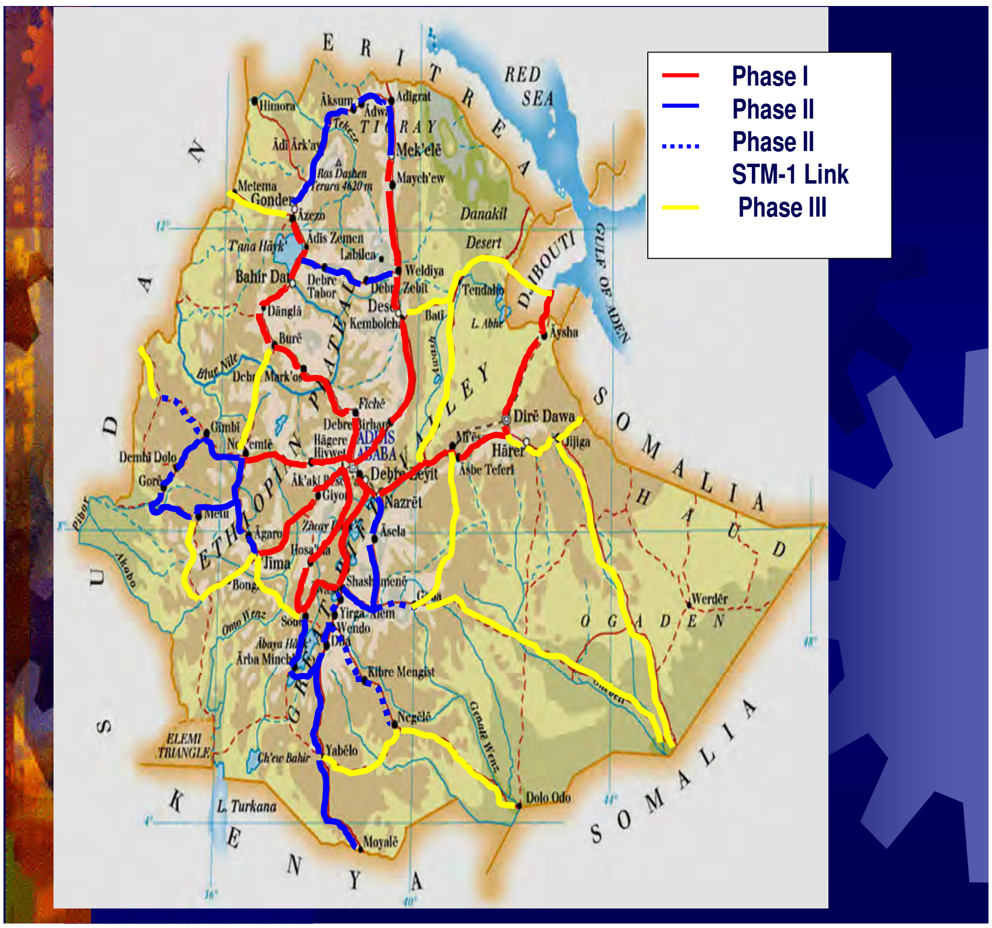


Figure 3.1 ETC Optical Backbone Network

The next portion will describe the technologies exploited in the transmission network of ETC and the current trends of the technology.

3.2.1 PDH and SDH.

Multiplexing digital transmission system is based on the Plesiochronous Digital Hierarchies (PDH) and Synchronous Digital Hierarchies (SDH).

Plesiochronous Digital Hierarchies (PDH)

Practically when 2 Mbps streams from various sources are multiplexed to yield higher order stream, it is possible for the various constituent streams to have slightly different actual bit-rates. Since higher-order streams are created by bit interleaving these 2 Mbps streams, “dummy” bits are added to make the bit rate of all streams equal. The purposes of the “dummy” bits are known to the demultiplexer and they are discarded at the receiver. This introduction of justification “dummy” bits occurs at all levels of the multiplexing in the hierarchy, and this Plesiochronous Digital Hierarchies (PDH).
Plesiochronous Digital Hierarchy (PDH) was intended as a solution for voice telephony channel multiplexing and is not suited for the high-bandwidth connections that the applications of today need. Additionally, PDH is not capable of efficiently multiplexing streams of different bit rates. The topology of a PDH network is the mesh topology where every multiplexer in each site worked with it own clock. In order to synchronize between two multiplexers that work together, usually the transmission was according to the local clock and the reception was made according to the recovered clock that was recovered from the received data.
The PDH contains 4 basic bit rates. These are: E1 – 2.048 Mbps, E2 – 8.448 Mbps, E3 – 34.368Mbps, and E4 – 139.264 Mbps. There is no in band management in the PDH protocol if we need to know the status of one of the multiplexers, or if we need to change the route of one of the trails we have to go to the site or build an outside network that allows us to manage the PDH network.[3]

Synchronous Digital Hierarchy (SDH)

In a transmission system based on SDH, all equipment is synchronized to a common network clock. But there is a mechanism to handle the inevitable delay associated with any transmission link.

The SDH contains the following bit rates: *STM-1 – 155 Mbps, STM-4 – 622 Mbps, STM-16 – 2.5 Gbps, and STM-64 – 10 Gbps.* In SDH, the basic transmission rate is the (STM-1) Synchronous Transport Module –1 that has a bit rate of 155.52 Mbps. This basic rate was chosen to provide direct interoperability with the American transmission hierarchy SONET(Synchronous Optical Network) which has a basic rate of 51.84 Mbps, which is exactly one-third the STM-1 rate. The long-standing interconnection problem between North American and European transmission systems, beginning with the T and E carriers respectively, can finally be brought to an end. Very importantly, an STM-1 signal can carry a number of lower rate signals as payload, thus PDH signals to be sent in an SDH network. This ensures that PDH equipment is not made obsolete by an advent of SDH.

An important difference between PDH and SDH is that higher order streams are obtained in SDH by byte interleaving the lower streams, instead of bit-interleaving as is done in PDH. Therefore, obtaining higher transmission rates in SDH is relatively straight forward. STM-4, corresponding to 622 Mbps and STM-16 corresponding to 2.4Gbps are existing rates that many operators are working at. The space occupied by SDH equipment was a fraction of that occupied by PDH equipment, which in fact supported a lesser bit-rate. Another thing, which instantly obvious is the ease of maintenance. The SDH STM structure has provision for sending maintenance information, and this makes it possible to remotely configure equipment. Using OA&M software running on a general purpose computer suitably interfaced with the transmission equipment, it is possible to view the status of all repeaters and sections in the link, with the option of changing 'drop-insert' configurations at various points in the link. And all this can be operated sitting in O&M center. [3]

3.2.2 The Current Trend

Fiber has been the transmission medium of choice for several years for long hauls as well as for metropolitan area networks (MAN) in inner-city and inner-campus applications. Based on a demand by end customers for higher bandwidth, fiber penetration in the loop plant starts becoming noticeable as well. SONET/SDH technology, based on a time division multiplexing (TDM) approach, has paved the fiber way for ultrahigh bit rates and ultra bandwidths. Many thousands of kilometers of fiber are installed each year around the world. Advances in solid state and photonic technologies have made the idea that certain things "can't be done" a thing of the past; that is, bit rates at 2.5 Gb/s, 10 Gb/s, and 40 Gb/s over many kilometers of single-mode fiber. In addition to traditional TDM services (for example, voice and low-speed data), new services (such as the Internet, high-speed data, video, wireless, etc.) have triggered a big appetite for bandwidth. Currently, voice traffic keeps increasing at approximately 10 percent per year whereas data increases at a rate of 80 percent; because of this difference in growth rates the expectation is that in a year or two the world data aggregate bandwidth will have surpassed the world voice aggregate bandwidth.

However, the existing "legacy" communications systems and network that are based on TDM technology are not able to deliver the bandwidth demand at the quality of service and cost expected by the customer. There must be a new technology based on new systems and networks. Such systems and networks must be able to process and transport incredibly large volumes of voice and data (video, high-speed data, interactive multimedia, etc); they must be able to manage a continuously increasing of bandwidth; and the information conduits in this network must be able to pass an enormous amount of bits per second from one system to another.

Therefore, the following challenges and questions arise.

- As the bandwidth keeps increasing, how do we ensure there is a transmission medium with a scalable bandwidth capacity?
- a technology fast enough to deliver the bandwidth in demand?
- a cost-effective and reliable technology that is also able to match the above two challenges?

Currently, there are few technological options to answer these questions. Nevertheless, all these answers are centered on a key new emerging technology -- WDM, also known as dense-WDM (DWDM) and coarse-WDM (CWDM), depending on the applicability of the technology.

Wavelength division multiplexing

The bandwidth demand continues to increase. And due to the Internet and other data services that require high bandwidth and thus extremely fast bit rates, it has experienced, in recent years, exponential growth in bandwidth demand. Although additional fiber can accommodate this, it does not address the explosive scalability of the communications optical network. Fortunately, super-pure glass fiber with low-loss properties over the wavelength spectrum of 1.3 and 1.55 nm have enabled to put more than a single wavelength in the same fiber (multiplex). The technology, called wavelength division multiplexing (WDM), exhibits an inherent flexibility: it transports all types of traffic and services. Thus, one wavelength may carry Internet traffic, another may carry voice or video. Having put a large number of wavelengths in the same fiber, the aggregate bandwidth per fiber is now multiplied by this number. For example, with 40 wavelengths at 40 Gb/s each, an aggregate bandwidth of 1.6 tera bits per second per fiber (1.6 Tb/s) is realized. In SONET/SDH terms, this is equivalent to 20 million simultaneous conversations per fiber. In addition, WDM systems are becoming more "optical." That is, functionality that was previously implemented with electronics is now achieved with all-optical devices. For example, multiplexing and demultiplexing of wavelengths is accomplished with passive optical prisms or gratings. Optical amplification is also accomplished using specialized doped fibers (for example, erbium-doped fiber amplifiers, or EDFA) or semiconductor optical amplifiers (SOA). And there are many more optical components not mentioned here -- and more to come -- that will eventually transform the communications network to an all-optical WDM network.

Depending on the number of wavelengths used a WDM system is termed DWDM (dense WDM), if many wavelengths are used, and CWDM (coarse WDM), if few wavelengths are used. Moreover, each wavelength may transport different services, such as Internet (IP), SONET/SDH, ATM, or other. WDM technology is evolving and work continues to improve and integrate optical components and, in addition, to draft standards such as wavelength operation, administration, management and provisioning (OAM&P), fault management, network management, reliability and survivability of the system and network, latency, and quality of service. Current systems support fixed wavelength assignment for each node or they are manually reconfigurable. Dynamic wavelength assignment is another area to be nailed down. Wavelength protection is another area for which each network provider provides its own solutions.

3.3 The Data Network

Initially the ETC was having a data cloud called by the name Digital Data Network (DDN) and which supported the following connectivity services: -

- **ISDN**
- **Frame Relay**
- **Leased Line**

With its limited capacity, the DDN has been offering herein listed services.

- Point-to-point digital leased line N X 64 Kbps
- Point-to-point digital leased lines at sub rate speeds
- Frame Relay permanent virtual circuits NX8 Kbps committed information rate
- ISDN BRI service extension from existing Ericsson AXE switches

The National Digital Data Network has the schematic diagram shown in figure 3.2.

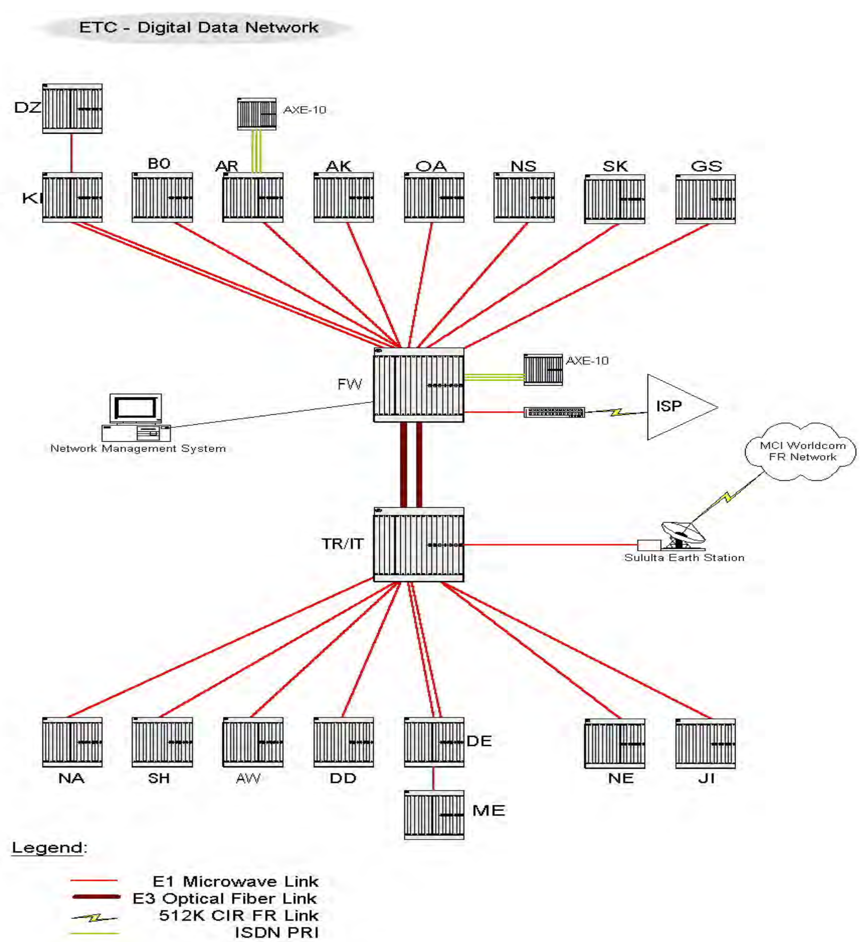


Fig 3.2 Digital Data Network

Additionally, ETC has currently introduced to run in parallel to the DDN, a broadband Multimedia Backbone Network that supports fully the following network services

- ➢ Optical services
- ➢ IP VPN services
- ➢ Voice trunking
- ➢ ADSL access
- ➢ Wireless cable access
- ➢ Wireless radio access

The Broadband Multimedia network consists of three major components

- ➢ **The Core Layer (The ONS Metro Optical Network in Addis Ababa)**
- ➢ **The Edge Layer (Multiservice Switches that support IP, ATM and MPLS)**
- ➢ **The Access Layer**

  (The XDSL and Broadband Fixed Wireless)

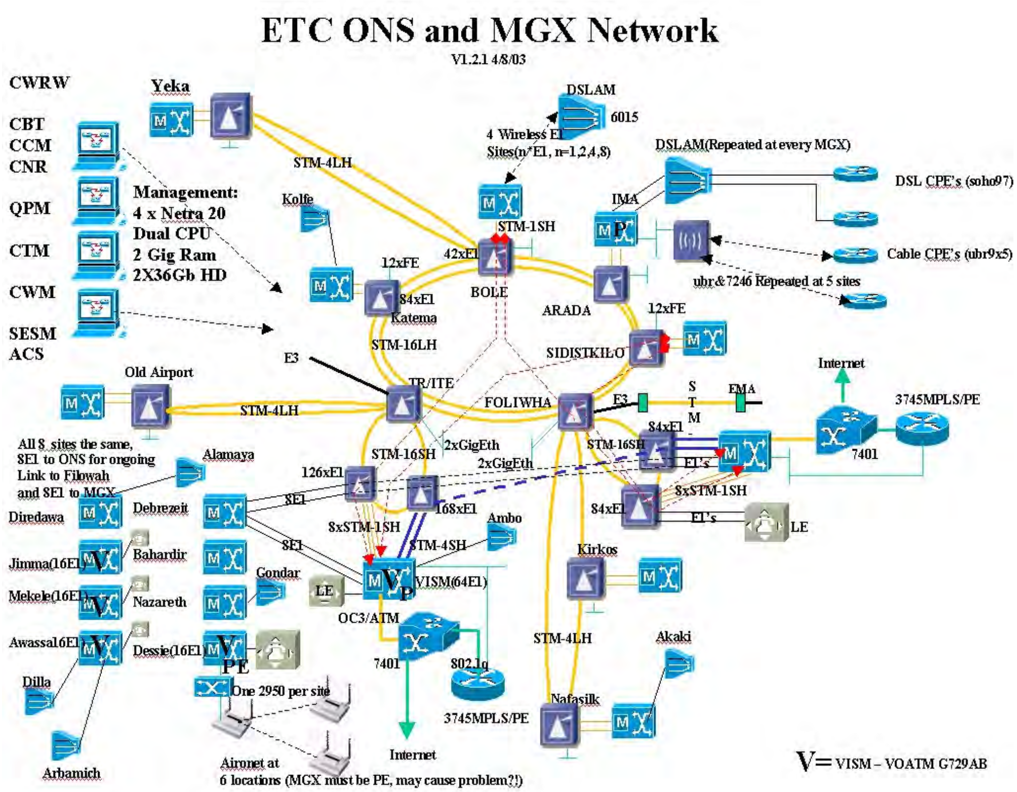


Figure 3.3 Broadband Multimedia Network

The broadband multimedia backbone is an infrastructure based on a SDH core network and on an ATM core network. This infrastructure will support different services such as E1 pass-through, IP MPLS/VPN, voice trunking. The optical network is based on Cisco ONS platforms. The ATM network is based on Cisco MGX8850 switches.
Ethiopia Telecom SDH core network requirement is to support mixed type of services between several location in Addis Ababa and its surroundings. Traffic type is mainly TDM (2 Mbit/s, E1 and E3), necessary for the internal mobile network, but the core network will also serve the ATM network detailed via STM4 and STM connectivity. The SDH network is composed of several nodes of Cisco ONS15454 MSPP platform, connected together in a multiple ring fashion. The core ring is made of two fiber pairs, allowing 4 fiber MS-SPRING protection schemes. The ATM core network requirement is to support both layer 2 services and layer 3 services. Layer 2 services will be based on Private Network to Network Interface (PNNI) protocol. The Layer 3 services will be based on MPLS. Services such as ADSL access, voice trunking will rely on PNNI while the IP VPN services will rely on MPLS.

3.4 The Switched Network

Switched networks offer two services through PSTN: Plain Old Telephone Service (POTS) and Integrated Services Digital Network (ISDN). In the year 2004 the total installed exchange capacity has been more than 720,000 which has shown an 11.2% increase over the previous year. Of the total capacity the digital exchanges accounts for 97.3% and the remaining being analog automatic and manual exchanges. And during this year the total number of telephone subscription lines had reached just over 480,000. Switched networks are based on the oldest principles of telecommunications where circuit switching involved the setting up of a physical path from one point all the way across a transmission network to an end point. These networks were developed to provide basic telephone service, which involves the two-way, real-time transmission of voice signals. In its most basic form, this service involves the transfer of an analogue signal of a nominal bandwidth of 4 kHz across a sequence of transmission and switching facilities.

3.4.1 Structural Components of the PSTN

Modern digital telephone networks combine the circuit-switching approach to operating a network with digital transmission and digital switching. In an over-simplified description of the current telephone network it consists of four structural components:

- Copper lines that connect subscribers to the telephone system (local loop)
- Hierarchy of digital telephone exchanges (local, transit and international)
- Trunk lines (2 Mb/s leased lines) that interconnect the exchanges
- Control and signaling system

The local loop

The legacy copper network provides access through dedicated Unshielded Twisted Pair (UTP) lines that run underground from all subscribers to the nearest Remote Subscriber State (RSS) or directly to the Local Exchange (LE). Along the way from subscribers to LEs, UTP cables are grouped together in Street Cabinets (SC) from where they run in bundles to the LE.

The Digital Telephone Exchanges

The digital exchanges in ETC were primarily composed of AXE 10 from Ericcson, DMS10/100 from Telrad, and Huawei C & CO8 digital exchanges. The AXE 10 exchange holds the major share of the subscribers and the only functional local and international transit switch. AXE 10 switches are composed of modulated software and hardware and a control system (APZ), which is responsible for the operating system and the I/O functions. APZ Control Systems are available with different capacity and memory.

The more the traffic gets and the larger the software system becomes, more memory is needed and more powerful the APZ needs to be. The hardware in an exchange takes care of transmission and switching and is not upgraded much, unless new software requires it. The software on the other side, which takes care of signaling and advanced services, is upgraded regularly (on average each year or every second year).

All local loops terminate in a Remote Subscriber Stage (RSS). If the subscriber is located in the proximity of an LE, the RSS is positioned in the LE. If however a group of subscribers is located far away from the LE, they connect to a detached RSS that connects to the nearest LE with trunk lines. The figure below describes the current ETC switches layered in hierarchal structure.

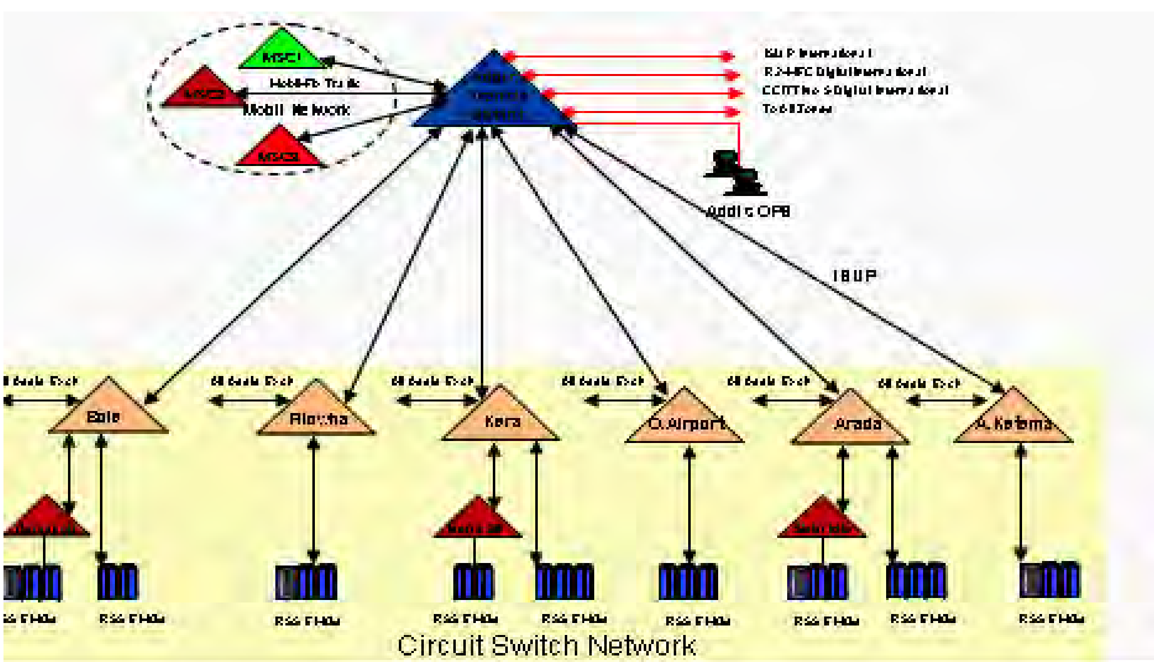


Figure 3.4 ETC Circuit Switch Network

3.5 Overview of Packet Based Networks

In packet based networks information is split up into cells or packet. Each packet contains a part of the transmitted information in addition to a header that defines where the packet comes from, where it is going and the number of the packet. The packet based network then transmits each packet from node to node through the network until it reaches the destination. Packet based networks use different methods for transmitting the packet but in general they are either connection oriented, like ATM, or connectionless datagram oriented, like IP. In ATM and other connection oriented networks a channel is set-up between the originating and destination nodes before the transmission starts. Many channels can exist on the same path and each can have different QoS parameters. This differs from the method used in IP networks where each packet is sent from the originating node without checking if and how the network can get it to the destination node. Benefits of packet based networks include:

- Better utilization of bandwidth since it is shared
- Connection speeds of nodes do not have to be the same everywhere.
- Switched networks have a defined peak value of connections after which they block all incoming calls.
- Packet based networks accept more information although the time delay can increase.
- In packet based networks the information can be prioritized, i.e. limiting delay on special information.

3.5.1 An Overview of ATM

Asynchronous Transfer Mode (ATM) is an International Telecommunication Union standard for cell relay, where information for multiple service such as voice, video, or data, is conveyed in small, fixed-size cells. ATM was originally conceived as a high-speed transfer technology for voice, video, and data over public networks. ATM was standardize by the ITU and was further extended by the ATM Forum for use over public and private networks. ATM is a cell-switching and multiplexing technology that combines the benefits of circuit switching (guaranteed capacity and constant transmission delay) with those of packet switching (flexibility and efficiency for intermittent traffic). It provides scalable bandwidth from a few megabits per second (Mbps) to many gigabits per second (Gbps). Because of its asynchronous nature, ATM is more efficient than synchronous technologies which often use time-division multiplexing (TDM). With TDM, each user is assigned to a time slot, and no other station can send in that time slot. If a station has a lot of data to send, it can send only when its time slot comes up, even if all other time slots are empty. If, however, a station has nothing to transmit when its time slot comes up, the time slot is sent empty and is wasted. Because ATM is asynchronous, time slots are available on demand with information identifying the source of the transmission contained in the header of each ATM cell. ATM transfers information in fixed-size units called *cells.* Each cell consists of 53 octets, or bytes. The first 5 bytes contain cell-header information, and the remaining 48 contain the "payload" (user information). Small fixed-length cells are well suited to transferring voice and video traffic because such traffic is intolerant of delays that result from having to wait for a large data packet to download. ATM networks are therefore connection oriented.

ATM Services: Two types of ATM services exist: *permanent virtual* circuits (PVC) *switched virtual* circuits (SVC).

- PVC allows direct connectivity between sites; a PVC is similar to a leased line. Among its advantages, a PVC guarantees availability of a connection and does not require call setup procedures between switches. Disadvantages of PVCs include static connectivity and manual setup.
- A SVC is created and released dynamically and remains in use only as long as data is being transferred. In this sense, it is similar to a telephone call. Dynamic call control requires a signaling protocol between the ATM endpoint and the ATM switch. The advantages of SVCs include connection flexibility and call setup that can be handled automatically by a networking device. Disadvantages include the extra time and overhead required setting up the connection.

ATM Virtual Connections: ATM networks are fundamentally connection oriented, which means that a *virtual channel* (VC) must be set up across the ATM network prior to any data transfer. (A virtual channel is roughly equivalent to a virtual circuit.) Two types of ATM connections exist: *virtual paths*, which are identified by virtual path identifiers (VPI), and *virtual channels*, which are identified by the combination of a VPI and a

virtual channel identifier (VCI). A virtual path is a bundle of virtual channels, all of which are switched transparently across the ATM network on the basis of the common VPI. All VCIs and VPIs, however, have only local significance across a particular link and are remapped, as appropriate, at each switch. A transmission path is a bundle of VPs.
ATM Switching Operations: The basic operation of an ATM switch is straightforward: The cell is received across a link on a known VCI or VPI value. The switch looks up the connection value in a local translation table to determine the outgoing port (or ports) of the connection and the new VPI/VCI value of the connection on that link. The switch then retransmits the cell on that outgoing link with the appropriate connection identifiers. Because all VCIs and VPIs have only local significance across a particular link, these values are remapped, as necessary, at each switch.

3.5.2 An Overview of MPLS

Multi-protocol Label Switching (MPLS) is a high-performance method for forwarding packets (frames) through a network. It enables routers at the edge of a network to apply simple labels to packets (frames). ATM switches or existing routers in the network core can switch packets according to the labels with minimal time to waste. MPLS integrates the performance and traffic management capabilities of Data Link Layer 2 with the scalability and flexibility of Network Layer 3 routing. It is applicable to networks using any Layer 2 switching, but has particular advantages when applied to ATM networks. It integrates IP routing with ATM switching to offer scalable IP-over-ATM networks. In contrast to label switching, conventional Layer 3 IP routing is based on the exchange of network reachability information. As a packet traverses the network, each router extracts all the information relevant to forwarding from the Layer 3 header. This information is then used as an index for a routing table to determine the packet's next hop. This is repeated at each router across a network. At each hop in the network, the optimal forwarding of a packet must be again determined. The information in IP packets is usually not considered when forwarding packets. Thus, to get maximum forwarding performance, typically only the destination address is considered. However, because other fields could be relevant, a complex header analysis must be done at each router that the packet meets. The main concept of MPLS is to include a label on each packet. Packets or cells are assigned short, fixed length labels. Switching entities perform table lookups based on these simple labels to determine where data should be forwarded to. The label summarizes essential information about routing the packet, which are Destination, Precedence, Virtual Private Network membership, Quality of Service (QoS) information from RSVP, the route for the packet, as chosen by traffic engineering (TE). With Label Switching the complete analysis of the Layer 3 header is performed only once: at the edge label switch router (LSR), which is located at each edge of the network. At this location, the Layer 3 header is mapped into a fixed length or label. At each router across the network, only the label need be examined in the incoming cell or packet in order to send the cell or packet on its way across the network. At the other end of the network, an edge LSR swaps the label out for the appropriate header data linked to that label. A key result of this arrangement is that forwarding decisions based on some or all of these different sources of information can be achieved by means of a single table lookup from a fixed length label. Label switching integrates switching and routing functions, combining the reachability information provided by the router function, plus the traffic engineering benefits achieved by the optimization of capabilities of switches.

MPLS Applications : MPLS networks as shown in have three main applications. Typically, two or all three of these capabilities would be used simultaneously:

- **IP+ATM Integration**

MPLS fully integrates IP services directly on ATM switches. The IP routing resides directly on ATM switches. Thus MPLS allows ATM switches to optimally support IP multicast, IP class of service, RSVP, and Virtual Private Networks.

- **IP Virtual Private Network (VPN) Services**

A VPN service is the infrastructure of a managed Intranet or Extranet service offered by a provider to many corporate customers. These are often massive IP networks. MPLS with BGP offers a very flexible, scalable, and manageable way of providing VPN services on both ATM and packet-based equipment.

- **IP Explicit Routing and Traffic Engineering (TE)**

The IP Traffic Engineering capability of MPLS uses special Label Switched Paths (LSPs) to finely adjust IP traffic flows.

CHAPTER IV
NGN ARCHITECTURE AND SERVICES

4.1 Background in Communications Network

In traditional network architecture, different voice, data and television services are integrated with the particular technology that transports and switches them. The networks are separate, and vertically integrated. This means that within one large telecom operator there can exist many independent networks, PSTN, Mobile Telephony, data and Cable Television, all comprise the same architectural components: management, switching, transport and access layers

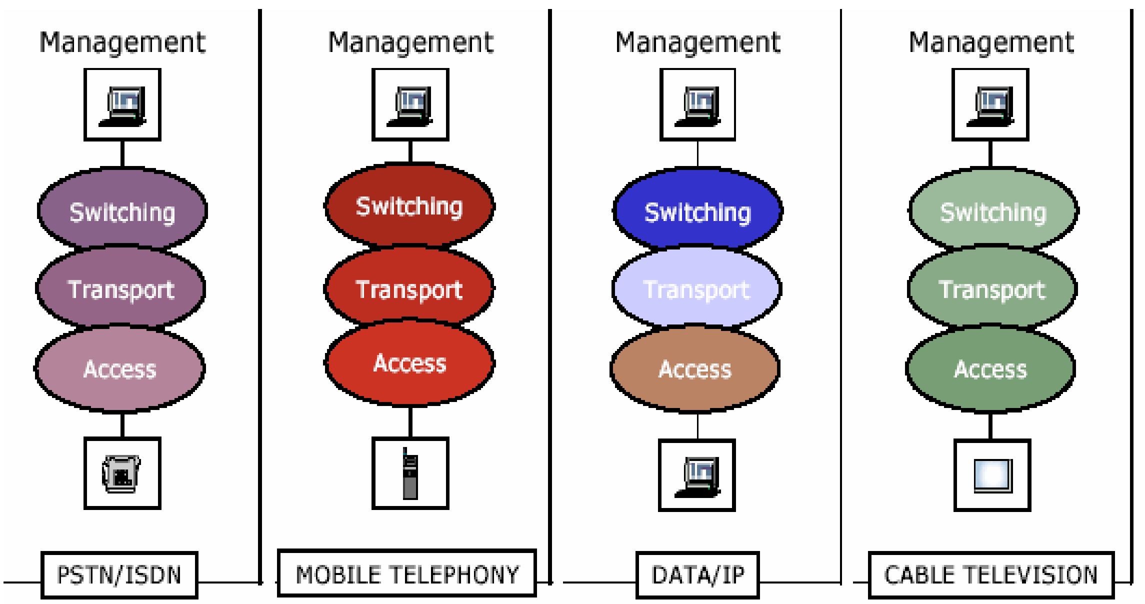


Figure 4.1 Traditional communications network architecture

In the past this was necessary because the requirements of each service were different and each network was designed to meet certain Quality of Service (QoS) parameters that the other networks were incapable of meeting. The thought of constructing a single network that consolidates all the others is therefore at hand and has been around for some time now. Recent advances in fibre optics and computer networks have now made this technically possible. The proposed solutions are based on using one packet based transport network that can guarantee QoS requirements of all the formerly separated

services. To understand and be able to estimate the importance of implementation options fundamental understanding of the technical function of the current as well as the future network is essential. Alongside the development of the PSTN, other communications networks have been built. Having little to do with each other in the beginning, telephone-, mobile-, and data networks are starting to show the same characteristics through modernization. Each system is built up from its own fundamental structure components; management, switching, transport and access networks. The NGN concept is based on combining the distribution of all these Medias in one multipurpose network that fulfils the needs and requirements of all. According to ITU, *Next Generation Network (NGN) is* a packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It offers unrestricted access by users to different service providers. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users.
NGN enables the network operators to run all services on one network, i.e. voice, data and video. This forthcoming converged multipurpose network has the potential of reducing operational cost through better utilization and economies of scale, as well as offering new and improved services.

4.2 Next Generation Networks

The PSTN is a centralized architecture while softswitch architecture offers a distributed architecture. The next section will describe this architecture for softswitch solutions, detailing the major components that comprise this new and flexible architecture. Although many of the rules of construction change drastically with the shift to a packet-switched client-server architecture, the basic functions remain the same. Much of the current architecture for softswitch networks reflects attempts to protect an investment in the legacy aspects of a network, and softswitch components bridge the legacy technology with next generation technologies. As economic models dictate new strategies, legacy equipment will be replaced by softswitch architectures. New market entrants will have an advantage by deploying softswitch solutions from the start. Incumbents will be disrupted by an architecture that is cheaper, smaller, and more convenient to use.

Legacy, Converging, and Converged Architecture

VoIP and softswitch technologies rose out of the economic necessity for distance service providers to switch to the least expensive means of transport, which is IP. By bypassing TDM and ATM networks, long-distance service providers greatly reduced their costs of long-distance transport which made them more competitive and more profitable than their TDM or ATM equipped competitors. Softswitch and VoIP got their beginnings in the transport aspect of the network. Long-distance service providers needed an intelligence that would perform call control over the IP network they used for their transport. In addition, softswitch needed to interface SS7 to the IP network, and then finally it had to control the transmission of features across the IP network. Thus was born the Class 4 replacement softswitch. Service providers speak of a telecommunications market where voice, data, and perhaps video and other broadband services are provided over a single network, presumably based on IP. The subscriber consequently enjoys highly efficient IP services desktop to desktop. This is called a *converged* network. The vast majority of the Class 4 and 5 switch market was designed and installed when voice and data were handled via separate channels. These are referred to as legacy networks. The transition network is called a *converging* network. To define the markets for Class 4 and 5 versus softswitch, it is important to understand that legacy markets apply to legacy networks where voice and data are separate networks. A converging market applies to converging networks, where, in most instances, the legacy infrastructure of Class 4 and 5 switches remains at the periphery of the network while the core of the network is IP, which provides efficient voice transport. A converged market applies to a converged network where voice and data are handled on one network. In the converged market, voice switching is performed by "classless" switches. This is because the limitations of geography defined a Class 5 switch as providing local service and a Class 4 switch as providing long distance. If geography is irrelevant, then a Class designation is irrelevant.

4.2.1 Softswitch and Distributed Architecture: A "Stupid" Network

Figure 4.2 illustrates the distributed architecture that is generally agreed upon as the model for softswitch networks. This model decouples the underlying packet-switching hardware from the call control, service logic, and service creation. This distribution enables flexibility in hardware choices as well as the innovation of new services without requiring changes in the switching fabric or structure. This model also opens up the opportunity for third-party developers. The bottom layer is considered to be the bearer or transport plane, which physically transports both voice and data traffic. This plane consists of the media gateways in the softswitch solution. What makes this possible is the client-server architecture of as opposed to the mainframe architecture of the Class 4 and 5 switches as shown in figure 4.3. One advantage is that it enables a service provider to start small and grow with the demand, as opposed to a large upfront investment in a class 4 switch. Internet is the inverse of the PSTN in that the intelligence of the Internet resides at the periphery of the network, instead of residing at the core of the network as it does in the PSTN. Thus, Softswitch architecture reflects a "stupid" network. Softswitch is a sum of its parts distributed across an *Internet Protocol* (IP) network, opposed to the PSTN where a few large, highly centralized Class 4 and 5 switches operate. Softswitch can be considered as a stupid solution as it utilizes distributed architecture (intelligence at the periphery), which is different than the smart or centralized architecture of the class 4 and 5 switches.

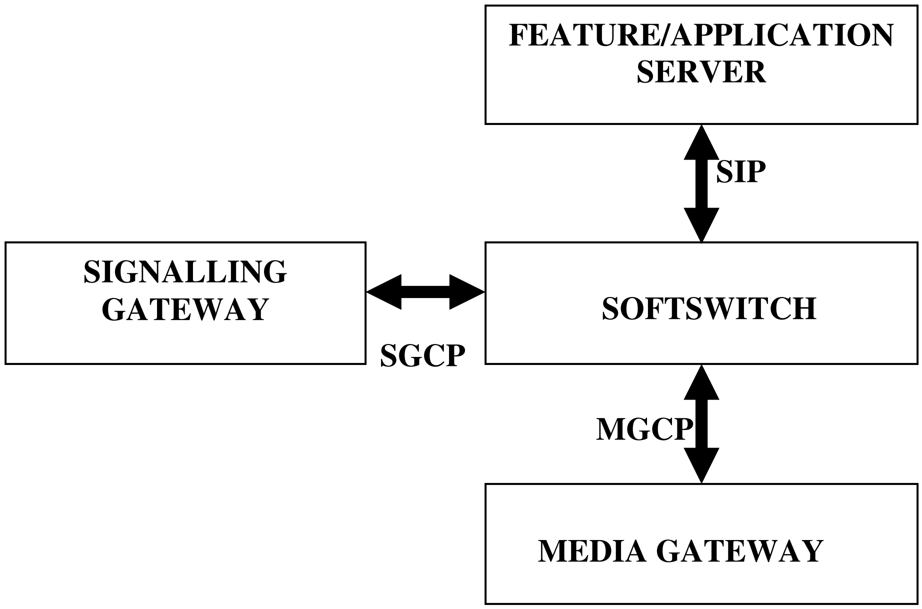


Figure 4.2 Softswitch Architecture Components

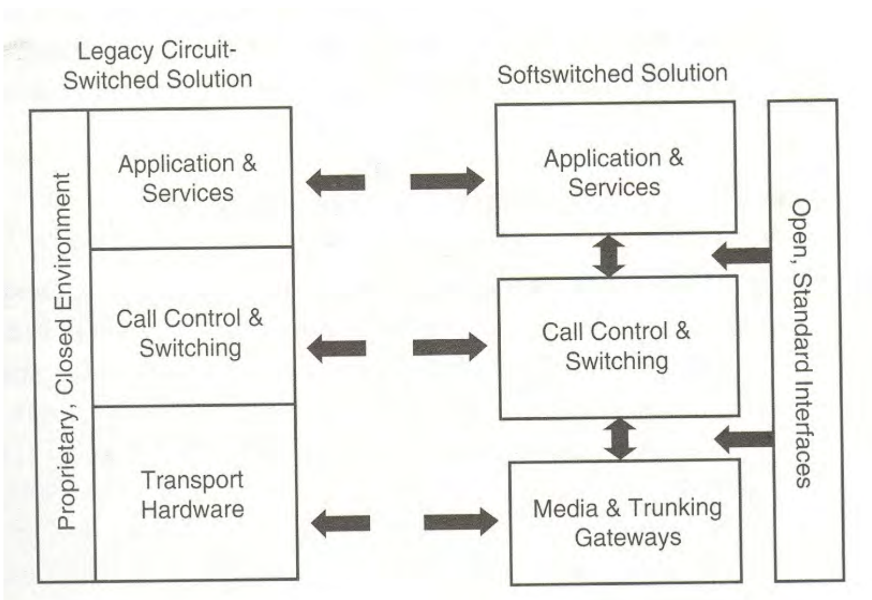


Figure 4.3 Mainframe of Class 4/5 versus Softswitch Architecture

Softswitch architecture, like the PSTN, can be described as having three elements: (1) access, that is, how a subscriber gains access to the network (2) switching, how a call is controlled across the network; and (3) transport, or how a call is transported across the network. In the case of access a VoIP network, access can be gained either from an IP source (PC or IP phone) or from a legacy, analog handset via a media gateway.

4.2.2 Access

The access network can be seen as PC to PC and PC to phone, IP phones (Phone to Phone VOIP) and Media Gateways.

PC to PC and PC to Phone

The first VoIP applications used PCs equipped with speakers and microphones as terminals for access to a VoIP network. Initially, the Quality of Service (QoS) left much to be desired and, as a result, this form of access did not immediately catch on in the market. This service is often referred as PC to PC. It is also possible to complete phone calls PC to Phone. PC-to- PC and PC-to-phone applications are now used most widely by consumers for long-distance bypasses as shown in figure 4.4. The market drivers for this form of access has been saving money on long distance, specifically on international long distance. Although often touted as an enterprise telephony solution, the use of PC as a telephony terminal has not seized any significant market share. Even where the QoS was acceptable for the task, anthropological issues remained. PCs do not resemble telephones in appearance, feel, or function. This presents a psychological barrier to the user for using a PC as readily as a telephone handset.

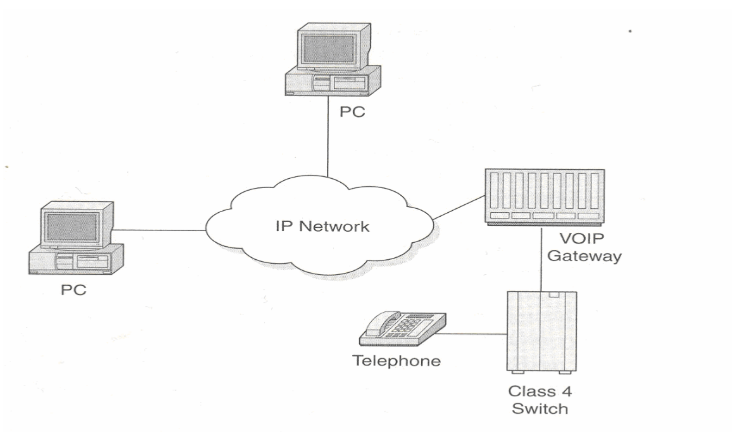


Figure 4.4 PC to PC and PC to PHONE

IP PHONES (IP HANDSETS) Phone-to-Phone VOIP

The IP handset incorporates all the computer hardware necessary to make an IP phone call possible. Another strong advantage of the IP handset is that it removes anthropological objections to VoIP calls. The IP handset looks and functions like a telephone as opposed to a PC. IP handsets are stand-alone devices and present an IP desktop-to-desktop solution. IP handsets offer further benefits in that they do not require a gateway and its incumbent investment and management responsibilities. The chief advantage of an IP phone to an enterprise is that the phone requires a minimum of network configuration and management. Each employee equipped with an IP phone can take his or her phone anywhere on a network with no reconfiguration of the phone or the network. IP phone-equipped employees are potentially more productive because the *graphic user interface* (GUI) on the IP phone makes using features much easier than with a 12-button conventional telephone handset and its list of star codes.

The IP handset has its own IP address, which is recognized wherever it is connected on an IP network as in Figure 4.5. IP phones are available in two flavors (or VoIP protocols) H.323 and the Session Initiation Protocol (SIP), and they require no gateway. Physically, the IP phone connects to the network via an Ethernet connection (RJ-45). In a business environment, an Ethernet hub serves to concentrate VoIP phone lines; although in a legacy network there would be an expensive PBX. The advantage to a VOIP service provider is that it need not maintain a Class 4 or 5 switch or VoIP gateway.

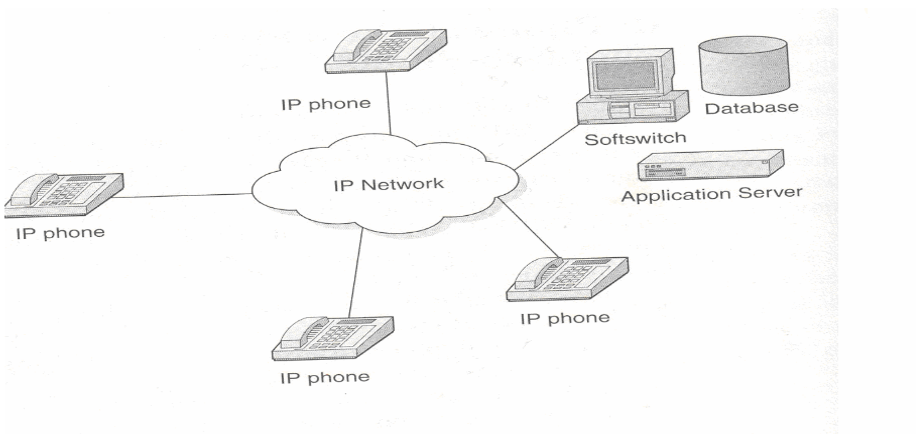


Figure 4.5 IP Phones on an IP Network

Most IP phones have liquid crystal display (LCD) screens with GUIs that enable expanded functions over a 12 button analog handset. With a conventional handset, the user must memorize long reams of number codes to perform functions such as conferencing, voice mail retrieval, call forwarding, and so on. For many users, this presents a psychological barrier that limits them to using only a handful of the features available on a PBX, thus preventing them from being as efficient in their communications as they could be. An IP phone with a GUI overcomes a number of these shortcomings by presenting the user with graphic choices to access their features. A disadvantage to the IP phone is that currently the IP phones on the market are very expensive relative to a conventional handset. That high cost makes this technology unattractive to the residential market. Price competition will drive the price of IP phones to lower values. In short, the voice network of the not-so-distant future will consist of IP phones that connect to IP networks where the intelligence for that service is provided by a softswitch located anywhere on the network.

Media Gateways (a VOIP Gateway Switch)

The most successful commercial form of access has been the use of a VoIP gateway. The gateway provides a connection between an end point on a data network and the PSTN or circuit- switched network. The gateway translates between transmission formats and the communication procedures that are used on each side. Gateways can be provided as stand-alone devices or be integrated into other systems. In this form of access, an existing telephone handset interfaces a gateway either via a direct connection, a PBX, or a Class 5 switch. The gateway packetizes the voice and routes it over the IP network. The Table below details the range of gateways and the scale of access they provide. The media gateway resides on the edge of a network and interfaces between Time Division Multiplexing (TDM) and IP networks. It is here that analog or digital signals from a handset (PBX or Class4 switch) are digitized (if analog), packetized, and compressed for transmission over an IP network. In the inverse, incoming calls are translated decompressed from an IP network for reception on digital or analog telephone devices. The media gateway interfaces directly with a TDM switch. The design of a gateway includes three key elements: an interface for the TDM side of the network (described in terms of DS0s or Tls), an interface for the packet side of the network (usually an Ethernet connection) and the necessary signal processing between these two sides.

Gateway	Scale
Residential/SOHO	2 to 8 ports (DS0s)linking a telephone handset via an RJ-11 connector
Enterprise	From two ports to multiple T1s/E1sconnecting to a PBX or directly to a handset
Carrier Grade	High Density, multiple T1/E1 line side, an OC3, and a trunk side with 3000 plus DS0s in Seven foot rack

Table 1 Gateways and their markets based on scalability

Signal processing is done with digital signal processors (DSP) on circuit boards designed to support voice. Signal processing functions include echo cancellation, coding/decoding of the analog signal with codec algorithms such as G.711 or G.723.1, adapting the digitally encoded information into a series of IP datagrams, and transmitting those datagrams via a network to its ultimate destination. On softswitch architecture, a media gateway can also be a part of the switching function depending on the amount of intelligence contained in the gateway. The trend is toward less intelligence in the media gateway and more intelligence in the softswitch. In the early days of the VoIP industry, the gateway had to contain a good deal of intelligence to make calls possible. However, the evolution of gatekeeper technology into carrier-grade softswitch has drawn the intelligence out of the gateway and on to the softswitch. Perhaps the most important issue for media gateways is their scalability. The density (the number of DS0s or ports in one chassis) determines its classification. Depending on its density, a media gateway falls in to one of the three following classifications: Residential/ SOHO, Enterprise, or Carrier-Grade.

Residential or SOHO Gateways*:* Figure 4.6 shows a residential grade of a VOIP gateway. Residential gateways are configured in number of DS0s which could range from 2 to 8 ports linking a telephone handset via an RJ-11.

Enterprise Gateways: Enterprise gateways aggregate legacy telephone infrastructures for interface with VoIP networks. This is usually done by connecting a gateway to the trunk side of a legacy PBX. Users retain their existing handsets. This has the effect of

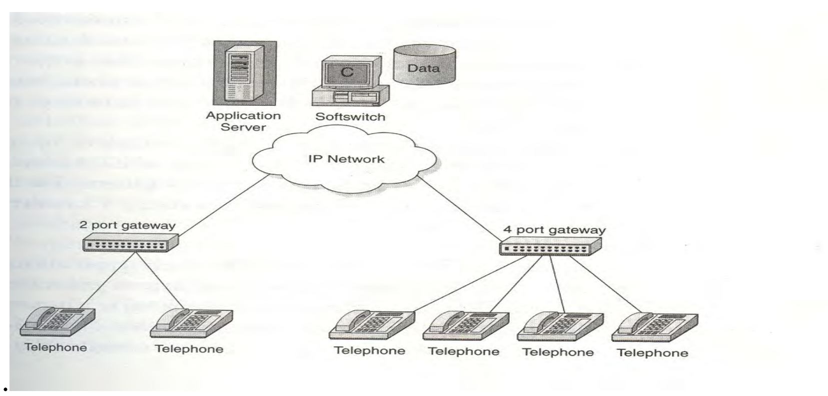


Figure 4.6 Residential Gateways in a VOIP Network

making VoIP indistinguishable to the end user. The business need not train its staff to use new hardware or software to work their existing telephone handsets. This option also offers investment protection in that the business retains its expensive PBX and PBX-associated telephone handsets. The only thing that has changed is that the company reduces or eliminates interoffice phone bills. Enterprise gateways are usually configured in multiples of Tl or E1 cards in a single chassis that interface with the trunk side of the PBX. Tl trunks connect to the line side of the gateway. The trunk side of the gateway is its Ethernet connection to a router if a router is not built in to the gateway.

Carrier-Grade Gateways: The early applications for VoIP gateways were for international long-distance bypass and enterprise interoffice long distance. Success in these applications led to a demand for expanded density gateways for carrier operations. These gateways needed to be densely populated (have enough DS0s or ports) enough to interface Class 4 or Class 5 switches (up to 100,000 DSOs in one node with an OC-3 trunk-side interface).

4.2.3 Switching

"Soft"switching is a break-through technology that has empowered VoIP to eclipse TDM as a telephony technology. Prior to the development of softswitch, VoIP was handicapped by a lack of intelligence necessary to route calls across the network. Without this intelligence, an evolution to IP alternative to the PSTN would not be possible.

Softswitch (Gatekeeper and Media Gateway Controller)

A softswitch is the intelligence in a network that coordinates call control, signaling, and features that make a call across a network or multiple networks possible. Primarily, a softswitch performs call control. Call control performs call setups and teardowns. Once a call is set up, connection control ensures that the call stays up until it is released by the originating or terminating user. Call control and service logic refer to the functions that process a call and offer telephone features. A softswitch coordinates the routing of signaling messages between networks. Signaling coordinates actions associated with a connection to the entity at the other end of the connection. To set up a call, a common protocol must be used that defines the information in the messages and which is intelligible at each end of the network and across dissimilar networks. The main types of signaling a softswitch performs are peer to peer for call control and softswitch to gateway for media control. For signaling, the dominant protocols are SIP, Signaling System 7

(SS7), and H.321. For media control, the predominant signaling protocol is the Media Gateway Control Protocol (MGCP). The precursors to softswitch are media gateway controllers (MGCs) and Gatekeepers. MGCs and Gatekeepers (essentially synonymous terms for the earliest forms of softswitch) are designed to manage low-density (relative to a carrier-grade solution) networks. MGC communicates with both the signaling gateway and the media gateway to provide the necessary call-processing functions. The MGC uses either MGCP or MEGACO/H.248 for intergateway communications.

Gatekeeper technology evolved out of H.323 technology (a VoIP signaling protocol). H.323 was designed for local area networks where an H.323 gatekeeper would manage activities in a zone. A zone is a collection of one or more gateways managed by a single gatekeeper. A gatekeeper should be thought of as a logical function and not a physical entity. The functions of a gatekeeper are address translation (a name or email address for a terminal or gateway and a transport address) and admissions control (authorizing access to the network). As VoIP networks got larger and more complex, management solutions with far greater intelligence became necessary. Greater call-processing power became necessary, as did the capability to interface signaling between IP networks with the PSTN (VoIP signaling protocols to SS7). Other drivers included the need to integrate features on the network and interface disparate VoIP protocols, thus was born the softswitch.

A significant market driver for softswitch is protocol intermediation, which is necessary to interface H.323 and SIP networks, for example. Another market driver for softswitch is to interface between the PSTN (SS7) and IP networks (SIP and H.323). The other function of softswitch is the intermediation between media gateways of dissimilar suppliers. Despite an emphasis on standards such as H.323, interoperability remains elusive. A softswitch application can overcome intermediation issues between media gateways.

Softswitch provides usage statistics to coordinate the billing, track operations, and administrative functions of the platform while interfacing with an application server to deliver value-added subscriber services. The softswitch controls the number and type of features provided. It interfaces with the application server to coordinate features (conferencing, call forwarding, and so on) for a call.Physically, a softswitch is software hosted on a server chassis fitted with IP boards and includes the call control applications and drivers. Very simply, the more powerful the server, the more capable the softswitch. That server need not be collocated with other components of the softswitch architecture.

Signaling Gateway

Signaling gateways are used to terminate signaling links from PSTN networks or other signaling points. The SS7 signaling gateway serves as a protocol mediator (translator) between the PSTN and IP networks. That is when a call originates in an IP network using H.323 as a VoIP protocol and must terminate in the PSTN, a translation from the H.323 signaling protocol to SS7 is necessary in order to complete the call. Physically, a signaling function can be embedded directly into the MGC or housed within a stand-alone gateway.

Application Server

The application server accommodates the service and feature applications made available to a service provider's customers. Examples include call forwarding, conferencing, voice mail, forward on busy, and so on. Physically, an application server is a server loaded with a software suite that offers the application programs. The softswitch accesses, enables, and applies these application programs to the appropriate subscribers as needed. A softswitch solution emphasizes open standards as opposed to the Class 4 or 5 switches that have historically offered proprietary and closed environments. This usually translates into less competitive pricing for those components. Softswitch open standards are aimed at freeing service providers from vendor dependence and the long and expensive service development cycles of legacy switch manufacturers. Features reside at the application layer in softswitch architecture. The interface between the call control layer and specific applications is the application program interface (API). Writing and interfacing an application with the rest of the softswitch architecture occurs in the service creation environment.

4.2.4 Transport

Transport is the means by which voice is transported across the network. It connects the switches in the network. Simply put, three modes of voice transport are in use today: IP, *Asynchronous Transfer Mode* (ATM), and TDM. This has been previously discussed in chapter 3 on overview of packet based networks.

4.3 Next Generation Network Services

4.3.1 Characteristics of NGN Services

Although it is difficult to predict what the next killer applications will be, we can infer the types of service characteristics and capabilities that will be important in the NGN environment by examining current service-related industry trends. It is certainly true that we are moving from Time Division Multiplex (TDM)-based, circuit switched networks to packet-, cell-, and frame based networks. However, these changes in the transport networks are merely enablers for the dramatic changes we will see at the service level. The major thrust of traditional network service providers has been to offer the mass market basic transport of information between end users, with various value-added capabilities. These services tended to involve narrowband voice calls, with a single point-to-point connection per call. However, this view of services is rapidly changing as the world's economies are becoming increasingly reliant on information as a basic resource. While existing services will remain part of service providers' offerings, customers' expectations will migrate towards more advanced broadband multimedia and information intensive services. End users will interact with the network via sophisticated CPE, and be able to select from a wide range of Quality-of-Service (QoS) and bandwidth.

The current evolution of telecommunication services points to a world where service providers will have the flexibility to focus on micro-marketing (as opposed to mass-marketing). Decisions about their service offerings may have as much to do with packaging (e.g., pricing, bundling, marketing, and convenience), as they will with the actual services offered. As multiple carriers, service providers, equipment suppliers, and other business entities all become involved in providing services to end users, merged network and business systems will become increasingly important. The primary goal will be to enable users to get the information content they want, in any media/format, over any facilities, anytime, anywhere, and in any volume. Based on the above mentioned trends, the following is a summary of several service characteristics likely to be important in an NGN environment:[16]

• **Ubiquitous, real-time, multi-media communications** - The only hope for dramatically increased reliability, similar to communicating in person, is high-speed access and transport for any medium, anytime, anywhere, and in any volume.
• **More "personal intelligence" distributed throughout the network** - This includes applications that can access users' personal profiles, learn from their behavior patterns, and perform specific functions on behalf of them
• **More "network intelligence" distributed throughout the network** - This includes applications that know about, allow access to, and control network services, content, and resources. It can also perform specific functions on behalf of a service or network provider
• **More simplicity for users** - This shields users from the complexity of information gathering, processing, customization, and transportation. It allows them to more easily access and use network services/content, including user interfaces that allows for natural interactions between users and the network. It involves providing context-sensitive options/help/information, transparently managing interactions among multiple services, providing different menus for novices vs. experienced users, and providing a unified environment for all forms of communication.
• **Personal service customization and management** - This involves the users' ability to manage their personal profiles, self-provision network services, monitor usage and billing information, customize their user interfaces and the presentation and behavior of their applications, and create and provision new applications.
• **Intelligent information management** - This helps users manage information overload by giving them the ability to search for, sort, and filter content, manage messages or data of any medium, and manage personal information (e.g., calendar, contact list, etc.).

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4.3.2 Description of some NGN Services

For the purpose of this work the following services have been identified for more description.

1. Multimedia Conferencing

Videoconferencing has been possible for some time. Telephone companies offer videophone and videoconferencing services, but most of these require the purchase of special equipment and the setting up of special rooms for videoconferencing. They also require a Multipoint Control Unit (MCU) which causes multi-way conferences to be set up as a series of point-to-point connections between the MCU and each participant. This is an expensive and often cumbersome way of holding remote meetings, and does not scale to supporting large user groups who want to have interactive meetings. The Internet provides an alternative that can be scaled more economically. The Internet is mainly used for supporting asynchronous communication such as e-mail and the WWW (World Wide Web), but parts of it can also be used for transmitting voice and video in real time, making multimedia conferencing on the Internet possible. Conferencing on the Internet is different because it uses multicast transmission instead of the point-to-point communication used by the telephone companies. It is this multicast technology that gives the Internet community an economic way of supporting multi-way multimedia conferences between large numbers of participants. A computer-supported conference involves the transmission of audio, video and data – hence “multimedia”.

On the Internet, networks and hosts are located by means of addresses. IP (Internet Protocol) addresses consist of four numbers which, between them, identify the network and host. Transmission between sites is enabled by routers which communicate with one another and hold tables of routes between addresses. There may be several possible routes between two hosts. [3]

2. IP Centrex

Centrex is a set of specialized business solutions where the equipment providing the call control and service logic functions is owned and operated by the service provider and hence is located on the service provider's premises. Since Centrex frees the customer from the costs and responsibilities of major equipment ownership, Centrex can be thought of as an outsourcing solution. Call control and service logic refer collectively to the functions needed to process a telephone call and offer telephone features

In traditional Centrex service (i.e., analog Centrex and ISDN Centrex), call control and service logic reside in a Class 5 switch located in the Central Office. The Class 5 switch has also responsibility for transporting and switching the electrical signals that carry the callers' speech or other information (e.g., faxes).

In IP telephony, voice conversations can be digitized and packetized for transmission across the network. IP Centrex refers to a number of IP telephony solutions where Centrex service is offered to a customer who transmits its voice calls *to the network* as packetized streams across a broadband access facility. IP Centrex builds on the traditional benefits of Centrex by combining them with the benefits of IP telephony. One of these IP telephony benefits is increased utilization of access capacity. In IP Centrex, a single broadband access facility is used to carry the packetized voice streams for many simultaneous calls. When calls are not active, more bandwidth is available for high speed data sessions over the LAN, like Internet access. This is a much more efficient use of capacity than traditional Centrex. In analog Centrex, one pair of copper wires is need to serve each analog telephone station, regardless of whether the phone has an active call; when the phone is not engaged in a call, the bandwidth capacity of those wires is unused. IP Centrex solutions are being developed on a number of platforms, including Class 5 switches and softswitches. [21]

3. Video On Demand

Modern day Television offers programmes from a number of available channels and is very simple to use. The Cable TV (CATV) makes it possible to choose programmes from large number of channels. They became video rental business in combination with a video recorder, which provides customers to select movies at their will. This service may be called video on demand.

Nowadays Video-on-Demand (VoD) includes much wider services and opportunities. Today s technology allows telecommunication network operators to offer such services as home shopping, games, and movies on demand. These services should have a competitive price compared to the video rental, and customers do not need to travel for the services. These possibilities have been reached by the development of the telecommunication and electronic industry. The capacity of a hard disk has doubled almost every year at near-constant cost. The useful compression ratio for video has been increased considerably; MPEG-formatted video can be transported at a bit rate of few Mbit/s. The digital signal processing techniques permit the transport of a few Mbit/s over existing copper wires for a distance of a few kilometers. Finally, Asynchronous Transfer

Mode (ATM) systems allow the switching of any reasonable bit rate to a single or multiple customers among a large number of connected customers. However, today's transmission bandwidth is large only downstream towards the customer with narrow upstream bandwidth. But upstream bandwidth will also become wider in the future, then interactivity between the customer and the service provider will increase.

A Video-on-Demand system has many elements that are necessary for the use of the complete service. This includes video servers, community network, switching office, set-top unit, and backbone network. VoD system providers will offer services which select the right technology, features, performance, price, reliability, and ease of use. Equipment are developed so that they will allow to operate in different environments and in a variety of services.

The main VoD scenario consists of a local database and server connected to the user via a communications network. The data is stored on local distribution sites which are connected through high speed backbone network to information archives and video servers. This distribution scheme serves many purposes. First, it is possible to implement it in a distributed fashion, increasing availability and reliability. Second, a provider can tailor the information delivery to the specific tastes of a user community in a particular geographic area, reducing costs. Third, it is easier to manage, as each local system is responsible for its own billing and accounting. Fourth, the system can be constructed in a regional, piecewise fashion. [3]

4. Unified Messaging

Unified Messaging is the integration of several different communications media, such that users will be able to retrieve and send voice, fax, and e-mail messages from a single interface, whether it be wireline phone, wireless phone, pc or internet-enabled PC.

The essence of communication is breaking down barriers. The telephone breaks distance and time barriers so that people can communicate in real time or near real time when they are not in the same place at once. There are now other barriers to be overcome. For example, people use different terminals to communicate, and there are new forms communications, such as e-mail, voice mail, fax machines, and pagers. The unified messaging concept involves breaking down the terminal and media barriers so that people using different technologies, different media, and different terminals can still communicate to anyone, anywhere, at anytime.

Frequently people have a message that they want to communicate, but the intended recipient of the message cannot be reached. Technology is helping people overcome this problem as well; products are available that are powerful as well as flexible to meet these needs. With the current developments in communication, standards are important. Also products are needed that offer interoperability. These products may not be from the same vendor, but they must operate together to form powerful solutions for customers.

Unified messaging is a personal agent for the individual user. It can help send and receive messages, whether they are voice, email, or fax. It also will notify the user whenever mail arrives. The concept of notification is becoming a large part of messaging. Some people want to be reached at all costs, anywhere, anytime. Whether they at home or on vacation, they want to be notified of messages. Others are more protective about their privacy. They do not want to be reached when, for example, they are sleeping or having dinner. Unified messaging technology provides the power to reach people almost anywhere at anytime and the flexibility to allow people to control when they can be reached. This is based on a concept of “your time” communications, where subscribers can interface with messages how and when they want. With unified messaging, subscribers reduce the number of places they must check for incoming voice, fax, and email messages. From a single interface, they can check for all message types. [15]

CHAPTER V

NEXT GENERATION NETWORK ASSESSMENT

This chapter focuses on the assessment of the next generation network dividing the topic in to three major parts. The first part addresses an assessment made on the main elements of both the legacy PSTN and NGN, which are Class 4/5 switch and Softswitch respectively based on some performance metrics.

When considering aspects of NGN network deployment it is useful to review deployments of new technologies by other operators in other markets. From the network operator perspective this assists in gaining technical knowledge from other operator experiences regarding the planning, deployment and operation of new voice switching solutions as well as giving an awareness of other service providers strategies and product offerings. The case studies have been taken for Incumbents. The other main players in the evolution to NGN are vendors. It was also tried to assess the position of softswitch product vendors.

There are general considerations that wireline NGN operators should consider before deploying next generator networks. The major ones are quality of service of the IP Network, Choice of protocol used for voice services and network security issues. And the major part of the wireline subscribers in the Ethiopian Telecom Market got service from Ericsson AXE 10 exchange. Ericsson's NGN migration strategy, with its plan to ETC will be seen at last.

5.1 Softswitch as an alternative to Class 4 and Class 5

New technologies have now arrived on the market that provide a low-cost alternative to Class 4 and 5 switches in both purchase price and cost of maintenance. These technologies are Voice over Internet Protocol (VoIP) and Softswitch. Softswitch provides the call control or intelligence for managing a call over an Internet Protocol (IP) or other network. Class 4 and 5 switches have been well known in the technology for such qualities as reliability, scalability, quality of service (QoS), features, and signaling. Many have argued that VoIP and softswitch technologies must match Class 4 and 5 switches in all of these qualities before their deployment in a market environment is feasible. This section addresses the assessment of Softswitch technologies in relation to Class 4 / Class

5 switches using the above metrics. It is seen now that the time may have come. Not only do VoIP and softswitch compare favorably in function and quality with Class 4 and 5 switching, but also they deliver services not possible with Class 4 and 5 switches. Novel services could potentially generate additional revenues for service providers, making them at least as profitable as the incumbents and introducing true competition to the local loop.

5.1.1 Reliability

Reliability is plainly the chief point of comparison for service providers evaluating competitive technologies. Class 4 and 5 switches have a reputation for the "five 9s" of reliability. That means they will be in service 99.999 percent of the time. Building a voice-switching solution to achieve five nines is a matter of meticulously engineering into the solution the elements of redundancy, no single point of failure, and Network Equipment Building Standards (NEBS**)**. A recurring objection to Voice over IP (VoIP) and softswitch solutions is the perception that such a solution would not match the "five 9s" of reliability provided by the Class 4 and 5 switches.

World Trade Center Attack: A need to redefine reliability

The September 11th, 2001 attack on the World Trade Center has served to focus attention on the vulnerabilities of the legacy, circuit-switched telephone network. Verizon, the largest telephone company, has suffered phone and communications service interruption for 20,000 residential and 14,000 businesses customers. The question a subscriber inevitably has to ask is "Where are the five 9s of reliability in this system?" Public Switched Telephone Network (PSTN) has a centralized architecture. The entire network across the nation rely on one hub or central office, meaning that if that hub were to be destroyed, that city would lose all land-line telephone connectivity with the outside world. Without a drastic change in the architecture of the PSTN, it will remain vulnerable to major outages. The chief reasons for this are not that the Network elements in the form of Class 4 or Class 5 switches are less than reliable, but rather the architecture centers on single points of failure (SPOF): the central offices and the star network architecture that comprises the PSTN.

How Does a Switch, PSTN or Softswitch, Achieve Five 9s?

Although a number of softswitch vendors claim to achieve five 9s, much uncertainty exists among service providers that the five 9s of the new technology have the same experience as a Class 4. Carriers require high system availability and are concerned with the effects of possible softswitch downtime. Carriers demand low MTBF and employ traffic overload control, the shedding of call-processing capacity, the event of component failures, and quick failure detection and recovery mechanisms. The softswitch answer is to design redundant softswitch hardware nodes at different locations throughout the network, which contributes to the overall network reliability. NEBS addresses the physical reliability of a switch. NEBS testing include electrical safety, immunity from electromagnetic emissions, lightning and power faulting, and bonding and grounding evaluations.

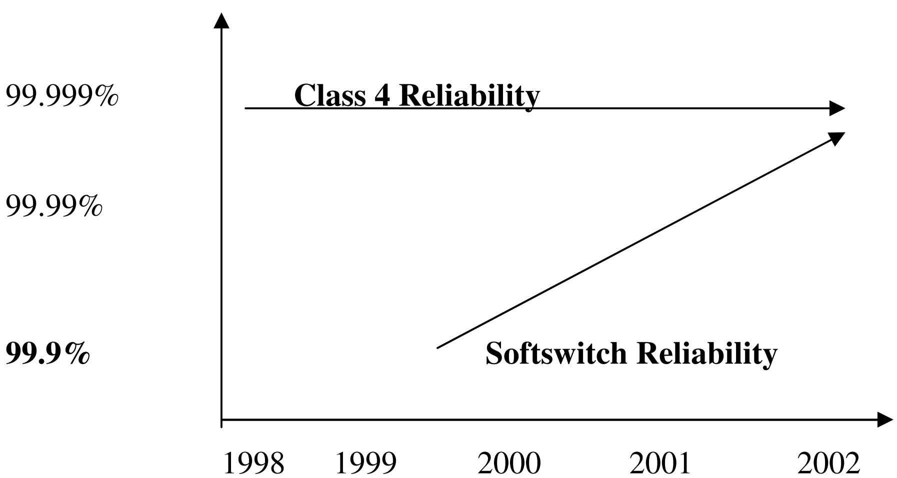

Figure 5.1 Softswitch Product Performances in Reliability

By using many of the mechanisms that Class 4 and Class 5 switches have utilized over the years (redundancy, fault tolerance, and NEBS) to achieve five 9s of reliability , softswitch is achieving the same levels of reliability. Distributed architecture can also improve the reliability of a softswitch solution. With a distributed architecture, no SPOF exists on a network. Any redundant component on the IP network can pick up where the primary component failed. Starting from year 2002, many softswitch vendors have claimed 99.999% reliability. The figure 5.2 illustrates a dispersal of softswitch components around one sample country. If a MGC in Region 1 is destroyed in a force majeure, another MGC can pick up where the Region 1 MGC failed.

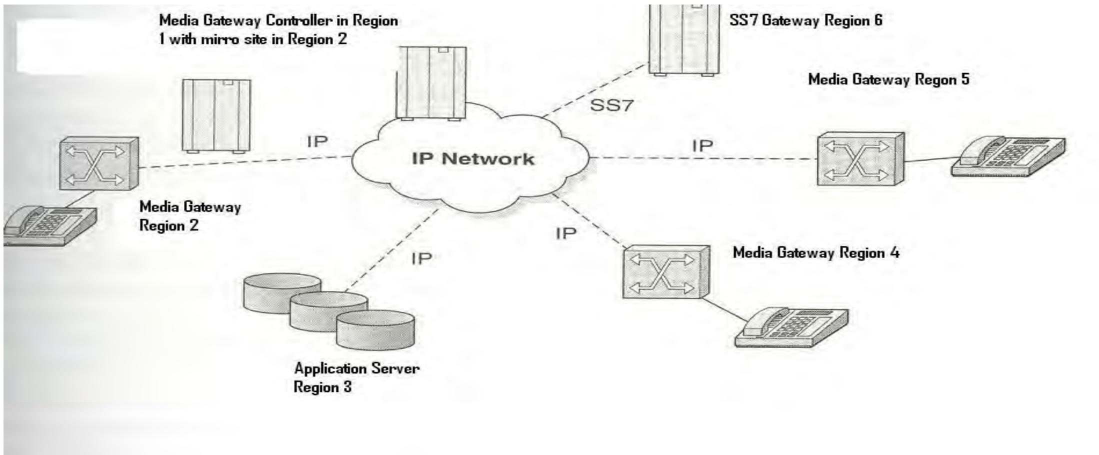

Figure 5.2 Distributed Architecture Provides Greater Reliability

5.1.2 Scalability

One of the most promising aspects of softswitch architecture is its flexibility in scaling that makes so many revolutionary applications possible. Scalability for a softswitch is contingent on two elements: the total number of ports and the call-processing capabilities. The softswitch industry and the media gateway industry in particular are only recently introducing high-density gateways that compete with Class 4 and 5 switches in terms of port density.

Scaling Up

Scaling up a media gateway is a factor of hardware and software. A high density design consists of call-processing modules, PSTN interface modules, and packet interface modules. The packet interface modules provide the interface to the packet-switched network. Packet interfaces include DS-3, OC-3 optical interfaces, multiple 100 BaseT, and Gigabit Ethernet interfaces. DS-3 /OC-3 capacities are relatively recent improvements in the upward scalability of media gateways. The table below and figure 5.3 illustrate the progression of media gateways (and softswitch solutions) in port density.

Media Gateway	**Density (DS0s/7 ft rack)**
Cisco MGX 8260	16,000
Convergent Networks ISC2000	24,192
Sonus GSX9000	24,000

Table 3 Media Gateways offer greater density in one 7 foot rack than class 4

In 2005, most Softswitch vendors have claimed that they have reached a maximum number of lines of 2-5 million in terms of port density [5] & [20].

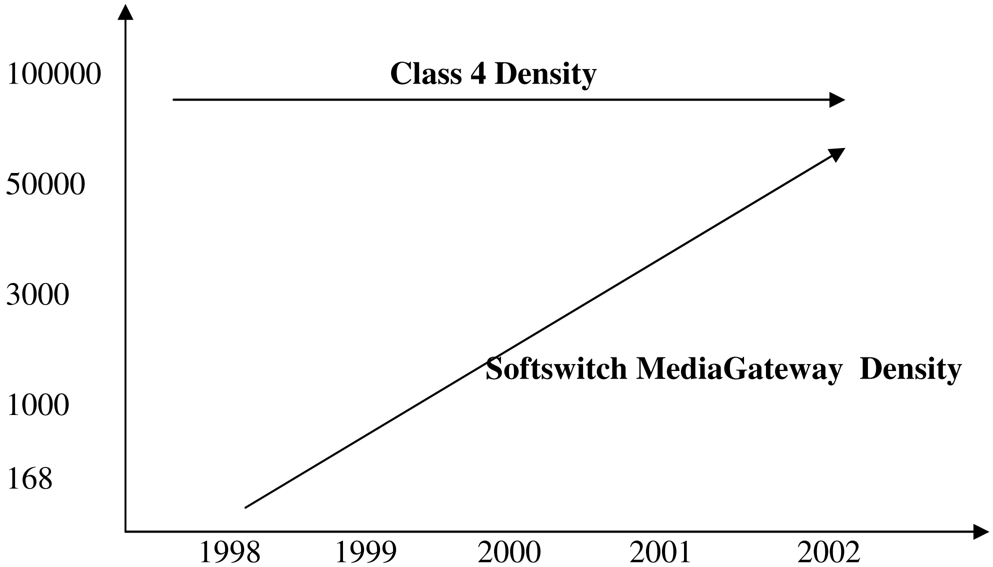

Figure 5.3 Softswitch Product Performance in DS0 Density

The other issue of scalability is the call-processing power of softswich. Early versions of softswitch could not compete with Class 4, for example interms of calls per second or BHCAs. In 2000, softswiches were reported to offer BHCA ranges from 250,000 to 500,000 [19]. ***And in 2005, there were reports that the call processing power ranges of most softswitches has gone up to 10 million BHCAs [19].*** Figure 5.4 illustrates this progression in BHCA counts. It should be noted that no independent verification of these BHCA counts exists either Class 4 or 5 switches or for softswitch solutions.

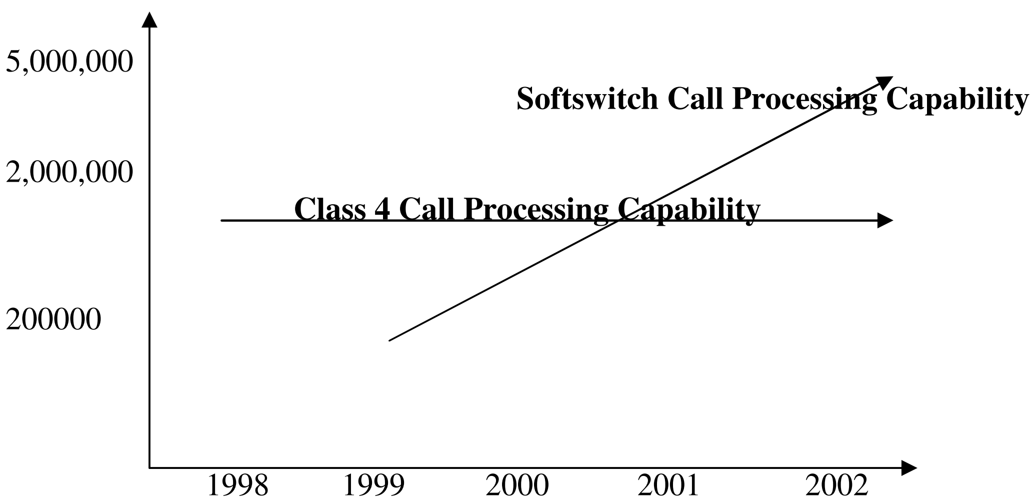

Figure 5.4 Softswitch Product performance in BHCAs

Scaling Down

Rather than focusing a network on a very dense, highly centralized switching architecture with its incumbent and excessive expense, softswitch architecture offers far greater flexibility over Class 4 or 5 switches. At the lowest end of the scalability range for softswitch architecture would be an IP phone, which constitutes the equivalent of one DS0 on a telephone switch. However the minimum configuration, for example, for a Nortel DMS-250 Class 4 switch is 480 DS0s. The service provider does not need to invest in a switch to service a customer with an IP phone. The service provider provides only the IP connection or, at a minimum, the proxy and application server to complete the call over the IP network.

The following table illustrates comparison made between a softswitch product and a class 4 switches on reliability and scalability terms. BHCAs refers to Busy Hour Call Attempts and MOS is the Mean Opinion Score.

Vendor and Product	DS0s/rack	BHCAs	Reliability	NEBS	MOS
Nortel DMS 250(Class 4)	2,688	800,000	99.999	Y	4.0
Lucent 4ESS (Class 4)	2,688	700,000	99.999	Y	4.0
Convergent ICS2000(SS)	108,864	1,500,000	99.9994	Y	4.0
SONUS GSX9000(SS)	24,000	2,000,000	99.999	Y	4.0
Neura NU_Tandem(SS)	6120	480,000	99.999	Y	4.0

Table 4 Comparing Softswitch Vendors to Class 4 switches

5.1.3 QOS

A major objection to Voice over IP (VoIP) and softswitched networks is the notion that their quality of service (QoS) is inferior to the Public Switched Telephone Network (PSTN). QoS covers a number of parameters, but a mostly concerned with the user's perception of the voice quality, the call setup, and keeping the call up for the intended duration of the call (avoiding dropped calls). It should be noted here that the context of this section focuses on private networks, enterprise and service provider-maintained and not the public Internet.

The most important network parameters for the effective transport of VoIP traffic over a softswitched network are bandwidth, delay, jitter, echo and packet loss. This presents a challenge for network designers who must first focus on these issues in order to deliver the best voice quality. It is necessary to examine the network for any element that might induce delay, jitter, packet loss, or echo. This includes first the hardware elements, such as router and media gateways, and second the routing protocols that prioritize voice packets over all other types of traffic on the IP network. Delay can be corrected via functions contained in the IP network routers, the VoIP gateway, and in engineering in the IP network. The shorter the end-to-end delay, the better the perceived quality and overall user experience.

Improving Quality of Service in IP Routers and VOIP Gateways

Packet delay is primarily determined by the buffering, queuing, and switching or routing delay of the IP routers. Packet capture delay is the time required to receive the entire packet before processing and forwarding it through the router. This delay is determined by the packet length, link-layer operating parameters, and transmission speed. Using short packets over high-speed trunks can easily shorten the delay. VoIP networks use packetization rates to balance connection bandwidth efficiency and packet delay. Routing delay is the time it takes a network element to forward a packet. New IP switches can significantly speed up the routing process by making routing decisions and forwarding the traffic in hardware devices instead of software. Due to the statistical multiplexing nature of IP networks and the asynchronous nature of packet arrivals, some delay in queuing is required at input and output ports of a packet switch. Over provisioning router and link capacities can reduce this delay in queuing time.

Voice-signal processing at the sending and receiving ends adds to the delay and includes the time required to encode or decode the voice signal from analog or digital form into the voice-coding scheme selected for the call and vice versa. Compressing the voice signal also increases the delay. The greater the compression, the greater the delay. When bandwidth costs are not a concern, a service provider can utilize G.711, which is uncompressed voice, and this impose a minimum of delay due to the lack of compression. On the transmit side, packetization delay is another factor that must be entered into the calculations. The packetization delay is the time it takes to fill a packet with data. The larger the packet size, the more time is required. Using shorter packet sizes can shorten this delay but will increase the overhead because more packets have to be sent, all containing similar information in the header. Balancing voice quality, packetization delay , and bandwidth utilization efficiency is very important to the service provider.

Gateways can be engineered to minimize impairments to QoS. Those impairments are echo, end-to-end delay, buffering delay, and silence suppression. Echo is a phenomenon where a transmitted voice signal gets reflected due to an unavoidable impedance mismatch and a four-wire/two-wire conversion between the telephone handset and the communication network. Echo can disrupt the normal flow of conversation and its severity depends on the round-trip time delay. If a round-trip time delay is more than 30 milliseconds, the echo becomes significant, making normal conversation difficult. A gateway should use an echo canceller so that when delay reaches above 30 milliseconds, the echo canceller circuits can control the echo. Voice communication is half-duplex, which means that one person is silent while the other speaks. A gateway can save bandwidth by halting transmission of cells at the gateway during these silent periods. This is known as silence suppression or voice activation detection (VAD).

Improving Quality of Service on the Network

QoS requires the cooperation of all logical layers in the IP network-from application to physical media-and of all network elements from end to end. Clearly, optimizing QoS performance for all traffic types on an IP network presents a scary challenge. To partially address this challenge, several Internet Engineering Task Force (IETF) groups have been working on standardized approaches for IP-based QoS technologies [2]. The IETF's approaches fall into the following categories:

- Prioritization using the Resource Reservation Setup Protocol (RSVP) and Differentiated Services (DiffServ)
- Label switching using multi protocol label switching (MPLS)
- Bandwidth management using the subnet bandwidth manager

Resource Reservation Setup Protocol (RSVP)

A key focus in this industry is to design IP networks that will prioritize voice packets. RSVP is the signaling protocol that is used to reserve bandwidth on a specific transmission path. RSVP is designed with routing protocols Open Shortest Path First

(OSPF) and the Border Gateway Protocol (BGP). RSVP currently offers two levels of service. The first level is guaranteed, which comes as close as possible to circuit emulation. The second level is controlled load, which is equivalent to the service that would be provided in a best-effort network under no-load conditions. What makes RSVP wonderfully simple in providing improved QoS for VoIP is that a network manager can invoke reservation, allocation, and policing of bandwidth in the RSVP protocol to improve QoS.

Differentiated Service (DiffServ)

A follow-up on the IETF initiative to IntServ was DiffServ. DiffServ sorts packets that require different network services into different classes. Packets are classified at the network ingress node according to service level agreements (SLAs). DiffServ is a set of technologies proposed by the IETF to allow Internet and other IP-based network service providers to offer differentiated levels of service to individual customers and their information streams. The preferential grade of service (GoS), which can only be attempted and not guaranteed, includes a lower level of packet latency as those packets advance to the head of a packet queue, should the network suffer congestion. DiffServ makes use of the IP version 4 Type of Service (ToS) field and the equivalent IP version 6 Traffic Class field. The portion of the ToS /Traffic Class field that DiffServ uses is known as the DS field. The field is used in specific ways to mark a given stream as requiring a particular type of forwarding. The type of forwarding to be applied is Per Hop Behavior (PHB), of which DiffServ defines two types: expedited forwarding (EF) and assured forwarding (AF). PHB is the treatment that a DiffServ router applies to a packet with a given Differentiated Service Code Point (DSCP) value. A router deals with multiple flows from many sources to many destinations. Many of the flows can have packets marked with a DSCP value that indicates a certain PHB. The set of flows from one node to the next that share the same DSCP code point is known as an aggregate. From a DiffServ perspective, a router operates on packets that belong to specific aggregates. When a router is configured to support a given PHB, then the configuration is established in accordance with aggregates rather than to specific flows from a specific source to a specific destination.

MPLS enabled IP Networks

MPLS has emerged as the preferred technology for providing QoS, traffic engineering, and VPN capabilities on the Internet. MPLS contains forwarding information for IP packets that is separate from the content of the IP header such that a single forwarding paradigm (label swapping) operates in conjunction with multiple routing paradigms. The basic operation of MPLS is to establish label-switched paths (LSPs) into which certain types of traffic are directed. MPLS provides the flexibility of being able to form Forwarding Equivalence Classes (FECs) and the capability to create a forwarding hierarchy via label stacking. All these techniques facilitate the operation of QoS, traffic engineering, and VPNs. MPLS is similar to DiffServe in that it marks traffic at the entrance to the network. The function of the marking is to determine the next router in the path from the source to the destination.

MPLS involves the attachment of a short label to a packet in front of the IP header. This procedure is effectively similar to inserting a new layer between the IP layer and the underlying link layer of the Open System Interconnection (OSI) model. The label contains all the information the router needs to forward a packet. The value of a label can be used to look up the next hop in the path for forwarding to the next router. The difference between this routing and standard IP routing is that the match is exact. This enables faster routing decisions in routers.

An MPLS-enabled network, on the other hand, is able to provide low latency and guaranteed traffic paths for voice. Using MPLS, voice can be allocated to an FEC that provides the differentiated service appropriate for this traffic type.

QoS in softswitched networks is corrected with mechanisms similar to those in TDM network. By engineering out deficiencies in the components (media gateways) and improving the network (DiffServ and MPLS}, QoS can be brought up to the standards of the PSTN. Although not as quantifiable as a Mean Opinion Score (MOS) score on a media gateway, significant progress has been made in recent years in engineering closed IP networks to deliver PSTN quality voice. Figure 5.5 illustrates the progress in improving QoS on those networks [2].

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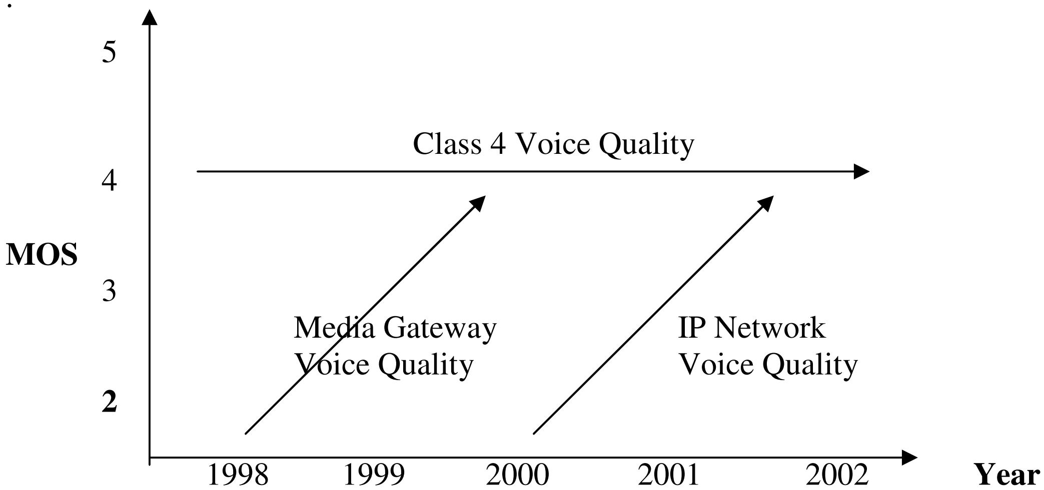


Figure 5.5 Softswitch Performance in QOS

5.1.4 Features

One objection many service providers have regarding Voice over IP (VoIP) and softswitch technologies is the perception that they do not duplicate the 3,500 features of a Class 5 switch. The first question many would have what are those features by name and what functions do they perform? A consensus in the industry is that a residential customer may use a maximum of four or five features. A business customer may use upwards of a dozen features. Subscribers access those features via their dual-tone multi frequency (DTMF) dial pads on their telephone handsets. Softswitches that replace a Class 5 switch utilize application and media servers that replicate the features found on a Class 5 switch. They potentially offer features not found in a Class 5 or not even possible with a Class 5 switch. These features are usually written in text-based languages using open standards. It is possible that given the flexibility in creating new features, some softswitch solutions that replace Class 5 switches may eventually offer more than 3500 features [22] & [19].

Custom Local Area Signaling Service (CLASS) features are basic services available in each local access and transport area (LATA). The features and the services they offer are a function of Class 5 switches and Signaling System 7 (SS7) networks. The Class 4 switch offers no features of its own. It transmits the features of the Class 5. With almost three decade of development, the Class 4 switch has a well-established history of seamless interoperability with the features offered by the Class 5 and SS7 networks. A softswitch solution emphasizes open standards as opposed to the legacy Class 4 or 5 switch that historically offered a proprietary and closed environment. Softswitch open standards are aimed at freeing service providers from vendor dependence and the long and expensive service development cycles of legacy switch manufacturers.

Features and Signaling

Features are a function of the Class 5 switch and the SS7 network. So how are the features of the Public Switched Telephone Network (PSTN) transferred to a converging market where the softswitch competes with the Class 4 or 5 switches? Firstly, SS7 information generated in the PSTN must be transported transparently across the IP network. It is possible with mechanisms that make it easier for service providers to quickly roll out new, high-margin services that the voice market will shift in favor of the service provider that can deliver those features quickly. It should be noted here that softswitch could enable a service provider to offer expanded features that are made available via the subscribers PC or IP handset. The PC offers greater flexibility than a telephone handset in the range of communications between the subscriber and the switch. The service creation environment (SCE) of softswitch is almost unlimited and has the potential to change much of the switch market in favor of softswitch. Softswitch has the capability to relay existing features of the Class 5 and SS7 networks. In addition, as figure 5.6 illustrates, it can offer a variety of new features, potentially leading to a total count of features in excess of the 3,500 features of the Class 5 switch.

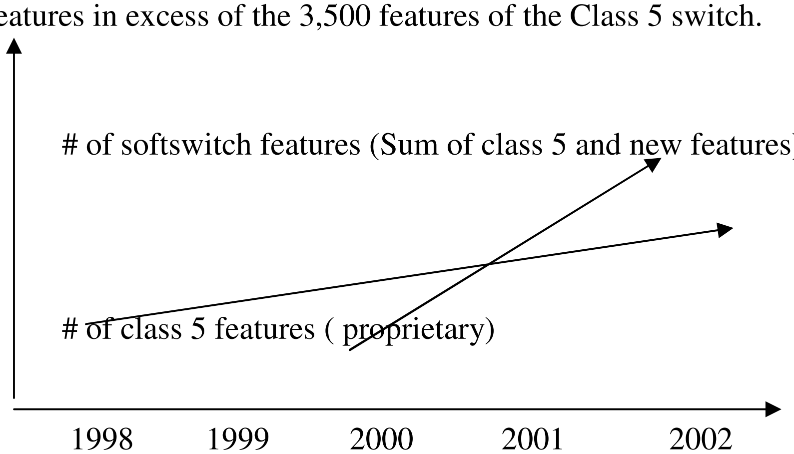


Figure 5.6 Softswitch Performance in number of features

By virtue of its open standards interface, softswitch enables service providers to quickly create and roll out new services. The inclusion of the option to write custom application programming interfaces (APIs) on a softswitch enables a service provider to write only those custom APIs that are applicable at a given service at a given location. APIs are the interface between the call control layer and specific application.

5.1.5 Cost Effectiveness

Softswitch technologies, simply put, are cheaper than circuit switches both in purchase price and in terms of OAM&P. This lowers the barriers to entry to new market entrants, which is critical for competition to enter the local loop market place. The following paragraph will delve into the cost factors of a voice service provider and how softswitch is the more cost-effective option.

Purchase and OAM & P

Softswitch is cheaper than Class 4 in purchase cost and Operation, Maintenance and Provisioning (OAM&P). It is estimated that a service provider's largest cost is the ongoing expense of running the network and the switches themselves. Networks must be deployed, maintained, repaired, monitored, and upgraded. The ability to manage the entire network via a softswitch solution (including adding and altering specific Customer services, upgrading capacity, and fixing faults) from a centralized location leads to considerable time and cost savings.

Bandwidth Saving

The PSTN transmits conversations at a rate of 64 Kbps. Using a variety of voice coder / decoders (codecs), a conversation can be compressed to use less bandwidth. Table below lists the various International Telecommunication Union (ITU) voice codecs and their associated data rates.

Standard	Data Rate(kbps)
G.711	64
G.721	16,24,32,40
G.723	
G.226	
G.728	16
G.729	8
G.723.1	5.3,6.3

Table 5 ITU codecs and data rates

The advantage of compressing a conversation is best illustrated by the early long-distance bypass industry. For example, let's say a service provider is providing long-distance service from the United States to Ethiopia using a 64 Kbps satellite circuit that costs $1,000 per month. Using G.711 uncompressed voice, the service provider can offer only one conversation at a time. Allowing for some degradation of voice quality, the service provider could compress the conversation to 8 Kbps using G. 729. Very simply put, the service provider can now get 8 simultaneous conversations over the same 64 Kbps satellite circuit. This has the potential to boost the revenue stream for the service provider eightfold and cut the cost of transmission per conversation by a factor of eight. The service provider can determine which compression algorithm offers the best tradeoff in voice quality and revenue.

Lower Barrier to Entry

The overarching issue regarding softswitch being cheaper than Class 4 is that it lowers the cost barrier to enter the converging market. The 1egacy market has required facilities-based carriers to build much of their network at very fixed high costs. The service providers who actually built long-distance networks could charge very high rates for their circuits as they are a scarce commodity. In the case of long-distance bypass carriers, new operators could enter the market because their cost to do so is much less than if they had to purchase, install, and maintain a Class 4 switch for their operations.

In addition to a lower barrier to entry as regards the acquisition of softswitch, the leasing of IP circuits has also lowered the barrier to entry. The two major factors for new market entrants are the cost of switches and IP circuits.

Foot Print and Power Savings

Softswitch is smaller than Class 4 and requires less physical space to deliver the same port density, that is, the equivalent number of phone lines as Class 4. Depending on the configuration, a softswitch may take as little as one-thirteenth of the space required by Class 4 to perform the same service [22]. Due to the smaller footprint, softswitch power consumption and cooling requirements are also less than the legacy switches. Smaller hardware size also translates into lower "real estate" expenses. The smaller footprint of the softswitch translates into lower real estate costs for the service provider. And this is critical for a new market entrant or an incumbent expanding into new markets.

5.2 Case Study for NGN Deployments

A wireline network VOIP configuration is presented in figure below showing the various functional entities that might come together in physical devices and logical interactions .The physical entities in this case consist of the MGC, Applications Server (AS), Trunk Gateway (TG), Access Gateway (AG), Signaling Gateway (SG) and a Media Server (MS). The MGC terminates all the signaling (either directly or transported over IP) and carries out signaling interworking (for example, a SIP phone wanting to signal to a PSTN network.)

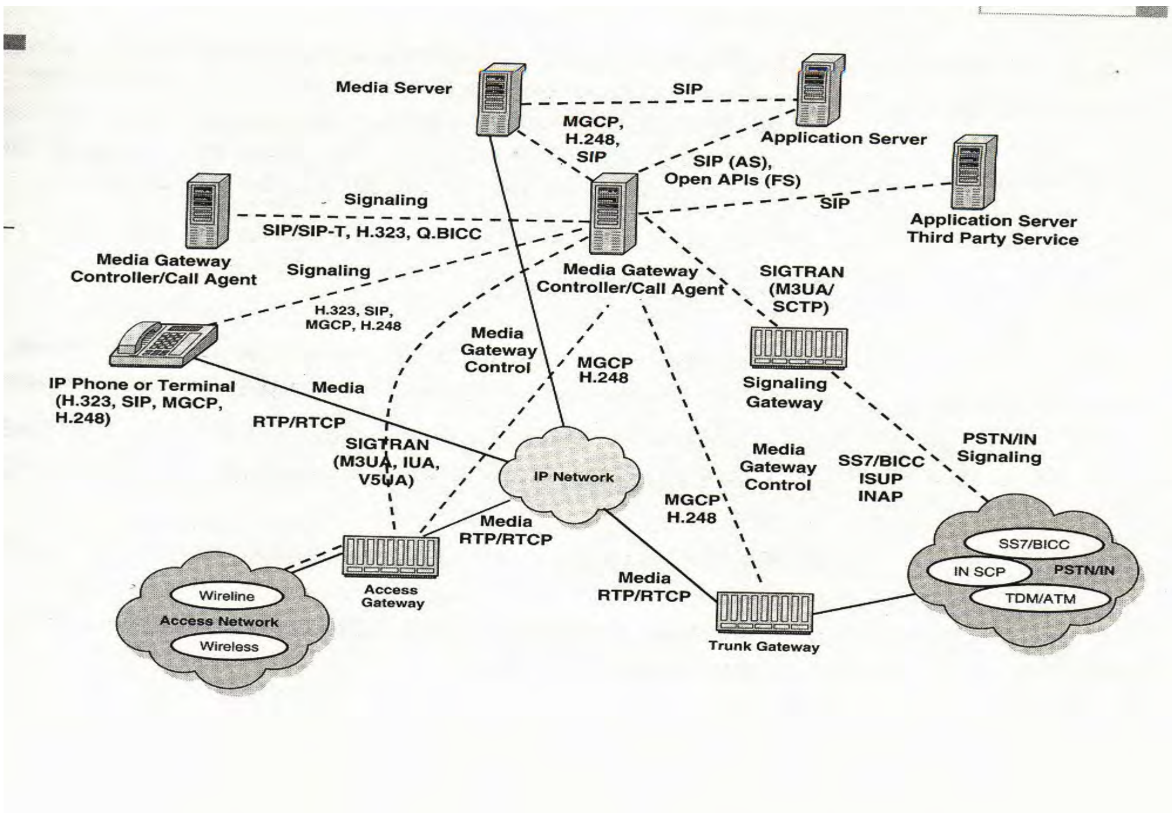


Figure 5.7 Reference Architecture for VOIP wireline network

It controls both the TG and AG for allocation of media resources. The MGC also authenticates and routes calls into the VoIP network and provides accounting information. Finally, the MGC interacts with other MGCs using either SIP/SIP-T, H.323, or Q.BICC. The AS has the service logic for applications, such as voice mail. Calls requiring these functions can be either handed over by the MGC to the AS for service control, or the application server can provide the information required for the service logic execution to the MGC. The AS can control the MS directly or pass control of the

MS to the MGC. The TG terminates the physical media carrying the bearer (voice streams) from the PSTN, transcodes the bearer, and transports it over IP into the IP network. The TG is controlled by the MGC. The AG serves as the interface between the IP network and any wireline or wireless access network. The AG transports signaling over IP to the MGC while transcoding the bearer and transporting it over IP to an IP end-point or to a TG for sending it to a circuit or other packet network. The SG terminates the physical media for the PSTN signaling and transports the PSTN signaling to and from the MGC over IP. The MS might perform tasks like announcements and digit collection, although the AG normally handles digit collection in most cases. The MS can be controlled either by the MGC, the AS, or both.

Class 4 Replacement Softswitch

The origins of Class 4 replacement softswitch solutions lay in the long-distance bypass industry. Long distance bypass operators used VoIP gateways for international transport, which enabled them to be competitive relative to long distance companies. Part of the success was due to the fact that they were able to avoid paying into international settlements. Initially, these service providers used enterprise-grade media gateways that interfaced with TDM switches in the PSTN. Technical challenges for these operators arose as their business flourished and demand grew. First, the media gateways were not dense enough for the levels of traffic they were handling. Second, the gateways that controlled these gateways were also limited in their capability to handle ever-increasing levels of traffic over these networks. Thirdly, international traffic called for interfacing different national variants of SS7 signaling with each nation having its own variant. In short, market demand dictated that a more scalable and intelligent solution be offered in the long-distance bypass industry. That solution came in the form of what is known as a Class 4 replacement softswitch solution, comprised of more densely populated gateways managed with greater intelligence than an MGC. The first applications involved installing a dense gateway on the trunk side of a Class 4 switch. The media gateway packetized the voice stream coming out of the Class 4 switch and routed it over an IP network, saving the service provider money on long-distance transport. The next step in the evolution of Class 4 replacement softswitch was the removal of the circuit-switched Class 4 switch from that architecture. That is, the Class 5 switch connected directly to a media gateway that routed the call over an IP network. The call control, signaling, and features were controlled by a softswitch, and the Class 4 switch was replaced in its entirety.

At this the service provider softswitch, as a Class 4 replacement switch, competes directly with the Class 4 switch. Figures 5.8 and 5.9 illustrate this evolution in architecture.

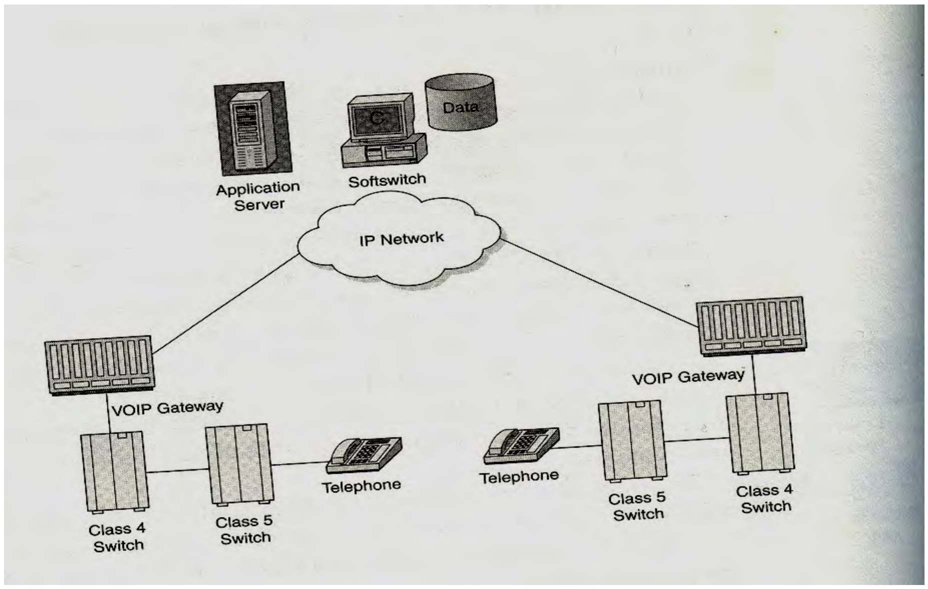


Figure 5.8Architecture of Class 4 Switches with VoIP Gateway

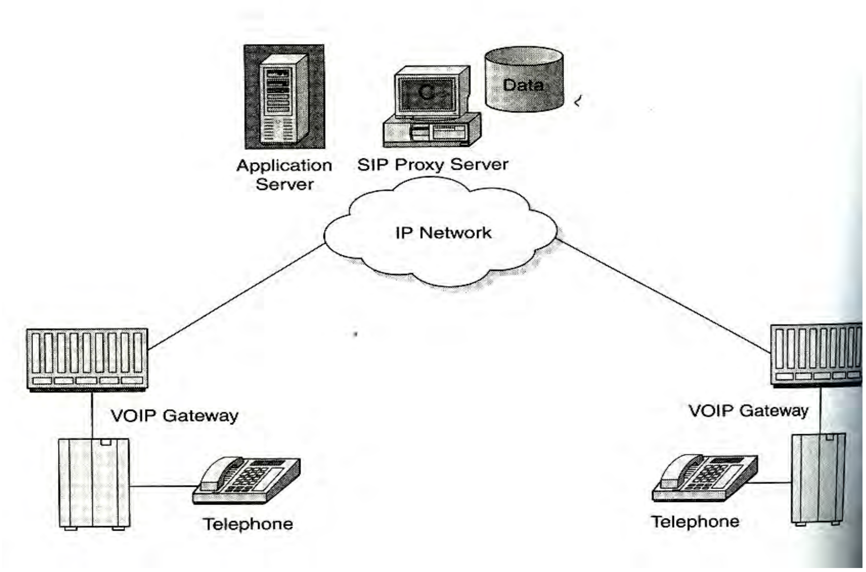


Figure 5.9 Class 4 Replacement softswitch solution

Class 5 Replacement Soft switch

The next level of progression in the development of softswitch technologies was the Class 5 replacement as shown in figure 5.10. This has led the most exciting debate over softswitch. The ability of the softswitch industry to replace the class 5 marks the final disruption of the legacy telecommunication infrastructure. The evolution of a successful Class 5 replacement softswitch has amazing implications for the world's local telephone service providers.

From the early days of the telephone industry, it was assumed that the cost of deploying local phone service with its copper pair access and local phone switches (most recently, a Class 5) was so expensive that only a monopoly could affect this economy of scale and scope. Enter a Class 5 replacement softswitch that does not cost tens of millions of dollars nor require a centrally located and expensive central office and the barriers to entry and exit crumble. The result is that new market entrants may be able to effectively compete with quasi-monopolistic incumbent service providers. This is potentially disruptive to incumbent local service providers and their Class 5 switch vendors.

In summary, the softswitches that replace PBXs and Class 4 and 5 switches (including Centrex) are differentiated in their scale. That is, their processing power is measured by the number of busy-hour call attempts or calls per second they can handle. Other differentiating factors include the capability to handle features from a feature server and interface disparate signaling protocols

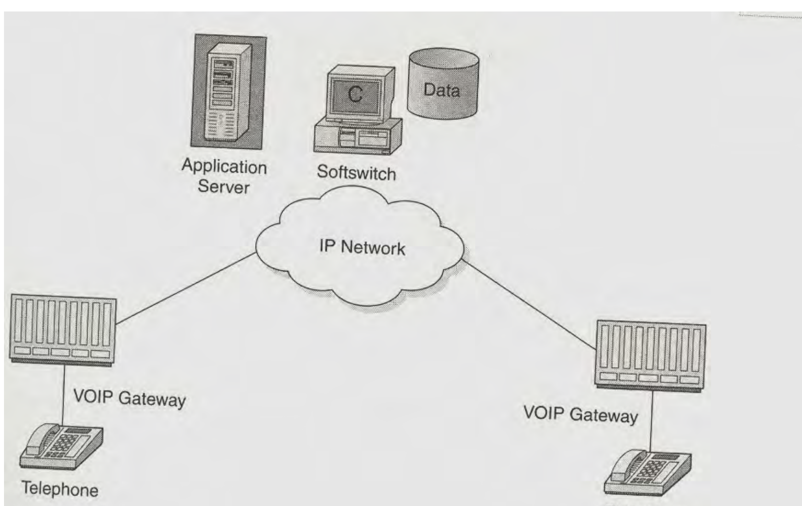


Figure 5.10 Class 5 Replacement Softswitch Solution

5.2.1 Case Study for Incumbent Operators

The incumbent fixed network operators have in general not been very fast in the uptake of the NGN network model. A principal reason for this is the large installed base of the circuit switching equipment that incumbent operators have. In some cases packet switching solutions have been used for specific product or geographic solutions such as the Telecom Italia national backbone network and British Telecom Business Services Solution. But incumbent operators across Europe have started to announce plans for deployment of NGN networks. The incumbent operator's chosen for the purpose of this work are British Telecom's 21st Century NGN network and the Italian Telecom BB Multimedia Network.

5.2.1.1 British Telecom

British Telecom (BT) is the incumbent operator in the United Kingdom. BT announced in 2004 the company's plans for the evolution of BT access and core networks to simplify and optimize network resources and to permit a platform for future service offerings. The plan entitled 21st Century Network consists of a major change of the network architecture with a reduction of number of network elements using a common packet switched backbone for both voice and data. On the access side all existing and new subscribers will be connected to what BT terms Multi Service Access Nodes (MSANs) which enable both voice and broadband services. This NGN solution is in contrast to the current BT network architecture where voice and broadband services are separated at the local exchange. BT's stated target is to ensure that, by 2009, broadband dial tone is instantly available to most BT customers in the UK. Consumers will be able to plug a broadband device into their phone line, and immediately be able to subscribe to BT's broadband service

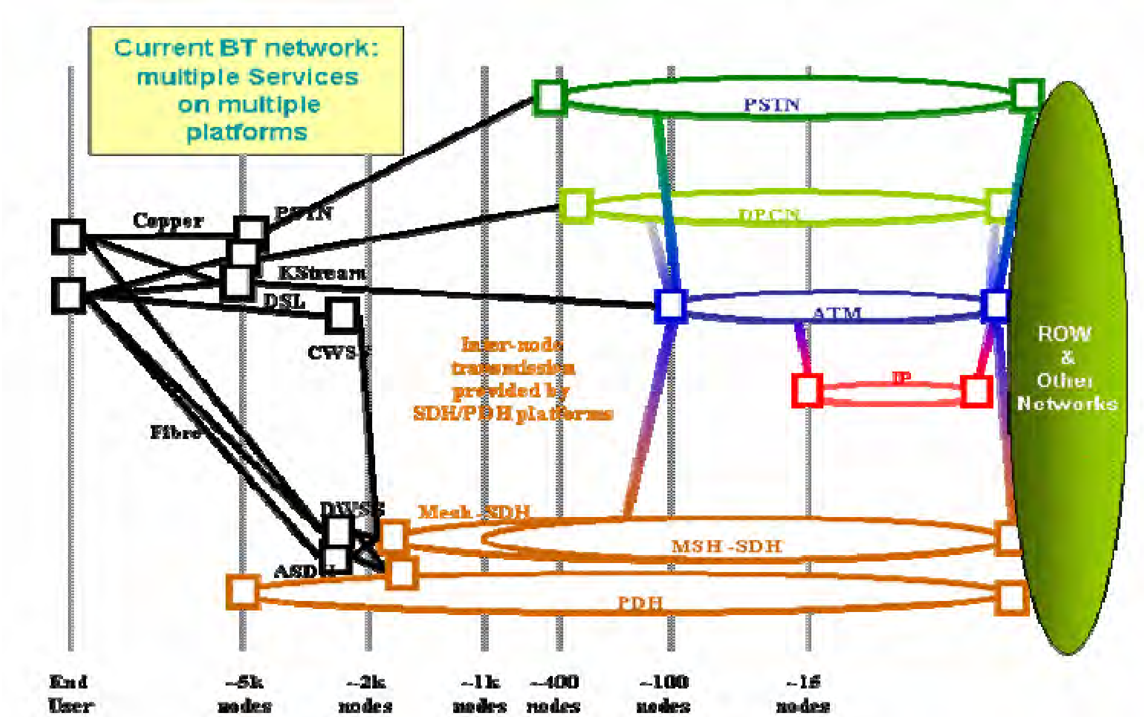


Figure 5.11 BT Network High Level Overview

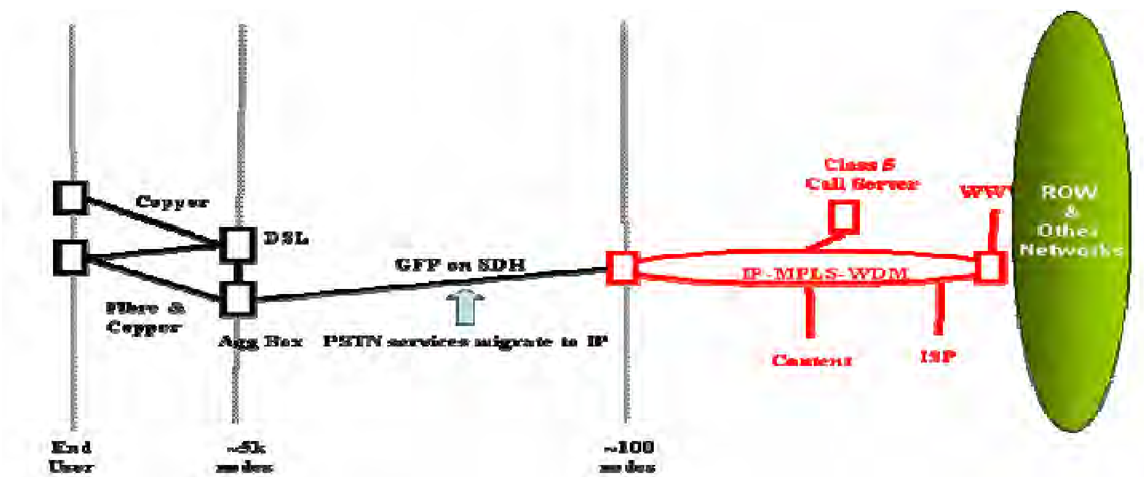


Figure 5.12 BT Next Generation Network High Level Overview

BT's stated aims with the new network architecture are to:

1. Make it easier to create new services (Faster and more people can create)
2. Make it easier to buy and use services
3. Make it simpler to deliver and maintain service
4. 30-40% cost reduction through use of fewer networks carrying more services

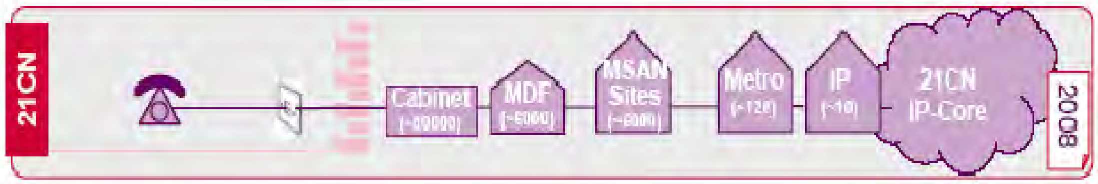


Figure 5. 13 BT 21st Century Network Model

The overall Architecture of the 21C network is based on:

- A converged Network based on IP and an MPLS core.
- An Intelligence layer that is mobile enabled based on 3GPP concepts permitting users to log on anywhere and use any device.
- Operation Support Service (OSS) systems integrated with a hub architecture permitting lower cost of integration and faster change.
- An open applications layer exposing re-usable capabilities – faster application provision and third party applications.

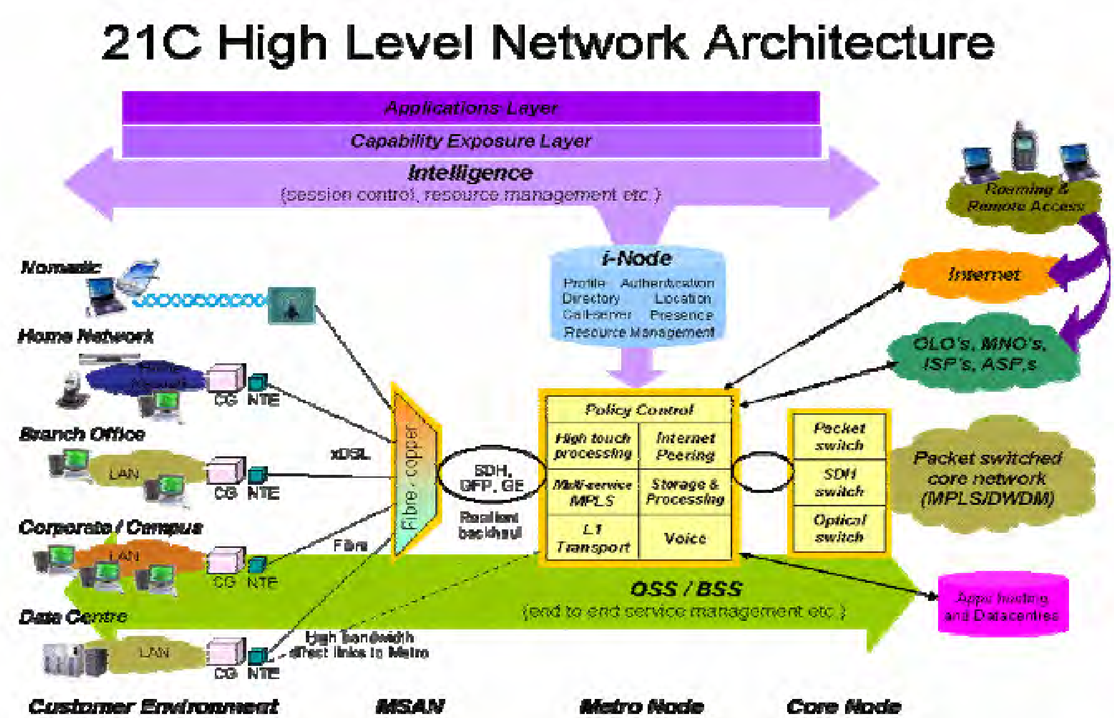


Figure 5.14 BT 21st Century Network High Level Architecture

BT has announced that its 21st Century Network will be deployed to the majority of its customers by 2008. This network has the potential to enhance competition in service provision and application design, if it uses open rather than proprietary standards.

The BT NGN network will be the one of the first major NGN network deployed in Europe and will be interesting to project to see further as it rolls out in the coming years.

5.2.1.2 Italian Telecom

Telecom Italia is the first major operator to undertake a wide-scale migration to a converged infrastructure. By 2006, a single national network will handle all the voice, data, and eventually video traffic generated across Telecom Italia's 27 million fixed subscriber lines in Italy and across its pan-European backbone, as Figure 5.15 details. Telecom Italia has already slashed 30 percent of its operational expenses in 2003 through its initial packet-voice implementation, prompting the company to accelerate its rollout plans.

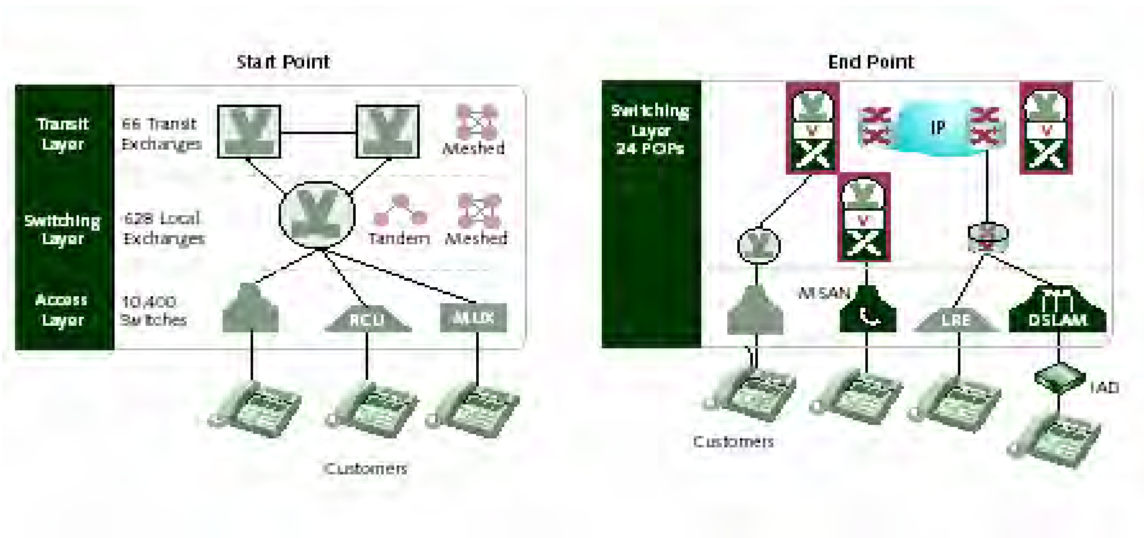


Figure 5.15 Telecom Italia Voice Network Evolution 2001-2006

Project Genesis

Telecom Italia's national Voice over IP migration project grew out of several earlier initiatives to reduce network complexity, as well as long-term relationships with both the equipment supplier Cisco Systems Inc., and the network integrator Italtel, itself a former division of the Italian operator. In 1996, Telecom Italia announced that it had chosen Cisco to implement a national Asynchronous Transfer Mode (ATM) network, with the aim to deliver various data communications services across a single network platform. The integration of voice, although mentioned in passing as a potential target for integration, was of secondary interest at the time. Commitment to voice and data convergence came to the fore in 1997, when Telecom Italia decided to converge its pan-European network. This first project served as proving ground for the more complex task of migrating Telecom Italia's national voice infrastructure. For the company's international network, the economic benefits of running a converged network were undeniable just from the sheer cost of buying and maintaining major voice and data transmission links outside Italy.

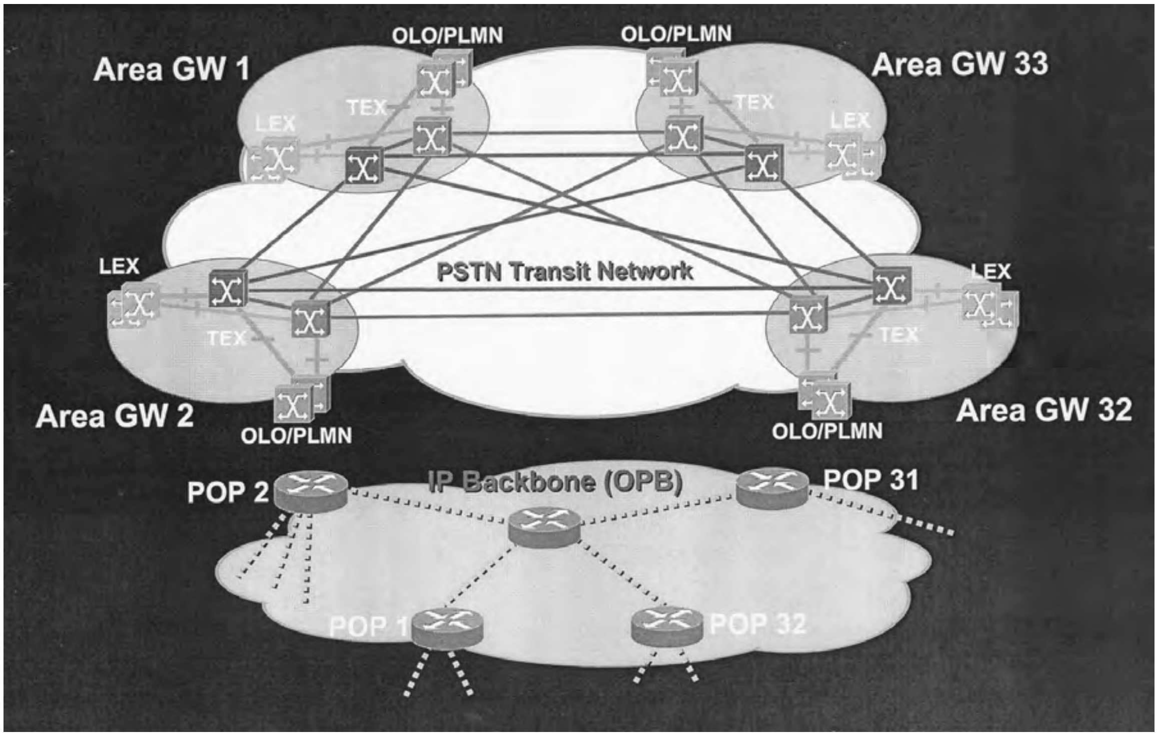


Figure 5.16 BBN Project: Merging TDM and IP Networks

PSTN Evolution is Opex and Market Driven, and Evolution phase towards NGN comprises of the following phases.

- Phase 0:Pan European backbone: VoIP on the International Network
- Phase 1: National Backbone Class 4 ; VoIP at National Transit Exchange Level
- Phase 2 : Native VoIP services over broadband access; Class 5 softswitch solution for broadband corporate customers and SOHO
- Phase 3: Next Generation Class 5; VoIP at the local exchange level

Telecom Italia Network Migration

Telecom Italia plans a dramatic cut in the number of connections operated on its national voice transit network from more than 1,100 to only 24 as shown in fig 5.17. Previously, maintaining its SDH-based national voice transit network meant that Telecom Italia had to complete thousands of tests and measurements to maintain the network at a decent level of performance. From 66 transit POPs and 628 local exchanges, Telecom Italia will downsize to only 24 PoPs. Currently, to provide resilience, the operator’s 10,400 national access nodes are each linked to two PoPs. The new plan will require a fundamental architectural shift from a highly meshed infrastructure with complex built-in redundancy to a streamlined hub-and-spoke configuration. The replacement of Telecom Italia's Class 4 and Class 5 switches with a Cisco-Italtel VoIP softswitch-based solution, and the installation of MPLS Traffic Engineering tunnels between different PoPs, based on a IP optical converged network that is to handle all voice and data traffic are currently under way.

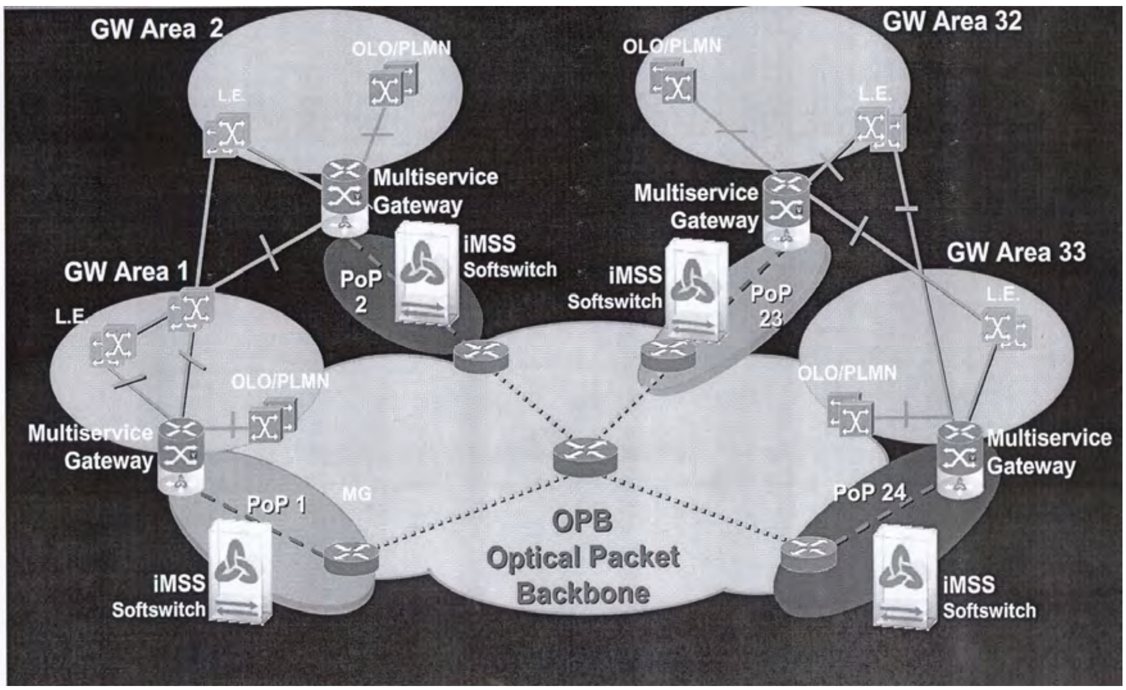


Figure 5.17 Italian Telecom BBN Multiservice National Backbone

5.2.2 Major Vendors NGN Solution Assessment [9]

The softswitch architecture market, composed of softswitch/call control and voice over Internet Protocol (VoIP) gateways, has gone through some iterations of industry hype, from an initial "packet will replace all legacy infrastructure" to a now more moderate network migration and packet overlay approach. Many vendors following the cycle have vanished or were acquired, and some have been able to build out their portfolio to have product solutions for greenfield Internet Protocol (IP) deployments as for network evolution.

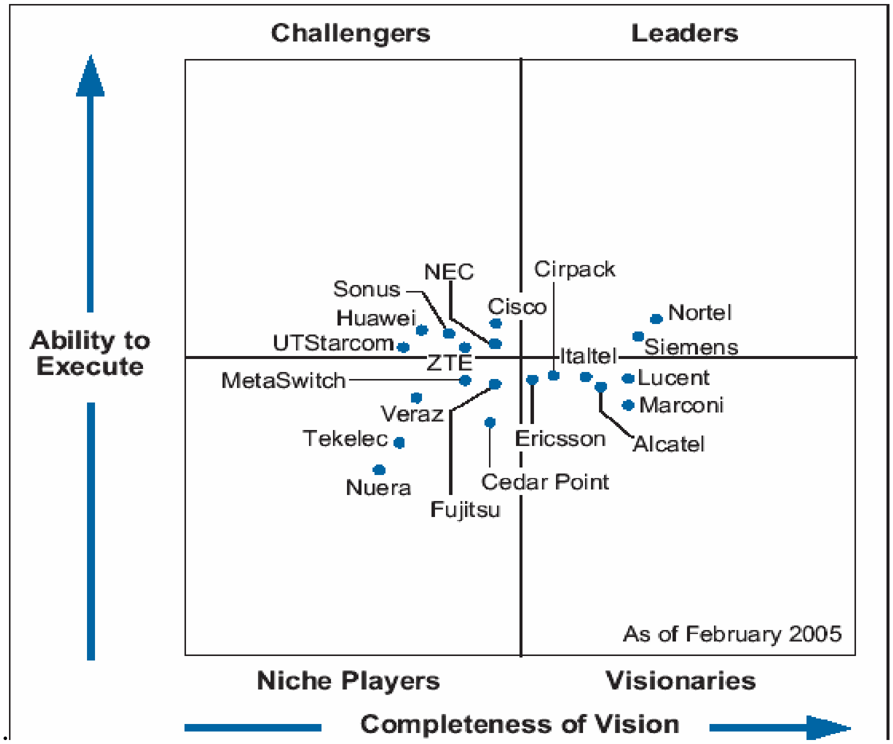


Figure 5.18 Magic Quadrant for Softswitch Architecture (Source Gartner Research)

As the carrier demand for VoIP solutions ramps up, vendors must follow through with their strategic commitments while enhancing their technology vision and market presence in different regions. Vendors must also build their support networks either through partnerships or through organic growth. Furthermore, companies such as ZTE, and Huawei are expected to expand their presence in international markets with lower than average selling prices.

Each company that passed the inclusion criteria is situated based on the company's current state as well as its strategic direction. The table below presents Gartner's report on the position of softswitch vendors in the market based on the Magic quadrant[9].

Nortel	➢ the strongest vendor in the market ➢ offers softswitch solutions to global and domestic carriers
Siemens	➢ a high reputation for carrier switching capabilities ➢ commitment to open interfaces for revenue-generating services
Cisco	➢ recognized for IP expertise ➢ Improving its marketing capabilities, and has entered partnership with Ericsson and enhanced the Italtel partnership
Alcatel	➢ good positioning in its strategy for NGN with its program called OPEN (Open Path to Enhanced Networking). ➢ strong recognition for its TDM switching legacy, hence good chances to move into the leader quadrant
Ericsson	➢ The company has not been very aggressive in marketing its wireline NGN switching strategy ➢ has strong carrier support and an outsourcing reputation that provides the company with the advantage of being able to offer a strong partnership and commitment to its customers.
Huawei	➢ maintains a cost-effective manufacturing and R&D process, which can translate lower prices in the global market
NEC	➢ strong challenger in the softswitch architecture market in Asia/Pacific, and holds 48 percent of the market in Japan.
Sonus	➢ one of the first companies shipping carrier-grade trunking softswitch architectures, and therefore it has high name recognition among carriers
Marconi	➢ a strong technology road map and commitment to VoIP
Lucent	➢ reputation in the industry as a technological innovator stems from its Bell Labs R&D operations
ZTE	➢ benefits from low-cost manufacturing, but it needs to increase the build-out of its support network and partnership globally to be able to meet worldwide operator requirements.

Table 6 Summary of Gartner's report on Major Softswitch vendors

5.3 NGN Migration: ETC Case Study

5.3.1. Next Generation Network Design Considerations

This section will see the main design considerations for NGN networks as deduced from the analysis of the technological, economical and regulatory considerations seen.

The following factors are considered to be important in the design of an NGN network: *Quality of Service, Protocol Choice, Security, Multimedia Options, Numbering and Regulatory considerations.*

I. Quality of Service

Quality of Service (QoS) is a broad term used to describe the overall experience a user or application will receive over a network. QoS involves a broad range of technologies, architecture, protocols. Network operators achieve end-to-end QoS by ensuring that network elements apply consistent treatment to traffic flows as they traverse the network. A number of QoS parameters can be measured and monitored to determine whether a service level offered or received is being achieved. These parameters consist of the following: *Network availability, Bandwidth ,Delay, Jitter ,echo and packet loss.*

II. Voice of Packet Protocol Choice

One of the first key design choices faced by operators when designing a packet switched network is the choice of protocol. On a core network level one of the first choices will be whether to have the backbone based on ATM or IP. The advantage of ATM was that many carrier backbones are already running on ATM technology and the DSL (including VoDSL) access has traditionally been provided over ATM as DSL is based on ATM. However for the foreseeable future IP will continue to dominate the market through its widespread ubiquity and versatility and although in scenarios with a high DSL deployment it may still make sense to keep some ATM in the network the future of voice packet switching will predominantly be based on IP networks. There is a range of voice over packet transport protocols available in packet switched networks. These will include Emulated Loop Control Protocol ELCP(Voice over ATM), MGCP/MEGACO. H.323, and SIP. The choice of protocol to use for voice services on a packet network is not a clear one and will depend a lot on the type of users the network aims to support as the requirements of large multi-user business customers are different to those of single user residential subscribers. For the time being at least it will be important that an operator network can support multiple VoIP protocols. ATM may still be used is some parts of the packet switched networks but the use of VoATM is seen as being replaced by more versatile VoIP protocols even if part of the transport layer is still over ATM. Historically VoIP traffic was done predominantly using the H.323 protocol and H323 is the protocol with the largest equipment installation at the moment. However the main development efforts the manufacturing industry are going into SIP at the moment as it is seen to be a more simple but versatile protocol for the multimedia services of the future. Current large scale VoIP deployments show a pattern of SIP and MGCP deployments as access protocols but ideal model is not clear: MGCP is used to control dumb terminal devices in a "client-server" mode where typical standard POTS analogue telephones will be connected whereas SIP and H323 are implemented on intelligent endpoints which may be connected to other softswitches, IP PABXs, or multimedia terminal devices. Taking into account that the service provider can control services and thereby revenue more easily with a 'dumb' endpoint this is an advantage for the MGCP protocol implementation. However some more advanced multimedia services involving voice may not be available with MGCP.

	H.323	SIP	MGCP/H.248/Megaco
Standards body	ITU	IETF	MGCP/Megaco—IETF; H.248—ITU
Architecture	Distributed	Distributed	Centralized
Current version	H.323v4	RFC2543-bis07	MGCP 1.0, Megaco, H.248
Call control	Gatekeeper	Proxy/Redirect Server	Call agent/media gateway controller
Endpoints	Gateway, terminal	User agent	Media gateway
Signaling transport	Transmission Control Protocol (TCP) or User Datagram Protocol (UDP)	TCP or UDP	MGCP—UDP; Megaco/H.248—both
Multimedia capable	Yes	Yes	Yes
DTMF-relay transport	H.245 (signaling) or RFC 2833 (media)	RFC 2833 (media) or INFO (signaling)	Signaling or RFC 2833 (media)
Fax-relay transport	T.38	T.38	T.38
Supplemental services	Provided by endpoints or call control	Provided by endpoints or call control	Provided by call agent

Table7 Cisco VOIP Protocol Comparison[13]

III. Security

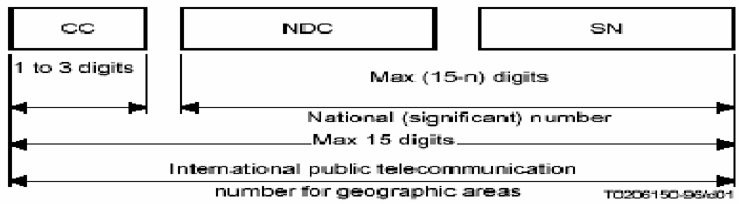
Hacking and replicating a Voice over packet call is easier to do than with circuit switched voice services as the more opportunities and ways to simulate and/or intercept IP packets than traditional circuit switched voice. Therefore it will be important to analyze possible threats, protocols weaknesses, and similar security risks and what can architecturally be done to ensure a secure environment.

IV. Multimedia Options Offered by a Packet Network

One of the advantages offered by a packet based voice switching network is the availability of a host of new service offerings taking advantage of the flexibility that packet networks offer. The widespread use of IP as the standard transport mechanism for data networks enables many advanced Internet applications which could be thought of as being enhanced voice services such as multimedia e-mail, real-time chat, streaming media (including music and video), and videoconferencing. Also with the increased bandwidth permitted by advances in access technology such as ADSL2 non voice services such as television and video on demand can also be provided over the same IP interface. The successful deployment of multimedia communications services in a commercially viable way to a diverse consumer population is a key challenge facing telecommunications service providers. The benefits of designing a network that is based on open interfaces and can support a range of media types is that new services can be interactive, On-demand & Personalized. By using an open protocol based packet switched network provides the network architecture for a service provider to deliver new advanced services quickly and economically.

V. Numbering/Addressing for packet based voice

Numbering is an important aspect of NGN design and depends on regulatory input to determine which numbering will be permitted and which will not. PSTN numbering follows the ITU-T E.164 recommendation. The ITU-T recommends that the maximum number of digits for the international geographic, global services, and network applications should be 15 (excluding the international prefix). The international public telecommunication number for geographic areas is composed of a variable number of decimal digits arranged in specific code fields. The international public telecommunication number code fields are the Country Code (CC) and the National (Significant) Number N(S)N as shown in Figure 5.19.



CC Country Code for geographic areas
NDC National Destination Code (optional)
SN Subscriber Number
n Number of digits in the country code

Figure 5.19 ITU International Public Telecommunication Number

Voice Subscribers on the packet network need to have an address that is accessible from the PSTN network. The type of number assigned should depend on the type of voice service associated to this number. There are fundamentally two types of application possibilities for voice packet switch subscribers:

Fixed Location VoIP: Voice services where the voice application is always in a fixed physical location like a voice port on a DSL IAD. The service is treated similarly to regular POTS services on the PSTN network.

Variable Location VoIP: Voice services where the voice application is not fixed to a single physical location such as soft phone client where users can logon and logoff from different physical locations to use the voice services. Subscribers of these types of services are also known as nomadic users.

Regarding numbering VoIP services can be addressed in many alternative forms such as through IP addresses, SIP addresses, H.323 addresses or E.164 numbers.

VI. Regulatory Considerations

Although the design of an NGN network is primarily the concern of the network operator certain aspects also have policy impacts and need be considered by policy makers such as regulatory bodies. Specifically the following areas of possible regulatory intervention related to the design of voice services on packet switched networks shall be considered:

- Whether changes in national & international numbering plans are required
- What VoIP interconnect policies if any should be implemented.
- Whether to impose rigidly the requirement for lifeline and emergency services access in packet switched networks

5.3.2 NGN Impacts on the Telecommunication Market Structure

The deployment of NGN packet switching networks can benefit both incumbent and alternative fixed network operators although the benefits are most substantial for network operators that do not have a large install base of TDM equipment which are not incumbent operators. Incumbent operators may have an advantage in the fact that they may find it easier to raise the capital investments required for the deployment of an NGN network. Principal NGN advantages are:

- . Lower network purchase and operating costs;
- Expanded geographic reach.
- New Services and Higher service differentiation;

NGN permits a reduction in operating costs for the wireline operators due to (a) reduced size and energy requirements and (b) due to increased lower switch port and software license prices due to increased competition among the manufacturers of softswitching equipment. Subscribers that might have been uneconomic or physically unreachable with circuit switch technology can be more easily reached in NGN networks as more access technologies are available to the network operators to use. But the main advantage of NGN is that the convergence of transport services in NGN networks, whereby Internet transport protocols are used to transport any type of information, and the convergence of content services of retrieving and displaying digitized information using the uniform Internet application protocols of the World Wide Web, together with the use of intelligent devices such as personal computers or integrated STBs at the edge of the network for multiple tasks , creates a new ubiquitous computing and communication platform permitting service providers to offer multiple and new services and move up the value chain by offering more than simple voice and data connectivity to customers.

With transport of packets becoming increasingly cheaper, the transmission of packets is becoming a commodity and very hard to make money out of a commoditized network in which complexity resides at the edges rather than the core. Telecoms companies are having to depart from the business of simply transporting bits and are trying to diversify their services. The NGN network technology could play a vital role in service differentiation, and in helping companies to make profits and consumers have more and better product offerings.

5.3.3 Ericsson NGN Description

Ericsson's vision of the future telephone networks is called ENGINE and relies on a packet based multi-service networks. The ENGINE concept is based on a migration of the current PSTN over to NGN, using an IP/ATM backbone network. While other NGN vendors such as Cisco propose a radical change to a completely new network, Ericsson focuses on a smooth migration towards NGN. By choosing a migration strategy, the current equipment is upgraded to obtain the benefits of NGN. There are three main components of an ENGINE network:

Telephony Server (TS)

The Telephony Server handles traditional PSTN and ISDN services over an IP/ATM connectivity network. The telephony server resumes the responsibility of the Softswitch in the current PSTN, handles call logic and controls the switching resources that are implemented in the MGW via the standardized H.248 control protocol.

Media Gateway (MGW)

The Media Gateways are at the edge of the connectivity network and connect to RSSs. The function of the MGW is to adjust, send and receive the information from users to the connectivity network and vice versa. It performs the real switching and media interworking functions between the TDM domain and the connectivity network.

Connectivity Network

This is a backbone IP/ATM network that transmits information between MGWs based on instructions it receives from the TS. Ericsson maintains the ENGINE can reduce total cost of ownership in a number of ways. There are however no documented or confirmed records of this and therefore they must be taken with care. The main savings that Ericsson lists are:

- Full integration of Local and Transit switching in one system
- Lower costs of software Operation and Maintenance due to processor centralization
- Lower transmission costs
- Smaller and less power consuming equipment
- Easy network planning
- Less spare parts , and higher functional flexibility and higher change readiness

5.3.4 Migrating from the current PSTN to ENGINE

The version of ENGINE that Ericsson offers now is called ENGINE Integral Network 3.1. Not much is known inside ETC about the technical implementation and very limited information is available from Ericsson. Hence a short description will follow then:

In essence, the migration to ENGINE is based on ***upgrading the existing Transit Exchanges to Telephone servers (TS) and the Local Exchanges to Media Gateways (MGW) and connect them all using a shared IP/ATM connectivity network.*** The system architecture of ENGINE then becomes that of Figure 20. Ericsson proposes that the structure of the current PSTN will be used as the basis for the structure of the ENGINE network.

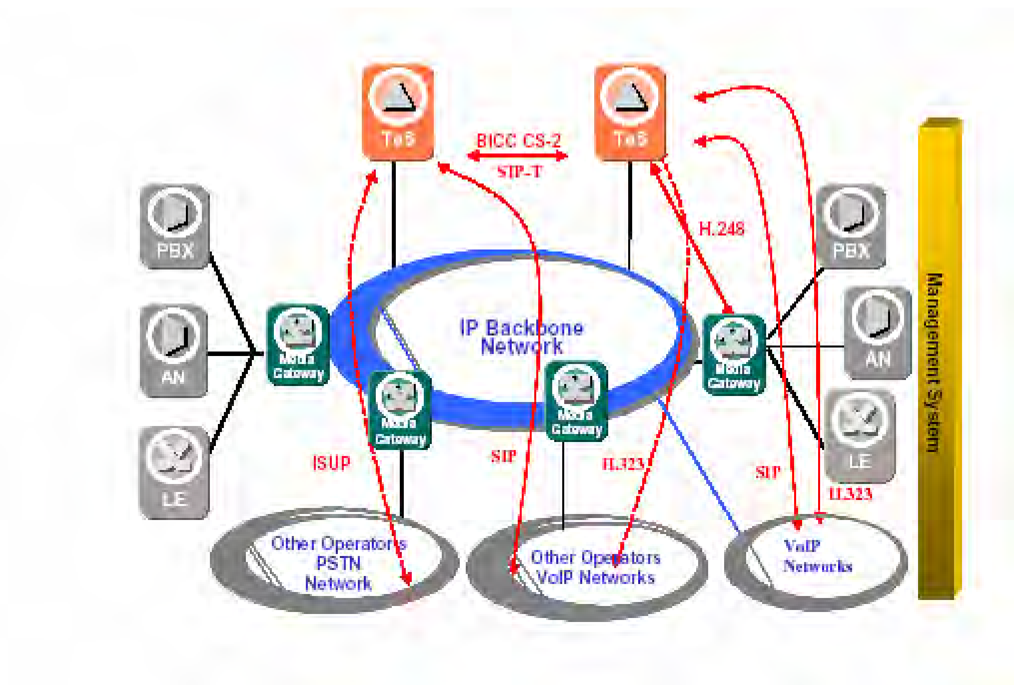


Figure 5.20 System Architecture of an Engine Network

While this work has been processing, Ericsson proposed its NGN Migration Plan to the Existing Network as follows:

I. Transit Layer in Addis Ababa with all destinations

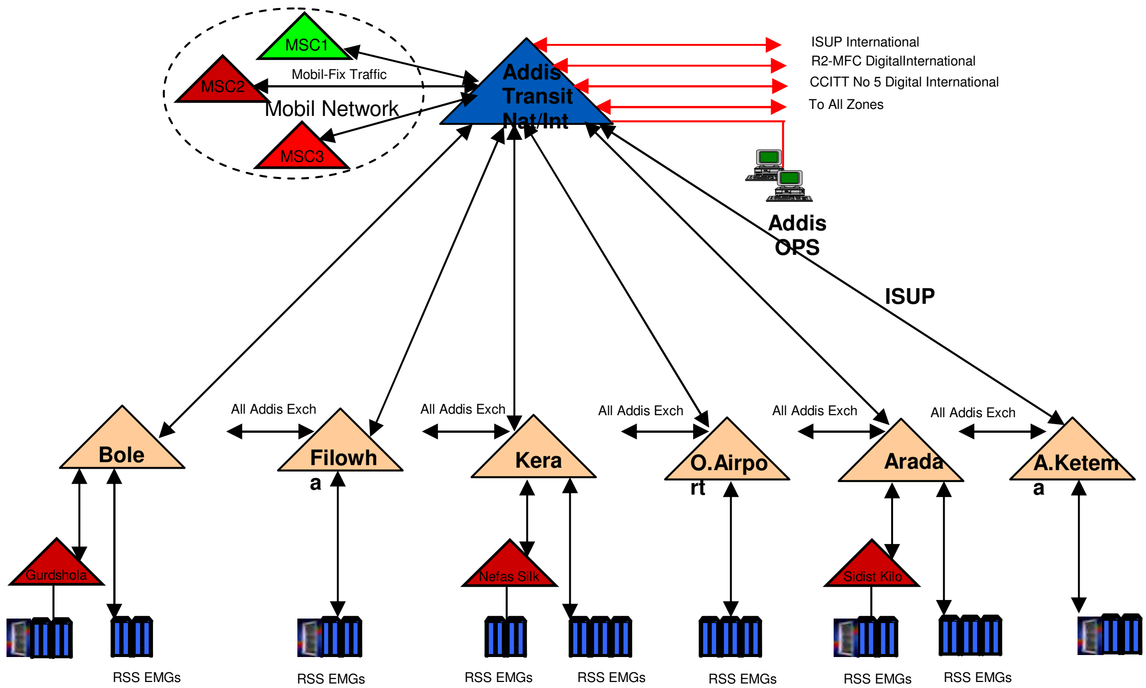


Figure 5.21 Current ETC Addis Ababa Circuit Switch Network

II. Proposed Next Generation Network based on IP/MPLS core

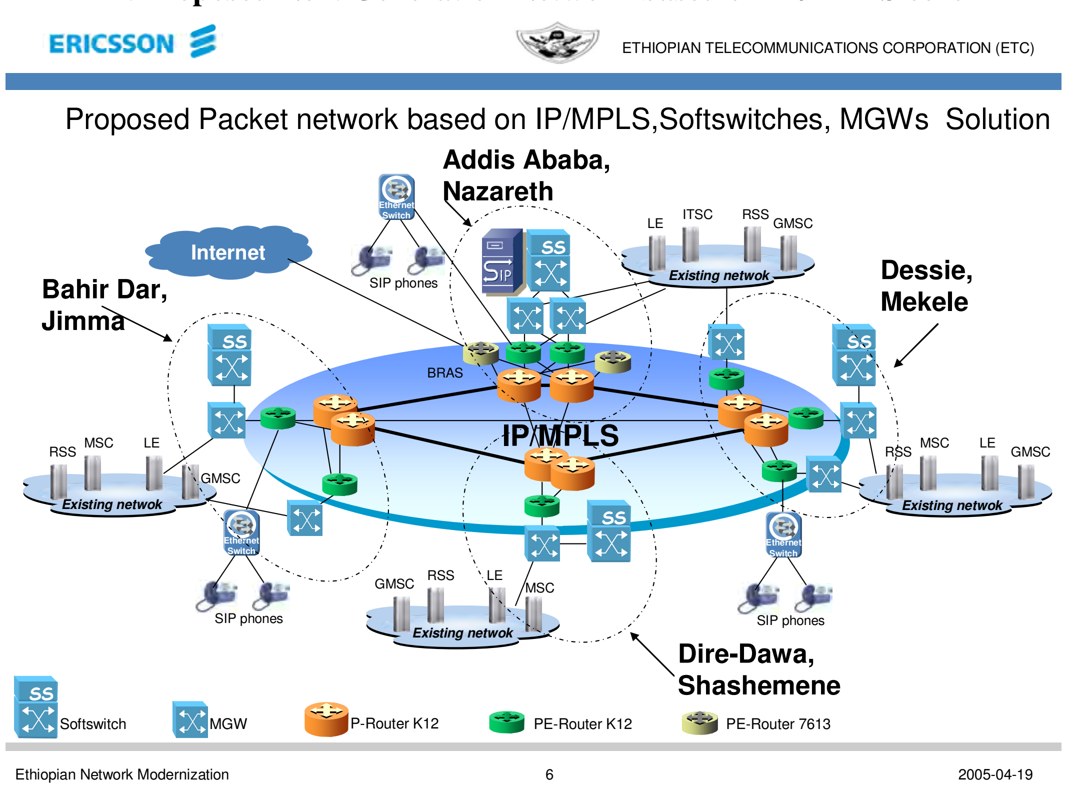


Figure 5.22 Ericsson's proposed packet network solution[12]

5.3.5 Maintaining the current PSTN

The alternative to migrating to ENGINE is maintaining the current PSTN. This will become more and more problematic and costly as the network becomes older. This is among other things due to service agreements with Ericsson that expire when equipment is not replenished. If the service agreements expire all spare parts and maintenance become more expensive. To stay functional, the current network would also have to be updated and replenished. The main cost components of this are according to Ericsson [11]:

- **Costs based on capacity demand**:

  New switch and interface boards to cape with traffic growth

  Switch replacements

- **Costs based on continuous functionality upgrades:**

  Network upgrade to allow for new functionality

  Control System (APZ) upgrade of all exchanges to cope with new SW demands

- **Costs because of old HW/SW**

CHAPTER VI

NGN SERVICE MODELLING

This chapter focuses on the economics of the next generation network services based on the case made on the Ethiopian Telecommunications Corporation market. For this the chapter has been presented in three phases. First it will overview the basic terms for investment analysis and describes the time value of money. Here the investment analysis parameters like NPV, IRR, ROI, and payback will be explained. Second, basic descriptions will be made on the Alcatel Service Modeling tool, describing to the services to be modeled and their revenue and cost drivers. Lastly the results of the modeling tool will be displayed in terms of Cash Flow Statement and Service Analysis. In general the Cash flow statement describes the results of the investment made on the whole service based the parameters NPV, Pay Back, and IRR. Sensitivity Analysis made on the NPV value by varying the discount rate and the terminal value are also presented. The service analysis shows the results of the revenue and cost of the services as piled up for the next five consecutive years. The NPV and IRR values for each of the services will be also presented.

6.1 Time Value of Money: Investment Analysis

The basic concept of a net present value procedure is that a dollar in hand today is worth more than a dollar to be received sometime in the future. A dollar is worth more today than tomorrow because today's dollar can be invested and can generate earnings. In addition, the uncertainty of receiving a dollar in the future and inflation make a future dollar less valuable than if it were received today. The procedure for accounting for the delay in receiving funds or the income given up is to discount, or penalize, future cash flows. The longer you must wait to receive them the more heavily you must discount them. This discounting procedure converts the cash flows that occur over a period of future years into a single current value so that alternative investments can be compared on the basis of that single value. This conversion of flows over time into a single figure via the discounting procedure takes into account the opportunity cost of having money tied up in the investment.

Investment Analysis is the method of determining if a given investment is profitable. Commonly used measures to determine the profitability of an investment include the investment's net present value, internal rate of return, and payback period. For this one must know the Initial Cost, Annual Net Cash Flow, Terminal Values and Discount Rates to be used.

Pay Back period

The Payback Period is defined as the length of time required to recover an initial investment through cash flows generated from the investment. The payback period provides visibility as to the level of profitability of the investment in relation to time. The shorter the time period the better the investment opportunity:

$$\text{Payback Period} = \frac{\text{Investment}}{\text{Cash Flow per Year}}$$

The payback period is a simple, straight forward tool, but does have its limitations. The payback period analysis does not address the time value of money, nor does it go beyond the recovery of the initial investment (ignores cash flows after the payback period.)

Net Present Value

The Net Present Value (NPV) of a project or investment is defined as the sum of the present values of the annual cash flows minus the initial investment. The annual cash flows are the net benefits (revenues – costs) generated from the investment during its lifetime. These cash flows are discounted or adjusted by incorporating the uncertainty and time value of money. In summary, NPV is one of the most robust financial evaluation tools to estimate the value of an investment.

The calculation of NPV involves three simple yet non trivial steps. The first step is to identify the size and timing of the expected future cash flows generated by the project or investment. The second step is to determine the discount rate or the estimated rate of return for the project. The third and last step is to calculate the NPV using the equations shown below:

$$NPV = Co + \frac{\text{Cash Flow Year 1}}{(1+r)^1} + \frac{\text{Cash Flow Year 2}}{(1+r)^2} \ldots + \frac{\text{Cash Flow Year n}}{(1+r)^n}$$

Or,

$$NPV = Co + \sum_{t=1}^{t = \text{end of project}} \text{Cash Flow } t \,/\, (1+r)^t$$

Co: Initial investment made at the beginning of the project. This value is usually negative, since most projects involve an initial cash outflow. The initial investment can include costs such as hardware, software licensing fees, startup costs, etc.

Cash Flows: The net cash flow for each year of the project: benefits - costs

Rate of Return: The rate of return **r** is calculated by looking at comparable investment alternatives given the risk of the project. The rate of return is often referred to as the discount, interest, hurdle rate, or company cost of capital. Companies usually use a standard rate for the project as they approximate the risk of the project to be on average the risk of the company as a whole.

t = Number of years depicting the lifetime of the project

The rule goes that a company should invest in the project if the NPV is greater or equal to zero. If the NPV is less than zero, the project will not provide enough financial benefits to justify the investment, since there are alternative investments that will earn at least the rate of return of the investment.

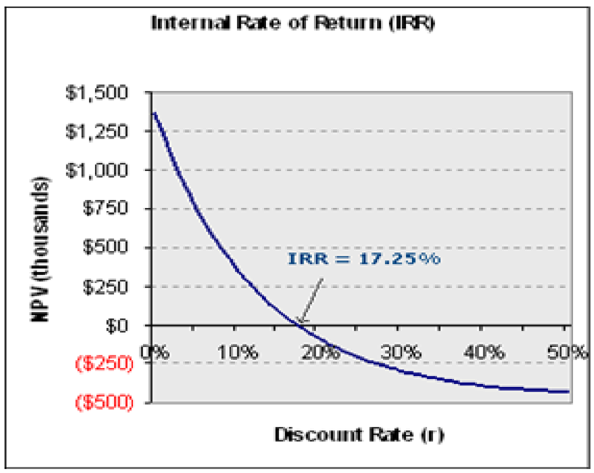
Internal Rate of Return

The Internal Rate of Return (IRR) is defined as the discount rate that makes the project have a zero Net Present Value (NPV). IRR is an alternative method to evaluate investments and is somewhat similar to NPV calculations but without the need to estimate the discount rate. IRR takes into account the time value of money by considering the cash flows over the lifetime of a project. The IRR and NPV concepts are related although they are not equivalent.

The IRR is calculated from the NPV equation as the starting point:

$$NPV = 0 = Co + \frac{\text{Cash Flow Year 1}}{(1+IRR)^1} + \frac{\text{Cash Flow Year 2}}{(1+IRR)^2} + \frac{\text{Cash Flow Year n}}{(1+IRR)^n}$$

The IRR is determined by "trial and error" by solving for the rate that would yield an NPV equal to zero. The "trial and error" calculation can be completed by using the solver function in a spreadsheet program or programmable calculator. The graph below was plotted for a wide range of rates until the IRR was found as shown below by the intercept with the x-axis that yields an NPV equal to zero.



Any project similar to the example above with a discount rate less than the IRR would yield a positive NPV. The higher the discount rate the more the cash flows will be discounted resulting in a lower NPV of the project. The company will approve any project or investment where the IRR is higher than the cost of capital as the NPV will be greater than zero. For example, the IRR for a particular project is 20%, but the cost of capital of the company is only 12%, then the company can effectively approve the project as the maximum value for the company to make money would be at the rate of 20%. If the company had a cost of capital for this particular project of 21%, then the company will have a negative NPV and the project will not be considered a profitable one.

The IRR is therefore the maximum allowable discount rate that would yield value considering the cost of capital and risk of the project. For this reason, the IRR is sometimes referred to as a "break even" rate of return. The IRR is then the rate at which the value of cash outflows equals the value of cash inflows.

Return On Investment (ROI): ROI is the most used metric when you need to compare the attractiveness of one business investment to another. The results of an ROI calculation are expressed in percentage terms and qualified by a time period. The main drawbacks of ROI are it ignores timing of benefits, the value is determined by its accounting income rather than actual cash flows and ROI requires a subjective discount rate. ROI figures alone don't give the full picture, and NPV can help to color your ROI analysis. Take for example two projects with equal ROIs, one with an NPV of $2 million, and the other with an NPV of $50,000. The first project is a more significant project to the company, since dollar value of your company's savings is more than forty times that of the second project. That is why the NPV is most often used in investment analysis.

ROI= [Cash Flow Year1 + Cash Flow Year 2 + … + Cash Flow of Year n] / C0; where C0 is the initial Investment made at the beginning of the project.

Terminal Value Multiplier*:* The terminal value is a multiple of the final year's cash inflow and represents the value of all cash inflows after the final year of computation.

Terminal Value: The product of the terminal value multiplier and the discounted cash flow in the last time of the valuation period. It is used to estimate the value of continuing cash flows.

Discount Rate: The discount rate is used to adjust future flows of income back to their present value. The discount rate chosen essentially indicates the minimum acceptable rate of return for an investment; it represents the "cutoff criterion" in judging whether or not an investment returns at least the cost of the debt and equity funds that must be committed or acquired by the business to obtain the asset.

COGS: Cost of Goods and services

Present Value (PV): The amount of money available or invested at the current time.

Future Value (FV): The value of amount of money to be received at some time in the future

EBITDA : Earnings Before interest and taxes plus depreciation and amortization.

As an illustration consider a hypothetical investment, where there is an initial cost of 10,000 ETB and a net return of 5,000 ETB annually for three years. We will assume a discount rate of 12 % and terminal value multiple of 5. The Cash flow analysis yields the following values:

	Year 0	Year 1	Year 2	Year 3
Cash Flow(CF)	-10,000	5,000	5,000	5,000
(1+12%) exp t	1	1.12	1.2544	1.404928
Discounted Cash Flow(DCF)	-10,000	4464.28	3985.97	3558.9
Cumulative DCF	-10,000	-5535.72	-1549.75	2009.15
Terminal Value Multiple	5			
Terminal Value	5 * 3558.9 = 10676.7			
NPV	TV + CDCF(Year 3)= **12,685.85**			
ROI	[CF(Y1)+ CF(Y2)+CF(Y3)] / CF(Y0) = **150%**			
Payback	2 years			
IRR	~23.5%			

The next section uses these economic values to measure the profitability of an investment made on Next Generation Network Services. The values are calculated using a modeling tool that Alcatel had delivered to ETC for the purpose of this thesis work. The basic reasons that Alcatel was chosen is because at the time of writing this, to the best of the authors knowledge only Alcatel have adopted such a tool (and which is recognized by ITU), and second the author has got a chance to be familiarized with the tool earlier, and also considers the cost of buying such a tool (if any) in the market.

6.2 **New Services Business Modeling**

Why models are used? Models are used because of the following points:

- They provide a coherent approach to a problem.
- They are a way to compare scenarios or findings
- A first way to detect opportunities or discard dead-ends
- They are methodologies to analyze further in detail a specific case
- They are a set of tools to assist decision makers

6.2.1 Description of the Alcatel Modeling Tool

General Model Structure

- The tool was developed on French Excel; hence commas are sometimes used as decimal points in captured screens.
- Taking into account a consistent development time frame for global telecommunication services, a 5-year model is used.
- Value beyond the 5 year window will be captured by Terminal Value.
- The model uses one year granularity. And this might be long for services with high growth. The model uses average demand per year, not end of year demand and these results in an approximate payback computation.
- And finally the model computes Costs and Revenues in the same period.

Model Architecture: The Alcatel Service model architecture with a description of each of the modules are presented below.

1. User Scenario: The tool generally considers two cases which are: Incumbent Operator in Africa and Incumbent Operator in Western Europe. For Each Scenario we will have two options to consider: Medium Network (1 million POTS) and Large Network (10 million POTS).

2. Financial Data

- The discount rate was taken as 12% as a default value. And the cash flows are discounted by the time value of money at the discount rate to compute today's value of future cash flows.

➢ The terminal value multiple (TVM) used to compute the Present Values after window of observation was taken to be 5 as the default value.

➢ For the corporate costs the model assumes as ***Internal Sales General Expenses and Administration (SG& A)*** which takes a percentage of the revenue as additional cost to support the company's corporate costs with a default value of 20%, and ***Promotion costs*** which is directly linked to the service and computed in addition as corporate costs

➢ The model is based on CASH FLOW ANALYSIS, not on Balance Sheet statement. Therefore Amortization and Depreciation (which are accounting values and not cash flow values) are NOT valued. And the financial model is BEFORE taxes. Taxes are NOT included

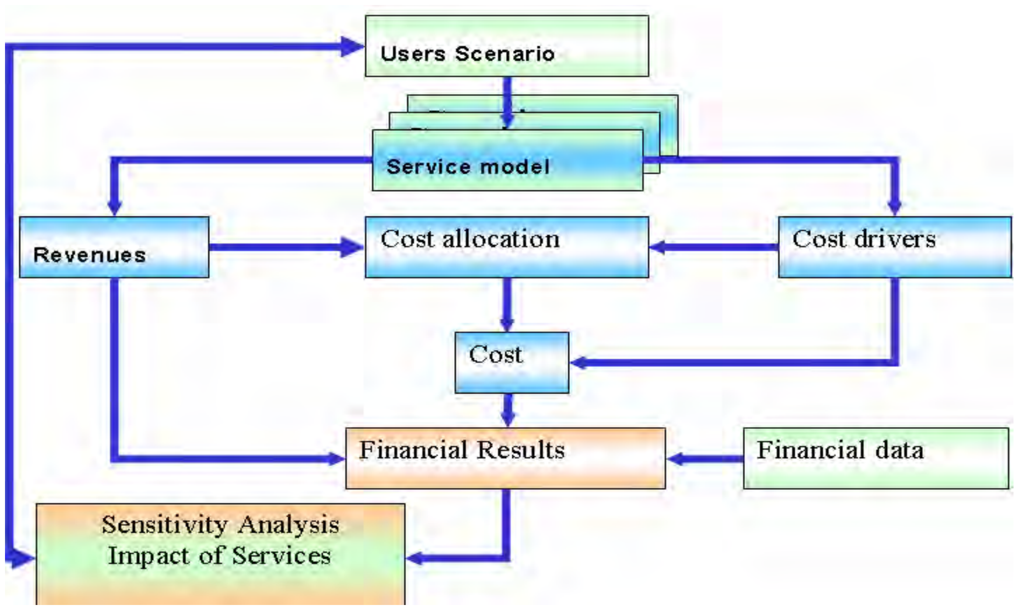


Fig 6.0 Alcatel Service Model Architecure

3. Services to be Modeled: The list of NGN services that are going to be modeled are: *IP Centrex , Multimedia Conference, Unified Messaging, Long Distance By-pass, IP Offload, VoIP, International PC to Phone, Video On Demand, and Content Delivery* Detail description and characteristic of each service like *Segment Drivers*, *Revenue Drivers* and *Cost Drivers* are identified and will be presented later on this chapter.

3. Revenues: Revenue per service is directly linked to service description (with tariff policy) and user's scenario (market assumptions)**.**

4. Cost Drivers: Each CAPEX cost item is associated to one cost driver, used as ***cost variable.*** Examples of Cost Drivers are CAPS (Call Attempts Per Second), Number of Users, Number of BB ports, Number of equivalent DS0, etc

5. Costs: The costs in the model will comprise of Capex Costs, Opex Costs, and Promotion Costs as described below.

- **CAPEX costs:** To account for the technological improvement and efficiency yearly decrease cost erosion was taken with a default value of 7%.The Capex costs includes:
    - ***Network based CAPEX*****:** This will include the following elements: Softswitch - MGC , Trunking Gateways, IAD , DSLAM, BRAS,ATM Aggregation, IP Transport, TDM service platform, NGN service platform, and Network Management
    - ***Service based CAPEX*****:** This will include the following elements: IP Centrex Application, LDB Application, VoD Application, Prepaid Application, Multimedia Conf. Applications, Content Delivery Application, VoIP application, IP Offload Application, VoD Set Top Boxes, and Billing Application
- **OPEX Costs:**
    - OPEX from Network-based elements is based on % of cumulated CAPEX (before cost erosion) and PY (Present Year) cost, based on empirical experience.
    - OPEX from Service-based elements is based on
        - number of persons to run the service and Present Year cost
        - Maintenance cost as a % of cumulated CAPEX
    - Model allows adding a yearly increase of OPEX (default: 2%)
- **Promotion Costs:**
    - Service-specific promotion is added, taking into account:

        - The necessity or not to advertise for a service
        - A reasonable promotion / revenue ratio
        - A reasonable absolute promotional budget
    - The full project promotion can be corrected to account for a cheaper global promotion campaign compared to many individual promotions

6. Cost Allocation:

- Some costs are shared among various services
- A Cost Allocation Table allocates each cost to the various services prorate either the service revenue Y5, or the number of Cost Drivers used by the service
- A few costs are not allocated to services, but to the full project. These are Common Costs.

7. Results: The results of the modeling tool will be specified by (as will be seen later):

- ***Global Analysis: Cash Flow Statement***
    - Cash Flow, Cumulated Cash Flow, Terminal Value, Net Present Value, Pay-back, Internal Rate of return
- ***Services Analysis***
    - Revenue per service per year
    - Cost per service per year

8. Sensitivity Analysis and Impact of services: There will be three market scenarios in the model described as:

- MAINSTREAM: Capture of service users and tariff as planned
- PESSIMISTIC: Entry of competition one year after. No impact on tariff, but 20% churn each year. First entrant takes 60% of the growth and of the churning users. Second entrant takes 40%
- CATASTROPHE: Competition enters first. Entry one year later. Symmetrical case from Pessimistic.

The sensitivity analysis will be done with macros accordingly as required by changing financial data and market variables. These will be done for example on NPV values by varying discount rate and terminal value multiple, impact of revenue shift , impact of cost erosion and impact of service mix.

6.2.2 Services to be modeled

After selecting the type of new services, the following key points have to be identified for each service.

The Value Chain: Here the major players like end users, service provider, content providers, access providers, and the business relation among each of these units will be identified.

Segmentation and bundles: Service features are bundled to best serve a uniform segment as Large Enterprises, Small and Medium Enterprises, Residential, Teens, etc.

Cannibalization: The new service contributes to lower the success of the existing service.

Diversification: The new service allows to recapture a substituted service that would have been lost otherwise.

For each of the NGN services the ***value proposition****, the different segments, the cost centers, the revenue drivers, and some other issues to be checked is identified. The Cost centers are the network based and service based capex values. Revenue drivers include the money value that the operator earns in providing such a service.*

1. VOIP

VOIP is access to telephone services from a data line (DSL). The different segments of VOIP are Corporate Users (for ex. VPN VOIP), SME/SOHO home workers (for ex. IP Centrex) and Residential Customers. The *Value proposition* of VOIP is identified such that; for the end user it will have access to different tariff schemes and enables for IP services like IP Centrex and Browse and talk and for the operator it will add value to broadband delivery. The major operator concern in delivering VOIP is cannibalization and QoS. The *cost centers* in delivering this service are Softswitch function, Trunking Gateways, Customer Premise Equipments to connect analog telephone sets and VoIP software package or IP phone cost. The *revenue drivers* are Connection fee, Subscription fee, Incremental CPE rental fee, and revenue from usage which could be on per minute or bulk price for national calls and IP phone lease. The major issues that an operator has to check in delivering this service are: speed of service penetration versus end-user value proposition, Service penetration compared to Internet or PC penetration, Global Voice ARPU (Average Revenue per User) compared to PSTN , and the competitive pressure of cannibalization or additional revenue .

2. IP Centrex

IP Centrex provides PBX-like voice and Computer Telephony Integrated (CTI) services to enterprises. The main targets of IP Centrex are SOHO/SME and Large Enterprises. Its *Value Proposition* is identified such that for the end user it will have no CAPEX investment from non-core business (telecom), no operation staff to be trained and employed, easier cost management (against move, obsolescence, traffic patterns), IP convergence, and converged service for home workers. For operators IP Centrex offers voice services on top of (already deployed) data-VPN, move up the value chain providing core-business (voice) services and has a possible brick for an SME bundle. The *cost centers* for the service are VoIP as pre-requisite (VOIP Cost Centers), IP Centrex applications, and service platform. The *Revenue Drivers are s*ervice subscription and setup fee. The operator has to check issues like the impact on PBX reselling business and tariff structure for usage on delivering this service.

3. **Multimedia Conferencing : MMC** is multi-party room-based or PC-based multiservice conference (with document sharing, Instant Messaging facilities,.). Its value proposition is identified such that for the end users it spares time and costs , lowers security hazards , low cost and user friendliness compared to ISDN room-based video conferencing. For the operator, MMC will increment value on data and voice; it will brick for package to business segment or vertical service (distance learning; home working...), and lower cost to operate the videoconferencing helpdesk. The cost centers for MMC are Softswitch, Trunking GW, VoIP functions to link non-IP users, and multimedia conferencing platform and bridges. The revenue drivers are setup fee, subscription fee, and consumption. The main issues that an operator has to check on this service are threat of "free" web conference, and cannibalization of ISDN video conferencing.

4. Unified Messaging

Unified Messaging (UM) is the type of service to retrieve from anywhere any message from any device. The main targets for UM are Business men, and teleworkers. Its value proposition to the end user is that it helps productivity gain, ease of use, and time-critical information management. For an operator UM will move up the value chain in Business segment and leverage mobile services. The main cost centers are UM platform and SW, and Individual messaging resources (Voice Mail, SMS server, etc). Issues that an operator should check in delivering this service are that the service is partly offered by advanced IP-PBX systems and benchmark with Mobile penetration.

5. Content Delivery.

Content Delivery is access to paid content (Music, Games, and Gambling). Its Value proposition for end-users are trusted account with operators, single wallet and seamless identification and for operators it will capitalize on their access to users, get share of e-commerce, content providers and ease of use for distribution channel; operators as trusted party. The cost centers for such a service include Payment and Authentication chain and hosting servers. The revenue drivers of Content Delivery include share of content revenue as "broker's" fee, and hosting services (servers) depending on business model. The major issues that an operator should check on delivering this service include relationship to content providers, and willingness to move into Access Service Provider (ASP) model.

6. Video On Demand (VoD)

VoD is access to movies on a per-demand basis from DSL line (VoD, NVOD, iTV). Its value proposition for end users are control / personalization of video content and for operators capture a part of entertainment value chain and uplifts DSL demand. The cost centers of VoD include Softswitch, Multimedia Gateways, Billing / Authentication solution, and capacity in transport network. The revenue drivers are share of revenue depending on value chain, and possible payment intermediation. The major issues that an operator should check on delivering these service are to see on alternate solutions like

CaTV, future digital TV , business model and value chain compared to traditional video delivery, and digital right management agreements with editors.

7. Long Distance Bypass (LDB)

LDB is IP transport of Voice for International Long distance. Its Value proposition for the operator are decrease the operational cost, higher bandwidth efficiency, scaleable solution, and step to target the NGN network. The cost centers of LDB are Softswitch, Trunking Gateways. LDB is an existing service; hence it has no revenue drivers except that if linked to a tariff decrease, consumption might increase. The main issue that an operator should check on this service is the regulation and bi-lateral agreements.

8. IP Offload

IP Offload is the type of service which will take dialup internet traffic away from the PSTN Class 5 switch. Its value proposition for the traditional operator is first it will have more efficient use of existing local/tandem switch capacity for profitable voice. It will offload high-volume low margin data traffic to dedicated RAS platforms, and most cost effective way to connect RAS equipment to switches and capture business from ISPs who want to outsource IP management. For an Internet Service Provider (ISP) the value proposition is that IP Offload is the most cost effective to connect to PSTN operator. The cost centers for IP Offload are trunking gateways and no direct revenue drivers (since an existing service). IP Offload is linked to dialup traffic only.

9. IP Phone Home

IP Phone Home offers PC to phone for incoming international calls. The value proposition for end-user is lower price for a customer having a PC access, and for operator it keeps higher share of value chain, win churn from call-backers, lower prices whilst increasing revenue per user and keep control of all investment. The cost centers are Softswitch, Trunking GW, Authentication / billing chain and Prepaid platform.
The main issues that an operator should check on are tariff decrease versus existing service, tariff origin-dependent, and the regulatory impact.

6.3 Results of the Modeling Tool

6.3.1 General methodology

The Service modeling tool is developed by Alcatel on Microsoft Excel and Visual Basic Software, consisting of 22 Excel sheets. The tool uses a Five year Model taking in to account a consistent development time frame for Global Telecommunication services. Inside some of the excel sheets there will be macros which are used to make the analysis and comparison for other scenarios than mainstream case and with variable parameters like the time shift between opex and revenues, cost erosion, discount rate, and terminal value multiple at different rates.

The recent data from ETC's National Traffic showed that the total number of connected fixed line subscribers as of April 2005 is 550,000. The annual statistical bulletin of ETC for the year 2003/2004 showed that from the total number of 484,368 customers, there were 80,203 business customers, 351,182 residential customers and 41,855 governmental customers. To fit the model the governmental customer's behavior was categorized under the business customers. This will result in the 75% to 25% ratio in terms of the penetration to the number of residential customers to business customers.

The total number of POTS lines for the last five years (from 1991 E.C. to 1996 E.C.) were shown in the table below.

Year	No.of POTS lines	Difference	Yearly Incremental rate (As % of Value)
1991	194 494		
1992	231 945	37451	22.3 %
1993	283 683	51738	24.7%
1994	353 616	69933	14.4%
1995	404 790	51174	19.6%
1996	484 368	79578	16.7%
1997 (as of Apr.)	549 679	90,000 (assumed)	

Table 8 ETC : Number of POTS lines 1991-1996

From the above data, considering the technological ease for deploying more number of POTS and rapid increment in number of POTS, the average yearly incremental rate were taken to be 15% for the next five consecutive years , i.e. 2005-2009.

For the cost of manpower, a well paid professional with a gross salary of 3000 ETB per year per month were considered. All the currency values were taken in USD. This will count to around 4000USD per year.

The segment drivers, Revenue drivers and cost drivers of each of the services to be modeled were analyzed and put as follows:

The Segment drivers will be automatically computed from the above input values.

Under the cost drivers, the variable *Persons x year* represent the estimated requirement to operate and deliver the service itself. Network and equipment maintenance cost are accounted automatically by the tool and service related promotion costs to be displayed in % of sales (not in value). The input values for the revenue drivers were taken considering the market of current related services that ETC is delivering and the world market values. A typical example that can be taken here is VoIP. This service may require no setup fee and the subscription fee that is taken is 5USD (around 45 ETB) per month. The Usage revenue assumed per subscriber per month is 7USD (63 ETB). Particularly for this service no personnel and promotion costs will be required since it is an existing service. All these input values have been entered in the "SIMUL INPUT " sheet of the modeling tool.

The values for other services have been analyzed and put as follows:

1. **IP Centrex**

Revenue Drivers

- external calls are not accounted here (See VoIP)

	Y0	Y1	Y2	Y3	Y4
Set-up fee (one time)	25.0	25.0	25.0	25.0	25.0
Subscription fee (per month)	15.0	13.5	12.0	12.0	12.0
Added Revenue / month for Advanced centrex	8.0	8.0	8.0	8.0	8.0

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	5	5	14	21	24
Promotion	15%	10%	5%	3%	3%

2. **VoIP**

Revenue Drivers

	Y0	Y1	Y2	Y3	Y4
2nd line: Set-up fee (one time)	0	0	0	0	0
2nd line: Subscription fee (per month)	5	5	5	5	5
2nd line: usage revenue (per month)	7	7	7	7	7
2nd line: cannibalization %	75%	15%	0%	0%	0%
IP Centrex VoIP: usage revenue (per month)	10	10	10	10	10
IP Centrex VoIP: cannibalization %	75%	15%	0%	0%	0%

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	0	0	0	0	0
Promotion	0%	0%	0%	0%	0%

3. Multimedia Conferencing

Revenue Drivers

	Y0	Y1	Y2	Y3	Y4
Multipoint service subscription (per month)	30.00	28.50	27.08	25.72	24.44
Fee per hour of usage (per connected leg)	5.00	4.75	4.51	4.29	4.07
No of hours per month per conf site	10.00	11.00	12.10	13.31	14.64

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	8	8	8	8	8
Promotion	20%	15%	5%	4%	3%

4. Unified Messaging

Revenue Drivers

	Y0	Y1	Y2	Y3	Y4
Setup fee per user (one time)	0	0	0	0	0
Service subscription per bus. user (per month)	6	6	6	6	6
Usage fee per access to the service	0.1	0.1	0.1	0.1	0.1
Nb of access per month per user	20	20	20	20	20
Service subscription per res. user (per month)	2	2	2	2	2

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	10	20	30	30	30
Promotion	20%	10%	6%	6%	6%

5. Video On Demand

Revenue Drivers

	Y0	Y1	Y2	Y3	Y4
Set-up fee per user (one time)	35	35	35	35	35
Service subscription per user (per month)	10	10	10	10	10
Average operator revenue per movie	1	1	1	1	1
Nb of movie per user per month	5	5	5	5	5

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	8	8	8	8	8
Promotion	15%	10%	6%	6%	6%

6. Content Delivery

Revenue Drivers

	Y0	Y1	Y2	Y3	Y4
Spending on music (per month)	5.00	5.50	6.75	7.80	8.70
Of which to operator	10%	10%	10%	10%	10%
Spending on Games (per month)	5.00	5.50	6.75	7.80	8.70
Of which to operator	10%	10%	10%	10%	10%
Spending on gambling (per month)	12.50	14.50	16.50	19.01	21.50
Of which to operator	15%	15%	15%	15%	15%

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	3	3	3	3	3
Promotion	15%	10%	6%	6%	6%

7. IP Phone Home

Revenue Drivers

	Y0	Y1	Y2	Y3	Y4
Current "Calling card" fee to user per min	0.35	0.35	0.35	0.35	0.35
Of which retained by terminating operator	0.145	0.145	0.145	0.145	0.145
Fee per minute for "IP phone home"	0.22	0.22	0.22	0.22	0.22

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Person x years	6	6	6	6	6
Promotion	10%	5%	5%	5%	5%

8. Long Distance Bypass

Revenue Drivers

- ***No Direct Revenue***

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
Transport lease per port	300	300	300	300	300
Person x years	0	0	0	0	0

9. IP Offload

Revenue Drivers

- ***No Direct Revenue***

Cost Drivers

	Y0	Y1	Y2	Y3	Y4
TDM Cost per port	40.0	37.2	34.6	32.2	29.9
Person x years	0	0	0	0	0

The “setup sheet “is there to enter the economical parameters and launch the visual basic macros that create the “Scenario Result” excel sheet. These parameters include cost erosion per year for capex, months of revenue and opex shift, discount rate to compute the NPV, the terminal value multiple, and the market scenario.

	Enter below		*What is currently used:*	
Cost erosion per year for CAPEX	7%		*7%*	
Months of revenue and OPEX shift	4	months	*4*	*months*
Discount rate to compute the NPV	12%		*12%*	
Terminal value multiple	5		*5*	
Scenario being active in non-summary sheets	0		*0*	

Table 9 Default Values for Economical Parameters

Each of the service sheets details the Revenue drivers, cost drivers, CAPEX and OPEX network and service costs using the input values and the tool default values.

6.3.2 Global Analysis: Cash Flow Statement

The economical output of the modeling is basically described by the Cash flow statement on the “Incremental P&L sheet” and Service analysis on the “Summary Sheet” of the modeling tool. The cash flow statement in figure 6. 1 shows on its first row the revenues from the total modeled services in millions of USD for the year from 2005 to 2009. Costs include Network Implementation Cost and Service /Applications Implementation Costs. COGS is the cost of goods and services which includes the sum of the above two values. The Gross Margin calculates the difference of the COGS and the revenues from services. The corporate costs incorporate the corporate and administration plus a specific service promotion. EBITDA describes the earnings before interest and taxes plus depreciation and amortization. It is calculated by subtracting the sum the network and service implementation OPEX costs, and corporate costs from the total service revenues. The free cash flow is then calculated by subtracting COGS and Corporate costs from the Service revenues. Accordingly the cumulative cash flow, the discounted cash flow, Payback, terminal value after the year 2009, NPV and IRR are computed.

From the results it can be seen that the payback period for the total investment is 24 months starting from the year 2005. The terminal value of the investment after the year 2009 will be around 62 MUSD. And with this TV, the Net present value of the investment is around 88 MUSD with an Internal Rate of Return IRR of 213% by the year 2010.The default values taken for the Terminal Value Multiplier and Discount Rate were 5 and 12% respectively.

Sensitivity Analysis was then made for the Net Present Value by varying the values of the Terminal Value Multiplier and Discount Rate as shown in figure 6. 2. From the sensitivity analysis the Discount rate was varied from 8% to 12% and the Terminal value Multiplier from 0 to 16, and as can be seen from the results the NPV values decrease with increasing the discount rate taking a fixed TVM value. For the default TVM value of 5, the NPV varies from 101 MUSD to 80 MUSD when varying the discount rate from 8% to 15% accordingly. The various NPV values obtained have all NPV values above zero, which strictly measures the ***profitability of the investment***.

The next figure 6.3 shows the graph of Cumulative Cash Flow and IRR Value for the time frame in one chart. And it can be seen that the Cumulative Cash Flow values for the first year are less than zero.

All the above analysis was made for the mainstream scenario. For the pessimistic and catastrophe scenarios (described in 6.2.1), the results have been displayed in figure 6. 4 in terms of the Cumulative Cash Flow, IRR, and Cumulative Discounted Cash Flow with the pay back period values. The payback periods for the pessimistic and catastrophe scenarios will be 25 and 37 months, and IRR values to 164% and 66% by year 2010 respectively as shown from the result.

6.3.3 Service Analysis

The service analysis was presented in the "Summary" and "Graph Impact" sheets of the modeling tool. Here the revenues and costs of each of the services were presented for the period 2005 to 2009 E.C. Then the total service revenue and total service cost was computed for each year. The Network Implementation cost consists of the capex network cost, opex network cost and Customer Premise Equipment (CPE) cost and the Service/Applications implementation cost consists of the sum of Capex service, Opex service and promotion costs. Additional corporate costs were taken 20% of the total service revenues and all other values computed like the cash flow statement.

The Service revenue chart in figure 6.5 shows the total revenue that will be obtained for the next five years (from 2005-2009) piled up for each of the services. As can be seen ***Multimedia conferencing*** and ***Unified Messaging*** will be the most profitable applications generating around 30MUSD and 26MUSD respectively with IP Centrex and VoIP following accordingly. Long Distance Bypass and IP Offload will not generate direct revenues since they replace other traditional services. Video on Demand and Content Delivery services have not shown to generate much revenue, the other aspect of their value proposition to the service provider should be seen as well.

The Service Cost per year chart in figure 6.6 shows the total cost that will be paid for the next five years piled up for each of the services. The results show that much cost will be expenditured on delivering the services of Video on Demand (around 9MUSD) and UMS (7 MUSD), followed by IP Centrex and VoIP.

The next chart in figure 6.7 is all about the total service revenues that would be generated in each year for the next five years. And by the end of year 2009, the total revenue that the service provider will generate will be around 34 MUSD.

Figure 6.8 reveals the payback period and net present value of each service, which could help the service provider to decide on delivering to the end user.

The “Graph Impact “sheet in the modeling tool is all about delivering the output of IRR and NPV on the same chart. It was shown in figure 6.9 that the NPV value of VoD and Content Delivery are less than zero, whereas multimedia conferencing, IP Offload and LDB have the highest NPV values.

CHAPTER VII
CONCLUSION AND FUTURE WORKS

7.1 Conclusion

This thesis has described issues of migrating a telecom operator in general and Ethiopian Telecom in particular towards the Next Generation Network. Two strategic areas have been addressed separately for the migration. These are evolution of the existing PSTN towards NGN, and modeling of NGN Services.

For the issue of migration of the existing network, the problem was addressed by assessing the technology levels of the Class 4/ Class 5 switch with that of a softswitch. It was seen that those metrics of reliability, scalability, QoS and features which have been the issues of PSTN network have now become the issues of the Softswitched network.

The key findings include:

- Given is distributed architecture, a softswitched network may prove more reliable than the PSTN as it doesn't suffer from a SPOF.
- Softswitch solutions incorporate low density media gateways that disrupt legacy class 4/5 switch vendors and legacy telephone service providers. Scaling down property of softswitches had made them more advantageous than the legacy switches.
- A properly engineered and managed softswitched network can deliver comparable voice quality with that of a PSTN, due different QoS technologies that have being adopted.
- Due to its open architecture and flexibility relative to a Class 4/5 switch, Softswitch solutions can provide as many services as that of a PSTN.

Both the giant European telecom operators: BT and Italian telecom are deploying NGN based on the IP/MPLS core network. The general trend of migration for incumbents now can be seen in phases as:

Step 1: use of today's TDM-based network for voice telephony and Internet access;

Step 2: consolidation of switching and access equipment;

Step 3: introduction of Voice-over-Packet technology for trunking;

Step 4: introduction of Voice-over-Packet technology in access and CPE;

Step 5: multimedia services and new applications;

Step 6: end-of-life replacement of legacy infrastructure and migration to all-IP

Next Generation networks permit the deployment of advanced services. Like the network NGN economics have also to be made for services. Services like IP Centrex, VoIP, Multimedia Conferencing, Unified Messaging, IP Offload, Long distance Bypass, IP Phone home, and content delivery have been assessed and modeled using the Alcatel Service Modeling tool, and the results shown in Cash Flow Statement and Service Analysis over the next five years for the Ethiopian Telecommunication Corporation market.

In the **Global Analysis**, the results of the work reveal that by investing on those services, the NPV and IRR values of the investment by the end of year 2009 will be around 88MUSD and 206% respectively with a payback period of 24 months starting from 2005, which effectively shows the feasibility of the investment. Sensitivity Analysis on the values of the default economical parameters and the case for a possible competitor entrance was also made to see the variation of the results.

The **service analysis** results shows the revenues and cost of each of the services per year (from 2005 – 2009), total service revenues and NPV and IRR values for each of the services. From the Service Analysis, the following points are concluded.

- UMS and MMC have the major share of the service revenues
- VoD, UMS, and IP Centrex have the major share on the service costs
- The service cost of LDB and IPO will decrease by 2MUSD and 0.5MUSD at the end of the timeframe respectively
- UMS and MMC have much higher NPV and IRR values, which measures the economic profitability of an investment.

7.2 Future Works

The following areas relevant to Next Generation Networks were identified during this work that requires further research:

- First it has to start with the limitation. The economics of the network part were not assessed due the lack of software (modeling tool). This will be done in terms of cost analysis in migrating to the Next Generation network and maintaining the current PSTN. It was identified that Alcatel have such a tool, and due its internal policy, the company is not able to deliver it for the work as educational version. One future work circles around this.
- Considerable studies should be done too on how to implement a next generation network in Ethiopia. This will include from the basic migration plan to detailed technical implementation of the Next Generation Network.
- This work tries to focus on the wired network architecture and services of the next generation network. The NGN aspect as seen from the concept of IP Multimedia Subsystem (IMS) could be also another possible future area of work. The IMS is an open, standardized, NGN multi-media architecture for mobile and fixed IP services.

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ANNEX

Figure 6.1 CASH FLOW STATEMENTS

In million USD	2005	2006	2007	2008	2009	Assumed 2010 (*)
REVENUE from services	**1**	**5**	**13**	**24**	**34**	37
COSTS						
Network Implementation Costs	**1.8**	**0.4**	**1.0**	**1.7**	**2.0**	**0.5**
- CAPEX	2.5	0.3	0.8	1.4	1.5	-
- OPEX	0.1	0.1	0.2	0.4	0.5	0.5
- Cost reduction services	(0.7)	(0.0)	(0.0)	(0.0)	(0.0)	
Services / Applications Implementation Costs	**0.7**	**1.1**	**2.4**	**3.9**	**4.4**	**1.6**
- CAPEX	0.9	1.2	2.1	2.9	2.9	-
- OPEX	(0.2)	(0.1)	0.4	1.0	1.6	1.6
COGS	2.5	1.5	3.4	5.6	6.4	2.0
GROSS MARGIN	(1.6)	3.5	9.7	18.1	27.7	35.4
as a % of revenue	-166%	70%	74%	76%	81%	95%
Corporate Costs	0.4	1.6	3.4	5.8	8.2	8.9
Corporate & Administration plus M&S	20.0%	20.0%	20.0%	20.0%	20.0%	20.0%
	0.2	1.0	2.6	4.8	6.8	7.5
Specific service promotion	17.2%	12.4%	6.3%	4.5%	4.0%	3.8%
	0.2	0.6	0.8	1.1	1.4	1.4
EBITDA	0.7	3.3	9.1	16.6	23.9	26.5
as a % of revenue	75%	67%	70%	70%	70%	71%
FREE CASH FLOW	(1.9)	1.9	6.3	12.3	19.5	26.5
Discount rate	12%					
Cumulated Cashflow	(1.9)	(0.1)	6.2	18.5	38.0	64.5
Discounted CF	(1.9)	1.7	5.0	8.8	12.4	15.0
CUMULATED DISCOUNTED CASH FLOW	(1.9)	(0.3)	4.7	13.5	25.9	40.9
PAYBACK	24	months starting 2005				
terminal value multiple from 2009	**5**					
TERMINAL VALUE after 2009	62					
NPV	**88**					
IRR			135%	187%	206%	213%

Figure 6.2 Sensitivity Analysis of NPV by Varying TVM and DR

Sensitivity Analysis		Discount Rate							
	NPV	8%	9%	10%	11%	**12%**	13%	14%	15%
Terminating Value multiplier	**0**	29	28	28	27	**26**	25	24	24
	1	44	42	41	40	**38**	37	36	35
	2	58	56	54	52	**51**	49	48	46
	3	72	70	68	65	**63**	61	59	57
	4	87	84	81	78	**76**	73	71	68
	5	**101**	**98**	**94**	**91**	**88**	**85**	**82**	**80**
	6	115	111	108	104	**100**	97	94	91
	7	130	125	121	117	**113**	109	105	102
	8	144	139	134	130	**125**	121	117	113
	9	159	153	148	143	**138**	133	129	124
	10	173	167	161	155	**150**	145	140	135
	11	187	181	174	168	**162**	157	152	147
	12	202	194	188	181	**175**	169	163	158
	13	216	208	201	194	**187**	181	175	169
	14	230	222	214	207	**200**	193	186	180
	15	245	236	228	220	**212**	205	198	191
	16	259	250	241	233	**225**	217	209	202

Figure 6.3 Graph of Cumulative Cash Flow and IRR Value for the time frame

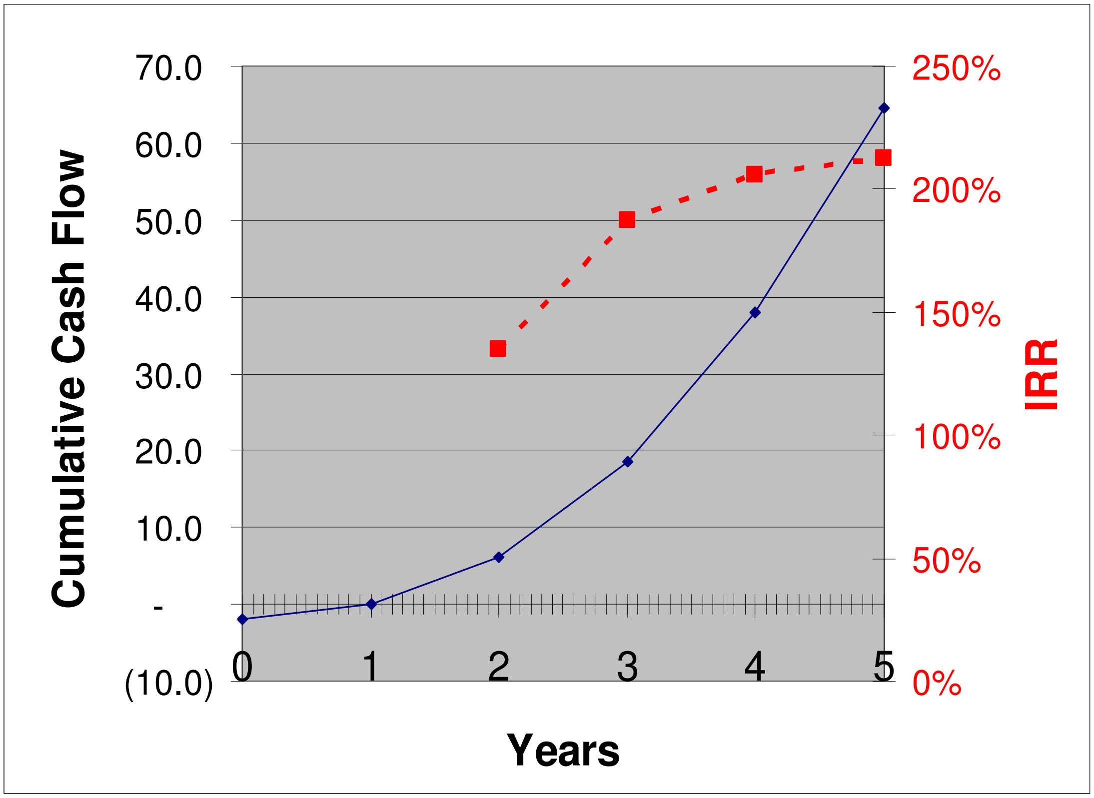

Figure 6.4 Mainstream, Pessimistic, and Catastrophe Scenario Results

Cumulative CF	2005	2006	2007	2008	2009	2010	Pay back month
Mainstream	-1.9	-0.1	6.2	18.5	38.0	64.5	24
Pessimistic - competitor after one year on same segment	-1.9	-0.3	3.7	10.8	21.6	36.3	25
Catastrophe - Operator second entrant		-3.0	-2.4	-0.3	4.2	11.2	37
Second entrant shifted one year earlier	*-3.0*	*-2.4*	*-0.3*	*4.2*	*11.2*		

IRR	2005	2006	2007	2008	2009	2010	Pay back month
Mainstream			134.7%	187.0%	206.0%	212.8%	24
Pessimistic - competitor after one year on same segment			93.2%	137.8%	156.0%	163.6%	25
Catastrophe - Operator second entrant			N/A	-4.7%	42.0%	65.6%	37
Second entrant shifted one year earlier		N/A	-4.7%	42.0%	65.6%		

Cumul Discounted CF	2005	2006	2007	2008	2009	2010	
Mainstream	-1.9	-0.3	4.7	13.5	25.9	40.9	88.0
Pessimistic - competitor after one year on same segment	-1.9	-0.5	2.8	7.8	14.7	23.0	49.0
Catastrophe - Operator second entrant		-2.7	-2.2	-0.7	2.1	6.1	16.2
Second entrant shifted one year earlier	*-2.7*	*-2.2*	*-0.7*	*2.1*	*6.1*		

Fig. 6.5 Services Revenues per year (years pile up from 2005-2009)

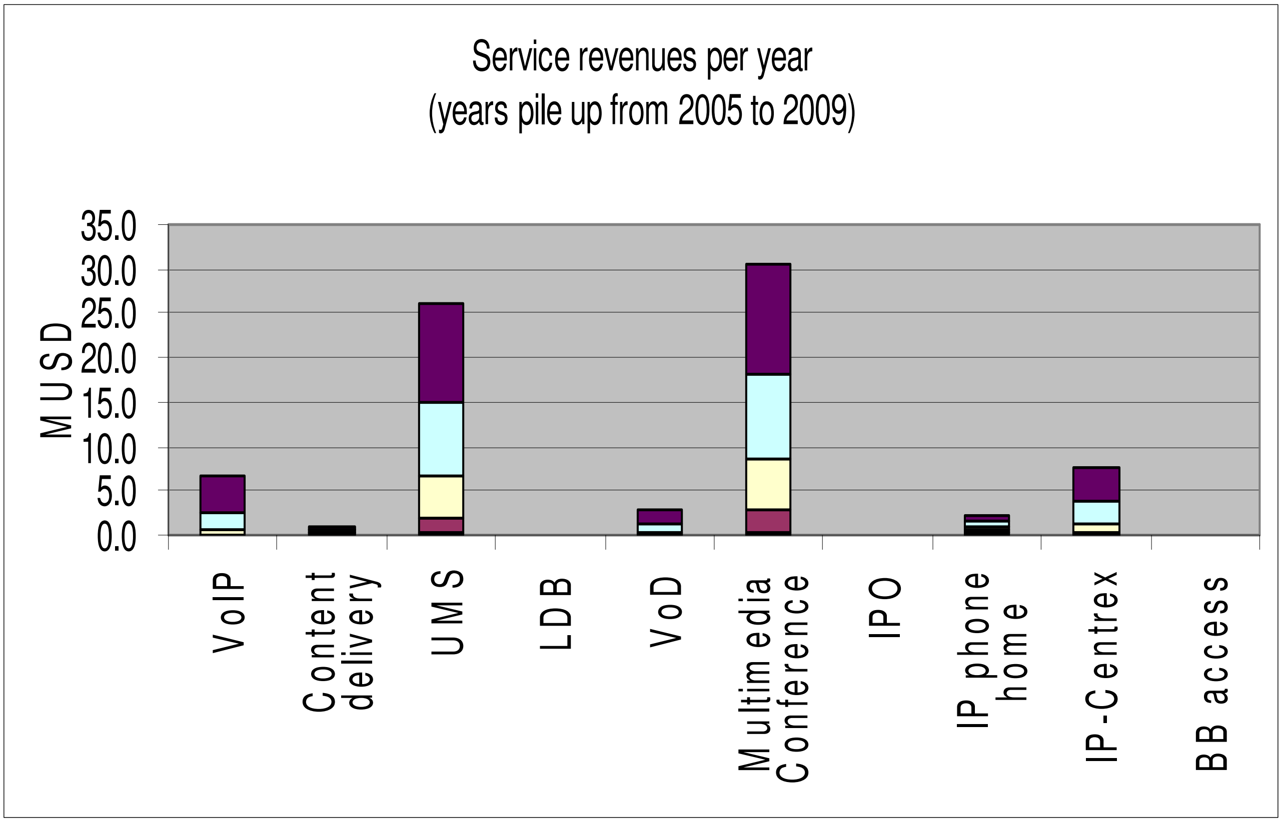

Figure 6.6 Services Cost per year (years pile up from 2005-2009)

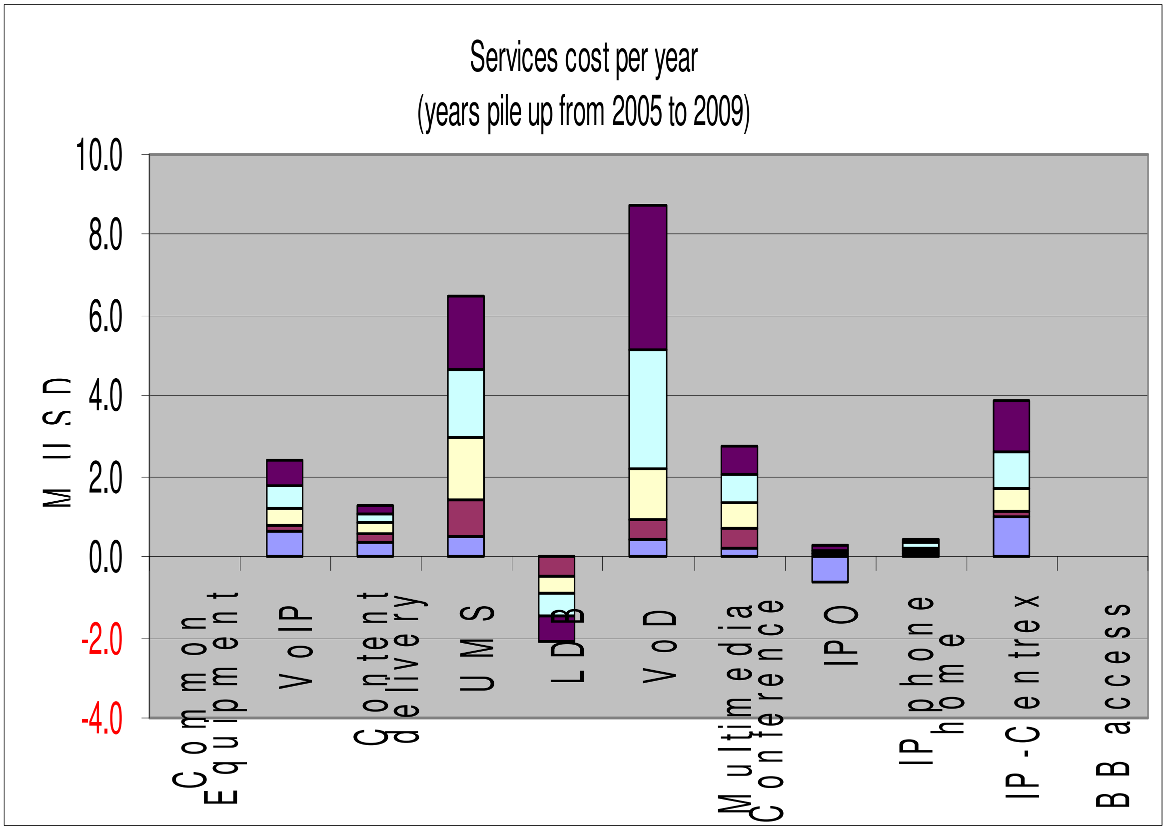

Figure 6.7 Total Service Revenues (service pile up) for the year 2005-2009

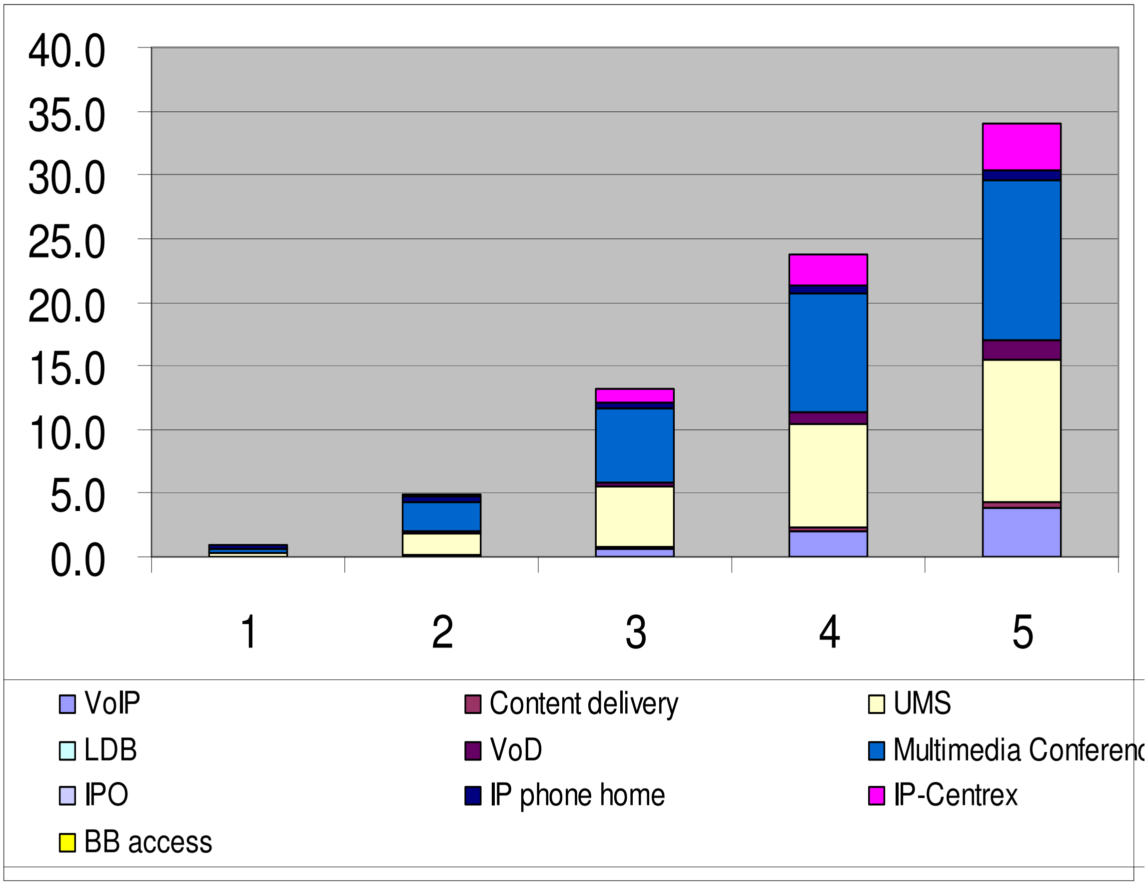

Figure 6.8 NPV and Pay Back values for Services

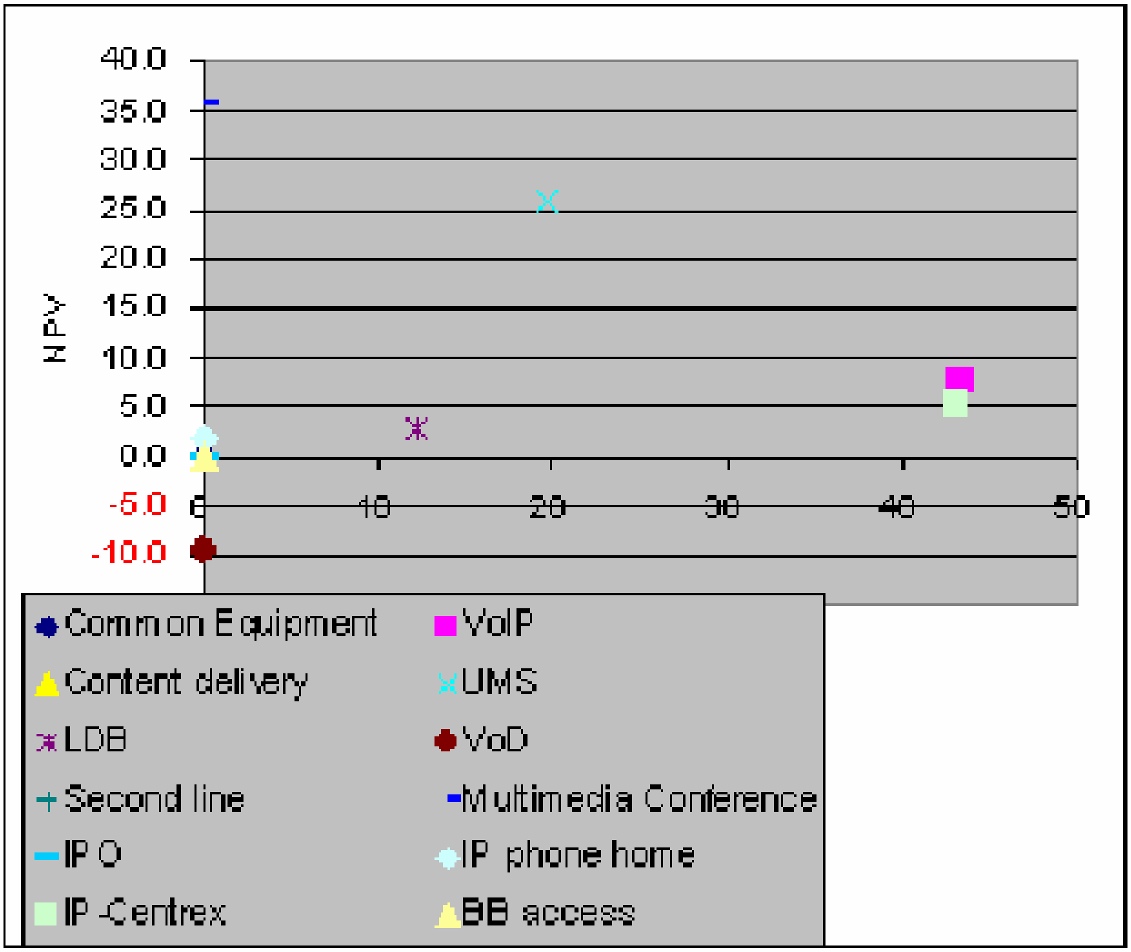

Figure 6.9   NPV and IRR Values for Services