DESIGN AND PERFORMANCE ANALYSIS OF A CORE NEXT GENERATION NETWORK FOR PSTN AND MOBILE COMMUNICATIONS IN BANGLADESH

A Thesis submitted

by

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in partial fulfillment of the requirements for the degree of

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DECLARATION

This is to certify that this thesis work is the outcome of the original work of the undersigned student. No part of this work has been submitted elsewhere partially or fully for the award of any other degree or diploma.

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APPROVAL

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ABSTRACT

The recent growth in data communications traffic placed significant strains on the capacity of traditional circuit-switched networks which could cause QoS problems for the basic voice services. Moreover, circuit switched networks limited the introduction of new services. It does not allow broadband services, high speed data communications. Under such circumstance, it becomes necessary to deploy NGN to support new generation services by using quality of service (QoS)-enabled transport technologies.

Capacity and Quality of Service (QoS) are two of the most important issues that need to be resolved before the commercial deployment of Voice over NGN. So, it is essential to determine the number of simultaneous users an IP based network can support simultaneously without significantly degrading the QoS. The capacity of NGN is highly dependent on the chosen speech codec and the QoS is partially dependent on the chosen codec and the other factors of quality such as delay, jitter and packet loss.

This thesis emphasizes the importance of converging network infrastructure to NGN. In this work, the performance of existing and proposed network is analyzed both theoretically and analytically. The effect of the quality factors (encoding mode, packet loss, jitter, throughput, and delay) on the available capacity, SNR, probability of error and QoS is analyzed and the estimated results are compared and verified by theoretical analysis.

Different possible strategies of migration are studied and the best strategy of migration at lowest cost and fastest speed is proposed.

VoIP traffic over NGN is simulated using Network Simulator to measure delay, jitter, throughput, packet loss contributing to QoS for varying number of nodes with different codecs (G.711 and G.729) and predicting the voice quality based on E-model. The simulation results are analyzed to evaluate the performance of VoIP over NGN. The voice quality and maximum simultaneous nodes supported by VoIP is estimated using the simulation results and verified by theoretical analysis.

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LIST OF IMPORTANT ABBREVIATIONS

3GPP	Third Generation Partnership Project
ACELP	Algebraic Code Excited Linear Prediction
ATM	Asynchronous Transfer Mode (ATM)
API	Application Programming Interface
ASM	Application Service Map
AMG	Access Media Gateway
AG	Access Gateway
ACL	Access Control List
BICC	Bearer Independent Call Control
BTCL	Bangladesh Telecommunication Company Ltd
BTRC	Bangladesh Telecommunication Regulatory Commission
BSC	Base Station Controller
CODECS	Coding and Decoding
CELP	Code Excited Linear Prediction coding
CS	Call Server
CS-ACELP	Conjugate Structure - Algebraic Code Excited Linear Prediction
CSN	Circuit Switched Network
CNG	Comfort Noise Generation
CAPEX	Capital Expenses
CS	Circuit Switched
C-BGF	Core Border Gateway Function
DLEs	Digital Local Exchanges
DSS1	Digital Subscriber Signal No.1
ECMP	Equal Cost Multipath Routing
E-Phone	Ethernet Phone
EC	Echo Cancellation
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
GSM	Global System for Mobile communications
GGSN	Gateway GPRS Support Node
ISDN	Integrated Services Digital Network
IAD	Integrated Access Device
IN	Intelligent Network
IVR	Interactive Voice Response
INAP	Intelligent Network Application Protocol
IPSS	International Packet Switched Service
IP	Internet Protocol
IMS	IP Multimedia Subsystem
IEEE	Institute of Electrical and Electronic Engineering
ICX	Interconnection Exchange

IGW	International Gateway
ITU	International Telecommunication Union
LAN	Local Area Network
MOS	Mean Opinion Score
MPLS	Multiprotocol Label Switching
MTA	Media Terminal Adapter
MRS	Media Resource Server
MGCP	Media Gateway Control Protocol
MAN	Metropolitan Area Network
M2UA	SS7 MTP2-User Adaptation Layer Protocol
M3UA	SS7 MTP3 User Adaptation Layer
MTP	Message Transport Protocol
MGC	Media Gateway Controller
MGCP	Media Gateway Control Protocol
MSC	Mobile Switching Center
NGN	Next Generation Network
NCS	Network-Based Call Signaling
NVR	Networked Virtual Reality
NAS	Network Access Server
NMS	Network Management System
NEs	Network Elements
OPEX	Operational Expenses
02	Outdegree 2
OSPF	Open Shortest Path First
OSS	Operation support system
PSN	Packet Switched Network
PCM	Pulse Code Modulation
PSTN	Public Switched Telephone Network
PBX	Private Branch Exchange
PNC	Public Network Computing
PS	Packet Switched
QoS	Quality of Service
RTP	Real-Time Transport
RTCP	RTP Control Protocol
RSVP	Resource Reservation Protocol
RACM	Resource and Admission Control Manager
RG	Residential Gateways
SPDM	Service Policy Decision Manager
SNR	Signal-to-Noise Ratio
SG	Signaling Gateway
SS7	Signaling System No. 7
SCTP	Stream Control Transmission Protocol
SCN	Switched Circuit Network
SIP	Session Initiation Protocol

SNMP	Simple Network Management Protocol
SBC	Session Border Controller
SCP	Service Control Point
SGSN	Serving GPRS support node
TRCM	Transport Resource Control Manager
TDM	Time Division Multiplexing
TCP	Transmission Control Protocol
TMG	Trunk Media Gateway
TG	Trunk Gateway
TDMA	Time Division Multiple Access (TDMA)
UDP	User Datagram Protocol
UMG	Universal Media Gateway
UMTS	Universal mobile telecommunications system
VoIP	Voice over IP
VAD	Voice Activity Detection
VPNs	Virtual Private Networks
VMGWs	Virtual Media Gateways
WLAN	Wireless LAN
xDSL	x Digital Subscriber Line

CHAPTER 1 INTRODUCTION

1.1 Introduction

Core network is the part of any network whose function is to carry out the signaling function and switching of a telecommunications system by connecting the appropriate incoming and outgoing lines. Core network employ two types of switching; Circuit Switching and Packet Switching. Switching describes the method by which the correspondents are connected. According to switching technology there are two types of core network; Circuit Switched Network and Packet Switched Network. Using the same access line, circuit and packet networks utilize different mechanism to provide services and to share resources.

The existing network of Bangladesh is TDM based Circuit switched network. Time Division Multiplexing (TDM) is a technique to combine digital signals into one signal. It can be used only for digital data. So, we need to convert the analog signals received in the end office (switching offices of the telephone co.s) from various local loops into digital signals by Codec and combine them into one signal that is transmitted on the digital trunk. This is done with the help of TDM.

The proposed NGN network involves digitization of voice streams and transmitting the digital voice as packets over conventional IP-based packet networks like the Internet, Local Area Network (LAN) or wireless LAN (WLAN). The goal of VoIP is to provide voice transmission over those networks. The digitalization process is composed of sampling, quantization and encoding. There are many encoding techniques that have been developed and standardized by the ITU such as G.711, G.729 and G.723.1. The encoded speech is then packetized into packets of equal size. Each such packet includes the headers at the various protocol layers such RTP 12 bytes, UDP 8 bytes, IP 20 bytes, Ethernet 26 bytes and the payload comprising the encoded speech for a certain duration depends on the codec deployed. As the voice packets are sent over IP networks, they incur variable delay and possibly loss.

The capacity of NGN is highly dependent on the chosen speech codec. As NGN Protocols specifies a series of audio codec ranging in bit rates from 5.3-64 kbps, increased capacity and revenue can be achieved. The QoS of NGN is partially dependent on the types of voice codec and the other factors as delay, jitter and packet loss.

Capacity and Quality of Service (QoS) are two of the most important issues that need to be resolved before the commercial deployment of Voice over NGN. Thus, it is necessary to examine if VoIP can provide a Quality of Service (QoS) comparable to that of the existing PSTN and cellular networks. Voice quality of NGN needs to at least be equal to the level ISDN telephony provides at the moment.

1.1.1 Circuit Switched Network (CSN)

A circuit-switched network [1] is a network in which there exists a dedicated connection as shown in Fig 1-1. A dedicated connection is a circuit or channel that is set up between two nodes so that they can communicate. In circuit switching, resources remain allocated during the full length of a communication even if no data is flowing on a circuit, hereby wasting link capacity when a circuit does not carry as much traffic as the allocation permits. . Generally, resources are frequency intervals in a Frequency Division Multiplexing (FDM) scheme or more recently time slots in a Time Division Multiplexing (TDM) scheme.

There are two basic types of circuit-switched networks: analog and digital. Pulse Code Modulation (PCM) scheme is used to convert the analog signal to digital format. On the receiving end of the connection, the digital signal must be decoded or converted back into an analog signal format.

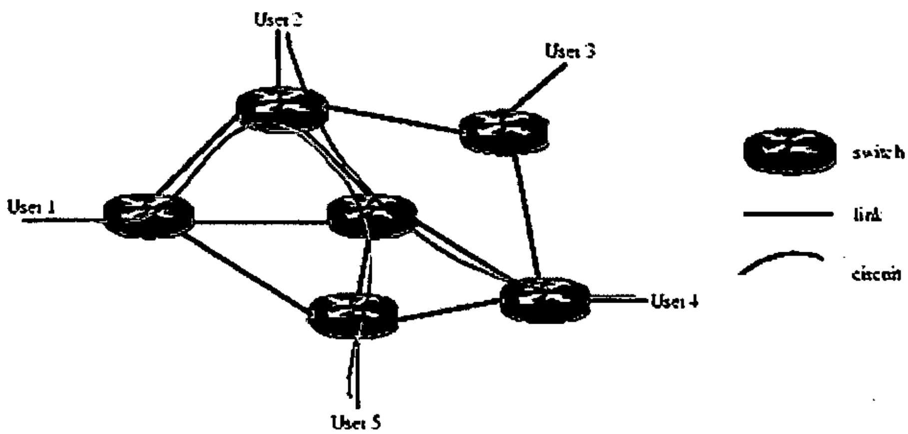


Fig 1-1: Circuit switched Network

Examples of circuit switched networks

- Public Switched Telephone Network (PSTN)
- ISDN B-channel
- Circuit Switched Data (CSD) and High-Speed Circuit-Switched Data (HSCSD) service in cellular systems such as GSM
- X.21 (Used in the German DATEX-L and Scandinavian DATEX circuit switched data network).

Advantages

Circuit switching has one big advantage over packet-switched networks. In a circuit-switched network when you use a circuit, you have the full circuit for the time that you are using the circuit without competition from other users.

Disadvantages

Circuit-switched networks have several disadvantages.

- Circuit-switched networks can be relatively inefficient, because bandwidth can be wasted.
- Another disadvantage to circuit-switched networks is that you have to provision for the maximum number of telephone calls that will be required for peak usage times and then pay for the use of the circuit or circuits to support the maximum number of calls.

1.1.2 Packet Switched Network (PSN)

Packet switching [1] is a technique that divides a data message into smaller units that are called packets as shown in Fig 1-2. Packets are sent to their destination by the best route available, and then they are reassembled at the receiving end. In packet-switch networks such as the Internet, packets are routed to their destination through the most expedient route. This almost guarantees that the packets will arrive at different times and out of order. Packet switching introduces the idea of cutting data on a flow into packets which are transmitted over a network without any resource being allocated. If no data is available at the sender at some point during a communication, then no packet is transmitted over the network and no resources are wasted.

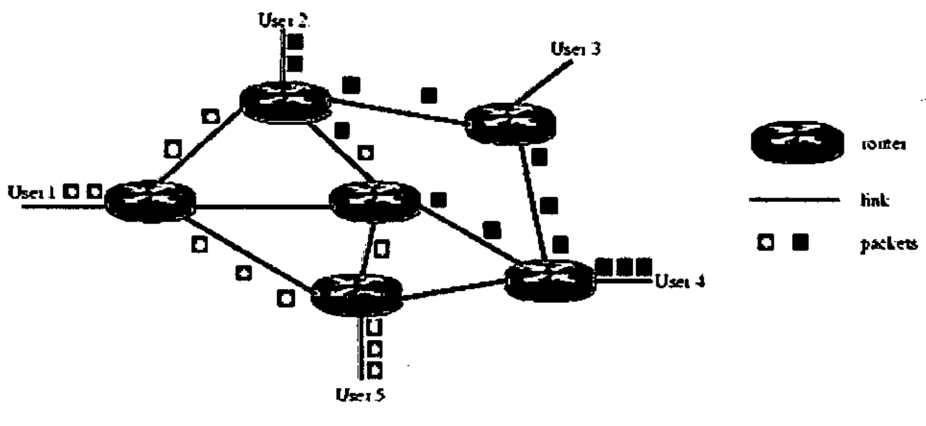


Fig 1-2: Datagram Packet Switching.

Examples of PSN

- The most well-known use of packet switching is the Internet and local area networks. The Internet uses the Internet protocol (IP) suite over a variety of data link layer protocols. For example, Ethernet and Frame relay are very common. Newer mobile phone technologies (e.g., GPRS, I-mode) also use packet switching.
- X.25 is a notable use of packet switching in that, despite being based on packet switching methods, it provided virtual circuits to the user. These virtual circuits carry variable-length packets. In 1978, X.25 was used to provide the first international and commercial packet switching network, the International Packet Switched Service (IPSS).

- Asynchronous Transfer Mode (ATM) also is a virtual circuit technology, which uses fixed-length cell relay connection oriented packet switching.
- Datagram packet switching is also called connectionless networking because no connections are established. Technologies such as Multiprotocol Label Switching (MPLS) and the Resource Reservation Protocol (RSVP) create virtual circuits on top of datagram networks. Virtual circuits are especially useful in building robust failover mechanisms and allocating bandwidth for delay-sensitive applications. MPLS and its predecessors, as well as ATM, have been called "fast packet" technologies. MPLS, indeed, has been called "ATM without cells".
- Modern routers, however, do not require these technologies to be able to forward variable-length packets at multigigabit speeds.

Advantages

- The outstanding advantage of the PSN is that as it shares the available bandwidth with all other network applications and makes more efficient use of the available bandwidth, the two correspondents can communicate at different rates, permitting much more efficient use of the communication channel.
- Packet-switched networking has made it possible for the Internet to exist and, at the same time, has made data networks—especially LAN-based IP networks—more available and widespread.

Disadvantages

- QoS may be affected due to packet loss, delay and jitter.

1.1.3 Next Generation Network (NGN)

Next Generation Network is a concept that has been introduced to take account of the new situation and changes in telecommunications fields. NGN can support new generation services by using quality of service (QoS)-enabled transport technologies and the huge amounts of bandwidth (tens of Mbps) they offer.

According to ITU-T definition of NGN in recommendation Y.2001 [2] "A Next Generation Network (NGN) is a packet-based network able to provide services including Telecommunications 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 is a common platform, which gives above services. Therefore, NGN is economic as compared with its services.

NGN is a service-oriented network. Through the separation of service and call control, as well as call control and bearer, the service-independent architecture is implemented, which makes services independent of network. Today, most services are tightly coupled with a specific transport network and signaling protocol, so regulation has been applied mainly in a vertical direction (e.g., regulation for service always also applies to the transport network). The separation between the service and transport layer changed with NGN to the horizontal direction, so there may be different regulations between services and transport networks. The separation of access capabilities with core transport capabilities of NGN has influenced changing business environments. The business of an access network provider domain dynamically expanded according to the various access technologies, and users may have much more freedom to choose access capabilities based on their specific requirements. Furthermore, another important aspect is to stimulate convergence between fixed and mobile communications. Thus, users may choose some fixed and some mobile access capabilities, and combine either, or both, with core transport capabilities, using single (or at least minimum) user subscription identification. [3]

Basic Characteristics of NGN

Basic Characteristics of NGN are [2, 4] as follows:

- All IP or packet-based networks (Migration from circuit based PSTN to packet based NGN)
- Separation of application services from the transport networks. NGN adopts the hierarchical architecture, which is divided into media access layer, transport layer, control layer and service/application layer.
- Open and Distributed Network Structure
- Converged or integrated broadband networks
- Ubiquitous network
- Independent network control layer
- Diversified Access Modes
- Internetworking and gateways
- NGN is based on standard protocols

Advantages of NGN

NGN makes use of best of both the worlds (flexibility, efficiency & Innovativeness of IP and QOS, Security, Reliability, Customer-friendly features of Proven PSTN) [5].

- Simplified network management.
- Reduced cost of voice calls.
- Improved network scalability to accommodate business growth.
- Better customer service.
- Expanded number of access types.
- Increased capacity and revenue.
- Save transmission
- Save power

- Better network redundancy.
- Better ability to launch and manage new applications and services.

1.2 Background

In Bangladesh, the whole network was TDM based circuit switched network. Circuit-switched networks (CSN) were initially deployed to handle only voice communications and were not designed to handle data efficiently and cannot scale cost-effectively to accommodate the growth in data traffic. It does not allow broadband services, high speed data communications.

The recent growth in data communications traffic placed significant strains on the capacity of traditional circuit-switched networks [1]. Even without the data traffic, the growth in subscriber-generated minutes of network use has caused mobile networks to grow beyond initial design standards. Even if the equipment can be upgraded to support new data standards, it still suffers from legacy issues that are embedded within the equipment design. It increased costs for both inter-connection facilities and for routine support and administration [6].

Internet Telephony first broke into public view in 1996. The migration has started in the mid 1990s, by installing gateways into the public telephone network. This allowed them to offer low-priced overseas calls in the traditional telephone networks. The possibility of combining data, voice and multimedia services into one facility assures huge benefits in added value to companies and users deploying the service of VoIP. But, the quality of IP telephony at that time was not high. The PSTN is the ultimate benchmark for voice quality in communication networks. When the first VoIP solution was introduced in 1996, the Internet's audio transport capabilities were far below the benchmark set by public network standards and therefore much lower performing than traditional telephony. [7]

These practical network issues, along with the requirement to support future data traffic, have caused Service Providers to begin to look at alternative network structures. The concept of a new, integrated broadband network has developed over the last few years and has been labeled next-generation network (NGN). NGN is a service oriented packet based network supporting new generation services by using quality of service (QoS)-enabled transport technologies. As NGN Protocols specifies a series of audio codec ranging in bit rates from 5.3-64 kbps, increased capacity and revenue can be achieved.

One of the main issues related to NGNs, which has been the focus of several works and still require further research and development, is the end-to-end QoS guarantee. Although the quality of VoIP does not yet match the quality of a circuit-switched telephone network, there is an abundance of activity in developing protocols, speech encoders and optimization services for the implementation of the high quality voice service. For instance, the NGN QoS signaling and control solutions are still at the development stages. PacketCable, proposed a QoS solution, as outlined in [8], which focuses mainly on some precise problems related to packet-based cable access networks. Similarly, the 3GPP also proposed an end-to-end QoS solution for the 3rd generation mobile networks, as described in [9]. It was developed for specific category of networks and lacks several functionalities that can be deemed necessary for a standard NGN QoS model.

One of the problems to evolve NGN are there are many legacy equipments which we can not replace at once and various QoS differential service among networks. To address this, there should be proper admission control mechanism to protect core network and support end-to-end QoS in access network. While Planning-based admission is simple but not efficient, and probed-based admission control is efficient but overhead in network, Egress resource prediction-based admission control is efficient and less overhead compared to the previous mechanisms. An efficient end-to-end QoS mechanism using egress node resource prediction via probe method in NGN has been proposed in [10].

A migration strategy integrating new entities for guaranteeing QoS has been proposed in [11]. In the strategy, the call management and gate control functionalities are sub-divided into Call Server (CS) and Resource and Admission Control Manager (RACM). This subdivision allows for the application of call admission control (CAC), achieving reliable and accurate resource management and improving the transport network scalability and resilience. The RACM is further decomposed into distinct Service Policy Decision Manager (SPDM) and Transport Resource Control Manager (TRCM) entities in order to enable the coverage of large domains. Finally, the necessary interfaces are defined and the corresponding open and mature standards are specified.

High availability and even distribution of traffic over the network are a prerequisite for the economical provision of QoS services. A novel routing approach has been proposed in [12], meeting both the above requirements and providing an order of magnitude improvement in availability. The proposed 02 routing appears to be very similar to equal cost multipath (ECMP) routing, which also requires additional entries in the routing table, although the implementation of unequal distribution weights may be slightly more complex. The information required to compute the routing is fully contained in the information needed for link state routing anyway. Therefore, an well-known link state routing protocol like OSPF can be used as the basis for 02 routing simply by replacing the routing algorithm. The paper also suggests a simple method to optimize distribution weights using information about the anticipated (or measured) traffic demand.

A model for the end-to-end negotiation and dynamic adaptation of QoS parameters for Networked Virtual Reality (NVR) services, addressing the heterogeneous NGN environment has been proposed in [13]. To identify common generic functionality related to session-level QoS signaling for advanced multimedia applications and to design a high-level application programming interface (API), which invokes this functionality, the paper[14] describes the Dynamic Service Adaptation model on which the proposed API is based. The IP multimedia subsystem (IMS) has been recognized as a reference next-generation network architecture for offering multimedia services over an Internet Protocol

(IP)-based infrastructure. To this extent, the approach aims to further enhance the IMS objectives of providing users with customized and enhanced service quality has been described in [14].

The paper [15] presents two QoS class mapping methodologies over heterogenous networks to provide end-to-end QoS support for application services and proposed a QoS class mapping method in different transport techonologies mixed circumstance using Application Service Map (ASM) which classifies application services based on performance requirements.

The QoS control [16] can be implemented either in the centralized or distributed model. In the centralized model, the call setup signaling and resource control functions are implemented in the control/signaling servers while distributed model implements the same functions in the transport equipment. The paper [17] identifies the three major parts of QoS processing – signaling, resource control, and policy setup and the control procedure of both models and provides a performance comparison on both models. Based on the analysis, the performance of distributed model is better when traffic load is high and traffic is uniformly distributed. When the majority of calls are initiated or destined to one edge node, centralized model has better performance.

The common reasons that affect the voice quality in the IP packet network include: delay, jitter, packet loss and echo. To reduce the effect of delay, jitter, packet loss and echo, Huawei proposed [18] advanced voice quality assurance technologies in the UMG. The third Generation Partnership Project (3GPP) introduces IMS on the basis of the packet domain. Technical problems, however, continue to affect the IMS that is applied on the IP network. The technical problems include: Network security, NAT traversal, QoS assurance of packets and media streams. Huawei proposed [19] the solution using Session border controller (SBC), serving as a gateway, provides a channel for all incoming or outgoing information streams or media streams which can solve the above three problems.

Previously, work has been done on proper admission control mechanism, migration strategy integrating new entities for guaranteeing QoS, High availability and even distribution of traffic, end-to-end negotiation and dynamic adaptation of QoS parameters, session-level QoS signaling for advanced multimedia applications and QoS control mechanisms to support end-to-end QoS guarantee. For instance, the NGN QoS signaling and control solutions are still at the development stages. Still, the quality of VoIP does not yet match the quality of a circuit-switched telephone network. Thus the advanced voice quality assurance technologies in the universal media gateway should be adopted to reduce the effect of delay, jitter, packet loss and echo.

Capacity and Quality of Service (QoS) are two of the most important issues that need to be resolved before the commercial deployment of VoIP. As VoIP technology is still in the early stages of commercial deployment, it is necessary to examine if VoIP can provide a Quality of Service (QoS) comparable to that of the existing PSTN and cellular networks.

Thus factors affecting the capacity and QoS should be analyzed to determine the number of simultaneous users supported by the proposed NGN simultaneously without significantly degrading the QoS.

1.3 Motivation

The Challenges Faced by the Quality of IP Network Services is as follows:

1. Operators Enter All-service Operation Age

In all-service age, the operator will bear various services on the same network, such as data, voice, video, and VIP customer. Various services have different requirements for the network quality. If the services are superimposed, the result must be 1+1<2.

2. Poor Service Quality Leads to Decrease of Customers

- The average performance of the traditional IP network is better, but the transient feature is poor. However, new services are very sensitive to the network index, so they cannot be directly deployed on the traditional network.
- When the network is busy, the number of the on-line users increases greatly, so the network speed decreases. There is no system to guarantee high-level users and value service QoS. The high-end users are provided with poor quality services.

3. Investment ≠ Income

- The number of users increases fast, and each user requires greater bandwidth. The investment of operators in bandwidth increases, but the income does not increase. The P2P traffic occupies 60% of the bandwidth. The high-end users spend more money, but the services still cannot be ensured by QoS.
- High-value services do not obtain corresponding resources with the investment of operators in bandwidth. The quality of services is not perfected. Free services occupy a great number of network resources.

Capacity and Quality of Service (QoS) are two of the most important issues that need to be resolved before the commercial deployment of Voice over NGN. As VoIP technology is still in the early stages of commercial deployment, it is necessary to examine if VoIP can provide a Quality of Service (QoS) comparable to that of the existing PSTN and cellular networks. The capacity of NGN is highly dependent on the chosen speech codec. As NGN Protocols specifies a series of audio codec ranging in bit rates from 5.3-64 kbps, increased capacity and revenue can be achieved [9]. The QoS of NGN is partially dependent on the types of voice codec and the other factors as delay, jitter and packet loss. So, it is essential to analyze the delay, jitter and packet loss of VoIP and determine the number of simultaneous users the proposed NGN can support simultaneously without significantly degrading the QoS. Once NGN are in service, most traditional communication networks will become obsolete as they are. Instead of wasting this huge volume of the expensive and functional resources of traditional networks, it is more efficient and economical to devise ways to transform them into NGN compatible systems so that they continue to be functional and viable.

1.4 Objective

The thesis will emphasize the importance of converging network infrastructure to NGN. The objective of the thesis is as follows:

1. Existing and proposed Network analysis:
   Performance of the existing and proposed network will be analyzed both theoretically and analytically to determine the effects of different codec and other quality factors on the service performance of existing and proposed network and to compare that.
2. Best strategy of NGN migration:
   To design migration strategy aims for assisting established operators to capitalize on their existing resources and migrate smoothly towards converged NGN architecture while satisfying requirements in terms of QoS guarantees.
3. Simulation and results:
   Voice traffic will be simulated to measure delay, jitter, throughput, packet loss and analyze the measurement results and their effects on the voice quality and to determine the number of simultaneous users an IP based network can support simultaneously without significantly degrading the QoS.

1.5 Scope of Work

This thesis is aimed at stimulating discussions and identifying issues related to NGN. The thesis begins with a general introduction to the exiting and proposed network in chapter 1. In chapter 2, next generation network overview will be discussed. In chapter 3, the effect of different codec and other quality factors (packet loss, jitter, and delay) on the service performance (capacity, SNR, probability of error and QoS) of existing and proposed network will be analyzed both theoretically and analytically. The analytical results will be compared and verified by theoretical analysis. In chapter 4, Different possible strategies of migration will be studied. The best strategy of migration at lowest cost and fastest speed will be designed. In this work, we propose a migration strategy, which takes several factors into consideration. In particular, the proposed migration strategy aims for assisting established operators to capitalize on their existing resources and migrate smoothly towards converged NGN architecture. Most importantly, the desired upgrade is achieved while satisfying requirements in terms of QoS guarantees. In chapter 5, VoIP traffic over NGN is simulated using Network Simulator to measure delay, jitter, throughput, packet loss contributing to QoS for varying number of nodes with different codecs (G.711 and G.729) and predicting the voice quality based on E-model. The simulation results are analyzed to evaluate the performance of VoIP over NGN. The voice quality and maximum simultaneous nodes supported by VoIP is estimated using the simulation results and verified by theoretical analysis. In Chapter 6, the thesis will be concluded with suggestion and future work.

CHAPTER 2 NGN OVERVIEW

2.1 Architecture of NGN

NGN is a service-oriented network. It provides a rather independent service system by splitting the service module from the call control, and the call control from the bearer. In this way, it frees the service from the original network model. NGN employs open and integrated network structure. With abundant service models, NGN is able to provide a variety of services, such as voice, data and multimedia services, or integrated services. The whole network transmits and switches service data in IP packet mode. NGN comprises four planes [20]:

- Edge access
- Core switching
- Network control
- Service management

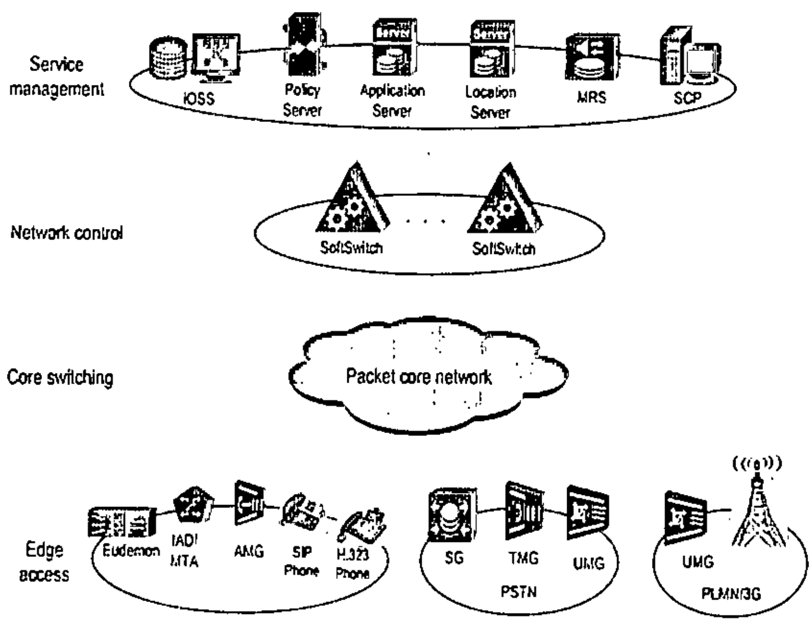


Fig 2-1: Architecture of NGN

Fig 2-1 illustrates a layered architecture bringing intelligence at every layer. Along with traditional voice and data equipment, the NGN architecture contains converged network equipment types such as Session/Call Controllers (*e.g.* IP Multimedia Subsystem (IMS) or

Call Agent/SIP Server), Media Gateways, Signaling Gateways, Feature Servers, Application Servers, Media Servers and Management Servers, Provisioning and Billing Interfaces. Core technologies include packet transmission technology, traffic engineering control protocol, technology which guarantees quality of service such as MPLS (Multi Protocol Label Switching), multi-party telecommunications technology such as real-time multicasting, session control technology, etc.

2.2 Major network components of NGN

2.2.1 Edge Access Layer

The edge access layer connects subscribers and terminals to the network through a number of styles, and converts the format of information so that it can be sent over the network. A gateway is a network point that acts as an entrance to another network.

For interworking with other networks, there are three types of gateways, that is, Trunking Gateway, Access Gateway and Signaling Gateway, which are located in the service provider's network.

Here are the components of the edge access layer: [20]

- NGN Terminal equipments

  The NGN terminal equipment includes E-Phone, IAD, and video terminals. The E-Phone and the IAD serve as a small AG, connecting one or multiple users. The video terminal interacts with the softswitch through H.323 or SIP, processing video and voice calls.

  - Integrated Access Device (IAD): It is a device used to access subscribers in the NGN. It converts the format of the subscriber data such as audio, and video so that it can be sent over the IP network. An IAD provides as many as 48 subscriber ports.
  - Media Terminal Adapter (MTA): It is an access device working in Network-Based Call Signaling (NCS) (developed and extended from MGCP). It accesses subscriber data, voice, and video through cable network to the IP network.
  - SIP Phone: It is a multimedia device working in the Session Initiation Protocol (SIP).
  - H.323 Phone: It is a multimedia device working in the H.323 protocol.

- Access Media Gateway (AMG): The AG provides the narrowband and broadband service access function. It provides subscribers with a diversity of service access, such as analog subscriber access, Integrated Services Digital Network (ISDN) subscriber access, V5 subscriber access and x Digital Subscriber Line (xDSL) access.

The AG transfers subscriber line data such as voice, modem, and fax over the NGN core network through conversion of media streams.

The AG interacts with the softswitch through H.248 or MGCP. Controlled by the softswitch, the AG reports the status of the subscriber line and completes call processing.

- Signaling Gateway (SG): It connects the Signaling System No. 7 (SS7) network with the IP network. It converts the signaling between the PSTN (SS7) and the IP network (packet signaling). The converts the PSTN signaling between the TDM bearer mode and the IP packet mode.

  The SG can be an independent device or embedded in the TG.

- Trunk Media Gateway (TMG): It resides between the circuit switched (CS) network and the IP network. It converts format between pulse code modulation (PCM) signal flow and IP media flow.

  The TG interacts with the softswitch through H.248 or MGCP. Controlled by the softswitch, the TG sets up and disconnects calls and implements other services.

- Universal Media Gateway (UMG): It converts the media stream and signaling between different formats. UMG8900 can be used as various service gateways of the access layer like Trunk Gateway (TG), Access Gateway (AG), NGN enabled switch, and extended integration of fixed network and mobile network services. It can connect many types of devices, including PSTN exchange, private branch exchange (PBX), access network devices, network access server (NAS), and base station controller (BSC).

- Eudemon: It is an IP gateway which is often used at the portal of residential network and enterprise network, or at the convergence layer of metropolitan area network (MAN). Eudemon can be used as a status firewall, or as a gateway. Eudemon provides traversal of private network and QoS.

2.2.2 Core Switching Layer

The core switching layer works in the packet switching technology, and is composed of devices [20] distributed in the backbone network and the MAN, such as routers and layer 3 switches. It provides subscribers with a common and integrated platform of data transport, ensuring a high reliability, Quality of Service (QoS) assurance and a large capacity.

2.2.3 Network Control Layer

The network control layer provides the call control feature. Its core technology is software switching or soft switching, which is used to achieve primary real-time call control and connection control functions.

The softswitch is the core equipment in NGN. It provides the following functions and features:

- Call control
- Media gateway access control
- Resource allocation
- Protocol processing
- Routing
- Authentication
- Charging
- Application Programming Interfaces (API)

The softswitch provides basic voice services, mobile services, and multimedia services. The softswitch serves as the call processing center in NGN. It supports narrowband and broadband call control signaling such as Signaling System Number 7 (SS7), Digital Subscriber Signal No.1 (DSS1), V5, H.323, and Session Initiation Protocol (SIP).

The softswitch controls gateway devices such as the trunk gateway (TG), access gateway (AG), integrated access device (IAD), Ethernet phone (E-Phone), and media resource server (MRS) through H.248 and Media Gateway Control Protocol (MGCP) for related call service processing.

The softswitch connect with the application servers at the service management layer through standard and open interfaces, enabling fast and flexible access for new services [20].

2.2.4 Service Management Layer

The service management layer provides value added services and operation support.
Here are the components of the service management layer.

- iOSS (means integrated operation support system): It includes two parts; a network management system (NMS) and an integrated charging system. The network management system (NMS) manages and maintains the whole network centrally, and monitors status of all devices in the network. The NMS manages network elements (NEs) through the standard network management interfaces such as the Simple Network Management Protocol (SNMP). The operation support system (OSS) is responsible for service release, charging, and subscriber line test management. It usually works with the NMS to implement its functions.

- Service node:

    - Policy Server: It manages the policies of the subscribers, such as access control list (ACL), bandwidth, traffic, and QoS.
    - Application Server: It produces and manages logics of value added services and intelligent network (IN) services, providing a platform for a third party to develop services through open APIs. The application server is the result of the separation of service from call control. It helps develop supplementary services.

- Location Server: It dynamically manages the routes between softswitches; indicates the reach ability of the destinations of calls; ensures the efficiency of call routing; prevents the routing table from being oversized and impractical; and simplifies the routing.
- Media Resource Server (MRS): It processes media streams in the basic and enhanced services. It provides functions of service tone playing, conference service, interactive voice response (IVR), recorded announcement and advanced tone service, digit collecting, recording, and audio mixing. Its function is equal to the intelligent peripheral (IP) in the traditional IN. Controlled by the softswitch through H.248, MGCP, or SIP, the MRS provides multiple intelligent services based on voice or video [10].
- Service Control Point (SCP): It is the core component in the traditional IN, which is used to store subscriber data and service logics. The SCP starts a service logic based on the call events reported from the Service Switching Point (SSP), and queries the service database and the subscriber database using the started service logic. It then sends proper call control instructions to the SSP on the next action, thus realizing various intelligent calls. That is the main function of SCP.

The traditional IN device and the softswitch implement traditional intelligent services by the Intelligent Network Application Protocol (INAP). These intelligent services are used in the initial establishment of the NGN and can inherit the traditional intelligent service resources. The App Server is an application server under the NGN architecture. It interacts with the softswitch through SIP. It also provides open third-party interfaces for external use. This open service providing mode helps to introduce third-party services [20].

2.3 Interface and Protocol of NGN

An interface is the connection point between two adjacent network entities, and a protocol specifies the principles to be followed for information interchanging over such connection points (interfaces). Different protocols are usually used on different interfaces and maybe on the same interface as well. [20]

The Fig 2-2 illustrates the interfaces and their corresponding protocols of NGN.

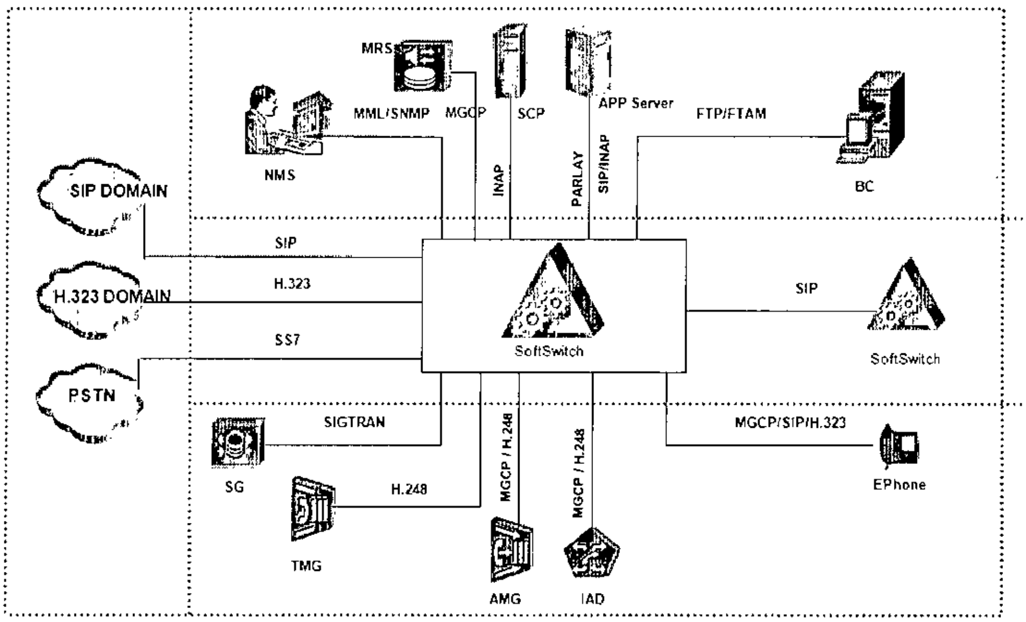


Fig 2-2: NGN using Standard Protocol

Table 2-1 represents the corresponding Relations between Soft Switch Interfaces and Protocols.

Fig 2-3 illustrates the correspondence between the protocols and the OSI model.

Table 2-1 Corresponding Relations between Soft Switch Interfaces and Protocols

Interface	Application Protocols	Signaling Transmission Protocols
Soft Switch-SG	SIGTRAN	M3UA/M2UA/SCTP/IP
SoftSwitch-TMG	H.248	UDP/IP、SCTP/IP、TCP/IP
SoftSwitch-AMG	MGCP、H.248	UDP/IP
SoftSwitch-IAD	MGCP、H.248	UDP/IP
SoftSwitch-MRS	MGCP	UDP/IP
SoftSwitch-Terminal	MGCP	UDP/IP
	SIP	UDP/IP
	H.323	UDP/IP、TCP/IP
SoftSwitch-SoftSwitch	SIP	UDP/IP
SoftSwitch-PSTN/ISDN	SS7	MTP
SoftSwitch-SIP	SIP	UDP/IP
SoftSwitch-H.323	H.323	UDP/IP、TCP/IP
SoftSwitch-NMS	MML or SNMP	-
SoftSwitch-BC	FTP or FTAM	TCP/IP

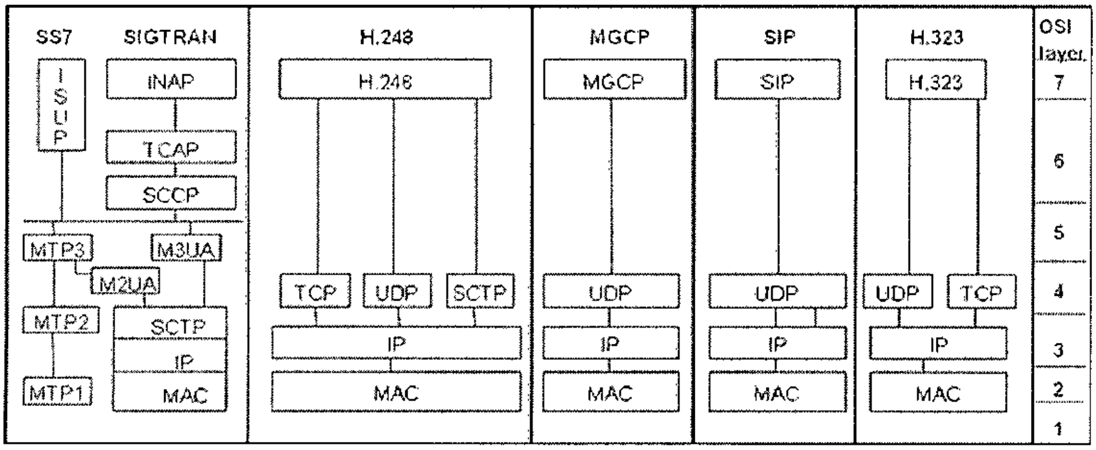
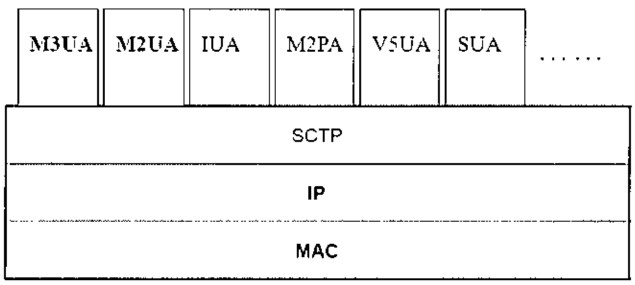


Fig 2-3: Correspondence between the Protocols and the OSI model.

2.3.1 SIGTRAN Protocol

As we can see in Fig 2-4, SIGTRAN is a protocol stack rather than a protocol. It includes transmission protocol, (SCTP), adaptation protocols (M2UA and M3UA). It supports transmission of SCN (Switched Circuit Network) signaling via IP network. [20]



M3UA: MTP3 User Adaptation layer	M2UA: MTP2 User Adaptation layer	IUA: ISDN Q.921 User Adaptation layer
M2PA: MTP2 Peer Adaptation layer	V5UA: V5 User Adaptation layer	SUA: SCCP User Adaptation layer

Fig 2-4: SIGTRAN Protocol Stack Structure

2.3.2 SIP Protocol

The Session Initiation Protocol (SIP) is an application-layer control protocol used to establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, remote education, Internet telephony and similar applications. SIP can be used to initiate sessions as well as inviting members to sessions that have been advertised and established by other means. SIP transparently supports name mapping and redirection services, allowing the implementation of ISDN, IN, and subscriber mobile services. The protocol stack of SIP has been shown in Fig 2-5. [20, 22]

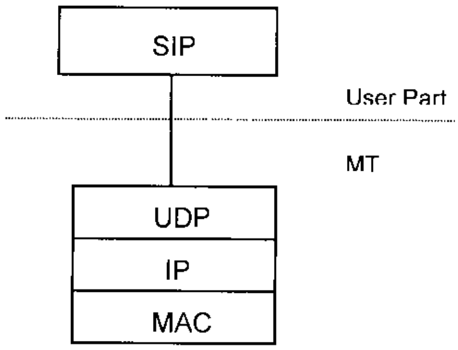


Fig 2-5: SIP Protocol Stack

The NGN Soft-Switch system interconnects with other Soft-Switch systems and SIP domain devices via SIP signaling, achieving the call control functions between them.

Support of SIP to Multimedia

SIP is a lightweight signaling protocol, which can be used in conjunction with other call setup and signaling protocols. SIP does not offer conference control services such as site control or voting and does not prescribe how a conference is to be managed, but SIP can be used to introduce conference control protocols. SIP does not allocate multicast addresses. SIP can invite users to sessions with and without resource reservation. SIP does not reserve resources itself, but can convey to the invited system the information necessary to do this. SIP supports five facets of establishing and terminating multimedia communications:

User location: Determination of the end system to be used for communication.
User capabilities: Determination of the media and media parameters to be used.
User availability: Determination of the willingness of the called party to engage in communications.
Call setup: "Ringing", establishment of call parameters at both called and calling party.
Call control: establish, modify and terminate multimedia sessions or calls.

2.3.3 H.323 Protocol

H.323 is a communication control protocol put forward by ITU [23]. It provides multimedia communications services over Packet Based Networks (PBNs). Call control is one of the major parts of H.323 and can be used to establish point-to-point media conference and multipoint media conference.

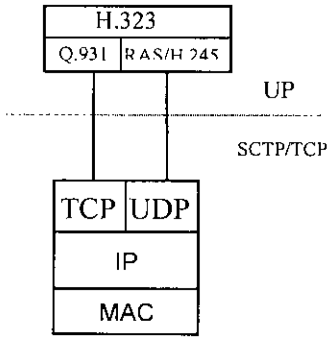


Fig 2-6: H323 Protocol stack

H.323 defines entities such as H.323 Gateways, Gatekeepers, Multipoint Controllers, Multipoint Processors and Multipoint Control Units. H.323 Gateways are between Switched Circuit Network (SCN) and packet switched network. Gatekeepers provide access control and address translation services. Multipoint Controllers (MCs) provide the multipoint control function for the multiparty conference. Multipoint Processors (MPs) enables mixing of multipoint media streams. H.323 itself is a protocol set, including such protocols as RAS, Q.931 and H.245 as shown in Fig 2-6. The RAS message is used for registration, admission and status inquiry between GW and GK. Q.931 is the ISDN user-network interface layer 3 specifications for basic call control by ITU-T. Q.931 protocol is used for call control. The functions of H.245 protocol include capability negotiation, master-slave control and opening and closing of logical channel One H.323 protocol session is accomplished through three protocols: RAS first, then Q.931, and finally H.245. RAS is transmitted on UDP, Q.931 is transmitted on TCP and H.245 is transmitted on UDP. [20]

2.3.4 Bearer Control Protocols

Bearer control protocols are used for the communication between a Media Gateway Controller (**MGC**) and a Media Gateway (**MG**) as shown in Fig 2-7. As control-layer equipment, Softswitch supports two bearer control protocols:

- MGCP/H.248.

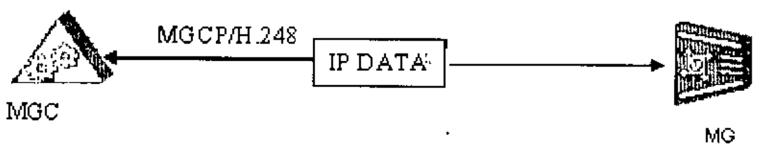


Fig 2-7: MGCP/H248 Implementation in Soft-Switch

MGCP (Media Gateway Control Protocol)

Media Gateway Control Protocol assumes a call control structure in which the call control functions are gateway-independent and are processed by the external call control unit. It is a master/slave protocol. The gateway needs to execute the commands sent from the Media gateway controller. [20]

Terms of MGCP

Gateway: A network element that provides interconnection and inter-working between networks of various architectures. In NGN architecture, NGN inter-works with other networks via certain gateways.

Trunk Media Gateways (TMG): It provides the interfaces between the traditional telephone network (PSTN) and a Voice over IP (VoIP) network.

Access Media Gateways (AMG): It provides a traditional analog subscriber line interface or a digital PBX interface to a Voice over IP network.

Residential Gateways (RG): It is an entity that provides traditional analog (RJ11) interfaces to VoIP network. Examples of residential gateways include cable modem/cable set-top boxes, xDSL devices, and broadband wireless devices.

Call Agent: Handles the signaling and call processing functions, and it is external call control element controlling Telephony Gateways. Soft-Switch system provides MGCP call agent functions. SoftSwitch can act as the access point for MGCP E-phones and soft phones in the network.

End point: It refers to the originating end or receiving end of data. It can be a physical concept or a virtual concept.

Media Resource Server (MRS): It is a type of gateway that supports endpoint types such as announcement server access point, interactive voice response access point, conference bridge access point, etc.

H.248

H.248 refers to the same kind of protocol. It is an achievement from the efforts of both ITU [24] and IETF. ITU-T by IETF names it H.248. H.248 comes into being based on MGCP and is combined with features of other media gateway control associated protocols. The function structure of H.248 is similar to that of MGCP. In NGN, both H.248 and MGCP can be used between Soft-Switch and most components. MGCP is deficient in its descriptive capability, which restricts its applications in large gateways. For those large-scaled gateways, H.248 is a much better choice. MGCP message transportation depends on UDP packets over IP network, and H.248 signaling messages may be based on multiple bearers such as UDP/TCP/SCTP. [20]

Terms of H.248

Media Gateway (MG): An MG converts media provided in one type of network to the format required in another type of network.

Media Gateway Controller (MGC): It controls the call state pertaining to connection control of media channels of MG.

Termination: A Termination is a logical entity on an MG, capable of sending and/or receiving one or more streams. A Termination is described by a number of characterizing properties, which are grouped in a set of descriptors included in commands. One termination belongs to one and only one context at any time.

Context: A context is the association among terminations. It describes topology relationships among terminations and media-mixed/switched parameters.

MGCP and H.2 48

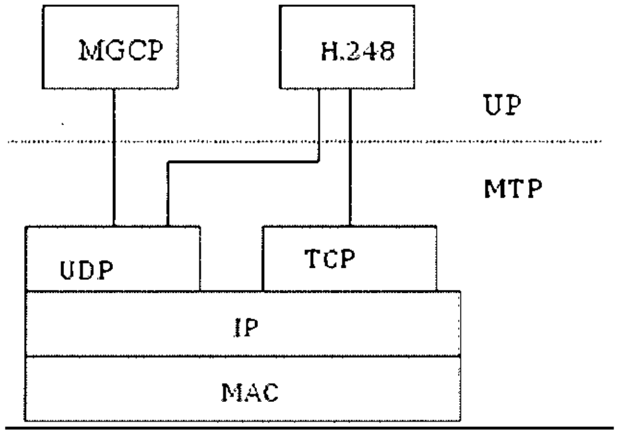


Fig 2-8: MGCP and H.2 48

Fig 2-8 illustrates the correspondence between the Protocols and the OSI model.

2.3.5 Bearer Independent Call Control (BICC)

Bearer Independent Call Control (BICC) is a signaling protocol based on N-ISUP that is used to support NB-ISDN service over a BB backbone network without interfering with interfaces to the existing network and end-to-end services. Specified by the ITU-T in recommendation Q.1901 [25], BICC was designed to be fully compatible with existing networks and any system capable of carrying voice messages. BICC supports narrowband ISDN services independently of bearer and signaling message transport technology.

ISUP messages carry both call control and bearer control information, identifying the physical bearer circuit by a Circuit Identification Code (CIC). However, CIC is specific to time-division multiplexed TDM networks. BICC was developed to be interoperable with any type of bearer, such as those based on asynchronous transfer mode ATM and IP technologies, as well as TDM.

BICC separates call control and bearer connection control, transporting BICC signaling independently of bearer control signaling. The actual bearer transport used is transparent to the BICC signaling protocol - BICC has no knowledge of the specific bearer technology.

The ITU announced the completion of the second set of BICC protocols (BICC Capability Set 2, or CS 2) in July 2001; these are expected to help move networks from the current model - which is based on public-switching systems - to a server-based model. The BICC deployment architecture comprises a proxy server and a media gateway to support the current services over networks based on circuit-switched, ATM, and IP technologies, including third-generation wireless.

The completion of the BICC protocols is a real and important ITU step toward broadband multimedia networks, because it will enable the seamless of circuit-switched TDM networks to high-capacity broadband multimedia networks. The 3GPP has included BICC CS 2 in the UMTS release 4. Among the future ITU-T plans for BICC are the inclusion of more advanced service support and more utilization of proxies, such as the SIP proxy.

2.3.6 VoIP Protocol Stacks

Fig 2-9 shows the basic IP network protocol stack used to implement VoIP. In order for the internet to provide useful services, Internet telephony required a set of control protocols for connection establishment, capabilities exchange as well as conference control.

H.323/SIP
RTP, RTCP, RSVP
UDP, TCP
Network Layer (IPv4, IPv6)
Data Link Layer
Physical Layer

Fig 2-9: VoIP protocol stack

H.323 and SIP is standard that specifies the components, protocols and procedures that provide multimedia communication services such as real-time audio, video, and data communications over packet networks, including Internet Protocol (IP) based networks. Real-Time Transport (RTP) protocol provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast networks. The RTP control protocol (RTCP) is used to monitor the quality of real-time services and to convey information about participants in an on-going session. There are components called monitors, which receive RTCP packets sent by participants in a session. These packets contain reception reports, and estimate the current quality of service for distribution monitoring, fault diagnosis and long-term statistics. Both TCP (Transmission Control Protocol) and UDP (User Datagram Protocol) enable the transmission of information between the correct processes (or applications) on host computers. IP is responsible for the delivery of packets (or datagram) between host computers. IP is a connectionless protocol and it does not establish a virtual connection through a network prior to commencing transmission because this is the task of higher level protocols. IP makes no guarantees concerning reliability, flow control, error detection or error correction. [22], [23]

2.4 Services of NGN

NGN is a common platform, which gives above services [20, 26].

Voice Telephony – NGN will likely need to support various existing voice telephony services (e.g., Call Waiting, Call Forwarding, 3-Way Calling, various IN features, various Centrex features and etc.).

Data Services – Allows for the real-time establishment of connectivity between endpoints, along with various value-added features.

Multimedia Services – Allows multiple parties to interact using voice, video, and/or data.

Virtual Private Networks (VPNs) – Voice VPNs improve the inter-location networking capabilities of businesses by allowing large, geographically dispersed organizations to combine their existing private networks with portions of the PSTN, thus providing subscribers with uniform dialing capabilities.

Public Network Computing (PNC) – Provides public network-based computing services for businesses and consumers.

Unified Messaging – Supports the delivery of voice mail, email, fax mail, and pages through common interfaces.

Information Brokering – Involves advertising, finding, and providing information to match consumers with providers.

E-Commerce – Allows consumers to purchase goods and services electronically over the network. Home banking and home shopping fall into this category of services. This also includes business-to-business applications.

Call Center/Web Contact Services – A subscriber could place a call to a call/Web contact center agent by clicking on a Web page.

Interactive gaming – Offers consumers a way to meet online and establish interactive gaming sessions (e.g., video games).

Distributed Virtual Reality – Refers to technologically generated representations of real world events, people, places, experiences, etc.

Home Manager – With the advent of in-home networking and intelligent appliances, these services could monitor and control home security systems, energy systems, home entertainment systems, and other home appliances.

CHAPTER 3 NGN MIGRATION

3.1 Existing network architecture of Bangladesh

More than 50% core network of Bangladesh was TDM based circuit switched network. Fig 3-1 illustrates the Pi Chart of Telecom Network in Bangladesh, where red remark means IP network and blue remark means TDM network. [27].

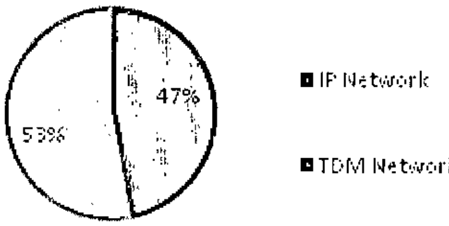


Fig 3-1: Telecom network condition in 2007

Fig 3-2 illustrates the percentage of TDM and IP based network of all PSTN and mobile operators in Bangladesh. It can be observed that comparatively new operators used more IP network than the old operators did. [27].

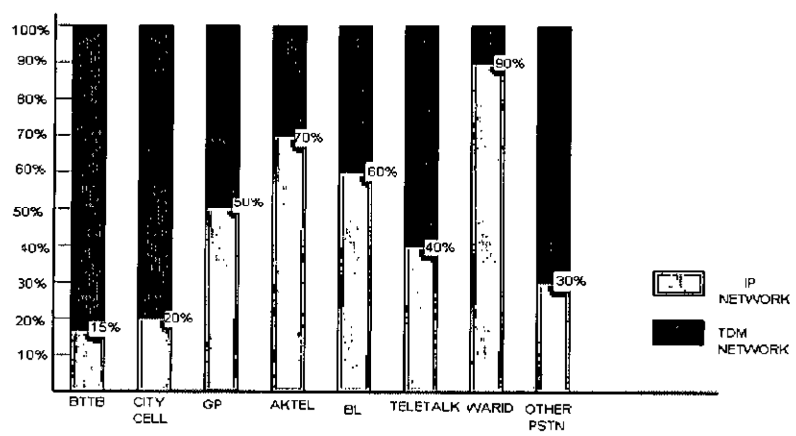


Fig 3-2: Operator wise network condition

Fig 3-3 illustrates the IP based NGN Architecture of Bangladesh in 2008. We can see that all the operators (Fixed and Mobile) are connected to the Interconnection exchange (ICX). The International Gateway (IGW) is connected to the ICX in domestic side and to the BTCL VOIP common platform/all ISP in international side. The IP based Core NGN has already been implemented in the ICX and IGW. The core switching technology is being converted from circuit switched to packet switched.

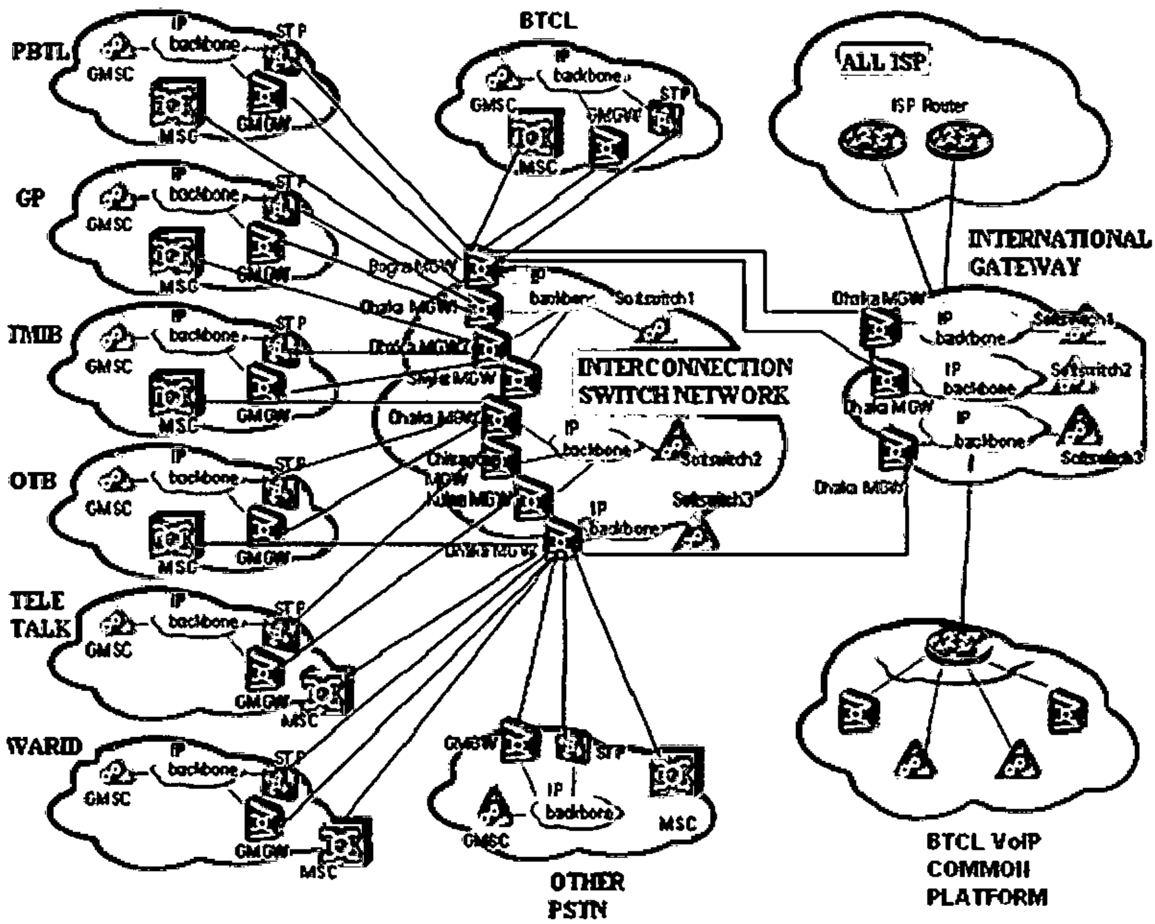


Fig 3-3: IP based NGN Architecture of Bangladesh in 2008

From the above observations the network of Bangladesh can be explained as follows:

1. IP based core NGN has been implemented in Interconnection switches (ICX), international gateway (IGW). Some of the networks of the operators have been migrated to core NGN.
2. Through the separation of service and call control, as well as call control and bearer the service-independent architecture is implemented in the above networks.
3. The access network of the operators and ICX is TDM based. So, they are interconnected to other office through TDM. Thus the intra-office call is going through IP. But the inter-office call is going through TDM.
4. The IGW is interconnected to the domestic nodes through TDM and to the international nodes through IP common platform. Thus, the call is going to the international nodes from the IGW through IP thus reducing the cost of the international outgoing call.

3.2 Proposed All-IP based NGN for Bangladesh

All-IP based NGN is proposed for Bangladesh. ALL-IP based NGN can be implemented by migration of Core and Access network to IP based NGN as shown in Fig 3-4. First, the IP based Core NGN have to be implemented for all operators in Bangladesh. Then the access network should be converted to IP common platform. By migrating NGN, the network will be service oriented, the resource will be properly utilized, capacity will be increased, network cost and management cost will be reduced and subscriber will enjoy the service with reduced cost.

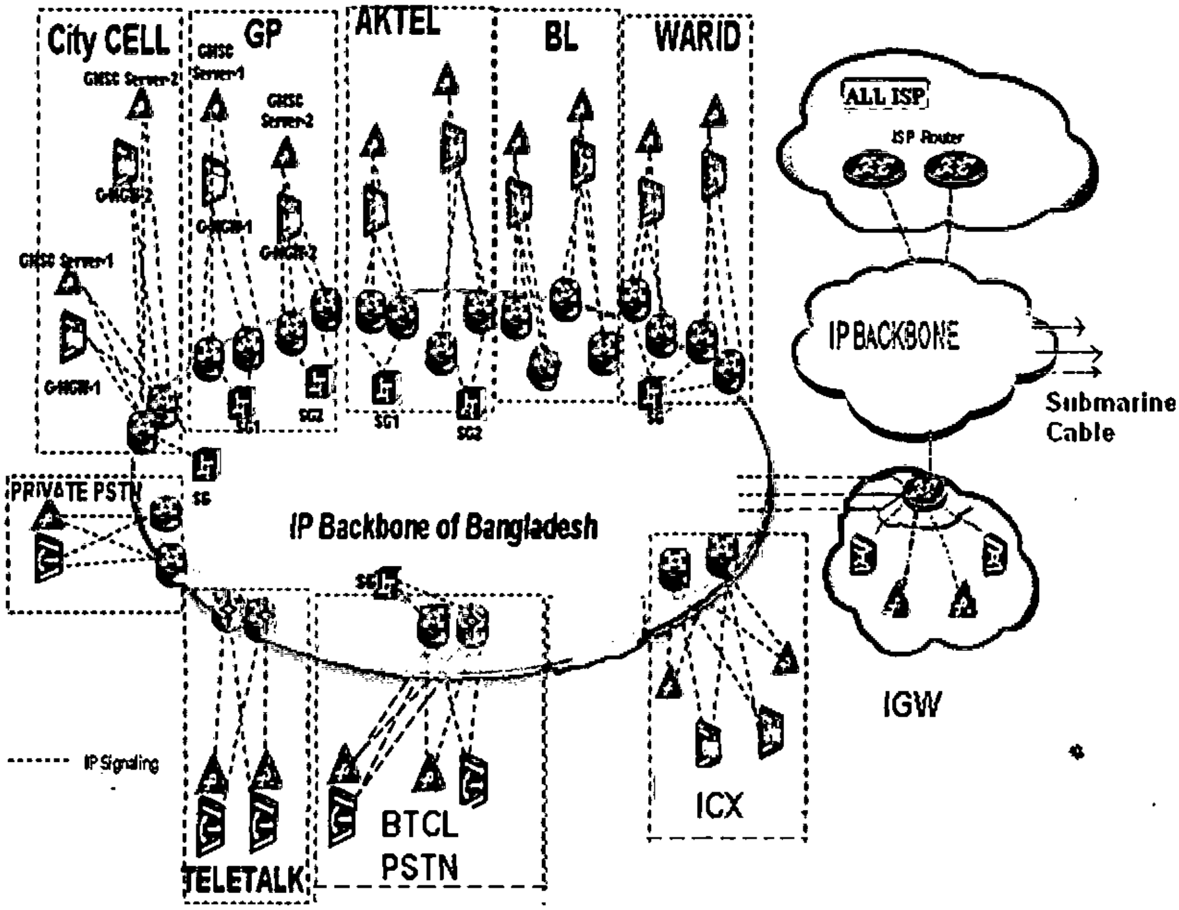


Fig 3-4: Proposed ALL-IP Based Core NGN for Bangladesh

There should a common IP platform for national calls and another common IP platform for international calls. As a developing country in the world, Bangladesh needs quickly to build their backbone as an All-IP based NGN but also as a cheapest way. So, the migration steps should be as follows:

1. The entire switching network of the operators should be migrated to Core NGN first by implementing NGN equipments (Softswitch, various media gaterways, packet switch, router and AS). Then it should be expanded from central (Dhaka) to out of central (Dhaka) under replace or expansion program. They should build their own local IP backbone by implementing routers. In this way the internal calls of the operators will go through IP.
2. Then the IP edge access resource (media gateway interfaces and router) capacity of the operators should be expanded.
3. After that all local TDM access network of the operators of divisional cities and big cities need to convert to ALL-IP network in cheapest way. Thus the access network of

common IP platform will be implemented for the national calls. It should be expanded as per the requirement.

4. The ICX and IGW should also expand their IP resources. They should expand local IP backup by implementing new IP core switching equipments (packet switches, routers and session engines). The local IP backbone will be connected to the common IP platform. In this they will carry the signaling and voice over IP.
5. Then all the operators and ICX local IP backbone will be connected to the common IP platform.
6. The TDM resources will be replaced by the IP resources. In this way, the operators will be connected to the ICX through the national common IP platform. Thus, they will be connected to each other through IP. All the intra and inter office call will go through IP.
7. All the ICX is connected to the IGW for international calls. Now the IGW will be connected to the ICX through common IP platform. IGW is connected to international nodes through common IP platform. So, all mobile and PSTN operators will send their international calls through VOIP common platform which is connected with Submarine cable.

Finally, ALL-IP based Next generation network will be established in Bangladesh.

3.3 Overall Principles of Migration

The overall principles of migration are as follows:

a) Migrate from easy steps to difficult steps: Migrate the easy steps or the end offices with abundant resources. After the migration, the released transport ports can be used in other aspects.

b) Migrate by planes: Separate the network to several tandem planes first and then migrate the planes by sequence. Migrate the overlapped plane between the new trunk gateway and the old tandem office. The migration by planes requires relatively less workload and is consistent with the principle mentioned in (1).

c) Migrate by phases and by batches: In the same migration, migrate by phases and by batches based on the conditions of the end office. This can reduce the transmission difficulties and risks.

d) Simplification: To simplify the migration in the end office and the cooperation times, when the end office is migrated from the old tandem office to the NGN network, cancel the calls between the end offices in once if the transmission conditions permit.

e) Flexibility: The calls in the end office can be sent to the NGN network in different period according to the requirement of the transmission condition, security and project duration. Sending the calls in the end office in once in the migration of the trunks can reduce the cooperation between the end offices. Implement the migration of the calls after the traffic is normal and the transmission is adjusted. This can increase the security of the migration.

f) Security: Seamless migration without interruption of the calls can be realized as follows:

- Make use of the new transmission of the smart network.
- Add the bridge router between the new tandem plane of the NGN network and the old tandem plane.
- Enable the circuits from the end office to the NGN network.

During the migration, the traffic is load shared by the new tandem plane and the new tandem plane. After the migration, the circuits of the old tandem office are released

3.4 Migration Scenario from PSTN to NGN

The Migration scenarios from PSTN to NGN may be as follows [28]:
Scenario 1. Network consolidation
Scenario 2. Deployment of overlay packet based network
Scenario 3. Technology replacement

3.4.1 Scenario 1. Network consolidation

1. Maximum utilization of the installed capacities in the TDM switches:
– Optimal utilization of the already installed digital local exchanges (DLEs)
– Expansion of their service area
– Replacement of analogue exchanges with subscriber capacities, served by DLEs
– Optimization of the connectivity on regional level, reducing the number of nodal service areas

2. Limited deployment of multi-service access systems
– provision of POTS, ISDN BA, ISDN PA, digital LL (n x 64 k), xDSL (ADSL, SHDSL), served by MSANs and xDSLs by DSLAMs – splitting the dial up Internet traffic from the PSTN and routing it to the data network.

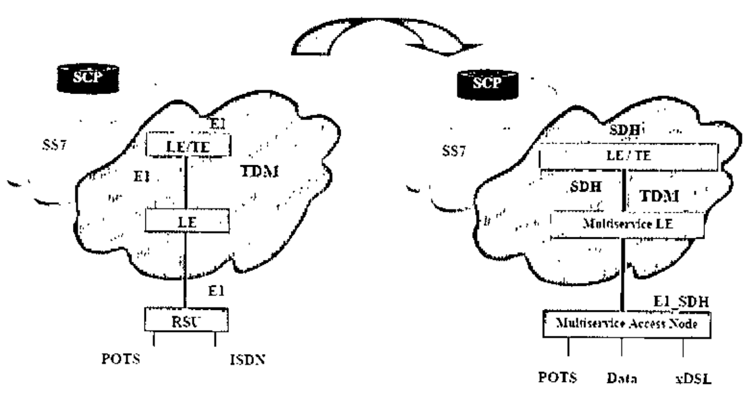


Fig 3-5: Network Consolidation

Fig 3-5 illustrates the Network consolidation scenario.

Major Advantages

– Optimal utilization of the existing TDM equipment, thus reducing the analogue part of the network, network infrastructure optimization.

– Significant CAPEX and OPEX reducement, due to the expansion of existing DLEs, decreasing the number of analogue exchanges in operation.

Major Disadvantages

– IP Network development delay

– Limited number of services to be offered

– Possible PSTN overload, due to the prevailing dial up Internet access and limited deployment of MSANs and DSLAMs.

3.4.2 Scenario 2. Deployment of Overlay Packet Based Network

1. Ongoing network consolidation (as for scenario 1 - optimal utilization of the already installed TDM equipment)
2. Deployment of IP-based overlay network
3. Deployment of Multi-service access systems and DSLAMs for broadband services provision
4. Initial (limited) deployment of VoP services for enterprise and business customers

Deployment of overlay Packet Based Network scenario has been illustrated in Fig 3-6. The general overview of the overlay scenario has been shown in Fig 3-7.

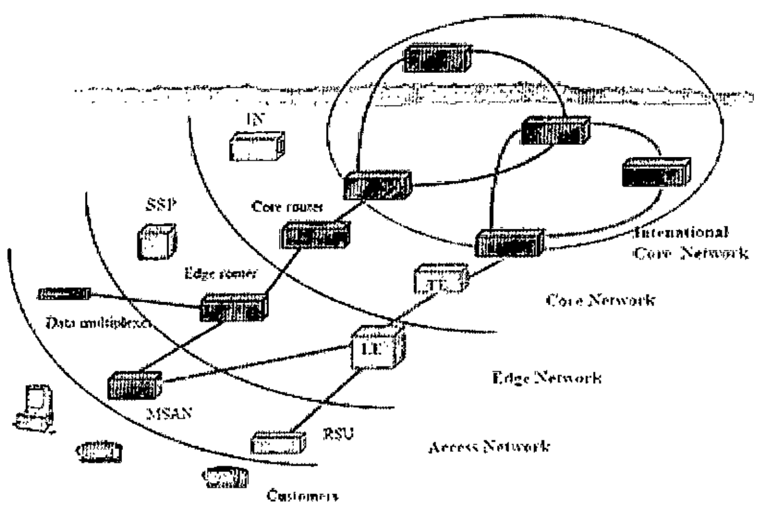


Fig 3-6: Deployment of overlay Packet Based Network

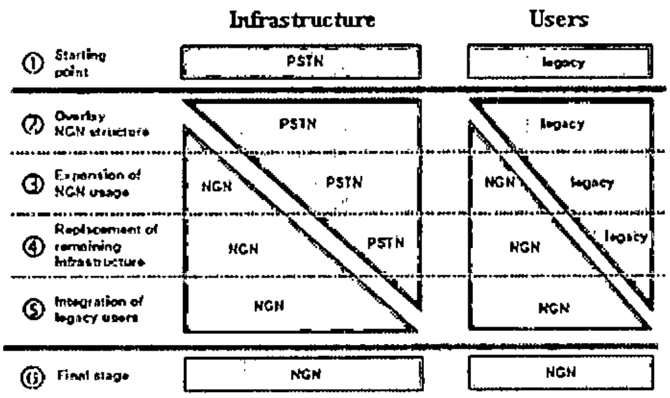


Fig 3-7: General overview of scenarios- overlay network

Major Advantages

1. Optimal utilization and Capitalization on the existing TDM equipment
2. The IP overlay network, combined with the Multiservice access systems - initial step towards the future common packet based network
3. Better services portfolio, especially for business and enterprise customers
4. Reduced OPEX in the TDM part of the network
5. Future save investments

Major Disadvantages

1. Increased CAPEX
2. Increased OPEX

3.4.3 Scenario 3. Replacement of Legacy TDM Equipment

Start point of:

- Replacement of the existing PSTN equipment with packet based one
- Building up a common packet based network for voice, data and video
- Accelerated deployment of multiservice access systems
- Offering voice services via softswitch with local exchange functionality

Major advantages

- Deployment of an unified packet based network for voice, data and video
- Investments are in a prospective technology
- Rich services portfolio, including multimedia services

Major disadvantages

- Part of the NGN equipment is still under research and development,
- IP based equipment is deployed mainly in enterprise networks
- major concerns about QoS
- CPEs require significant investments, if mass deployed

Replacement of legacy TDM equipment scenario has been illustrated in Fig 3-8. The general overview of Infrastructure replacement scenario has been shown in Fig 3-9.

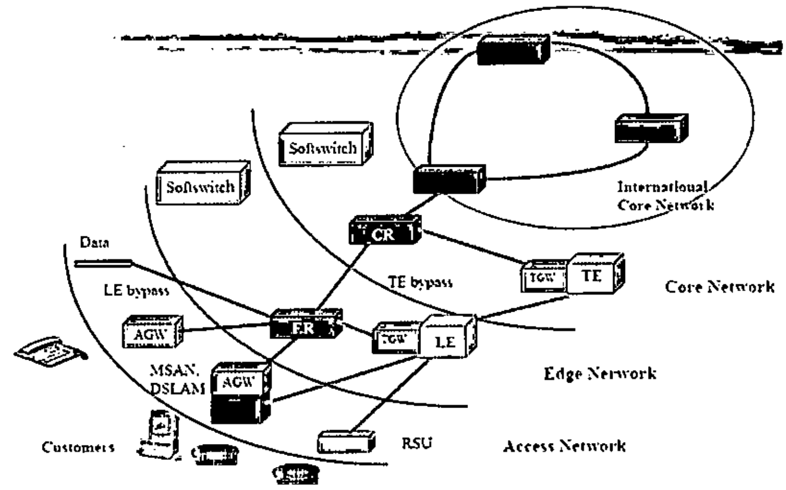


Fig 3-8: Replacement of legacy TDM equipment

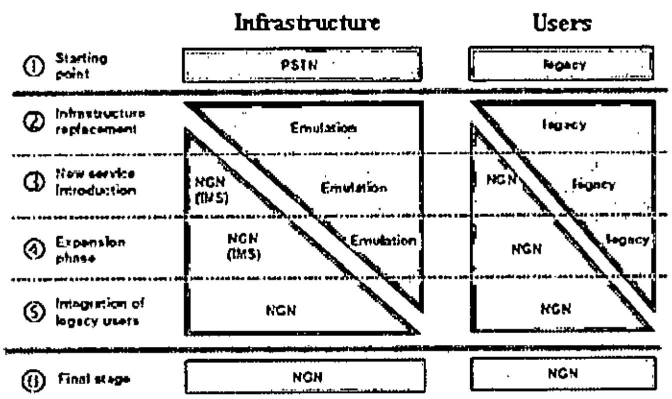


Fig 3-9: General overview of scenario - Infrastructure replacement

3.5 Appropriate Scenario for Network Development

The choice of scenario for network development depends on:

1. Existing infrastructure
   - The major part of the existing analogue exchanges are based on Step by Step technology (1929)
   - The installed TDM switches have capacity, which exceeds the forecasted demands
   - Extension of the TDM switches is accomplished only by adding new subscriber modules, thus reducing the CAPEX and OPEX

2. Demand on services

– The demand on voice telephony and dial up Internet access still prevails on the Bulgarian telecommunications market

– Limited demand on broadband services

3. Overall economics situation in the country
4. Investment capabilities
5. Telecommunications market liberalization
6. Major concerns about QoS, reliability in case of traffic volumes, similar to those served by the PSTN
7. Economic benefits.
8. Standards - interoperability between different vendors equipment under question mark
9. Integration between NGN equipment and the existing PSTN infrastructure

The analysis of the PSTN development, customer demands, standards and international incumbent operators experience, as well NGN element development are the basis for choosing Scenario 2 - Deployment of overlay Packet Based Network for short term development of network. The Deployment of an Overlay Packet Based Network is low risk scenario. The network consolidation allows capitalizing on the existing PSTN equipment, reducing the OPEX. It is IP based overlay network - initial step toward NGN. Important customers are provided with broad range of services, based on xDSL and IP technologies. It is a future proven technology deployment [28].

3.6 Best strategy of Migration for developing countries as Bangladesh

3.6.1 Fixed Core network migration

This describes the next generation network (NGN) evolution based on the orientation of the Universal media gateway (UMG) as shown in Fig 3-10.

Current Situation

Development of the communication technology promotes integration of the telephony network, computer network, and cable TV network. Due to variety of demands on services, carriers must provide more services to attract and keep customers.

The current network architecture is simple and a long time is required to introduce new services. As the trend is to provide integrated services based on IP packets, a network architecture that can suit this requirement is desired.

The NGN is a kind of service-driven network. It realizes the service system relatively independent of the network through the separation of service and call control, and the separation of call control and bearer.

This open architecture can meet the ever-increasing demands on services, strengthen the competitiveness of operation networks, and realize continuous development.

NGN Migration

As an integrated, large-capacity, and carrier-class gateway device, the UMG supports series of hardware and smooth evolution for the sake of carriers' capital expenditure (CAPEX).

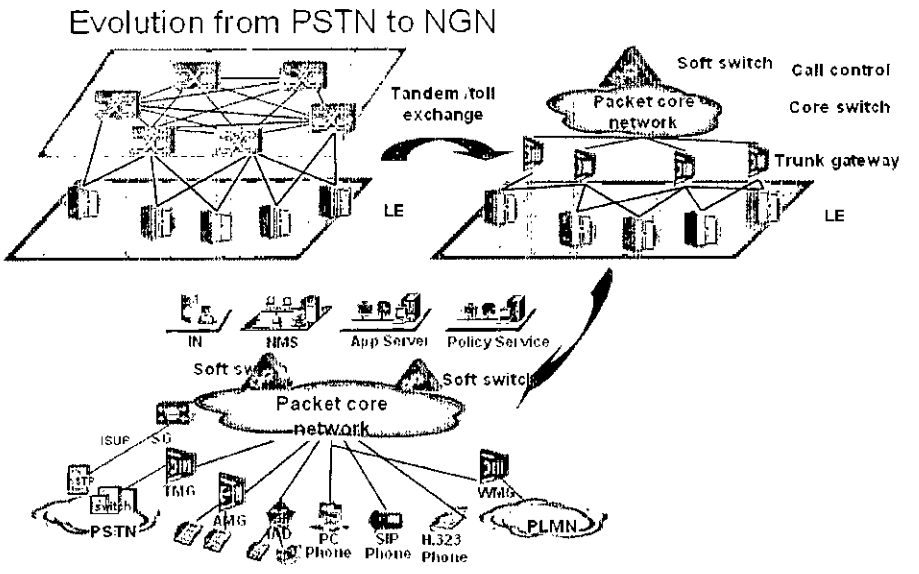


Fig 3-10: Migration from PSTN to NGN

The UMG realizes expansion and upgrade of the C4/C5 exchange in the existing PSTN by serving as an NGN-enabled switch and working with the softswitch.

The UMG adopts the Bearer Independent Call Control Protocol (BICC) architecture. It connects with the softswitch through standard interfaces. When networks evolve into the NGN, the UMG serving as a C4 exchange can be smoothly upgraded to a TG and the UMG serving as a C5 exchange can be upgraded to an AG. The IP-based core network can be achieved. Only software upgrade is required during the evolution while the UMG hardware can still perform its usual functions.

3.6.2 Mobile Core Network Migration

This describes the global system for mobile communications (GSM)/universal mobile telecommunications system (UMTS)/TD-SCDMA network evolution based on the orientation of the UMG as shown in Fig 3-11.

Current Situation

The current GSM network adopts the traditional time division multiplexing (TDM) transmission technology. Network elements (NEs) connect with each other in the star topology, and NEs with different functions connect with each other in the hierarchical topology. The network topology is complex. In addition, because TDM transmission devices are relatively complex, the cost for network construction and maintenance is high.

The development of the IP network makes it an inevitable choice for network evolution to provide integrated voice, data, and video services based on the IP packet technology.
The packet transmission technology will be introduced gradually during the evolution from the GSM network to the UMTS/TD-SCDMA network. The final aim is the all-IP network.
The existing GSM network needs to introduce the IP-based packet transmission technology during evolution processes such as capacity expansion, upgrade, and device replacement. On one hand, it meets the trend of network evolution. On the other hand, because the IP network technology is simple and universal, it can effectively reduce the cost for network construction and operation.
The UMG is designed based on the softswitch separated architecture. The UMG networks with the media gateway controller (MGC) to completely support various narrowband voice and data services of the existing GSM network. In addition, the IP packet transmission technology is introduced to achieve smooth evolution from TDM to IP.

NGN Migration

Fig 3-11 shows the applications of the UMG at different network evolution phases. The UMG is based on the standard separated architecture. It can work with the softswitch to implement core switching applications at different network phases.

In actual networking applications of different networks, the UMG can support smooth network expansion and evolution through addition of related hardware boards and software upgrade

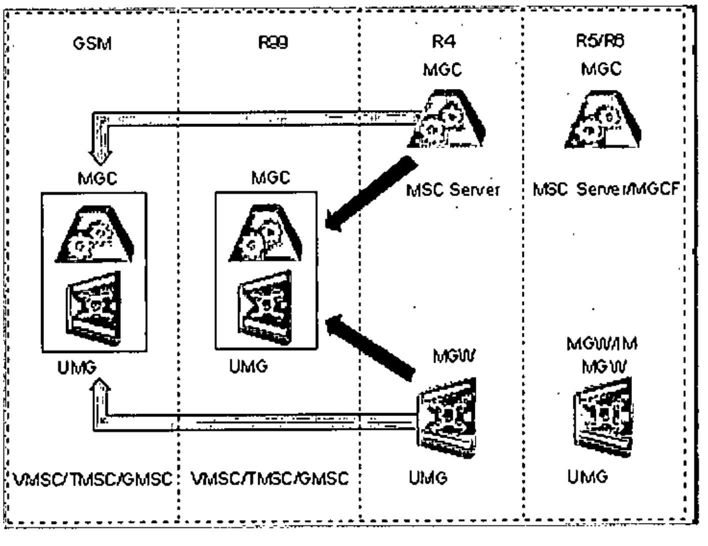


Fig 3-11: Applications of the UMG during the GSM network migration

Different phase of network Migration has been illustrated in Table 3-1. For GSM operators, Upgrade the 2G to 3G is a problem. The extra expenses of operator's equipment investment are predictable because these equipments can not be re-used in the future 3G evolutions. These problems solution is the R4 based mobile softswitch. To upgrade R4 to 3G, only software upgrade of the existing 2G is required.

Table 3-1 Different phase of network Migration

Phase	Characteristics
In the GSM	The air interfaces adopt the frequency division multiple access (FDMA) and the time division multiple access (TDMA) modes. All the transport networks adopt the TDM mode and comply with GSM series of specifications. The core network adopts the hierarchical architecture. The network connections are complex, and the end-to-end delay is low.
GPRS	Two physical entities, namely, gateway GPRS support node (GGSN) and serving GPRS support node (SGSN), are overlaid on the existing GSM network to meet requirements of high-speed packet services. The air interfaces in the access network do not change. Packet service processing interfaces are added to base station controllers (BSCs).
R99	The air interfaces adopt the CDMA mode. The transmission between access networks and the core network is based on the asynchronous transfer mode (ATM). The architecture of the core network has no change. The R99 network complies with relevant 3rd Generation Partnership Project (3GPP) R99 standards. The core network is divided into the circuit switched (CS) domain and the packet switched (PS) domain. The CS domain provides voice and narrowband data services. The PS domain provides high-speed packet services. The CS domain is based on the TDM mode, the same as that in the GSM network.
R4	The access network of R4 is the same as that of R99. The softswitch architecture is introduced to the CS domain of the core network. The original mobile switching center (MSC) is divided into two entities, MSC Server and media gateway (MGW). The core network supports TDM, ATM, and IP connections. The network evolves toward a flattened architecture, and the networking is more flexible. The PS domain of the core network of R4 is the same as that of R99.
R5	Based on R4, the IP multimedia subsystem (IMS) domain is introduced. The R5 network achieves the integrated access of voice, data, and video services and provides IP multimedia services.
R6	The R4 CS domain is retained to provide voice and narrowband data services. The IMS domain is enhanced to achieve the interworking with different networks and the interworking between different IMS domains.

3.6.3 Integrating of Fixed network and Mobile network

Fig 3-12 shows the Fixed Mobile convergence solution where all PSTN network and mobile network are interconnected each other by the help of NGN equipments (softswitch

& media gateways). Network convergence has become the urgent requirements of integrated carriers. The FMC networking can save investment and reduce operation cost for full-service carriers. The FMC networking also helps to enhance the differentiated competition capability of carriers. With the FMC networking, carriers can deploy services more flexibly, provide overall solutions, and offer personalized services.

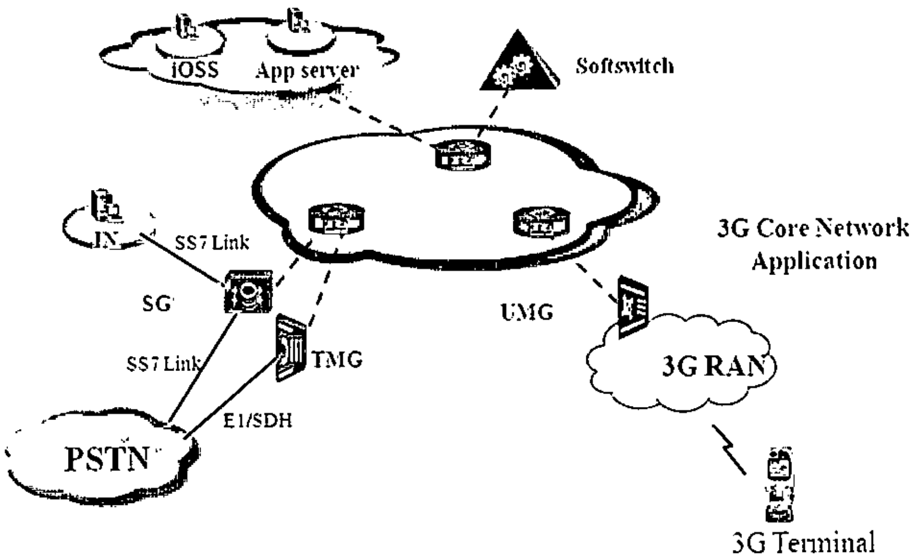


Fig 3-12: Integrating of Fixed network and Mobile network

Fig 3-13 shows network convergence solution based on the softswitch separate architecture. [11]

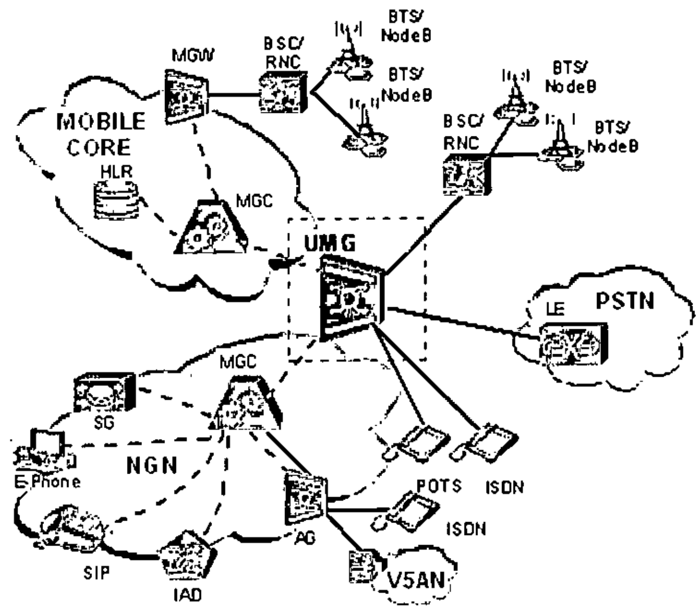


Fig 3-13: Network convergence solution based on the softswitch separate architecture

The UMG, serving as the FMC MGW, can enable the fixed network and the mobile network to share resources of one physical MGW. Under the softswitch separate architecture, the UMG is controlled by the MGC. For the UMG, a physical MGW can be logically divided into multiple virtual MGWs (VMGWs) that provide independent functions and are managed by different softswitches. For example, after a UMG is divided into two VMGWs, they are controlled by the fixed softswitch and the mobile softswitch separately so that the FMC application is implemented.

In addition to the application shown in Fig 3-12, the applications of the UMG as an FMC MGW include but not limited to:

- AG to access broadband and narrowband services
- TG to connect the public switched telephone network (PSTN) and the NGN and convert service stream formats and bearer modes from the PSTN to the IP network
- Mobile local exchange to connect the mobile access network
- Mobile tandem exchange to implement tandem between exchanges
- Mobile gateway exchange to exchange services with external networks
- IP interworking gateway to implement user service interworking between IP networks

Moreover, the UMG can serve as the IM-MGW in the IMS network, to provide abundant MGW functions, serve multiple networking modes, and flexibly and dynamically dispatch various service resources based on service development.

3.7 Technologies adopted to improve the Service Performance of NGN

3.7.1 Advanced Voice Quality Assurance Technology

The common reasons that affect the voice quality in the IP packet network include: delay, jitter, packet loss and echo. UMG improves the bandwidth utilization by adopting different kinds of encoding/decoding (G.711, G.729, G.726, AMR, T.38, and VBD) according to network conditions and user characteristics. It also improves the bandwidth utilization by Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) technology. The UMG reduces delay and jitter by the dynamic buffering technology and reduces the effect of packet loss on the voice quality by the lost-packet compensation technology. The echo is avoided by the echo cancellation technology. The UMG further improves the voice quality by the mute detection and comfortable background noise generation technologies.
In the packet service transmission, the system supports priority of the IP packet service stream. It realizes hierarchical transmission of different service streams through the cooperation with the bearer network and thus provides reliable protection.

3.7.2 Session Border controller

The Session Border Controller (SBC) is placed at the edge of the IP network to help build a service-based intelligent network that guarantees network security and QoS. It also implements the traversal of signaling and media streams on the NAT or firewall. The SBC also supports interconnecting between NGN networks and can serve as Core Border Gateway Function (C-BGF) in IMS.

3.7.3 QoS Optimization Services

- Analysis of service traffic model: Analyze traffic models of services on the current network and the newly added services to form the basis for path adjustment of following service traffic and deployment of QoS policies.
- Path adjustment of service traffic: The QoS policies take effect when the resources are preempted. You need to first adjust the traffic paths based on the traffic models and then balanced different types of service flows to available links to decrease the probability of the congestion.
- QoS policy deployment: Plan QoS policies on the whole network to implement reasonable utilization of the resources. Thus, the quality of high value-added services is ensured.
- Topology and device optimization: Topology and link optimization Suggestions for device optimization

3.8 NGN Network design for different types of office

3.8.1 Packet Multimedia End office

The Softswitch can act as a multimedia end office to provide different multimedia applications: The Softswitch supports SIP and H.323 protocol.

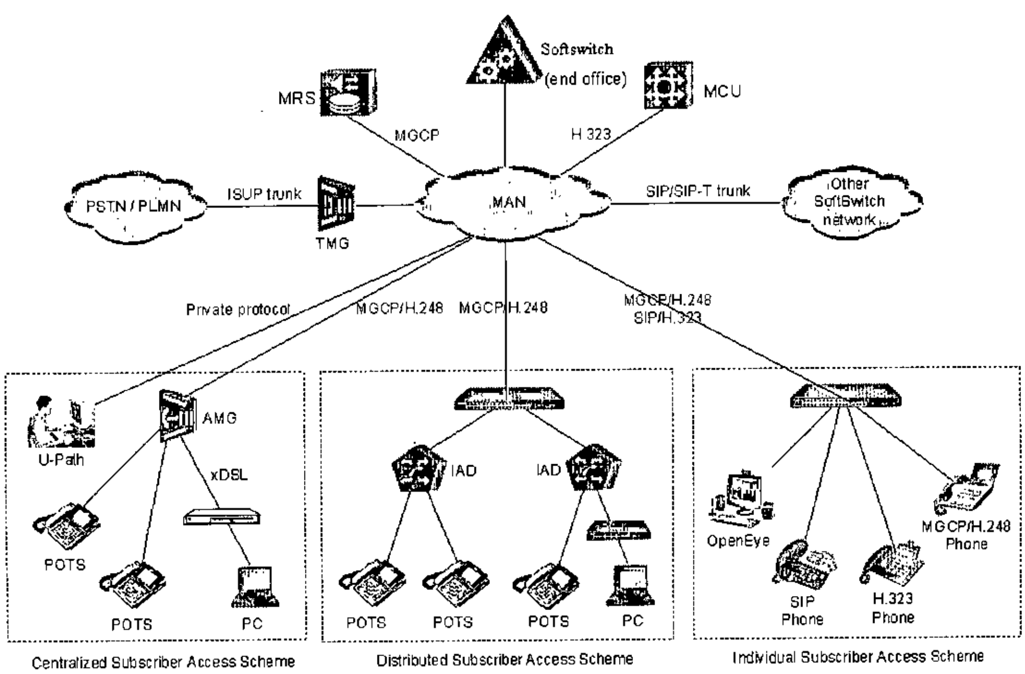


Fig 3-14: Packet Multimedia End office

It can function as H.323 gatekeeper (GK) or SIP Server to control and access multimedia terminal. The Softswitch also supports MGCP (include NCS and TGCP) and H.248 protocol. It can access voice media gateway. A typical networking model is illustrated in Fig 3-14.

3.8.2 Packet Tandem office

The Softswitch supports SIP and H.323 protocol. The Softswitch can act as the Packet Tandem office in the IP network. A typical networking model is illustrated in Fig 3-15.

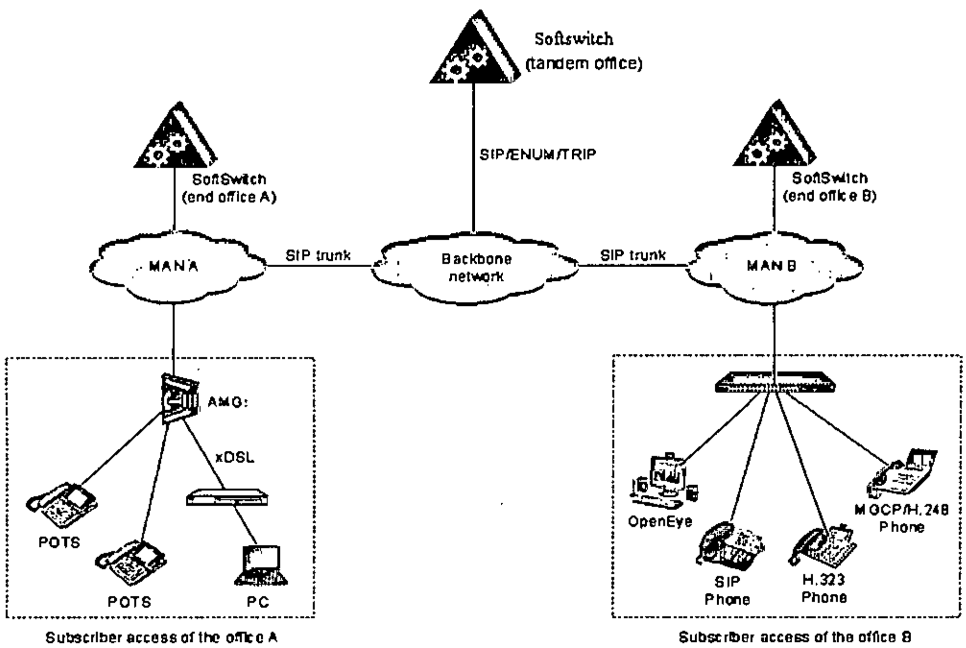


Fig 3-15: Packet Tandem Office

3.8.3 C5 office (End office)

The Softswitch supports a number of signaling transport adaptation protocols, including M2UA, V5UA and IUA, and supports a number of PSTN signaling including MTP, ISUP, R2, V5.2 and DSS1. When working with UMG or TMG, the Softswitch can act as a C5 office (end office) in the PSTN. A typical networking model is shown in Fig 3-16.

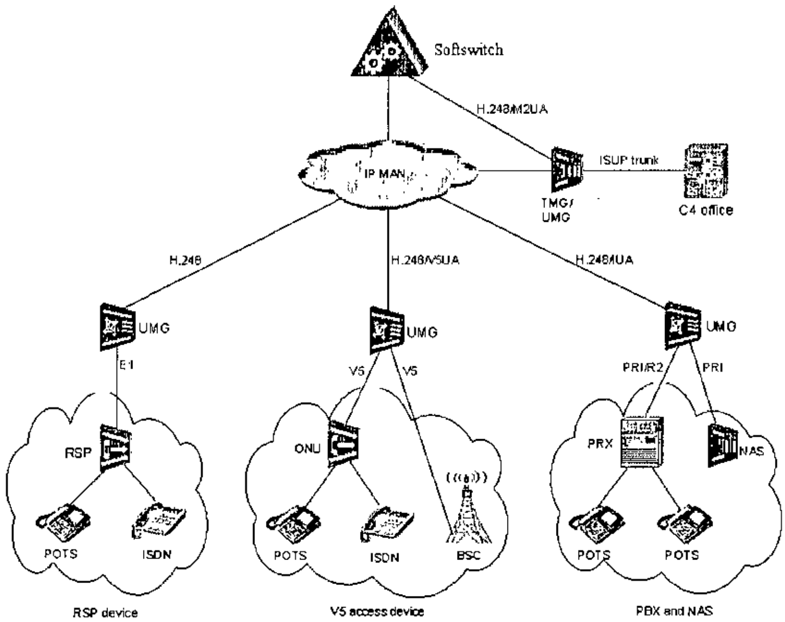


Fig 3-16: C5 office (end office)

3.8.4 C4 office (Tandem Office)

The Softswitch supports M2UA, M3UA, MTP and ISUP. When networking with UMG, TMG and SG, the Softswitch can act as the C4 office (tandem office) in the traditional PSTN network. A typical networking model is illustrated in Fig 3-17.

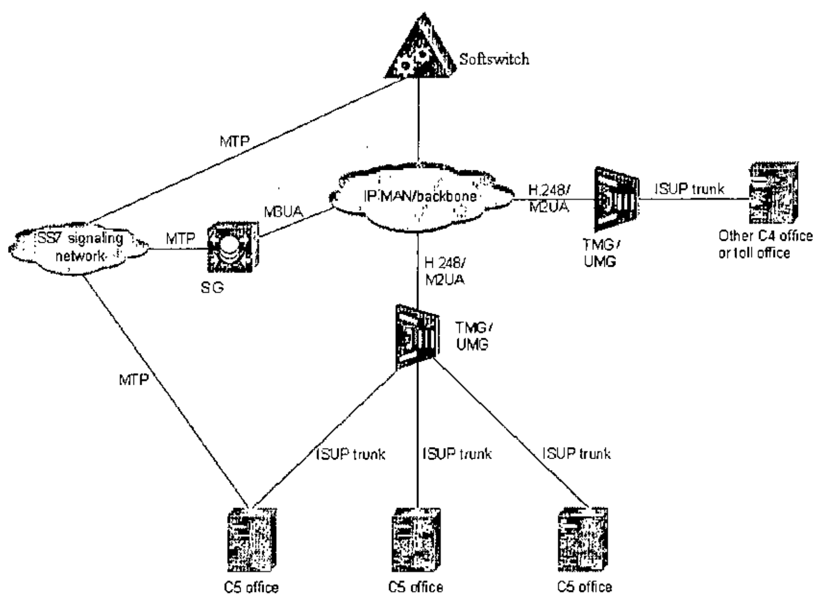


Fig 3-17: C4 office (Tandem Office)

CHAPTER 4 PERFORMANCE ANALYSIS

4.1 Performance Evaluation Criteria

The service performance can be evaluated by the following measures of any network:

1. Bandwidth & Capacity
2. Signal-to-Noise ratio (SNR)
3. Probability of bit error
4. Quality of Service (QoS)

4.2 Theoretical analysis

4.2.1 Bandwidth & Capacity

i) Voice over TDM

The existing network of Bangladesh is TDM based circuit switched network. G.711 codec is primarily used in telephony or TDM based network.

Network Capacity is the maximum number of sessions that can be supported. Lets N be the maximum number of sessions that can be supported. N can be expressed as

Network Capacity, N = T/B **(4-1)**

Where,

T = Transmission Capacity= Number of Frames X Number of channels/frame X Bit rate of channel

B = Channel Capacity/Required Bandwidth= (Symbols/Sec) X (Bits/Symbol) = (Sampling Rate X Bits per Sample) = (S X R) b/sec

S = Sampling rate = 2 X f

f = bandwidth of the base band signal

R = bits per sample= ln (L)

L= Number of quantization levels

ii) **Voice over IP**

In an NGN, after voice has been encoded/decoded, it will be packetized by using the RTP and sent to an IP network. If there are different encoding/decoding types and actual packetization durations, bandwidth demands in bearer network will be different:

For a VoIP packet, the header overhead, OH_{hdr} consists of the headers of RTP, UDP, IP and MAC Layer:

$$OH_{hdr} = H_{RTP} + H_{UDP} + H_{IP} + H_{MAC} \tag{4-2}$$

$$\text{Packet length} = OH_{hdr} + \text{payload} \tag{4-3}$$

$$\text{Payload} = \text{number of payload bits per second} \times \text{packetization duration (s)} \tag{4-4}$$

$$\begin{aligned}\text{Bandwidth} &= \text{packet length} \times \text{number of packets per second} \\ &= \text{packet length} \times (1/\text{packetization duration}) \\ &= (OH_{hdr} + \text{payload}) \times (1/\text{packetization duration}) \\ &= (OH_{hdr} + (\text{number of payload bits per second} \times \text{packetization duration (s)})) \\ &\quad \times (1/\text{packetization duration})\end{aligned}$$

$$\text{Bandwidth} = (OH_{hdr}/\text{packetization duration}) + \text{number of payload bits per second} \tag{4-5}$$

Network Capacity

Let n be the maximum number of sessions that can be supported. Let T be the average time between the transmissions of two consecutive packets. That is, in one second, there is totally 1/T packets transmitted. We can define 1/T as follows:

$1/T$ = Number of Streams X Number of packets sent by one stream in one second.

$$1/T = n \times N_p \tag{4-6}$$

Thus, $n = 1/(T \times N_p)$

$$\text{Where, } T = \frac{(\text{Payload} + \text{OHhdr}) \times 8}{\text{datarate}} \tag{4-7}$$

4.2.2 Signal-to-Noise ratio

To optimize system performance in the presence of channel noise, we need to minimize the average probability of symbol error. For this evaluation, it is customary to model the channel noise as additive, white and Gaussian. The effect of channel noise can be made practically negligible by ensuring the use of an adequate signal energy-to-noise density ratio through the provision of short enough spacing between the regenerative repeaters in the PCM system. In such a situation, the performance of the PCM system is essentially limited by quantization noise acting alone.

The quantization noise can be made negligibly small through the use of an adequate number of representation levels in the quantizer and the selection of a companding strategy matched to the characteristics of the type of message signal being transmitted.

SNR is an important criterion for the performance of a communication system. For a system, it is desired that the value of SNR should be high. However, it should be remembered by a good designer that cost would be a vital factor while increasing SNR. So, optimum value of signal power should be chosen. [29]

Noise consideration in PCM system

Noise is any unwanted, random and unpredictable signal from environment, which can be of even infinite amplitude. The presence of noise superimposed on a signal tends to obscure or mask the original signal.

Noise can be described as a zero-mean Gaussian random process. A Gaussian process n(t) is a random function whose value n at any arbitrary time t is statistically characterized by the Gaussian probability density function [29].

$$p(n) = \frac{1}{\sigma\sqrt{2\pi}} \exp\left[-\frac{1}{2}\left(\frac{n}{\sigma}\right)^2\right] \tag{4-8}$$

Where σ^2 is the variance of n.

The performance of a PCM is influenced by two major sources of noise

1. Quantization noise, which is introduced in the transmitter and carried all the way along to the receiver output. Unlike channel noise, quantization noise is signal-dependent in the sense that it disappears when the message signal is switched off.
2. Channel noise, which is introduced anywhere between the transmitter output and the receiver input. Channel noise is always present once the equipment is switched on.

Quantization Noise

The use of quantization introduces an error defined as the difference between the input signal and the output signal. The error is called quantization noise [29].

Quantization Noise, $q = m - v$

For a uniform quantizer, the quantization error q will have its sample valued bounded by $-\Delta/2 <= q <= \Delta/2$.

Where, Δ= Step-size of quantization = $2\, m_{max}/L$ (4-9)

L = total number of representation levels. = 2^R

R= Number of bits per sample= log2L

Average noise power can be represented [30] by the following equation,

$$\sigma^2 = \int_{-\Delta/2}^{\Delta/2} q^2 f(q) dq = 1/\Delta \int_{-\Delta/2}^{\Delta/2} q^2 dq = \frac{\Delta^2}{12} = \frac{1}{3} m_{max}^2 2^{-2R} \tag{4-10}$$

Channel Noise

Noise power is evaluated by the bandwidth of the transmitted signal and noise spectral properties. Consider a zero mean signal which is band limited to B hertz. Then according to Nyquist theorem the transmitted signal bandwidth is 2B. These samples are transmitted in T seconds over a noisy channel, also band limited to B hertz. Hence, the number of samples is given by, K= 2BT. The channel output is perturbed by additive white Gaussian noise (AWGN) of zero mean and power spectral density No/2. Since the noise sample is Gaussian distributed, having uniform power spectral density $N_0/2$, the total noise power (variance) [29] for channel noise within the bandwidth of the transmitted signal is

$$N = \frac{N_0}{2} \times 2B = N_0 B \qquad (4\text{-}11)$$

The average noise power in the PCM system is increased by the R-fold increase in bandwidth B (As N=NoB), where R is the number of bits in a code word (bits per sample)

Signal-to-Quantization Noise ratio

SNR can be defined [29] as the following equation,

$$(SNR)_0 = \frac{P}{\sigma^2} = \left(\frac{3P}{m_{max}^2}\right) 2^{2R} \qquad (4\text{-}12)$$

Where P= average signal power

σ^2 = average noise power

m_{max} = Maximum signal amplitude

R= Number of bits per sample

SNR in dB,

$$SNR = 10\log_{10}\left(\frac{\text{Signal Power}}{\text{Noise Power}}\right) = 10\log_{10}(SNR)_0 \qquad (4\text{-}13)$$

For sinusoidal modulating signal of amplitude A_m, the average signal power is $P = A_m^2/2$ The total range of the quantization input is 2 A_m.

By replacing m_{max}^2 in equation (4-10), we get the average noise power (variance) as,

$$\sigma_Q^2 = \frac{1}{3} Am^2 2^{-2R} = \frac{Am^2}{3 * (2^{2R})} \qquad (4\text{-}14)$$

By replacing P and m_{max}^2 in equation (4-12), we get the output signal-to-noise ratio of the uniform quantizer as

$$(SNR)_0 = 3/2\ (2^{2R}) \qquad (4\text{-}15)$$

Thus, Signal-to-quantization noise ratio in decibels can be expressed as follows,

$$10\log_{10}(SNR)_0 = 1.8 + 6R \qquad (4\text{-}16)$$

Signal-to-Noise ratio in presence of channel noise

SNR in presence of channel noise can be expressed as follows

$$SNR = \frac{P_r}{N_0 B} = \frac{E_s}{N_0 B T_s} = \frac{E_b}{N_0 B T_b} = Eb/N0 . fb/B \tag{4-17}$$

Here T_s, T_b are symbol period and bit period respectively. f_b is the channel data rate (gross bit rate), and B is the channel bandwidth.

$$10\log_{10}(SNR)_0 = 10\log_{10}(f_b/B) + 10\log_{10}(E_b/N_0) \tag{4-18}$$

4.2.3 Influence of Channel noise on the Probability of error

Probability of error is the measure of probability of erroneous bits in received signal compared to the original information signal. Bit error happens due to the interferences of the channel, like noise, fading, path loss, shadowing, etc. The higher the probability of bit error, the more erroneous the system is. Therefore, it is desired that probability of bit error should be less for any communication system.

Probability of error in terms of complementary error function

Let's consider a binary PCM system based on polar non-return-to-zero signaling as shown in Fig 4-1. In this form of Symbol 1 and 0 are represented by positive and negative rectangular pulses of equal amplitude and equal duration. The channel noise is modeled as additive white Gaussian noise w(t) of zero mean and power spectral density $N_0/2$. In the signaling interval $0 <= t <= T_b$, the received signal is thus written as follows:

$$\begin{aligned} X(t) &= +A + w(t), \text{ symbol 1 was sent} \\ &= -A + w(t), \text{ symbol 0 was sent} \end{aligned} \tag{4-19}$$

Where Tb is the pulse duration and A is the transmitted pulse amplitude.

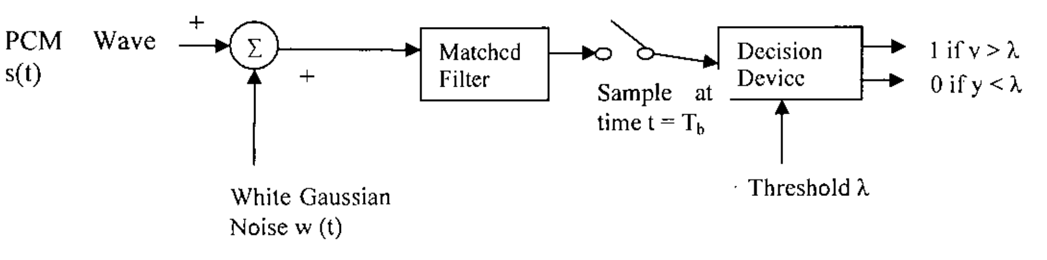


Fig 4-1: Receiver for baseband transmission of binary encoded PCM wave using polar NRZ signaling.

Let y denote the sample value obtained at the end of a signaling interval. The sample value is compared to a preset threshold λ in the decision device. There are two possible kinds f error t be considered:

1. Sybmol 1 is chosen when a 0 was actually transmitted; we refer to this error as an error of the first kind.
2. Symbol 0 is chosen when a 1 was actually transmitted; we refer to this error as an error of the second kind

Suppose that symbol 0 was sent. Then according to equation (4-19), the received signal is

$$x(t) = -A + w(t), \qquad 0<=t<=T_b \qquad (4\text{-}20)$$

Correspondingly the output of the Matched filter, sampled at time $t=T_b$ is given by,

$$y = \int_0^{Tb} x(t)dt$$

$$= -A + 1/Tb \int_0^{Tb} w(t)dt \qquad (4\text{-}21)$$

which represents a sample value of a random variable Y.
The random variable Y is Gaussian distributed with a mean of $-A$
The variance of the random variable Y is

$$\sigma_Y^{\ 2} = E[(Y+A)2]$$

$$= \frac{1}{T_b^{\ 2}} E[\int_0^{T_b}\int_0^{T_c} w(t)w(u)dtdu$$

$$= \frac{1}{T_b^{\ 2}} \int_0^{T_b}\int_0^{T_c} E[w(t)w(u)]dtdu$$

$$= \frac{1}{T_b^{\ 2}} \int_0^{T_b} R_W(t,u)dtdu \qquad (4\text{-}22)$$

Where $R_w(t,u)$ is the autocorrelation function of the white noise w(t). Since w(t) is white with a power spectral density $N_0/2$, we have

$$R_w(t,u) = N_0\ \delta(t-u)/2 \qquad (4\text{-}23)$$

Where $\delta(t-u)$ is a time shifted delta funsction. Hence substituting equation (4-23) into Equation (4-22) we get,

$$\sigma_Y^{\ 2} = \frac{1}{T_b^{\ 2}} \int_0^{T_b}\int_0^{T_c} \frac{N_0}{2}\delta(t-u)dtdu \qquad (4\text{-}24)$$

The conditional probability density function of the random variable Y, given that symbol 0 was sent, is therefore

$$f_Y(y \mid 0) = \frac{1}{\sqrt{\Pi N_0/T_b}} \exp(-\frac{(y+A)^2}{N_0/T_b}), \quad \text{symbol 0 was sent} \qquad (4\text{-}25)$$

$$f_Y(y \mid 1) = \frac{1}{\sqrt{\Pi N_0/T_b}} \exp(-\frac{(y-A)^2}{N_0/T_b}), \quad \text{symbol 1 was sent} \qquad (4\text{-}26)$$

The probability of this error, conditional on sending symbol 0, is defined by

$$p_{10} = P(y > \lambda \mid \text{symbol} \quad 0 \text{ was} \quad \text{sent})$$
$$= \int_{\lambda}^{\infty} f_Y(y \mid 0)dy$$
$$= \frac{1}{\sqrt{\Pi N_0/T_b}} \int_{\lambda}^{\infty} \exp(-\frac{(y+A)^2}{N_0/T_b})dy \tag{4-27}$$

The probability of this error, conditional on sending symbol 1, is defined by

$$p_{01} = P(y < \lambda \mid \text{symbol} \quad 1 \text{ was} \quad \text{sent})$$
$$= \int_{-\infty}^{\lambda} f_Y(y \mid 1)dy$$
$$= \frac{1}{\sqrt{\Pi N_0/T_b}} \int_{-\infty}^{\lambda} \exp(-\frac{(y-A)^2}{N_0/T_b})dy \tag{4-28}$$

The complementary error function is defined as follows

$$\text{erfc}(u) = \frac{2}{\sqrt{\Pi}} \int_{u}^{\infty} \exp(-z^2)dz \tag{4-29}$$

To express p_{10}, lets define a new variable $z = \frac{y+A}{\sqrt{N_0/T_b}}$

Thus we can write equation (4-27) as follows

$$p_{10} = \frac{1}{\sqrt{\Pi}} \int_{(A+\lambda)/\sqrt{\frac{N_0}{T_b}}}^{\infty} \exp(-z^2)dz$$
$$= \frac{1}{2}\text{erfc}(\frac{A+\lambda}{\sqrt{N_0/T_b}}) \tag{4-30}$$

To express p_{01}, lets define a new variable $z = \frac{A-y}{\sqrt{N_0/T_b}}$

Thus we can write equation (4-28) as follows

$$p_{01} = \frac{1}{\sqrt{\Pi}} \int_{(A-\lambda)/\sqrt{\frac{N_0}{T_b}}}^{\infty} \exp(-z^2)dz$$
$$= \frac{1}{2}\text{erfc}(\frac{A-\lambda}{\sqrt{N_0/T_b}}) \tag{4-31}$$

The average probability of symbol error P_e is given by

$$P_e = p_0 p_{10} + p_1 p_{01}$$
$$= \frac{p_0}{2}\,\mathrm{erfc}\left(\frac{A+\lambda}{\sqrt{N_0/T_b}}\right) + \frac{p_1}{2}\,\mathrm{erfc}\left(\frac{A-\lambda}{\sqrt{N_0/T_b}}\right) \quad (4\text{-}32)$$

$$\lambda_{opt} = \frac{N_0}{4AT_b}\log\left(\frac{p_0}{p_1}\right) \quad (4\text{-}33)$$

For the special case when 1 and 0 is equiprobable, we have $p_0=p_1=1/2$
In that case $\lambda_{opt}=0$ and $p_{01}=p_{10}$

Thus, the average probability of symbol error reduces to

$$P_e = \frac{1}{2}\,\mathrm{erfc}\left(\frac{A}{\sqrt{N_0/T_b}}\right) \quad (4\text{-}34)$$

The transmitted signal energy per bit is defined by

$$E_b = A^2 T_b \quad (4\text{-}35)$$

The probability of bit error can be defined in terms of complementary error function:

$$P_e = \frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right) \quad (4\text{-}36)$$

Where,
E_b/N_0 = the ratio of the transmitted signal energy per bit, E_b , to the noise spectral density, N_0.

Lets T is the time interval between two errors. T can be defined as:
T = 1/ Number of errors
T = 1/ (P_e X bit rate) (4-37)

4.2.4 QoS

Speech transmission over IP networks is highly sensitive to transmission impairments. This in turn arises from two characteristic attributes of the internet: (1) Limited bandwidth: when a connection between two routers operates at full capacity, the packets to be transmitted are stored in a queue (store-and-forward-principle), resulting in a transmission delay. If the capacity of the queue is exceeded, packets are dropped (packet loss). (2) Individual routing of packets: resulting from the fact that different packets may be routed differently through the network, packets may arrive at their destination in a different order than that in which they were sent.

Factors affecting the QoS are as follows:

1. Voice encoding modes.
2. Bandwidth: Unit: bit/sec.
3. Network Delay: Unit: ms.
4. Network Jitter.
5. Network Packet loss.
6. Echo

These characteristics of the internet yield six different kinds of transmission impairments:

1. Coding distortion

CODEC converts the analog signals to digital signal using PCM scheme.

- The codec makes 8000 samples per sec or one sample per 125 microsec. This is because Nyquist theorem says that the sampling rate should be equal to or less than the twice of the BW of the base signal. So, it is sufficient to capture all the information from the 4 KHz telephone channel BW. This technique is called PCM.
- All the time intervals (a pulse) within the telephone system are multiples of 125 microsec.

Table 4-1 and 4-2 illustrates different parameters of the codecs [30], [31], [32].

Table 4-1: TDM Codec

Parameters	G.711 Codec (using PCM Scheme)
Bit rate (Kbps)	64
Framing interval (s)	1
Payload (Bytes)	8000

Table 4-2: Audio/Voice codec parameters of VoIP Codecs

Parameters	G.711 Codec (PCM)	G. 723.1 (ACELP)	G.729 (CS-ACELP)
Bit rate (Kbps)	64	6.3	8
Framing interval (ms)	20	30	20
Payload (Bytes)	160	24	20
Packets/s, N_p	50	33	50

Each codec results in a perceivable degradation of speech quality. Table 4-3 & 4-4 gives an overview of common codecs used for VoIP [33].

Table 4-3 Equipment impairment factors for different codec

Codec	Bit Rate (kbps)	Equipment Impairment factor, I_e
G.711 a-law	64	0
G.711 i-law	64	0
G.723.1	5.3	19
G.723.1	6.3	15
G.729	8	10
G.729b	8	11

2. Delay

Time spent in going from the originator to the receiver for end-to-end packet transmission. Unit: ms. Generally, if delay exceeds 100ms, we will feel in our conversation that the peer does not speak naturally and reacts slowly; if delay exceeds 250ms, we will feel unbearable in our conversation [34].

- Propagation delay: It is determined by propagation speed and total distance.
- Transmission delay: It indicates the duration of transmitting the voice through all the network equipment.
- Packet conversion delay: It indicates the duration of digit-analog conversion through a coder.
- Jitter buffer delay: It indicates the duration to overcome packet arrival (or jitter).

Delay in VoIP-transmission results from several factors. Delay results from the following:

1) End-to-end voice delay = encoding delay
+ Compression & packetization delay
+ network transmission delay
+ unpacking & decompression delay
+ decoding delay.

2) Buffer (jitterbuf) set to eliminate network jitter.

The delay impairment factor can be defined as follows:

$$Id = 0.024d + 0.11\,(d-177.3)\,H\,(d-177.3) \qquad (4\text{-}38)$$

I_d: it is related to end to end delay
d=one-way delay (coding + network + de-jitter delay) [ms]
$H(x) = 0$ for $x<0$ $H(x) = 1$ for $x \geq 0$

3. jitter:

Delay jitter represents the variation of the delay due to variable packet routing over the internet. During a VoIP call, jitter refers to the difference in arrival time of all the sent data packets. VoIP calls may suffer from significant jitter. Jitter indicates how steady packet transmission is.

4. packet loss:

The number of packets lost during transmission over the network is the difference between the number of sent data packets and the number of received data packets.

When networks or parts of a network are used at their capacity limits, chances are that some packets do not arrive at their destination in time, or even never arrive at all.

Depending on the applied coding scheme, packet loss may lead to severe perceptual degradations of the transmitted speech signal.

$$Ief = Ie + 30\ln(1+15e) \quad (4\text{-}39)$$

$I_{ef:}$ it is related to packet loss

e= packet loss ratio

Using equation (4-39), Equipment impairment factors for different codec considering packet loss can be calculated as in table 4-4.

Table 4-4 Equipment impairment factors for different codec considering packet loss

%packet loss	G.729a	G.723.1a
0	11	15
0.5	13	17
1	15	19
1.5	17	22
2	19	24
3	23	27
4	26	32
8	36	41
16	49	55

5. packet doubling:

This effect may occur due to faulty router configurations.

6. echo:

Echo indicates that a speaker's voice goes through network equipment and loops back to itself. Features of echo: echo affects the perception quality of the speaker instead of that of the receiver;

As it has been shown that delays are high for VoIP transmissions, room echoes at the far end particularly degrade ease of communication.

E Model

The E-model [35] calculates the R from the network QoS factors as shown in Fig 4-2. It uses the sum of equipment impairment factors Ie and the equipment delay factor I_d, each one quantifying the distortion due to a particular factor and calculates R.

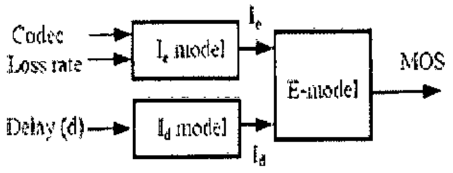


Fig 4-2: E- Model

As an example, in a system with distortion due to the codec, average one-way delay, packet delay variation (jitter) and packet loss, the rating R is computed as follows:

$R = R_0 - \text{Icodec} - \text{Idelay} - \text{Ipdv} - \text{Ipacketloss}$
$R = 93.2 - I_d - I_{ef}$ (4-40)

Once the R value is calculated, an estimate of the MOS can be directly calculated from it.

MOS (Mean opinion score)

MOS (Mean Opinion Score) is a method to evaluate the voice subjectively. MOS is to evaluate the voice quality into five levels (level 5 through level 1) according to the scoring standards defined by ITU-T [36] as listed in Table 4-5. For good voice quality, the MOS value should be 3.5 or higher.

Table 4-5: VoIP Voice Quality defined by ITU-T

Grade	MOS	Users' Satisfaction
Superior	4.0--5.0	Very good, able to be heard clearly, low delay, smooth communication.
Good	3.5--4.0	Slightly poor, able to be heard clearly, low delay, not smooth communication, some noise
Medium	3.0--3.5	Medium, unable to be heard clearly, with certain delay; communication is possible.
Poor	1.5--3.0	Poor, unable to be heard clearly, great delay, repeated communication.
Bad	0--1.5	Bad, unable to be understood, great delay, not smooth communication.

Mapping between MOS and E-Model

$MOS = 1 < 1 + (0.035*R) + (R(R-60)*(100-R)*7.0e^{-06}) < 4.5$ (4-41)

R < 0 MOS = 1

0 < R < 10, MOS = 1 + 0.035R + R(R - 60) (100 - R)7·10-6

R > 100 MOS = 4.5

The relation between the impairment factors and MOS has been shown in Table 4-6.

4.3 Analytical Results

4.3.1 Bandwidth & Capacity

i) Voice over TDM

G.711 codec is primarily used in telephony or TDM based network. For G.711 TDM Codec (Table 1), Channel Capacity/bandwidth = 8000 X 8= 64 kbps. We can calculate TDM network capacity for different data rates (Transmission Rate) using equation (4-1). Table 4-6 contains the calculated value of TDM capacity for different data rates. The Detail calculation is in Appx A.

Table 4-6: TDM Capacity for different Transmission/data rates

TDM Transmission	Bit rate (Mbps)	Maximum Simultaneous Nodes , N
E1	2.048	32
E2	8.448	132
E3	34.368	537
STS-1	51.84	810
STM-1	155.52	2430
STM-4	622.08	9720

ii) Voice over IP

The total bandwidth occupancy of NGN is related to encoding/decoding mode, packetization duration, and traffic. Audio/Voice codec parameters for different codecs have been illustrated in Table 4-2. The total traffic of a device is related to total number of users and the DSP path of this device. As compared with its voice bandwidth, the signaling bandwidth of a device can be ignored.

Ethernet header = 208bit (26byte)
IP header = 160bit (20byte)
UDP header = 64bit (8byte)
RTP header = 96bit (12byte)

Using equation (4-2)-(4-5) we can calculate the voice bandwidth occupied by various Kinds of Voice Encoding/Decoding. Table 4-7 contains the calculated bandwidth occupied by Various Kinds of Voice Encoding/Decoding. The Detail calculation is in Appx B.

Table 4-7: Bandwidth Occupied by Various Kinds of Voice Encoding/Decoding

Parameters	**G.711**	**G. 723.1**	**G.729**
Framing interval (ms)	20	30	20
Bandwidth occupied (kbps)	89.78	22.49	33.78

Replacing different VoIP codec parameters of Table 4-2 into equation (4-7), we can calculate the maximum simultaneous VoIP nodes supported by NGN using equation (4-6) as in Table 4-8. The Detail calculation is in Appx B. Fig 4-3 shows the comparison of supported capacity using different VoIP codecs.

Table 4-8 Maximum VoIP nodes supported using different codec (Theory)

	Maximum simultaneous VoIP nodes		
Bit rate	**G.711**	**G.723.1**	**G.729**
11Mbps	122	463	320
5.5Mbps	61	231	160
2Mbps	22	84	58
1Mbps	11	42	29

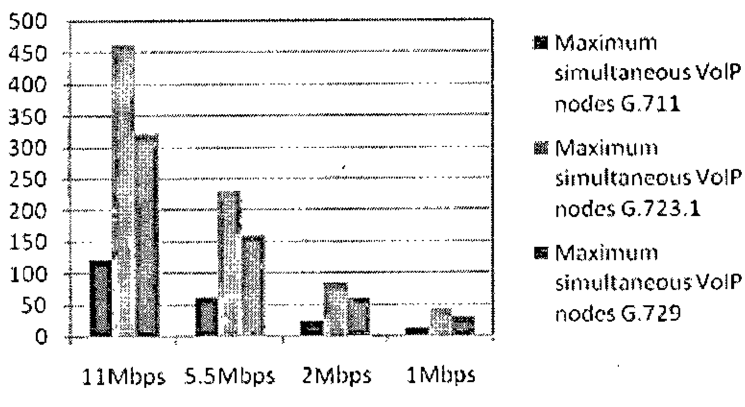


Fig 4-3: Maximum simultaneous VoIP nodes

From the above analytical results it can be observed that the maximum simulteneous VoIP nodes is supported by using G.723.1 codec.
Table 4-9 and Fig 4-4 shows the comparison of TDM and VoIP capacity

Table 4-9 TDM and VoIP capacity comparison

Bit rate	Maximum simultaneous nodes			
	TDM	VoIP		
	G.711	G.711	G.723.1	G.729
2Mbps	32	22	84	58

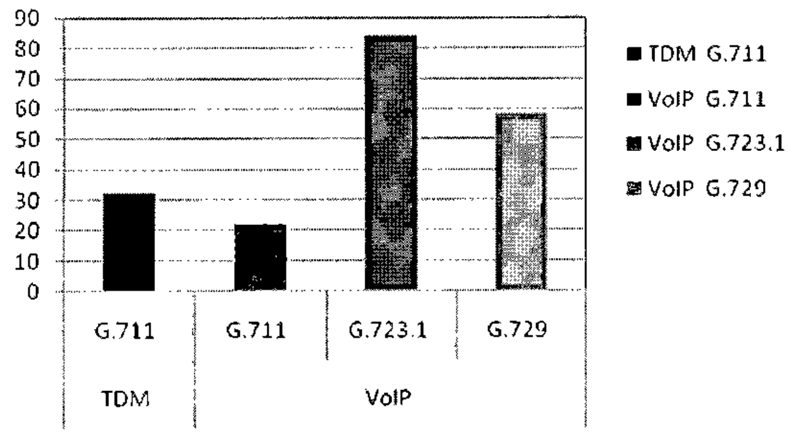


Fig 4-4: Comparison of TDM and VoIP capacity

4.3.2 Signal-to-quantization noise ratio

G.711

G.711 is applied for encoding telephone audio signal at a rate of 64 kbps with a sample rate of 8 kHz and 8 bits per sample. In an IP network, voice is converted into packets with durations of 5, 10 or 20*ms* of sampled voice, and these samples are encapsulated in a VoIP packet.

R = 8 bits per sample

G.729

G.729 coder is designed to operate with a digital signal obtained by first performing telephone bandwidth filtering (Recommendation G.712) of the analogue input signal, then sampling it at 8000 Hz, followed by conversion to 16-bit linear PCM for the input to the encoder.
The CS-ACELP coder is based on the Code-Excited Linear-Prediction (CELP) coding model. The coder operates on speech frames of 10 ms corresponding to 80 samples at a sampling rate of 8000 samples per second.

R = 16 bits per sample

G.723.1

G.723.1 is a standard for digital communications that employs 16-bit pulse-code modulation (PCM) at 5.3 or 6.3 kilobits per second (Kbps) with an input sample rate of 8 kilohertz (kHz). The encoding is done in frames having a duration of 30 milliseconds (ms). The look-ahead is 7.5 ms so the total algorithmic delay is 37.5 ms. The compression ratio can be as high as 12:1. In conjunction with voice activation detection (VAD), this provides exceptionally narrow signal bandwidth.

R= 16 bits per sample

Using Equation (4-16) we get the following signal to quantization noise ratio as in Table 4-10 using different codecs. The Detail calculation is in Appx C.

Table 4-10: SNR in dB using different codec

	G.711	G.729	G.723.1
Bits/ Sample	8	16	16
SNR in dB	49.8	97.8	97.8

From the above calculation and results it can be observed that better Signal to quantization noise ratio can be achieved for VoIP using G.723.1 and G.729 codec rather than Traditional TDM G.711 codec.

Since 6 dB of quantization noise is equivalent to 1 bit per sample by virtue of equation (4-16), the advantage of VoIP Codecs may also be expressed in terms of bit rate.

$10\log_{10} (SNR)_0 = 1.8 + 6R$
Where R = bit per sample

If R = 1 bit per sample then SNR will vary by 6 dB.
For voice signal sampling rate = 8000 samples/sec
Bit rate of voice signal using 1 bit per sample= 8000 X 1 = 8000 bits/sec = 8 kbps

Thus 1 bit can save 8 kbps bandwidth.

While in G.711, each sample is represented by 8 bit codeword, in G.729 and G.723.1 each sample is represented by 1 bit which provide a saving of about 56 to 58.7 kbps (i. e. 7 bits per sample) compared to the standard PCM. For G.729, its 56 kb/s and for G.723.1, its 57.7 to 58.7 kb/s.

4.3.3 Influenee of Channel noise on the Probability of error

Putting the values of E_b/N_0 in equation (4-36), we can calculate P_e. By replacing the value of Pe in equation (4-37) we can calculate the time interval between two bit errors. In this way, we can measure the influence of E_b/N_0 on the Probability of error.

Probability of error and the time interval between two errors has been calculated for different values of E_b/N_0. The results are as in Table 4-11. The results presented in the table assume a transmission bit rate 10^6 b/s. The Detail calculation is in Appx D.

Table 4-11 Influence of E_b/N_0 on the probability of error

E_b/N_0	Probability of error P_e	For bit rate of $1X10^6$ b/s, this is about one error every , T
4.3 dB	10^{-2}	10^{-4} Second
8.4	10^{-4}	10^{-2} Second
10.6	10^{-6}	1 Second
12.0	10^{-8}	2 Minutes
13.0	10^{-10}	3 Hours
14.0	10^{-12}	12 Days

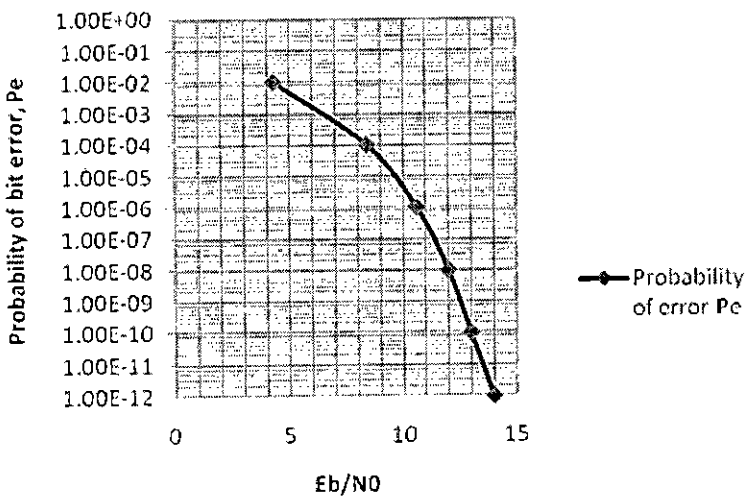


Fig 4-5: Probability of error, Pe Vs. Eb/N0

Fig 4-5 illustrates the Influence of E_b/N_0 on the probability of error. From the graph we can see that the probability of error decreases for increasing E_b/N_0.

<u>Error Threshold</u>

From the results in Table 4-11, it is clear that there is an error threshold (at about 11 dB).

Error threshold, $10\log_{10}(E_b/N_0) = 11\text{dB}$

For E_b/N_0 below the error threshold the receiver performance involves significant number of errors, and above it the effect of channel noise is practically negligible. In other words, if the ratio E_b/N_0 exceeds the error threshold, channel noise has virtually no effect on the receiver performance, which is precisely the goal of PCM. When, however, E_b/N_0 drops below the error threshold, there is a sharp increase in the rate at which errors occur in the receiver.

Signal to Noise ratio

The SNR in presence of channel noise can be calculated using equation (4-18) for different E_b/N_0. For sustainable probability of error E_b/N_0 should be equal to or more than error threshold. For E_b/N_0 equal to or more than the error threshold, the effect of channel noise will be negligible. In that case only the performance will be affected only by quantization noise. We calculate the SNR of different codec for different E_b/N_0 and compare the performance. Table 4-12 contains the calculated SNR and probability of error using different codec for different E_b/N_0. The Detail calculation is in Appx D. Fig 4-6 illustrates the comparison of the SNR using different codec.

The SNR using G.711, G.723.1 and G.729 for E_b/N_0 equal to error threshold can be calculated using equation (4-18) as follows:

For G.711, $10\log_{10}(SNR)_0 = 23dB$

For G.723.1, $10\log_{10}(SNR)_0 = 33dB$

For G.729, $10\log_{10}(SNR)_0 = 32dB$

Table 4-12 SNR and Probability of error using different codec for different E_b/N_0

E_b/N_0 dB	Signal-to-channel noise ratio in decibal (dB)			Probability of error P_e
	Traditional TDM (G.711)	NGN VoIP (G.723.1)	NGN VoIP (G.729)	
4.3	16.2	26.2	25.2	10^{-2}
8.4	20.3	30.3	29.3	10^{-4}
10.6	22.5	32.5	31.5	10^{-6}
12.0	23.9	33.9	32.9	10^{-8}
13.0	24.9	34.9	33.9	10^{-10}
14.0	25.9	35.9	34.9	10^{-12}

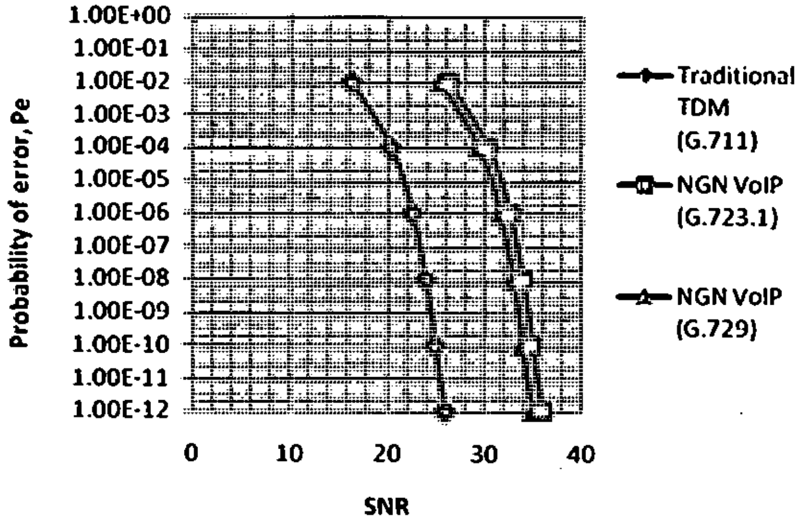


Fig 4-6: SNR vs. Probability of error of different codec

From the above calculation, results and graph, it is observed that for E_b/N_0 equal to or more than the error threshold, better Signal to channel noise ratio can be achieved using G.723.1 and G.729 codec rather than using G.711 codec.

It is found that the optimum signal-to-quantization noise advantage of G.723.1 and G.729 over standard PCM codec G.711 is in the neighborhood of 4 to 11 dB. For G.729 speech coder, it is 9 dB and for G.723.1 its 10 dB. The greatest improvement occurs in going from no prediction to first-order prediction, with some additional gain resulting from increasing the order of the prediction filter up to 4 or 5, after which little additional gain is obtained.

4.3.4 QoS

Using equation (4-38)-(4-41) and the equipment impairment factors for different codec as listed in Table 4-3 & 4-4; we can calculate the MOS value for different codec and compare it to ITU-T standard value as listed in Table 4-5 to acquire the service quality. Table 4-13 contains the calculated MOS value of different codec for different network conditions. The Detail calculation is in Appx E.

Table 4-13 Calculated MOS value of different codec for different network condition

Case	Delay (ms)	Jitter (ms)	Loss (%)	d (ms)	e	I_d	G.711			G.729		
							I_{ef}	R	MOS	I_{ef}	R	MOS
1	40	10	0.1	50	0.001	1.2	0.446	91.56	4.375	11.446	80.554	4.044
2	100	20	1	120	0.01	2.88	4.19	86.13	4.23	15	75.32	3.835
3	400	60	5	460	0.05	42.137	16.78	34.28	1.8	28	23.063	1.35

In Table 4-14, for the R factor values from the E-model are shown on the left, with their corresponding MOS values on the right. By comparing the MOS with ITU-T standard we can evaluate the service performance. The likely satisfaction level of human listeners is shown in the middle.

Table 4-14 Mapping between R values and estimated MOS

R	User Satisfaction	MOS
90-100	Very Satisfied	4.3-4.5 (Desirable)
80-90	Satisfied	4.0-4.3 (Desirable)
70-80	Some users dissatisfied	3.6-4.0 (Acceptable)
60-70	Many users dissatisfied	3.1-3.6 (Acceptable)
50-60	Nearly all users dissatisfied	2.6-3.1 (Not recommended
0-50	Not recommended	1-2.6 (Not recommended)

Table 4-15 represents the voice quality evaluation results (obtained by E-model analysis) in different network conditions using different codec.

Table 4-15 Voice quality performance in different network condition

<table>
<tr><th colspan="2">Parameters and services</th><th colspan="2">Good</th><th colspan="3">Poor</th><th colspan="2">Bad</th></tr>
<tr><td>ITU-T</td><td>MOS</td><td>4.0-5.0</td><td colspan="2">3.5-4.0</td><td>3.0-3.5</td><td colspan="2">1.5-3.0</td><td>0-1.5</td></tr>
<tr><td rowspan="3">Standard</td><td>Delay</td><td colspan="2">≤40ms</td><td colspan="3">≤100ms</td><td colspan="2">≤400ms</td></tr>
<tr><td>Loss</td><td colspan="2">≤0.1%</td><td colspan="3">≤1%</td><td colspan="2">≤5%</td></tr>
<tr><td>Jitter</td><td colspan="2">≤10ms</td><td colspan="3">≤20ms</td><td colspan="2">≤60ms</td></tr>
<tr><td rowspan="3">Voice</td><td>G.711</td><td colspan="2">Excellent</td><td colspan="3">Good</td><td colspan="2">Fair</td></tr>
<tr><td>G.729</td><td colspan="2">Good</td><td colspan="3">Good</td><td colspan="2">Poor</td></tr>
<tr><td>G.723.1</td><td colspan="2">Good</td><td colspan="3">Almost Good</td><td colspan="2">Fair</td></tr>
</table>

4.4 Cost Analysis of NGN and comparison with Existing Network

There are many reasons [5] why enterprises want to switch to converged network solutions. The two key driving forces behind corporate network convergence are cost and resource optimization.

1) Reduce OPEX due to proper utilization of transmission:

→In existing TDM network, 1 PCM or 1E1 (20048 kbps) has 32 (64 kbps) channel or time slot. One (64 kbps) channel or T.S. is used for synchronization and another 31(64 kbps) channels may use either 1 channel for signaling & 30 channels for bearer or total 31channels for bearer purpose. And every subscriber uses the specific channel during his conversation and other subscriber can not use or share that specific channel before disconnect his line. So by 1 PCM or 1 E1, maximum 31 (64 kbps) voice channels are used to transmit and receive the voice of the subscriber.

→But in NGN, as all are IP and there have no any specific channel. So by the technique of VAD (Voice Activity Detection) and silence suppression, new connected subscriber or existing subscriber can share or reuse other subscribers' bandwidth during their silence. And by using this process, approximately 60 to 62 (64 kbps) subscribers can use 2048 kbps bandwidth. So these technique we reduce the transmission cost from 100% to 50%. And by using various types of codec, we can compress the IP packet.

By using the coding/decoding technology G.729 a/b, we can compress the IP packet from 64kbps to 8kbps. So by this compression technique, NGN also can reduce the transmission cost from 50% to 85%.

2) Reduce OPEX due to transmission cost saving:

→In existing TDM network, to build national & international TDM channel based transmission route, our operators are connected to the national & international backbone through various HOP. And the operators have to pay large amount money for using these national & international TDM channel based routes and HOP. For this pure TDM call rate is high.

→But in NGN, as all are IP and the rent cost of IP bandwidth is so cheap. So by using the NGN, we can save the national & international transmission cost and the call rate of NGN will be less.

3) Reducc OPEX due to save the transmission link for signaling:

→In existing TDM network, as bearer (voice) and control (signal) are not separated. So same PCM or E1 are used for signal & voice and all channels of voice & signal arc connected to the main or mobile switching unit. For this transmission cost is high and not effective.

→But in NGN, as bearer (resource) & control are separated, so only controlling link is connected to the soft switch, which is in controlling layer, and the bearers (resources) are connected to media gateway. So need less transmission link for signaling which is cost effective.

4) Reduce OPEX & maintenance cost due to Centralized Service Control:

→In traditional TDM network, service control is not centralized. So to operate, maintain & upgrade every MSC, we need individual operator. As s result OPEX & maintenance cost is high.

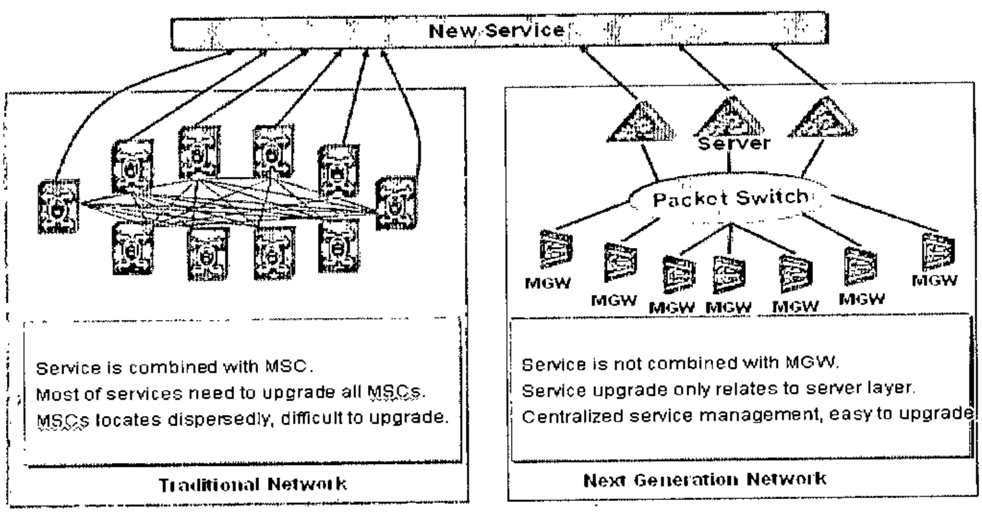


Fig 4-7: Reduce opex due to centralized service

But in NGN, due to centralized service control, we need less manpower to operate, maintain & upgrade the main server. Fig 4-7 shows the difference between distributed and centralized service control.

5) Reduce CAPEX & OPEX due to Distributed Architecture:

→As traditional PSTN network is not distributed, so all voice channel & signaling channel of PCM are connected to the main switching unit. From Fig 4-8 we can see that If any subscriber wants to make a local call from city C to City C, then voice & signaling both first go to City A, then it again come to City C; Which is not cost effective.

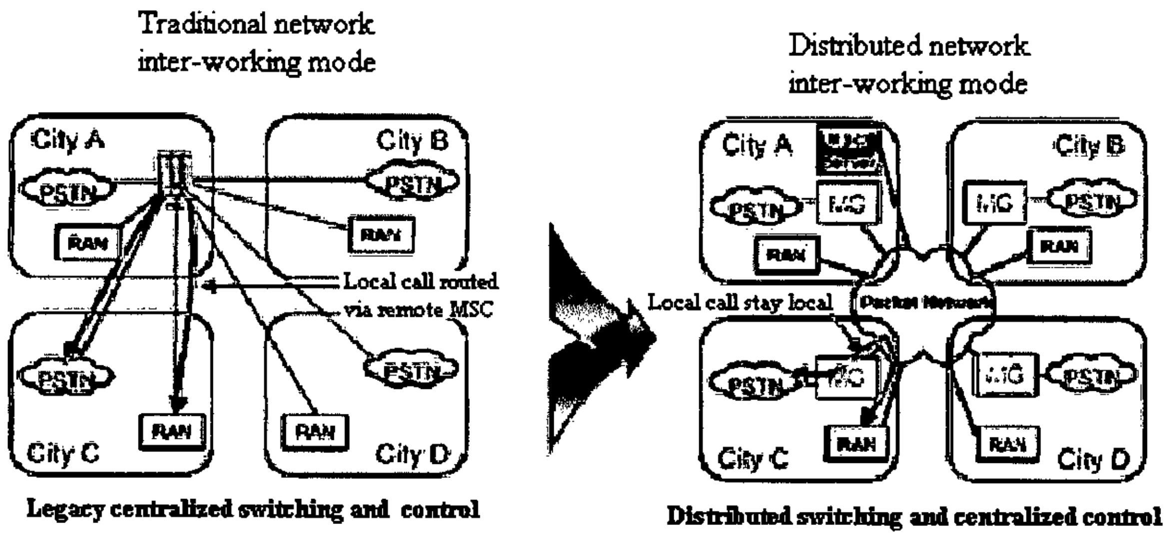


Fig 4-8: Advantage of distributed network

→But as NGN is distributed architecture. The advantage of distributed networking is as follows:

1. Traffic route is the best, network performance is the best.
2. Mostly suitable for the operators with wide coverage.

So transmission for signaling & voice are effectively used. Due to distributed architecture of NGN, if any subscriber wants to make a local call from City C to City C, only controlling signal goes to City A but voice not going to City A, it is locally connected. So here transmissions are effectively used.

6) Reduce OPEX due to less power consumption:

→As traditional TDM network is module based network (signaling module, trunk module & access module) and it is not centralized service control, so its power consumption & maintenance cost are high.

→But as NGN is gateway based distributed network & its service control is centralized, so its power consumption & maintenance cost are less.

Convergence gives the enterprise workforce the ability to make effectual decisions and act in real time using whatever tools are available from any location. InfoTech found that while many companies vary in why they choose VoIP, most enterprises have found the most common anticipated benefits as lowering total operating costs, enhancing end-user productivity, improving IT organization efficiency, reinforcing market differentiation and brand image11. Fig 4-9 [5] is the graphical representation of what companies say the initial interest is in deploying converged network infrastructure.

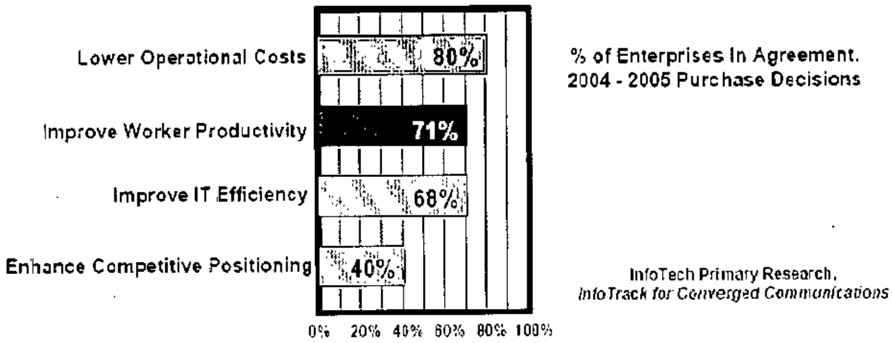


Fig 4-9: Business Drivers for Deploying Converged Solution

As it is made clear by Fig 4-9, cost cutting is one of the primary motivator and driving factors in the shift towards enterprise network convergence. But improving business efficacy and competence by revitalizing already available infrastructure, thus leading to resource optimization and enabling a user centric network design will be its accurate motive. [5]

CHAPTER 5 SIMULATION AND RESULTS

5.1 Simulation work

5.1.1 Approach

The Voice over 10 BASE-T Ethernet is simulated using the "Application Analyzer" Simulation tool of Huawei. Two widely used codecs for VoIP application are simulated, which are G.711 and G.729. Then incremented throughput test is done. It was achieved through generating traffic flow with changed packet size during each test period. In other words, the size of packet is increased gradually until it reaches at the maximum throughput. The highest traffic flow was carried out from 16:00:00 to 18:40:00.

From the simulation result, delay, jitter, packet loss, MOS and throughput is measured and compared. The Transmission Rating Factor, R and Mean Opinion Score (MOS) value is estimated from the simulation results by E-model analysis and compared to ITU-T standard to evaluate service performance. Then the maximum number of VoIP nodes supported simultaneously with acceptable R value is measured using throughput analysis. Moreover from the connected ratio measurement of the the traffic measurement report of two TDM and NGN operator, the capacity will be analyzed. The simulation approach flow has been illustrated in Fig 5-1.

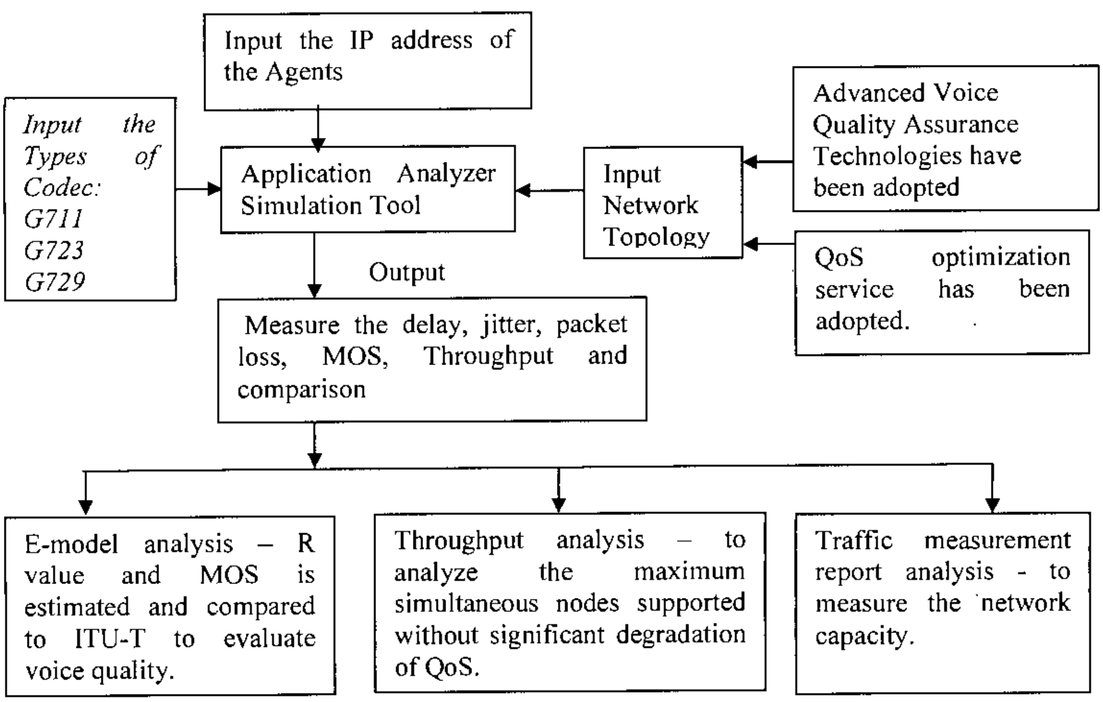


Fig 5-1: Simulation Approach

5.1.2 Network Topology

Traffic will be simulated using network simulator to acquire service performance. The test network topology is as in the Fig 5-2. In the simulation, five agents have been set for five VoIP nodes. In these cases, data rate depends on how much distortion presents in the environment as a function of distance to AP. Dotted lines indicate signaling path and solid line indicates voice path. Here form Agent 2 – Agent 5 and from Agent 4 – Agent 2 is carrying the voice both over the TDM and IP. Hence, from Agent 3 – Agent 4 the voice is going through IP.

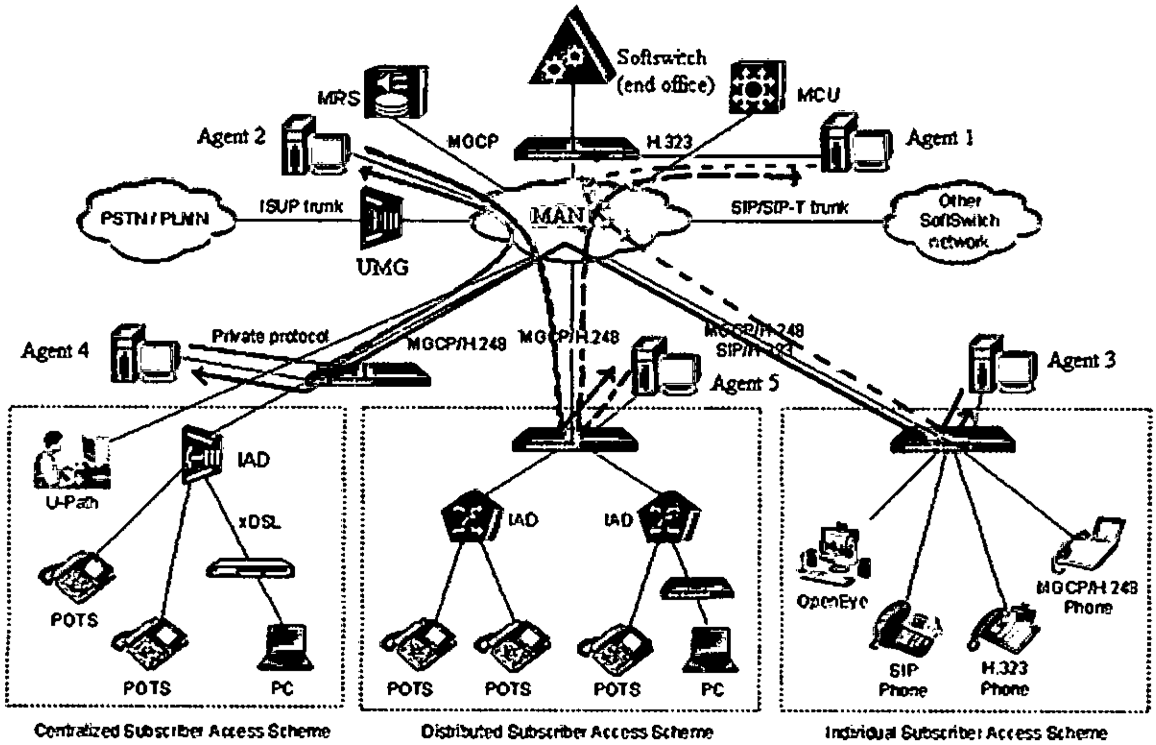


Fig 5-2: Network topology with 5 VoIP node

5.1.3 Audio/Voice Codec parameters

The main characteristics of the codec used in the simulation are summarized in Table 5-1.

Table 5-1 Audio/Voice codec parameters

Coding and decoding	Voice compression rate (kbps)	Packaging duration (ms)	Bandwidth coefficient (IP network)	Bandwidth coefficient (Ethernet network)
G.711 a/u	64	20	1.25	1.41
G.729 a/b	8	20	0.38	0.54
G.723.1 (6.3kbit/s)	6.3	30	0.27	0.37
G.723.1 (5.3Kbit/s)	5.3	30	0.25	0.36

5.1.4 Transmission parameters

The data rate for different IP transmission has been illustrated in Table 5-2.

Table 5-2 Ethernet and Wi-Fi data rates

Ethernet and WiFi technology standards	Data Rate (Mbps)
Gigabit Etherenet	1000
802.11n	540
Fast Ethernet	100
802.11g	54
802.11a	54
802.11b	11
10 BASE-T Ethernet	10
802.11	1

5.1.5 Test Parameters of the simulation tool

Default value of test parameters of the simulator has been listed in Table 5-3.

Table 5-3 Test Parameters

Parameter	Description	Default value
Sending Simulation Agents (Name, IP, Port)	It indicates a simulation agent that sends packets with time stamps.	
Receiving Simulation Agents (Name, IP, Port)	It indicates a simulation agent that receives packets.	
Conversation Duration	It indicates how long a VoIP conversation lasts.	60 seconds
Number of VoIP Calls	It indicates the number of concurrent VoIP calls on the same connection.	1
Number of Measurements	It indicates the number of measurements on the connection. By default, no measurement is conducted.	1
Frame Packing	It indicates the time spacing of sending VoIP packets (unit: millisecond) and the size of packet (sampling frame).	20 milliseconds (two samples

G711 Payload Type	Determines the type of analog to digital converter emulated by the test. To emulate VoIP sessions according to the method used in the USA, Select PCMU; to emulate VoIP sessions according to the method used in Europe and China, select PCMA.	PCMU (64000 bps)
Use PLC	It can facilitate in calculating the MOS. If it is selected, the PLC algorithm is activated.	Not used.
Silence Suppression	If it is selected, the G.711 VAD algorithm is activated.	
Jitter Buffer	If it is selected, the jitter buffer management is activated. If the jitter buffer management is activated, the simulation agent re-aligns the received IP packets with the voice sample packet passing through the IP network simultaneously.	
Jitter Buffer Length	It indicates the length of jitter buffer. It is validated only when the Jitter Buffer is activated.	4 packets
Initial Playout Delay	It indicates the minimum number of packets in the jitter buffer before a packet enters into an encoder/decoder. It is validated only when the Jitter Buffer is activated.	2 packets
Delay Threshold	If average delay exceeds the delay threshold, an alarm is generated.	150 milliseconds
Loss Threshold	If packet loss exceeds the loss threshold, an alarm is generated.	3%
Jitter Threshold	If the absolute value of jitter buffer exceeds the jitter threshold, an alarm is generated.	10
Throughput Low Threshold	If the actual throughput is less than the Throughput Low Threshold, an alarm is generated.	60 kbps
MOS Threshold	If the actual MOS is less than the MOS threshold, an alarm is generated.	4
Quality of Service	It indicates the transmission priority based on DiffServ or TOS.	DiffServ(0)
Base RTP Port	If the base RTP port is configured, the consecutive even-numbered ports will be occupied. If it is the default value, the system allocates ports automatically.	0
Max Network Delay	Max network delay is used for calculating test parameters, such as test time and timeout.	3000 milliseconds

5.1.6 Service Performance in Different Network Conditions

The tested results are compared with the ITU-T defined MOS value to acquire the service performance as in Table 5-4.

Table 5-4: Standard QoS Indexes for an IP Bearcr Network

Parameters and services		Good		Poor	Bad	
ITU-T	MOS	4.0-5.0	3.5-4.0	**3.0-3.5**	**1.5-3.0**	**0-1.5**
Standard	Delay	≤40ms		≤100ms	≤400ms	
	Loss	≤0.1%		≤1%	≤5%	
	Jitter	≤10ms		≤20ms	≤60ms	
Modem	Transparent transmission	Available		**Unavailable**	**Unavailable**	
Fax	Transparent transmission	Available		**Unavailable**	**Unavailable**	
	T.38	Available		Available	**Unavailable**	
Voice	G.711 a/u	Excellent		Good	**Fair**	
	G.729 a/b	Good		Good	**Poor**	
	G.723	Good		Almost good	**Fair**	
Video	384K	Not bad and available		Slightly bad and available	**Significant bad and unavailable**	
Dual dialing	RFC2833	Available Good or excellent voice quality		Available Almost good voice quality	Available **Fair or poor voice quality**	
Message		Available		Available	Available	
The bold fonts indicate that the QoS indexes do not meet service requirements.						

5.2 RESULTS

5.2.1 QoS Measurements Analysis

The QoS measurements resulted from simulation are analyzed in this section. The following VoIP test was carried out on the above network using the Application analyzer tool of Huawei. Two widely used codecs for VoIP application were simulated, which are G.711 and G.729.

1. G.711 codec simulation

Time: Oct. 29 (Wednesday 15:10–22:10)
Duration of each call: three minutes
Call interval: two minutes

2. G.729 codec simulation

Time: Oct. 30 (Thursday 14:00–22:01)
Duration of each call: one minute
Call interval: one minute

Now, the test measurement results will be compared to the ITU-T standard as in Table 5-3 to acquire the service performance as follows:

Delay

The time that voice spend in going from the originator to the receiver for end-to-end packet transmission. Unit: ms. While in circuit-switched networks end-to-end-delay only adds up to about 10 ms and therefore does not constitute a problem for voice quality in these networks at all, it is significantly higher when carried over packet-oriented networks (since there are additional sources of delay) as many small size packets are generated with variant inter-arrival time [7]. The simulation result of G.711 and G.729 as shown in Fig 5-3 and Fig 5-4 shows that delay for G.729 is bigger than delay for G.711.

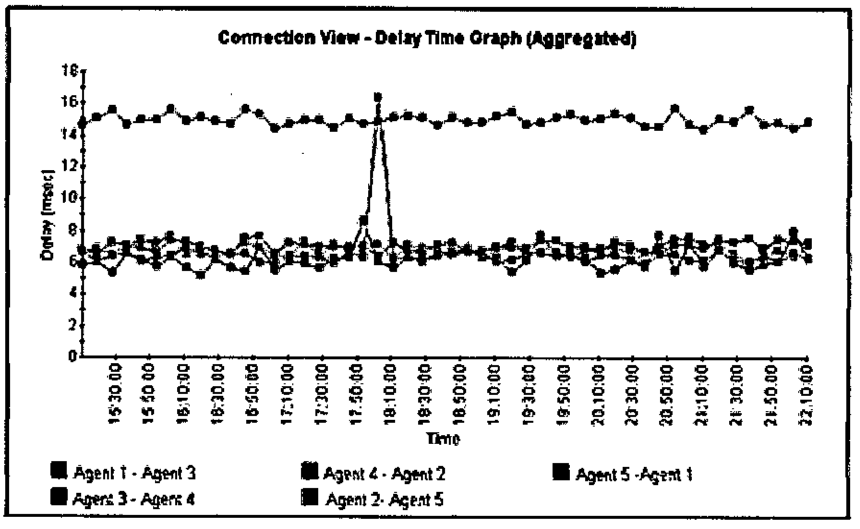


Fig 5-3: G.711 Delay Time Graph (Test result data available on Oct 29, 2008 3:10 PM-10:10 PM)

Analysis: All the end-to-end delay of tested pairs is less than 20ms. The end-to-end delay of the voice path from Agent 3–Agent 4 is from 6-9 ms. That indicates the indexes are good.

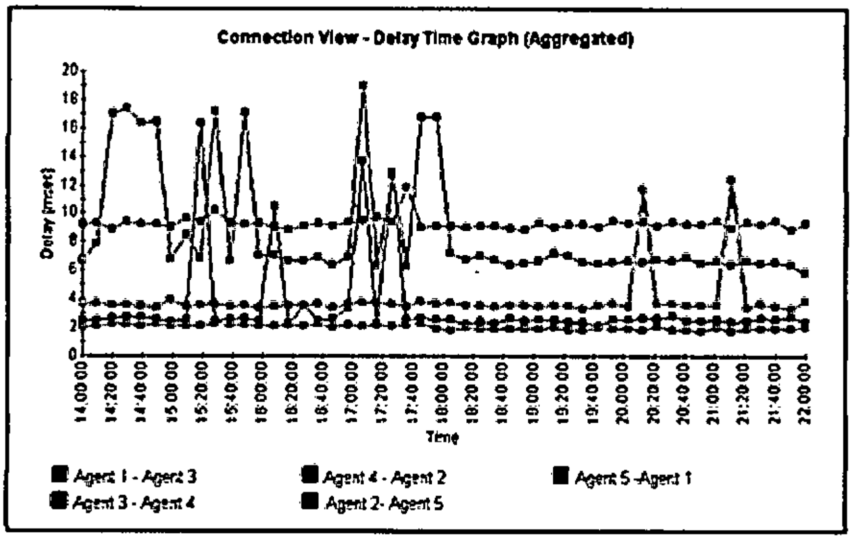


Fig 5-4: G.729 Delay Time Graph (Test result data available on Oct 30, 2008 2:00 PM to 10:00 PM)

Analysis: All the end-to-end delay of tested pairs is less than 20ms. From Agent 3–Agent 4, it is from 2-16 ms. that indicate the indexes are good.

Jitter

Jitter is defined as a variation rate in the delay of received packets. The simulation result as shown in Fig 5-5 and Fig 5-6, shows that jitter delay for G.729 is bigger than G.711 but the jitter is less significant.

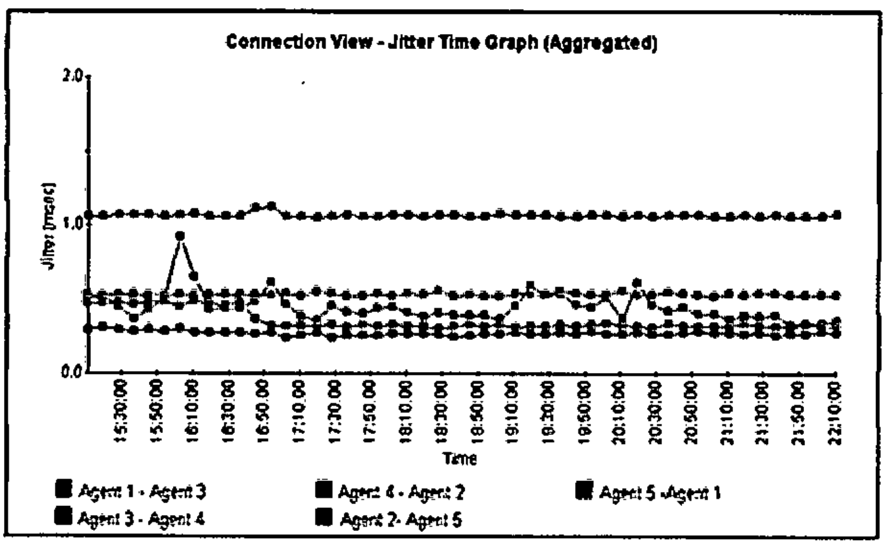


Fig 5-5: G.711 Jitter Time Graph (Test result data available on Oct 29, 2008 3:10 PM-10:10 PM)

Analysis: All the end-to-end jitters of tested pairs are less than 1ms or around 1ms. That indicates the indexes are good.

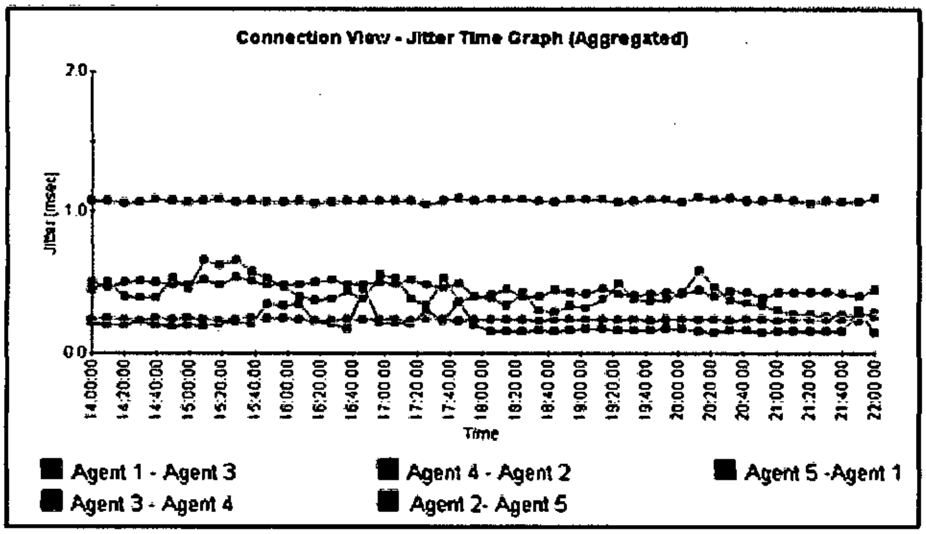


Fig 5-6: G.729 Jitter Time Graph (Test result data available on Oct 30, 2008 2:00 PM to 10:00 PM)

Analysis: All the end-to-end jitters of tested pairs are less than 1ms or around 1ms. That indicates the indexes are good.

Packet Loss Ratio

The loss of voice data in circuit-switched networks lie somewhere within one tenth of a percent. But loss of data packets is an absolutely normal occurrence in TCP/IP networks and occurs for two reasons; the first one being traffic blockages in the network that result in a rising number of queues and consequently in the rejection of data packets. The second reason for the loss of data packets is high delay, through which packets arrive behind schedule in the jitter buffers and consequently get discarded [7]. Packet loss is expressed as a ratio of the number of packets lost to the total number of packets transmitted. Packet losses results when packets sent are not received at the final destination. Generally, packet loss is related with the packet length, which is proportional to transmission time associated with each packet. The simulation result Fig 5-7 and 5-8 shows that the G.729 faces more packet loss than G.711.

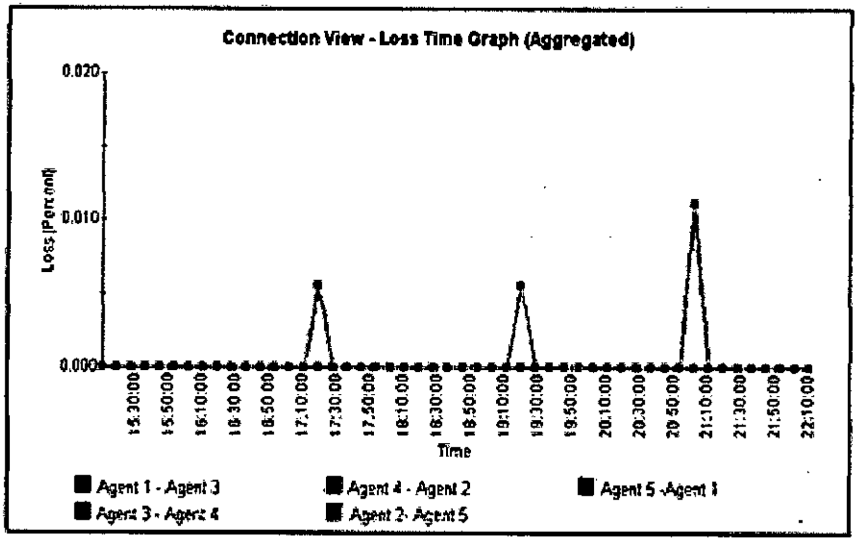


Fig 5-7: G.711 Loss Time Graph (Test result data available on Oct 29, 2008 3:10 PM-10:10 PM)

Analysis: All the end-to-end packet loss ratios of tested pairs are almost 0, even some instant packet loss ratios are much less than 0.1%. That indicates the indexes are good.

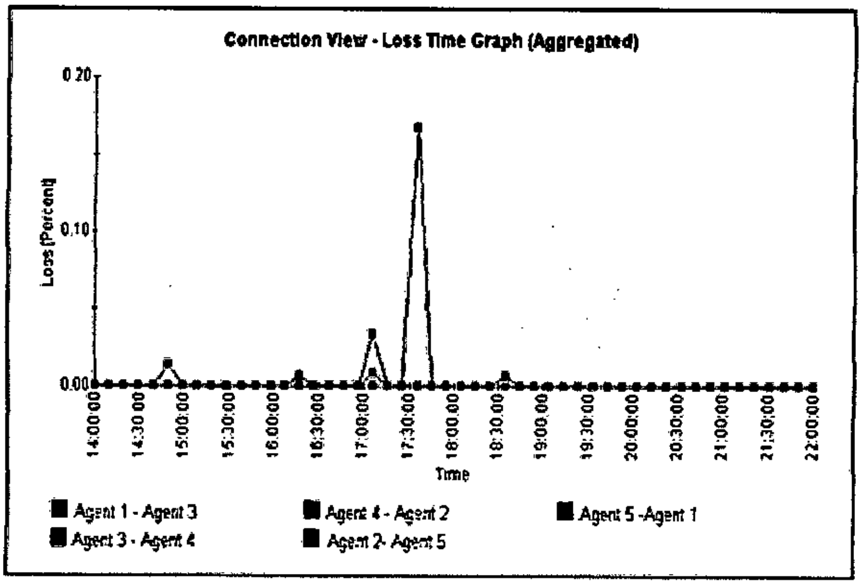


Fig 5-8: G.729 Loss Time Graph (Test result data available on Oct 30, 2008 2:00 PM to 10:00 PM)

Analysis: All the end-to-end packet loss ratios of tested pairs are almost 0, only Agent 3 – Agent 4 appears a 0.18% instant packet loss at 17:35. That indicates the indexes are good.

MOS

The simulation results as shown in Fig 5-9 and Fig 5-10 shows that MOS value for G.711 is bigger than that of G.729.

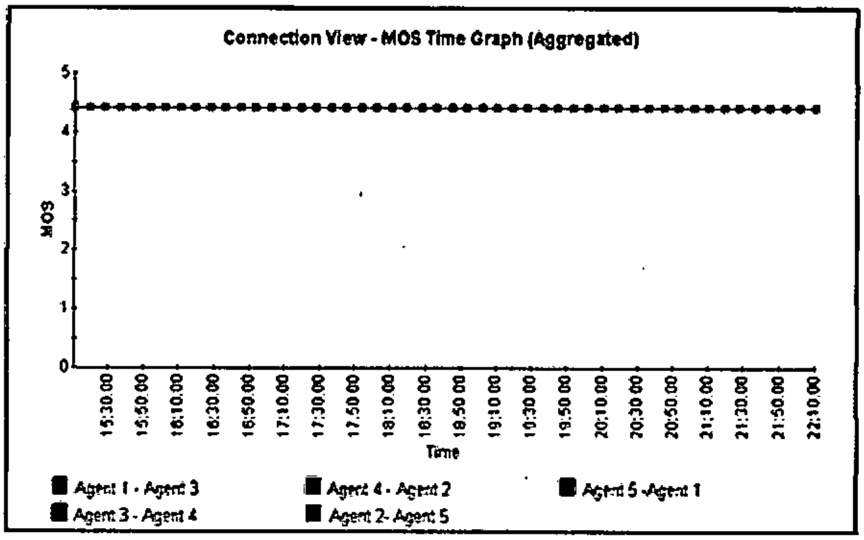


Fig 5-9: G.711 MOS Time Graph (Test result data available on Oct 29, 2008 3:10 PM-10:10 PM)

Analysis: All the end-to-end MOSs of tested pairs is about 4.4. That indicates the indexes are good.

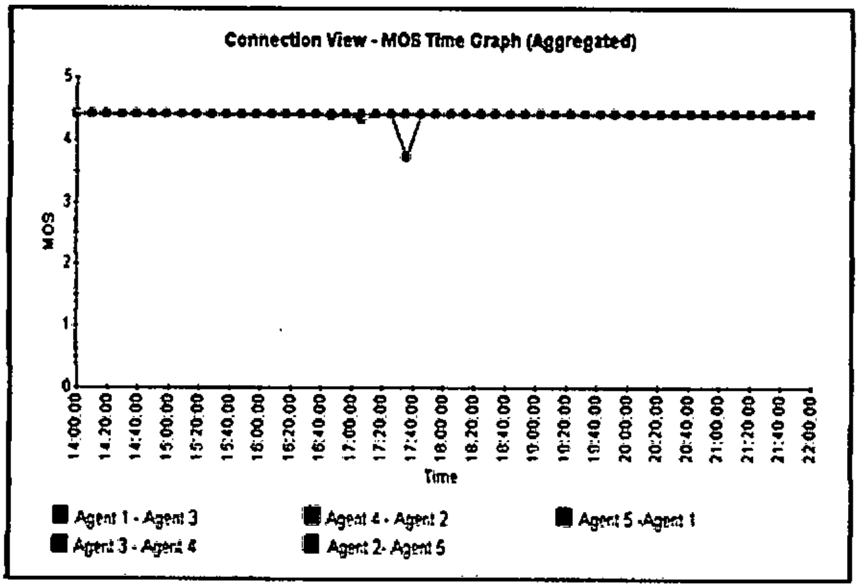


Fig 5-10: G.729 MOS Time Graph (Test result data available on Oct 30, 2008 2:00 PM to 10:00 PM)

Analysis: All the end-to-end MOSs of tested pairs is about 4.3. That indicates the indexes are good. (Affected by the 0.18% packet loss of Agent 3 – Agent 4 at 17:35, the MOS descended to about 4.00 at that moment.)

Throughput

The throughput (measured in bps) corresponds to the amount of data in bits that is transmitted over the channel per unit time. This test can suggest the maximum number of calls supported by simulation agents, as well as medium and final voice performance measurement.

From the simulation results as shown in Fig 5-11 and Fig 5-12, we obtain that the maximum throughput for G.711 is 0.75 Mbps and for G.729 is 0.35 Mbps. Thus, the throughput result for G.711 is better than G.729.

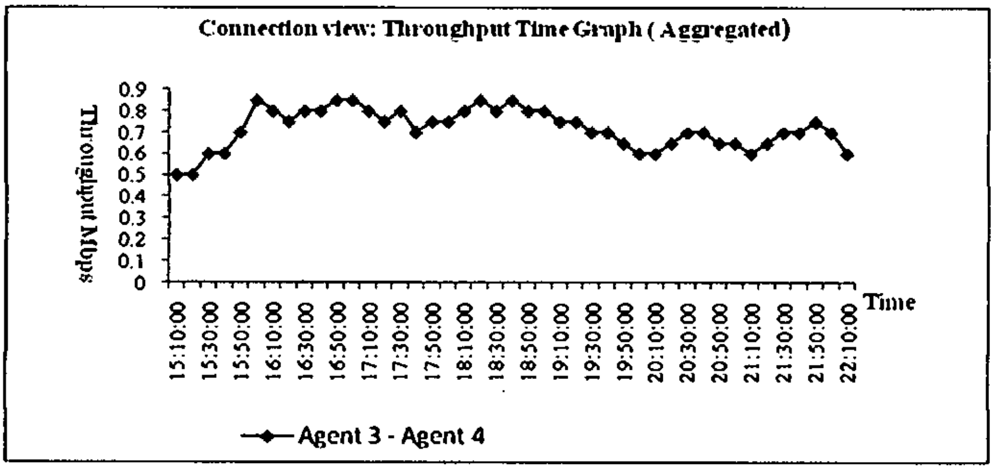


Fig 5-11: G.711 Throughput Time Graph

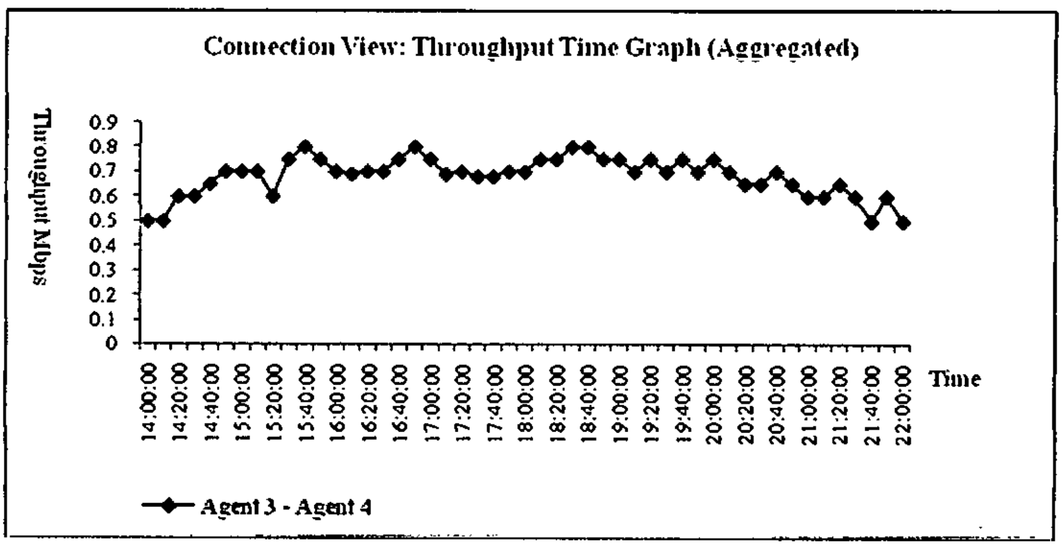


Fig 5-12: G.729 Throughput Time Graph

5.2.2 Simulation analysis (E-Model) of VoIP Quality of Service

The E-Model calculates the R, using the network impairment factors. Once the value of R is calculated from these factors, an estimate of the MOS can be calculated. Using the formula defined in equation (4-28)-(4-31) we can calculate MOS value from the network impairment factors delay, jitter and packet loss.

From the simulation measurement results at the peak hours (16:00 to 18:40), we can see that at All the end-to-end packet loss ratios of tested pairs are almost 0, only Agent 3 – Agent 4 appears a 0.18% instant packet loss at 17:35. The calculated R value and MOS from the simulation result of the VoIP traffic between Agent 3 – Agent 4 at 17:35 is as in Table 5-5. Detail calculation has been shown in Appx F.

Table 5-5: The voice quality calculated from simulation result (at peak hour) at some point where the packet loss value deteriorate

	From the simulation results			E-Model analysis	
	Delay (ms)	**Jittter (ms)**	**Packet Loss (%)**	**R-Value**	**MOS**
G.711	6.8	0.3	0	93	4.4
G.729	12	0.25	0.17	81	4

Comparison of the MOS to the ITU-T standard shows that the result is excellent and very satisfactory for G.711. For G.729, it is good and satisfactory.

Except the specific point where the packet loss value deteriorates, the average delay, jitter and packet loss (at peak hour) is 6.5 ms, 0.3 ms and 0% for G.711 and 7.5 ms, 0.3 ms and 0.09% for G.729. The calculated R value and MOS for the VoIP traffic between Agent 3 – Agent 4 for other points of the peak hour is as in Table 5-6. Detail calculation has been shown in Appx F.

Table 5-6: The voice quality calculated from the average of the simulation results (at peak hour).

	From the simulation results			E-Model analysis	
	Delay (ms)	**Jitter (ms)**	**Packet Loss (%)**	**R-Value**	**MOS**
G.711	6.5	0.3	0	93	4.4
G.729	7.5	0.3	0.09	81.6	4

Comparing the calculated MOS to the ITU-T standard, we see that the result is excellent and the users are very satisfied for G.711. For G.729, the result is good and the users are satisfied.

From the above E-Model calculation from the simulation results, we found that MOS value for G.711 is bigger than that of G.729. Thus better QoS or voice quality of VoIP can be achieved using G.711 codec rather than G.729. But we also found from the result that the QoS is not significantly degraded for G.729 codec.

5.2.3 Simulation analysis (Throughput) of VoIP Capacity

By using values of maximum achievable throughput from simulation, VoIP capacity can also be evaluated. The following formula is applied to get the maximum supportable Voice nodes over IP.

$$1/T_{avg} = \text{Maximum Throughput/ Data Rate} \qquad (5\text{-}1)$$

From the simulation result we observed the maximum throughput at peak hours is T_{avg} (G.711) = 0.85 Mbps and T_{avg} (G.729) = 0.8 Mbps.

The bandwidth required using G.711 is 89.78 kbps and using G.729 is 33.78 kbps as we calculated in Table 4-7. Thus by using the throughput value obtained from the simulation, we can calculate the maximum simultaneous nodes supported using G.711 and G.729.

Thus for G.711, $1/T_{avg}$ = 0.85 Mbps / 89.78 kbps = 850000/89780 = 9
For G.729, $1/T_{avg}$ = 0.8 Mbps / 33.78 kbps = 800000 / 33780 = 23

Thus it can be said that more VoIP nodes can be supported using G.729 codec rather than G.711 codec without significantly degraded the voice quality.

5.2.4 Simulation analysis (Traffic measurement report) of VoIP Capacity

By analyzing the call attempts and call connected ratio at peak hour the capacity can be calculated as follows:

$$\text{Call Connected ratio} = \frac{\text{Number of call attempts}}{\text{Number of call connected}} \text{ X } 100$$

We analyze the traffic measurement report of TDM based core network of Dhaka Telephone Company Limited (one of the PSTN operator of Bangladesh) and IP based GMSC network of Teletalk (one of the mobile operator of Bangladesh) for the outgoing call traffic. The tool used to illustrate this is the "Performance Measurement tool of HUAWEI Local Maintenance terminal" was used for the traffic measurement. The call connected ratio for TDM based network of DTCL and IP based network of Teletalk have been illustrated in Fig 5-13 and Fig 5-14.

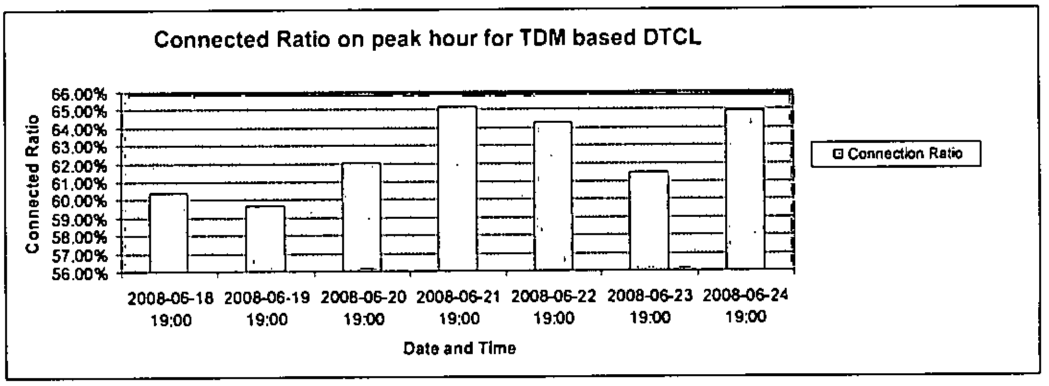


Fig 5-13: Call connected Ratio diagram at peak hour for TDM based network

The call connected ratio of IP based network of Teletalk has been illustrated as below:

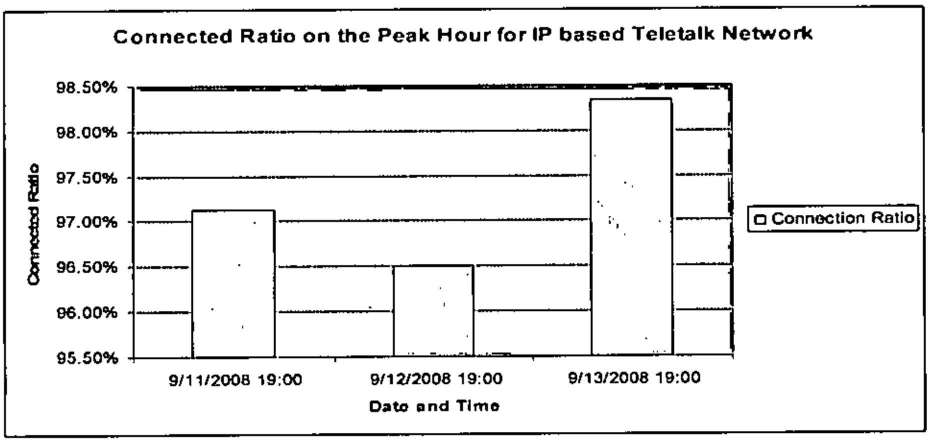


Fig 5-14: Call Connected Ratio diagram at peak hour for IP based network

We can see from Fig 5-13 and 5-14 that at the peak hour the call connected ratio for TDM based network is 59.5%-65% whereas for IP based network its 96.5%-98.5%. So, we can say that for IP based network the call connected ratio is far better than TDM based network. This achievement has been possible only because of the proper usage of the resource capacity by IP based network.

CHAPTER 6 CONCLUSION AND SUGGESTION

6.1 Conclusion

The performance analysis and simulation results show that due to the best utilization of the available capacity and resource improved capacity can be achieved using G.729 and G.723.1 codecs which are supported by NGN rather than using G.711 which is used in TDM. Thus, NGN is capable of supporting improved capacity than TDM based network.

In this work, the signal-to-quantization noise ratio has been analyzed and the analytical results depict that due to the higher degree of quantization, better SNR can be obtained using G.723.1 and G.729 codecs supported by NGN rather than the G.711 codec used in TDM. The bit rate has been analyzed and the result depicts that 1 bit can save 8 kbps bandwidth and by using G.729 or G.723.1 codec 56 to 58.7 kb/s (i. e. 7 bits per sample) of bandwidth can be saved compared to the standard PCM G.711.

The influence of channel noise on the probability of error has been analyzed and the result show that probability of error is increased for decreased E_b/N_0. From the result, an error threshold has been detected for which the probability of error is negligible. If E_b/N_0 is equal to or more than the error threshold, the effect of the channel noise is negligible. Thus to achieve better performance, E_b/N_0 should not be less than or equal to the error threshold. The Signal-to-Noise ratio in presence of channel noise has been analyzed and the result show that better SNR can be achieved using G.723.1 and G.729 codecs rather than G.711 codec with sustainable probability of error. The optimum signal-to-quantization noise advantage of G.723.1 and G.729 over standard PCM codec G.711 is in the neighborhood of 4 to 11 dB. For G.729 speech coder (CS-ACELP), it is 9 dB and for G.723.1 its 10 dB.

The effect of the quality impairment factors on the quality has been analyzed by E-Model analysis. The result shows that better quality of voice can be obtained using G.711 codec rather than G.729 codec.

The traffic over the proposed NGN has been simulated and simulation results have been verified by theoretical analysis. The results were compared to ITU-T standard to acquire service performance. The results depict that better voice quality can be achieved using G.711 codec and voice quality at satisfactory level can be achieved using G.729 codec.

In the NGN migration, the Advanced Voice Quality Assurance Technologies and QoS optimization services was adopted to reduce the affect of the quality impairment factors. The analysis and the simulation results shows that improved capacity can be achieved using G.729 (supported by NGN) rather than using G.711 without significantly degrading the QoS.

Thus for improved voice quality over NGN, Advanced Voice Quality Assurance Technologies and QoS optimization services should be adopted.

6.2 Further Recommendation for Future Work

The future work should be on the NGN QoS signaling and control solutions and development of protocols, speech encoders and optimization services for the implementation of the high quality voice service.

Several control architectures have been developed [16, 17] for supporting the QoS in the packet-based network. ITU-T RACF provides the general architecture covering both access and core networks. The current RACF specifies the functional architecture and control procedure in the IP level. There are still many open issues, and continuing effort is under way to solve the issues. QoS control in the transport technology dependent aspect is one issue. The general framework for flow aggregation and signal aggregation is one of the major issues for the overall QoS architecture. The high complexity and scalability of the control mechanism is another issue. Since the QoS control signaling is on a per-call basis, the QoS control in the core network will be a burden. For real implementation, the complexity of the QoS control should be optimized. For example, the call-by-call QoS control mechanism can be activated only when the network monitoring system detects that the performance degrades. To reduce the complexity, the part control function can be embedded in the transport equipment or combined with the management function. For the new services such as fixed mobile convergence and IPTV services, QoS control for mobility and the multicast condition must be developed.

APPENDIX

Appendix A TDM bandwidth and capacity Calculation

Channel capacity

The audio signal bandwidth is 15 kHz. In telecommunication, the components above 3400 Hz are eliminated by a low-pass filter.

So, f= 4000

Sampling rate, S = 2 X 4000 = 8000 samples/sec

The resulting signal is then sampled at a rate of 8000 samples per second.

Thus, G.711 uses a sampling rate of 8,000 samples per second. Non-uniform quantization with 8 bits is used to represent each sample.

$L= 2^8 = 256$

R= ln (L) = 8

Channel capacity of G.711, B = S X R = 8000 X 8 = 64 kbps

Transmission capacity

To allow better utilization of a given communication link, time-division multiplexing **(TDM)** systems, where voice signals from a number of users are digitized and then the resulting bits are time compressed for transmission over the same link.

T1

One of the first PCM standards - developed by Bell Labs

- 24 telephone channels, sampling rate of each channel is 8000 Hz, with 8 bit μ law samples
- Bits per frame = 24 x 8 + 1 framing bit = 193 bits each 1/8000 seconds
- Total bit rate (R) = 1.544 Mbit/s (This is called the "T1" rate.)

E1

- 32 channels : 30 for data + 2 for signaling

- Sampling rate of each channel is 8000 Hz, with 8 bit samples.

Bits per channel = 8 X 8000 = 64 kbps

- Each group of four frames provides 64 bits of signaling : half for channel specific + half for frame sync

- Total bit rate (R) : 32 X 8 X 8000 = 2.04 Mbps

TDM network Capacity

The network capacity can be calculated using equation (4-1). For G.711 TDM Codec (Table 4-1), Channel Capacity/bandwidth = 8000 X 8= 64 kbps. The TDM network capacity using E1 transmission data rates (Transmission Rate) is

TDM network Capacity = Transmission Capacity/ Channel Capacity = 2. 04 Mbps/ 64 kbps = 31.87 = 32

Appendix B VoIP bandwidth and capacity calculation

Calculation of Bandwidth Occupied by Various Kinds of Voice Encoding/Decoding

In an NGN, after voice has been encoded/decoded, it will be packetized by using the RTP and sent to an IP network. If there are different encoding/decoding types and actual packetization durations, bandwidth demands in bearer network will be different. Audio/Voice codec parameters of VoIP codecs have been illustrated in Table 4-2.

Ethernet header = 208bit (26byte)
IP header = 160bit (20byte)
UDP header = 64bit (8byte)
RTP header = 96bit (12byte)

Usinh equation (4-2) we get OH_{hdr} = 208+160+64+96 =528 bit
Therefore, Packet length =528bit+payload

Using (4-5) we get, Bandwidth = (528 / packetization duration) + number of bits per second
G711: 64k payload bits/second; G729: 8k payload bits/second.
G723: 5.3k and 6.3k

Therefore:
G711, packetization within 20ms,
bandwidth: [528/ (0.020 × 1024) + 64] kbit/s= 89.78125 kbit/s
G711, packetization within 30ms,

bandwidth: [528/ (0.030 × 1024) + 64] kbit/s= 81.1875 kbit/s
G729, packetization within 20ms,
bandwidth: [528/ (0.020 × 1024) + 8] kbit/s= 33.78125 kbit/s
G723, 5.3k, packetization within 30ms,
bandwidth: [528/ (0.030 × 1024) + 5.3] kbit/s= 22.4875 kbit/s

Exemplified Calculation of Bandwidth Occupied by Voice Encoding/Decoding

Bandwidth = total traffic × bandwidth occupied by each kind of encoding/decoding × 2 [2 indicates bidirectional]

Total traffic (less than or equal to the total number of DSP paths) = traffic per line × total number of users

Example:
AMG adopts the PVM board (68-path DSP). When there are 100-line users, the traffic per line is 0.2erl. AMG adopts G.711A for communication and has the packetization duration of 20ms

Average bandwidth occupied by voice:
100 × 0.2 × 89.78125 × 2=3591.25kbit/s=3.507mbit/s
Average bandwidth occupied by voice in full load:
68 × 89.78125 × 2=12210.25kbit/s=11.92mbit/s

Calculation of Signaling Bandwidth

Before calculating signaling bandwidth, we need to obtain the following data:

1. Average number of message packets necessary for completion of each call;
2. Average number of the bits in each message packet during a call;
3. Call Attempts Per Second (CAPS).
4. Protocol stack of signaling

MGCP/H.248/SIP/H323
UDP/TCP/SCTP
IP
MAC (Ethernet)

For the sake of simplicity, signaling bandwidth of transmission layer is calculated in compliance with UDP. Then, the overhead of each protocol packet is as follows:

Network overhead =UDP header+IP header+Ethernet header=64+160+208=432 bit

The formula for calculating signaling bandwidth is as follows:
Signaling bandwidth = (CAPS × average number of message packets in each call) × (average number of bits in each message packet+number of network overhead bits in each message packet)

Average payload in each protocol packet (empirical value):
MGCP/SIP/H323/H248 180byte M2UA: 48byte

On average, each call needs to interact with 16 messages (empirical value)

Example: If one AMG interacts by adopting the H248 message and its CAPS is 2, then the signaling bandwidth is
[(2 × 16) ×(180 × 8+432)]/1024=58.5Kbit/s

The total bandwidth occupancy of NGN is related to encoding/decoding mode, packetization duration, and traffic

The total traffic of a device is related to total number of users and the DSP path of this device
As compared with its voice bandwidth, the signaling bandwidth of a device can be ignored.

Capacity Calculation
Using equation (4-2) we get,

$$OH_{hdr} = (208+160+64+96) = 528 \text{ bits} = 66 \text{ Byte}$$

Using equation (4-7) we can calculate the average time between the transmissions of two consecutive packets T. If the datarate = 2 Mbps= 2000000 bit/sec

$$\text{For G.711, } T = \frac{(\text{Payload} + \text{OHhdr})\ X\ 8}{\text{datarate}} = \frac{(160+66)\ X\ 8}{2000000} = 0.000904$$

$$\text{For G.723.1, } T = \frac{(\text{Payload} + \text{OHhdr})\ X\ 8}{\text{datarate}} = \frac{(24+66)\ X\ 8}{2000000} = 0.00036$$

$$\text{For G.729, } T = \frac{(\text{Payload} + \text{OHhdr})\ X\ 8}{\text{datarate}} = \frac{(20+66)\ X\ 8}{2000000} = 0.000344$$

From equation (4-6) we get,

For G.711, 1/T= n X 50 = 1/0.000904 = 1106
So, n = 1106/50 = 22 sessions

For G.723.1, 1/T= n X 33 = 1/0.00036 = 2777
So, n = 2777/33 = 84 sessions

For G.729, 1/T= n X 50 = 1/0.000344 = 2906
So, n = 2906/50 = 58 sessions

Appendix C Signal to quantization noise ratio calculation

Using equation (4-16) the signal to quantization ratio can be calculated as follows:

G.711
Bits per sample = 8
Thus SNR is dB, $10\log_{10}(\mathrm{SNR})_0 = 1.8 + (6X8) = 49.8\mathrm{db}$

G.729
Bits per sample =16
Thus SNR is dB, $10\log_{10}(\mathrm{SNR})_0 = 1.8 + (6X16) = 97.8\mathrm{db}$

G.723.1
Bits per sample =16
Thus SNR is dB, $10\log_{10}(\mathrm{SNR})_0 = 1.8 + (6X16) = 97.8\mathrm{db}$

Appendix D Signal to channel noise ratio calculation

Let E_b/N_0= 10.6 dB, P_e= 10^{-6} which has been calculated using equation (4-36). Lets bit rate = 10^6 b/s. The time interval between two errors can be calculated using equation (4-37) as follows,

$T = 1/(P_e \text{ X bit rate}) = 1/(10^{-6} \text{ X } 10^6) = 1$ sec

Probability of error and the time interval between two errors can be calculated in the same way for different values of E_b/N_0. The results are as in Table 4-11. The SNR for G.711, G.723.1 and G.729 can be calculated using equation (4-18) as follows:

For G.711,

$$
\begin{aligned}
10\log_{10}(SNR)_0 &= 10\log_{10}(E_b/N_0) + 10\log_{10}(f_b/B) \\
&= 11 + 10\log_{10}(10^6/64000) \\
&= 11 + 11.938 \\
&= 22.9 = 23 \text{ dB}
\end{aligned}
$$

For G.723.1,

$$
\begin{aligned}
10\log_{10}(SNR)_0 &= 10\log_{10}(E_b/N_0) + 10\log_{10}(f_b/B)\\
&= 11 + 10\log_{10}(10^6/6400)\\
&= 11 + 21.9\\
&= 32.9 = 33 \text{ dB}
\end{aligned}
$$

For G.729,

$$
\begin{aligned}
10\log_{10}(SNR)_0 &= 10\log_{10}(E_b/N_0) + 10\log_{10}(f_b/B)\\
&= 11 + 10\log_{10}(10^6/8000)\\
&= 11 + 20.9\\
&= 31.9 = 32 \text{ dB}
\end{aligned}
$$

Appendix E Measurement of voice quality (MOS) in different network condition using different codec

Using equation (4-38)-(4-41) and the equipment impairment factors for different codec as listed in Table 4-3 & 4-4; we can calculate the MOS value for different codec and compare it to ITU-T standard value as listed in Table 4-5 to acquire the service quality.

Case 1: Lets Delay = 40 ms, Jitter = 10 ms, Packet loss= 0.1%=0.001
d= 40 +10 =50, e=0.001

I_d= 0.024d + 0.11 (d-177.3) $H_{(d-177.3)}$
= 0.024 X 50 + 0.11 (50-177.3) $H_{(50-177.3)}$ = 1.2 + 0, for x<0 Hx=0 =1.2

For G.711, I_{ef}= Ie + 30ln(1+ 15*0.001) = 0 + 0.446 = 0.446

Thus R-value = 93.2 –I_d - I_{ef}= 93.2 – 1.2 – 0.446= 91.56

MOS= 1 + (0.035*R) + (R(R – 60)*(100 – R)*7.0e-^{06}) = 1 + (0.035*91.56) + (91.56(91.56 – 60)*(100 – 91.56)*7.0e-^{06}) = 4.375

Result: User very satisfied.

For G.729, Ief = Ie + 30ln (1+ 15*0.001) = 11 + 30ln (1+ 15*0.001) = 11.446
Thus R-value = 93.2 –I_d - I_{ef}= 93.2 – 1.2 – 11.446= 80.554
MOS= 1 + (0.035*80.554) + (80.554(80.554 – 60)*(100 – 80.554)*7.0e-^{06}) = 4.044

Result: Satisfied.

Case 2: Lets Delay = 100 ms, Jitter = 20 ms, Packet loss= 1%=0.01
d= 100 +20 =120 , e=0.01

$I_d = 0.024d + 0.11\ (d\text{-}177.3)\ H_{(d\text{-}177.3)} = 0.024 \times 120 + 0.11\ (120\text{-}177.3)\ H_{(120\text{-}177.3)}$
$= 2.88 + 0$, for $x<0$ $Hx=0$

$=2.88$

For G.711, $I_{ef} = Ie + 30\ln(1+ 15*0.01) = 0 + 4.19 = 4.19$

Thus R-value $= 93.2 - I_d - I_{ef} = 93.2 - 2.88 - 4.19 = 86.13$
MOS= $1 + (0.035*R) + (R(R - 60)*(100 - R)*7.0e^{-06}) = 1 + (0.035*86.13) + (86.13(86.13 - 60)*(100 - 86.13)*7.0e^{-06}) = 4.23$

Result: Satisfied.

For G.729, Ief= $Ie + 30\ln (1+ 15*0.01) = 11 + 30\ln (1+ 15*0.01) = 15$

Thus R-value $= 93.2 - I_d - I_{ef} = 93.2 - 2.88 - 15 = 75.32$
MOS $= 1 + (0.035*75.32) + (75.32(75.32 - 60)*(100 - 75.32)*7.0e^{-06}) = 3.835$

Result: Some users dissatisfied

Case 3: Lets Delay = 400 ms, Jitter = 60 ms, Packet loss= 5%=0.05
d= 400 +60 =460, e=0.05

$I_d = 0.024d + 0.11\ (d\text{-}177.3)\ H_{(d\text{-}177.3)} = 0.024 \times 460 + 0.11\ (460\text{-}177.3)\ H_{(460\text{-}177.3)}$
$= 11.04 + 0.11 \times 282.7 \times 1$, for $x>0$ $Hx=1$
$= 42.137$

For G.711, $I_{ef} = Ie + 30\ln(1+ 15*0.05) = 0 + 16.78 = 16.78$

Thus R-value $= 93.2 - I_d - I_{ef} = 93.2 - 42.137 - 16.78 = 34.28$
MOS $= 1 + (0.035*34.28) + (34.28(34.28 - 60)*(100 - 34.28)*7.0e^{-06}) = 1.79 = 1.8$

Result: Not recommended.

For G.729, Ief= $Ie + 30\ln (1+ 15*0.05) = 11 + 30\ln (1+ 15*0.05) = 27.78 = 28$

Thus R-value $= 93.2 - I_d - I_{ef} = 93.2 - 42.137 - 28 = 23.063$
MOS $= 1 + (0.035*23.063) + (23.063(23.063 - 60)*(100 - 23.063)*7.0e^{-06}) = 1.348 = 1.35$

Result: Not recommended.

Appendix F Calculation of R and MOS from the simulation results.

For VoIP traffic of some specific point at 17:35 between Agent 3 – Agent 4

For G.711
We get the delay = 6.8 ms
Jitter = 0.3 ms
Thus delay = 6.8 + 0.3 = 7.1 ms
Packet loss, e= 0

I_d= 0.024 X 7.1 + 0.11 (7.1-177.3) H $_{(7.1-177.3)}$ = 0.1704
I_{ef} = Ie + 30ln(1+ 15*0) = 0
R-value = 93.2 –I_d - I_{ef} = 93.2-0.1704-0=93

MOS = 1 < 1 + (0.035*R) + (R(R – 60)*(100 – R)*7.0e-^{06}) < 4.5
= 1 + (0.035*93) + (93(93 – 60)*(100 – 93)*7.0e-^{06})
= 4.405 = 4.4

For G.729
Between Agent 3 –Agent 4 at 17:35, delay = 12 ms
Jitter = 0.25 ms
Thus delay = 12 + 0.25 = 12.25 ms
Packet loss, e = 0.17%= 0.0017

I_d= 0.024 X 12.25 + 0.11 (12.25-177.3) H $_{(12.25-177.3)}$ = 0.294
I_{ef} = Ie + 30ln(1+ 15*0.0017) = 11 + 0.755 = 11.755
R-value = 93.2 –I_d - I_{ef} = 93.2 - 0.294 – 11.755 = 81.1 = 81

MOS = 1 < 1 + (0.035*R) + (R(R – 60)*(100 – R)*7.0e-^{06}) < 4.5
= 1 + (0.035*81) + (81(81 – 60)*(100 – 81)*7.0e-^{06})
= 4.06 = 4

For VoIP traffic between Agent 3 – Agent 4 (using the average of the results)

For G.711
Thus d = 6.5 + 0.3 = 6.8 ms
Packet loss, e= 0

I_d= 0.024 X 6.8 = 0.1632
I_{ef} = Ie + 30ln(1+ 15*0) = 0
R-value = 93.2 –I_d - I_{ef} = 93.2-0.1632-0=93

MOS = 1 < 1 + (0.035*R) + (R(R – 60)*(100 – R)*7.0e-^{06}) < 4.5
= 1 + (0.035*93) + (93(93 – 60)*(100 – 93)*7.0e-^{06})

$= 4.405 = 4.4$

For G.729

$d = 7.5 + 0.3 = 7.8$ ms

Packet loss, $e = 0.09\% = 0.0009$

$I_d = 0.024 \text{ X } 7.8 = 0.1872$

$I_{ef} = Ie + 30\ln(1 + 15*0.0009) = 11 + 0.4 = 11.4$

R-value $= 93.2 - I_d - I_{ef} = 93.2 - 0.1872 - 11.4 = 81.6$

$$MOS = 1 < 1 + (0.035*R) + (R(R - 60)*(100 - R)*7.0e\text{-}^{06}) < 4.5$$
$$= 1 + (0.035*81.6) + (81.6(81.6 - 60)*(100 - 81.6)*7.0e\text{-}^{06})$$
$$= 4.08 = 4$$

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