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Transformation and Analysis of Wire line Networks to Wireless Networks Using OPNET Simulator

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Abstract

The telecom sector is undergoing rapid changes and with the emergence of a new technologies competition to provide a different range of services has increased. As now the customers are more demanding and they need to get the services at a very fast rate and in minimum time. A race to get high speed data is becoming very high. So in order to survive it is very much necessary to replace the old wire line systems with new wireless networks and the transformation should be such that we can get best network quality at optimum cost. Till now, the most important communications platform in most of the countries is the public switched telecommunication network (PSTN), which provides access to all households and buildings across most of the countries. This paper gives a description of the wire line and wireless networks, its basics, its transformation need for transformation towards wireless networks and analysis of wireless networks using OPNET Simulator. It will also describe why VOIP is a better technology over PSTN, its advantages over PSTN. To get the optimum performance of wireless networks it is necessary that its QOS parameters should be within the acceptable range and for that analysis and simulations of networks is needed. Here OPNET simulator is discussed taking wide different areas of applications and the OPNET tool is compared with the other available simulators in order to prove the versatility of application of the simulation tool. Finally the paper is concluded showing its edge over other simulator giving valuable directions for its employment in various other applications.

***Keywords**: PSTN, VOIP, OPNET, QOS*

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INTRODUCTION

More than 100 years for transmission technology copper has been used for connecting each home and building to the telecommunication network in local loop. Copper is increasingly being replaced by fiber in the local loop while packet-based technology using the Internet. The existing circuit-based switching technologies are replaced by protocol. In today's economic scenario communication networks have become a key economic and social infrastructure. High speed networks are gradually helping tenacity continuing societal anxieties in parts such as the milieu, health care and education, and are increasingly playing a role in social networking. Though, for the potential of novel network technologies to be realised, the market will require that these networks have universal, or close to universal coverage. Technological novelty, stimulated via digitalization, has been a chief factor in motivating alteration in the communications market. This revolution has reduced costs and improving the competence of networks to upkeep new services and solicitations. A main novelty which is likely to bring more important variations in the communications market is the transformation from circuit-based public switched telecommunication networks to packet-based networks using the Internet Protocol. Throughout the past decade, the communication conducts of people have altered extremely. Technologies such as e-mail, mobile telephony, and SMS messaging have become an imperative share of every-day communication for the preponderance of the individuals in developed countries[1-5]. Developing technologies like MMS prompt messaging, push-to-talk, and dissimilar types of VoIP services are presently being reserved into use in the markets. Entirely technologies are commonly aiding the equal purpose, the characteristic need of people to communicate with each other. Agility raises the importance of telephony services by letting people to uphold their ability to communicate irrespective of time or place. This “anywhere, anytime” dimension removes the location requirement of voice communications and carries price e.g. in augmented output of business users. Enlarged reachability increases the network value as the number of on-line users in the network increases furthermore to fixed and mobile networks, there are a numerous of wireless technologies that deliver local mobility exclusive e.g. a home or office building. The present-day structure of vertically self-governing, though interconnected, networks may be altered into a horizontal structure of networks based on Internet Protocol. The growth in new communication edifices and the motivation they are likely to give to the current process of junction in networks, services and terminals are likely to grow to a superior level in future.

LITERATURE OVERVIEW

Public Switched Telephone Network

The public switched telephone network **(**PSTN**)** is the network of the world's public circuit-switched telephone networks. It comprises of fiber optic cables, telephone lines, cellular networks, microwave transmission links, communications satellites, and undersea telephone cables, all inter-connected by switching hubs ,thus letting any telephone in the world to communicate with any other. Initially a network of fixed-line analog telephone systems, the PSTN is today nearly completely digital in its main and includes mobile as well as fixed telephones. Factually, obligatory operators characteristically ran one network - the Public Switched Telephone Network (PSTN). The PSTN was intended to carry voice when voice was the only communication carried. As mandate for data communications advanced the occupants improved their networks to also carry data traffic. Though, characteristically, somewhat than replacing the PSTN operators characteristically put up novel networks that they ran in parallel which is called the overlay network. These novel overlay networks were intended precisely to carry data traffic. As network technology sustained to grow, the amount of networks multiplied in step. As a consequence, nowadays, many operators run typically 5-10 different network platforms like ATM, IP, Frame Relay, ISDN, PSTN, X.25 etc. The problematic condition with this type of multi-network approach is that it has constructed a web of complexity effecting in management difficulty, operational inadequacies, slighter spend less of scale, maintenance issues, and duplicating capex[6-9].

One of these kinds of PSTN switch is shown in Figure 1 is of Alcatel Lucent OCB-283, which was castoff by BSNL and MTNL, TATA, etc operators currently to run their landline and broadband network. The 1000 E10 (OCB283) exchange, also known as as the E10 (OCB283), was intended for emerging networks and the need to vindicate equipment operation. Its integrated architecture means that fresh services can be added and processing capability can be augmented without interrupting operation of the exchange. OCB 283 is digital switching system which ropes a diversity of communication requirements like basic telephony, ISDN, interface to mobile communication, data communication etc. Telecommunications networks are continually altering. The swift development of the digital network, mobile network and intelligent network and the propagation of innovative services continually being obtainable to subscribers mean that equipment must be continuously adapted to fresh requirements[10-15].

The OCB283 conformation provides solutions for the following:

1. The application for which it is envisioned.
2. The environment.
3. The volume and type of the traffic to be controlled.
4. The resources of the telecommunications network to which it is connected. The E10 (OCB283) can be used for all switching applications local switching center, regional or national transit center, international transit center, intelligent network service access point, mobile service access point.
5. The E10 (OCB283) can also provide the STP (Signaling Transfer Point) function of the Nº 7 signaling network.

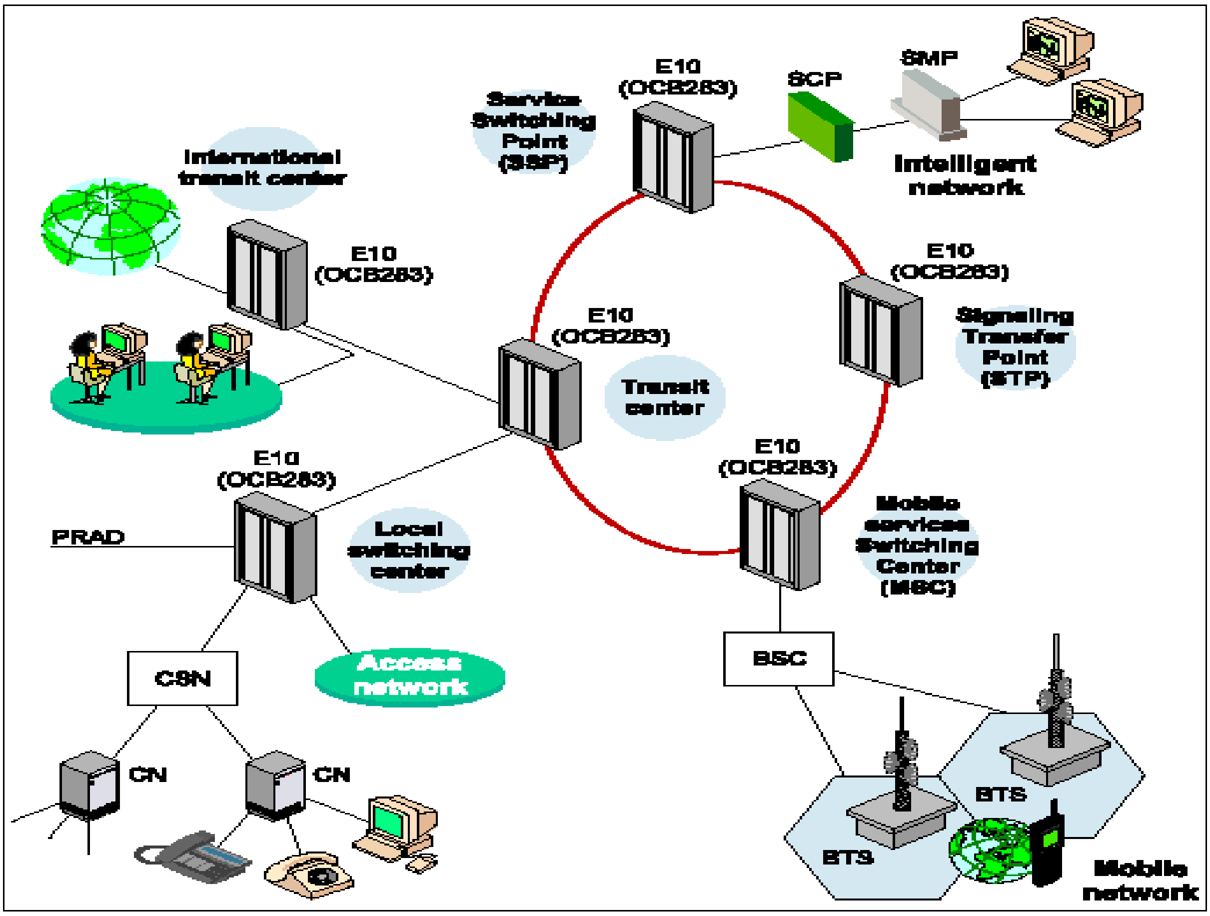

Fig. 1: *PSTN (OCB-283) Switch In a Wireline Network.*

BSC: Base Station Controller
BTS: Base Transceiver Station
CN: Digital Concentrator
CSN: Subscriber Digital Access Unit
PRAD: Primary Rate Access, Directly connected to the exchange
SCP: Service Control Point
SMP: Service Management Point

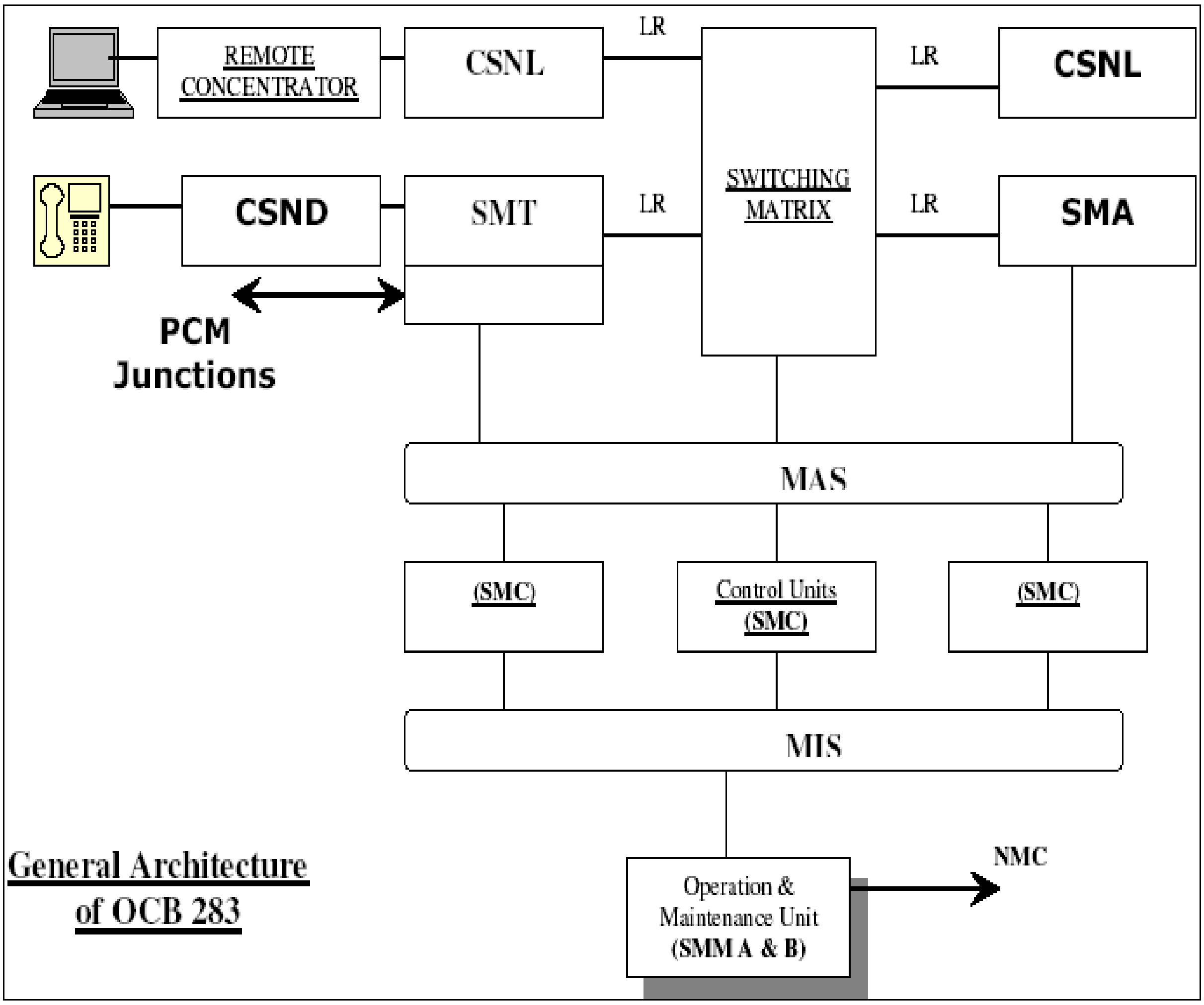

Fig. 2: *General Architecture of PSTN System.*

Common Architecture of PSTN Switch has three basic subsystems as depicted in Figure 2:
Subscriber Access Subsystem
Connection and Control Subsystem
Operation and Maintenance Subsystem

Subscriber Access Subsystem
This is preserved as an independent entity. Each Subscriber Connecting Equipment Rack is set a Signaling Point Number to function in Common Channel signaling mode with rest of the exchange Subsystems.

Connection and Control Subsystem
This unit comprises control functions and connection and switching Equipments. Control Functions has communal control equipments which process, monitor and control the call setup and release. It comprises of switching matrix equipment for accomplishment of digital time switching of speech path. Connection equipments for connecting PCM (DIGITAL) Junctions from other exchanges and RSU's. Auxiliary Equipments for Tones, Frequencies and other auxiliaries for signaling protocol handling[16-20].

Operation and Maintenance Subsystem
This unit is cast-off by operators for Operation and Maintenance of exchange. The functional architecture of the PSTN System as depicted in Figure 3. It encompasses of following different components:

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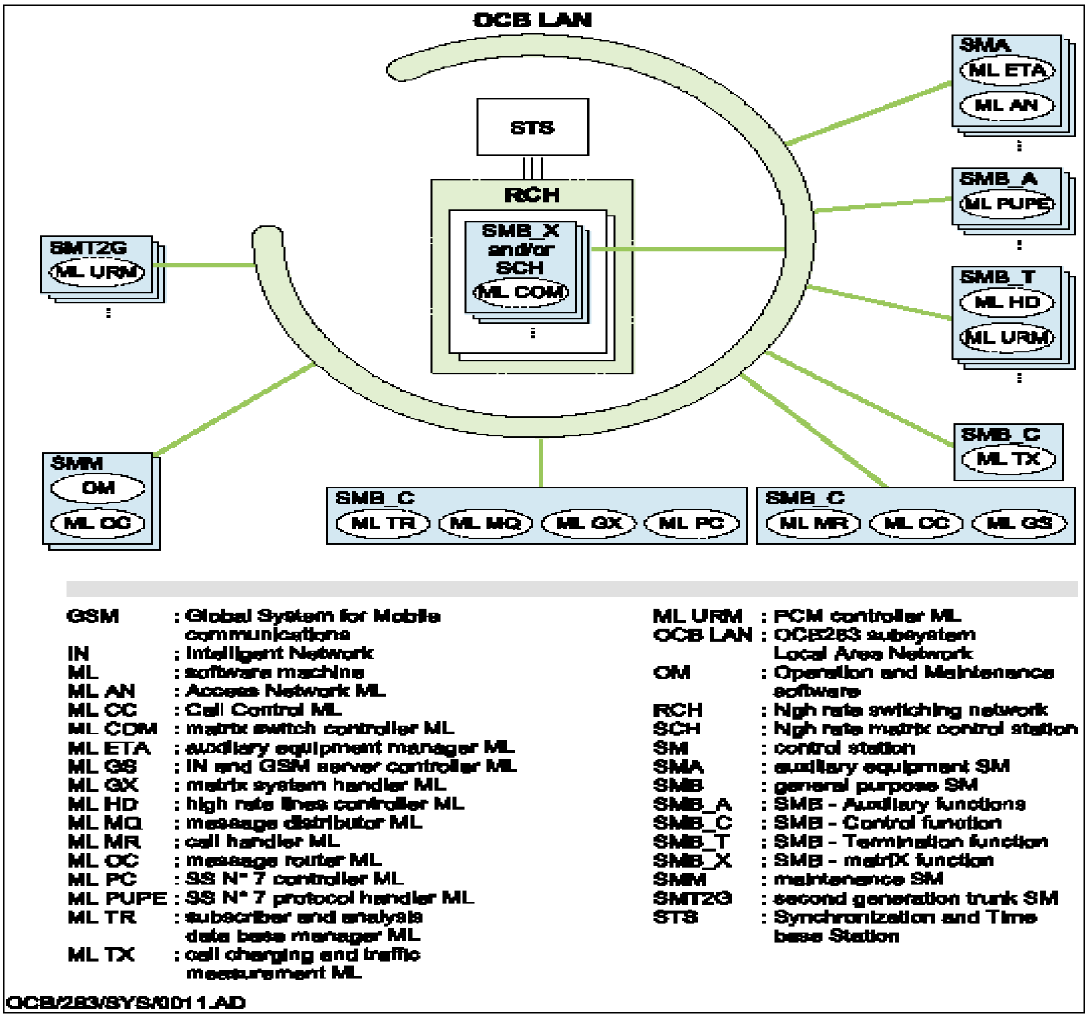

Fig. 3: *Functional Architecture of PSTN System*

Switching Network

This delivers capability for CONNECTING the LRs (internal PCM's) awaiting from connection units and performs Switching Operation for Calling Subscriber TS onto called Subscriber TS and vice versa for a two way connection per call of telephony.

In an Electronic Stored Programme Control Digital Exchange like OCB –283, all operation and maintenance activities are performed by a unit called O&M unit or OMC (Operation and Maintenance Centre).

Connection Units

These deliver capacity to attach a subscribers loop or circuits from an external PCM and transference of speech samples on to selected time slots called voice channels on a LR link (internal PCM) to switching matrix and vice versa.

Control Units

These units run control of calls on the origin of stored programs. They practice the calls on reception of dialed digits from calling subscriber/circuit and take share in handling of call setup and relief by processing, monitoring, measuring charging of calls and all further common control functions desirable for working of an Automatic Common control Exchange.

ABOUT VOICE OVER PACKET NETWORKS

In current ages, the propensity towards network conjunction has prime to an augmented interest in the transport of real time services (such as voice or video) over shared packet networks[2]. In Voice over Internet Protocol (VoIP) technologies, voice signals are digitized, packetized and then transmitted through a best effort packet network, such as the Internet. VoIP has many commercial advantages over TDM networks, as it makes use of the massive packet network infrastructure developed for the internet, effectively implementing a statistical multiplexing between different services, including voice, data and video. Goode in[3] summarized the challenges of transporting voice service over packet networks, which arise from the higher delay, delay variation, packet loss, and typically high compression algorithms to conserve bandwidth[10-14]. Typically, Internet applications use the TCP/IP protocol suite for their operation, where the IP protocol provides a connectionless best effort network communications protocol, and TCP protocol provides a reliable transport, which uses acknowledgments and retransmissions to ensure the delivery of information. This complement, however, is not suitable for the real time transmission of speech, because the nod and retransmission process results in extreme end to-end delay. For this reason, VoIP technologies use the UDP/RTP suite where UDP provides an unreliable connectionless delivery service using IP to transport messages between end points. Real Time Transport Protocol (RTP), utilized in conjunction with UDP, offers end-to-end network transport functions for applications transmitting real time data, such as audio and video. On the other hand, RTP does not backup of resources and does not assurance quality of service[3]. VoIP is a very speedy developing communication technology which ropes transportation of voice data via IP based networks.

It has an improved inherent quality of service (QoS) running with many assistances counting cost reduction, high quality as well as other value added network service solutions particularly for communications Service Providers with emphasis on real time services. Henceforth, this study tries to recognize some of the network performance parameters that will emphasis on to ripen VOIP as a communication tool that will attend as a voice communication broadband auxiliary technology to old circuit-switch voice communication. This technology enables the transmission of telephone calls through Internet or Intranet as opposed to PSTN by sending packetized voice signal via Internet Protocol (IP). Voice over IP entails that VoIP is based upon IP, hence, the transmission technology is basically in digital form this enables the application of complex encoding and decoding algorithms called codecs to digitize and split up voice data into packets. The quality of real-time applications like VoIP is commonly affected by several performance issues like voice packets end-to-end delays, voice packet delay variation and jitters as they are easily noticed irrespective of their percentage occurrence during voice/video data transmission.

Consequently, the deployment of VoIP technology demands that communication Service Providers should pay more strict attention into issues of Quality of Service with regard to voice, video, high speed data/access and mobility over an IP based networks. Hence, it is needed to intensively investigate, identify, analyze and evaluate the performance issues in this evolving technology by designing and simulating network scenarios using network simulator like OPNET Modeler in order to position VoIP as a voice replacement technology in the near future.

Because VoIP is still in its very tender stage in the commercial market today, this research through the study of the simulated network scenarios, will attempt to discover the performance (or QoS) weaknesses associated with this emerging technology and thereby propose necessary network add-on performance parameters that will give VoIP an edge over circuit-switched voice communication technology[15,16].

Majority of the research work in the past that have done on investigation of VoIP system performance used other network simulators such as NS-2, E modeling, Net-Sim etc. for designing, modeling and simulation of the network. Here OPNET Network Simulator which is a universal tool for students and communication professional to look at the performance criteria of the VOIP network will be discussed. IP telephony also known as VOIP is public internet through Internet Protocol which uses real time transmission of voice signals. One of the most important advantages of VOIP over traditional PSTN system is it may bypass a long distance phone call from toll charge for integrated voice/data solution. Apart from price change it also saves bandwidth 5 to 10 times than PSTN system. Performance analysis of VOIP networks can be done by simulation techniques for which some software is needed. If your network has real time traffic like voice, video etc., configuring and maintaining the right QoS parameters becomes all the more important. QoS obviously means Quality of Service. Queues send traffic proportionally to their weight, various Queuing mechanisms was suggested and the one which gives end to end QOS support while maximizing bandwidth utilization is concerned. The analysis and the results are drawn by using OPNET by considering various QOS parameters. As the number of user increases the performance and throughput degradation occurs for wireless network than the wired network with same speed due to the transmission limit, SNR (signal to noise) and bandwidth of the received signal[17].

Hence, to reduce congestion and collisions of the packets in the network it is better to use hybrid network which is the combination of both wired and wireless network. Another important parameter of a network that affects the QoS is congestion. Congestion is caused due to limited resources such as link bandwidth, processor capacity and buffer space. This results in bottleneck in the forwarding devices such as routers resulting in delay, jitter and loss. Congestion can be handled by appropriate admission control policy along with optimizing the usage of various resources within a network. One significant area of optimization is the scheduling algorithms used within the routers. Thus it is very much necessary that the QOS parameters for a wireless network or VOIP network should remain in acceptable limit and for that thorough analysis of the VOIP networks should be done[19,20].

The paper is organized as follows: Part I deals with Introduction, Part II deals with the Literature Overview, Part III deals with the advantages of VOIP technology over PSTN technology and need for transformation of PSTN to VOIP Networks. Part IV shows the importance of OPNET Tool in various areas of applications, and it is compared with other Simulator Tool and finally proving its edge as a Simulator Tool and the last section concludes the paper[21,22].

ADVANTAGES OF VOIP NETWORKS

There are many benefits of VoIP firstly, and most obviously, there are significant financial savings on running the network itself as same infrastructure is carrying both data and voice, provided by one supplier and it can be managed, maintained and upgraded much more efficiently than two separate networks for voice and data. Secondly, VoIP allows organizations to integrate their telephone, fax, e-mail and other applications to capitalize on the benefits of unified messaging. Such a system can eradicate unnecessary interruptions while ensuring individuals always receive information in the most convenient format wherever they are in the world. Thirdly, the system can be used to support flexible working practices, irrespective the members of staff work from home or in dispersed, 'virtual' teams. VoIP offers improved bandwidth capabilities and makes video-conferencing a viable and cost effective option for discussions between dispersed team workers. Fourthly, VoIP technology can contribute to an effective knowledge management strategy[21].

The larger the organization is, the more information that must be shared, so an efficient communications system is particularly important. The VoIP network provides individuals with the opportunity to tap into colleagues' areas of specialism, allowing them to search for experts according to specific criteria. Fifthly, an organization can also use VoIP to enhance relationships with its customers. In a conventional telephony network it is located within the network infrastructure. This allows enhanced services and features to exist in VoIP systems. For example, end points in a traditional telephone network are able to dynamically adjust the bit rate or compression algorithm of the codec to suit the current network characteristics. While usually the quality in traditional telephone calls cannot be adjusted in the event of a bad connection or to select a lesser quality for a lower price. Another fundamental difference between the two networks is the use of switching technologies.

VoIP systems are packet switched networks while traditional telephony networks are circuit switched networks. Packet switching allows multiplexing which makes more efficient use of bandwidth than does a circuit switched network. Signaling paths in VoIP systems also differ from those of traditional networks. In traditional networks, a separate network is often used for the signaling path (e.g., **SS7).** In VoIP systems, a single network is used for al1 services (voice, video, data and control). This results in cost savings to the operator. Finally, the networks differ in granting access to network resources. In a traditional telephony call, either the entire set of resources required to complete the call is allocated (i.e., 64 kbps bandwidth) or the call is not completed. In VoIP systems however the call is placed whether or not the network has sufficient resources. In VOIP systems to make further improvement it is important to improve its QOS parameters and for that we require analysis of VOIP systems which can be done by simulations. As such many Simulators have been designed and used by researchers such as NS2, Netsim, REAL, RTOS, Harvard Simulator, etc. Here OPNET (Optimized Network Engineering Tool) is considered as the Network Simulator which will be discussed and will be used for analysis purposes. Network Simulators are of various types but the tool which has wide applications and give accurate simulation results should be used and this condition is fulfilled by OPNET. In next section importance of OPNET over other existing simulators will be discussed[22].

What is The Meaning of Transformation

The path towards convergence was led mainly by the increasing digitalization of content, the shift towards IP-based networks, the diffusion of high-speed broadband access, and the availability of multi-media communication and computing devices. Transformation is taking place at different levels:

Network Transformation – It is driven by the shift towards IP-based broadband networks. It includes fixed-mobile convergence and 'three-screen convergence' (mobile, TV and computer).

Service Transformation – It enables stemming from network convergence and innovative handsets, which allows the access to web-based applications, and the provision of traditional and new value-added services from a multiplicity of devices.

Industry/Market Transformation – It carries collected in the same field industries such as information technology, telecommunication, and media, formerly operating in separate markets.

Legislative, Institutional and Regulatory Transformation – It includes protocols taking place between broadcasting and telecommunication regulation. Policy makers are considering converged regulation to address content or services independently from the networks over which they are provided (technology neutral regulation).

Device Transformation –It helps devices to include together a microprocessor, a screen, storage, input device and some kind of network connection so that together they provide multiple communication functions and applications.

Transformed User Experience-It makes unique interface between end-users and telecommunications, new media, and computer technologies. The process towards convergence has been based on an evolution of technologies and business models, rather than a revolution. This procedure has controlled to entry of new players into the market, increasing competition among players operating in diverse markets and the requirement for traditional operators to co-operate with companies previously in other fields. As a result, convergence has become an important part for the telecommunication sector.

Table 1: *Comparison between Existing Network and IP-Based Transformed Network.*

Existing Networks	VOIP Based Transformed Networks
Single purpose networks	Multi-purpose networks
PSTN, cellular, broadcast	IP network (providing voice, video and mobile services)
Narrowband	Broadband
Vertical Silos	Destroys compartmentalization i.e, Traditional boundaries between industry segments (*e.g.*, telephony, cable TV, broadcasting, wireless) are blurring and Need to re-think market definitions (product definition and geographic boundaries definition)
Network-service link	New services and content developed independently of the network
Operators control services to end users	Increased consumer control

IP Network Drivers and Its Impact on Telecom Sector

IP based Networks is an evolutionary process and it can be expected that operators will take different migratory paths, switching to NGN while gradually phasing out existing circuit networks, or building a fully-IP enabled network from the outset. Telecommunication operators across the world have been faced with a decline in the number of fixed-line telephone subscribers, coupled with a decrease in Average Revenue Per User (ARPU), as a result of competition from mobile and broadband services. Traditional sources of revenue (voice communications) have declined rapidly and fixed-lines operators are subject to an increase in competitive pressure in the market to lower tariffs and offer innovative services. This has generated pressure from the investors' community to decrease the cost and complexity of managing multiple legacy networks, by disinvesting from non-core assets and reducing operational and capital expenses. The investment in developing NGN is motivated by several factors, which are given below

Table 2: *Reasons for IP Based Networks.*

Technological Reasons	Economic Reasons	Social Reasons
Obsolescence of legacy networks, plus cost and complexity in managing multiple legacy networks.	Erosion of fixed line voice call revenues	Demand for innovative, high bandwidth, services (HDTV, VoIP, etc).
Lower capital and operational expenses.	Competitive pressure from new entrants in high-margin sectors of the market	Demand for more targeted or personalized content (on-demand multimedia services, mobility).
Increased centralization of routing, switching and transmission, lower transmission costs over optical networks.	Saturation of both Fixed and Mobile telephone services.	Demand for increased Interactivity: possibility to interact actively with the service, growing interest for user-created content.
IP-based networks enable the provision of cheaper VoIP services as a replacement for PSTN voice services.	Retain and expand users' base lower customer churn	Demand for evolved and more flexible forms of communications, including instant messaging, video-conferencing, P2P, etc.
IP-based networks enable the provision of a wider range of services, and allow bundling of services (triple and quadruple play).	Ability to expand into new market segments	Business demand for integrated services, particularly in case of multi-national structures.
Evolution and convergence of terminal equipment.	Possibility of investment in phased approach for investment, initially targeting more densely populated areas, and then gradually expanding in other areas	Guaranteeing a flexible and secure access to centralized resources and intelligence.

Migration towards NGN

IP Based Networks do not represent a specific technology but rather a system and a market concept enabled by combination of different advance technologies. The basic features of these networks are range of access platforms, independence and range of service architecture, different migration paths and timing.

The drivers for migrating to IP Based Networks are different for different players viz. reducing their network infrastructure and maintenance costs, enabling faster service deployment for the provisioning of enhanced services and therefore creating new sources of revenue. The high flexibility, low cost and wide support throughout the world for the Internet Protocol makes it the best option for building NGNs, even though it has some limitations that need to be overcome for example the lack of guaranteed QoS and security requirements. Network operators will potentially choose a different migration path depending on their existing assets. Individual path may consequently include dissimilar technologies and happen at dissimilar stride. For circuit-switched network operators, a multi-service network for joined services and further revenues signifies the main driver for the migration towards these networks. However, a migration also imposes various direct and indirect costs including network upgrades, staff retraining costs, organization process changes, etc. Moreover, some circuit-switched networks specially the cellular mobile systems are still in a very good state and provide quality telephony services, which are currently difficult to replicate on a large-scale in IP based networks. The migration of packet-based networks towards NGN follows numerous approaches that could possibly be combined. A migration to NGN could also involve a move towards the next version of the Internet Protocol, IPv6.

NGN concept also implies many different types of "Convergence";

1. PSTN/ISDN convergence with IP based networks.
2. Fixed – Mobile convergence (FMC).
3. Broadcast (Cable) – Telecommunications convergence.
4. Web based services convergence

The convergence towards Next Generation Networks also requires that customers of different market players, using different network technologies, can communicate with each other and access resources on another market player's network. In order to convert a PSTN network to NGN network three parts has to convert from PSTN to NGN.

Firstly call control mechanism of PSTN has to be converted to soft switch where all the functionalities are governed by a single call server, secondly circuit based switching has to be replaced by IP based switching governed by MPLS(Multi protocol Label Switching) where switching is done by routers so that the congestion of the data should be minimal and data can be forwarded by any path, several paths are designed by routers and the call/data will move by the shortest path available and with fastest speed. Thirdly PSTN interfaces are changed by NGN gateways. These gateways work on standard protocols as defined by industry standard so as to reduce the time taken by the data to reach the destination as well as they is supported by IP platform. Further the TDM transport network has been converted to a common IP MPLS platform in order to have a common platform for transporting all types of data and information. The designed migration of PSTN to NGN is shown in Figure 4.

PSTN to IP Based Networks

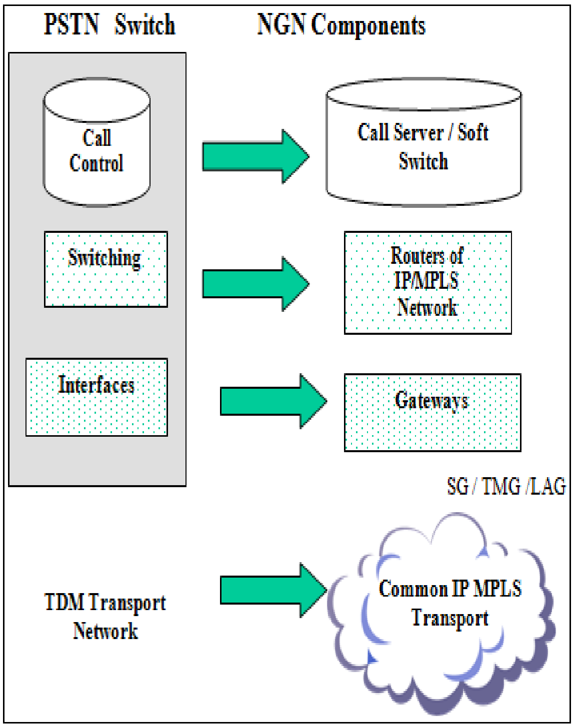

Fig. 4: *PSTN to VOIP Based Migration.*

A fully converged NGN Architecture is presented in Figure 5, which shows a converged network along with its protocols and its key IMS components

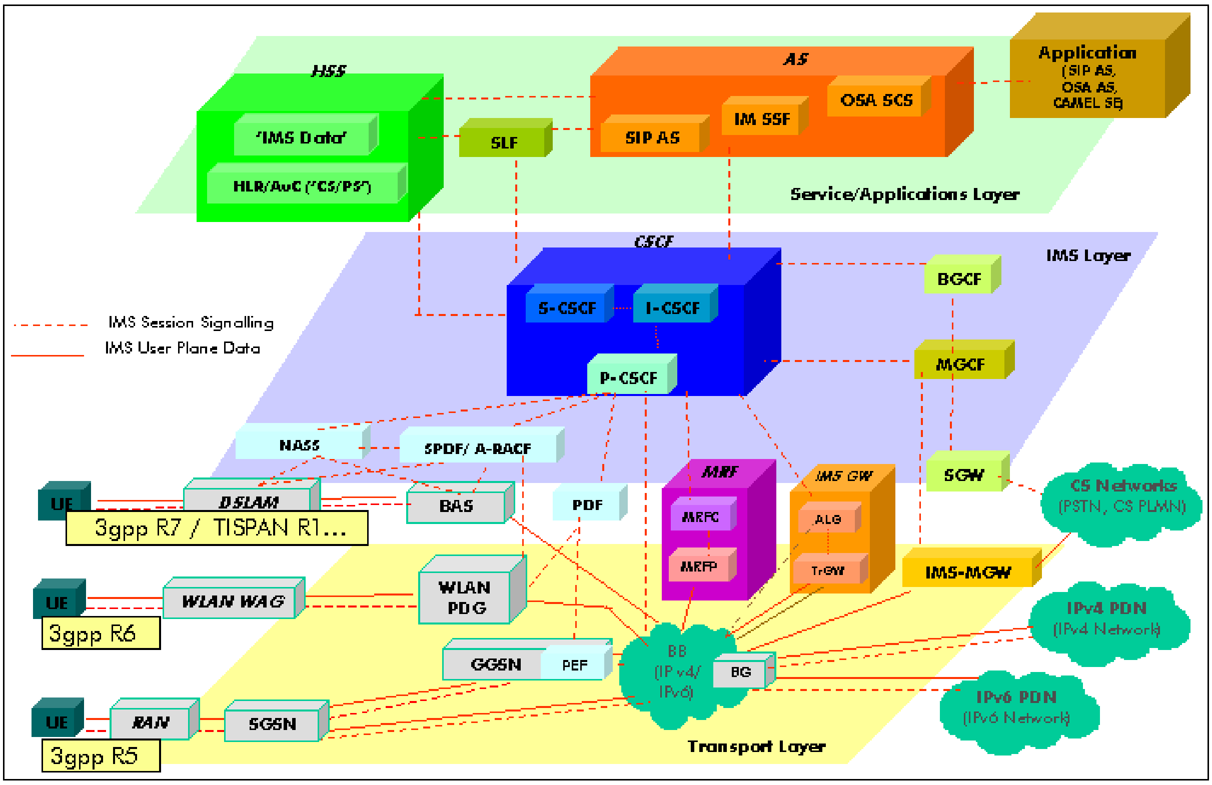

Fig. 5: *A Fully Transformed VOIP Based Network.*

The description of IMS components of VOIP Based network is given below

Table 3: *IMS Key Components.*

Network Function	**Role**	**Origin**
Call Session Control Function (CSCF)	Handles registration of end points, routing of SIP signaling messages	SIP Proxy Servers, Developed from scratch
Home Subscriber Server (HSS)	One-stop database for user information	HLR from mobile network
Media Gateway Control Function (MGCF)	Soft switch Media Gateway Control	Soft switch
Application Server (AS)	Provide service logic for applications	Feature Servers, Feature sets from Soft switches
Policy Decision Function (PDF)	Provides QoS	Existing Policy Managers, from scratch

ANALYSIS OF NETWORKS USING OPNET SIMULATOR

Several simulators have been designed for use such as REAL, NetSim, Harvard Simulator etc. Similarly OPNET (Optimized Network Engineering Tool) is one of them which are becoming a favorite tool for simulations among researches, industry professionals, etc. As such OPNET is being used by under- graduates, Graduates, Researchers, Professionals, etc. but we will gave its importance in terms of analysis for networks simulations. In case of high computation demands of application domains, researchers are employing efficient hardware platforms, versatile techniques, etc. Thus, due to its efficiency and detailed modeling, OPNET prove to be a useful tool for researchers. OPNET provides four tools called editors to develop a representation of a system being modeled. The editors, the Network, Node, Process and Parameter Editors, are organized in a hierarchical fashion, which supports the concept of model level reuse. Models developed at one layer can be used by another model at a higher layer. OPNET enables simulation of a large variety of networks, ranging from a small network to a large network consisting of many components, protocols, routing strategies etc. Here the analysis is based on application type, various applications are discussed and utility of OPNET simulator in that application is described. The ATM model suite in OPNET contains several client and server node models, such as workstations and servers, uni clients and uni-servers, uni-sources and uni-destinations, and intermediate switching elements (such as clouds and switches).The author conducts experiments to study various parameters of the simulated network, using ATM distance vector routing protocol, and different queuing schemes, such as round-robin and weighted round-robin. The results show that network performs almost similarly with these two queuing schemes. The frequency based detection approach is inveterate in OPNET by successively synthetic network intrusion data in simulated networks. Amid the numerous methods anticipated in the literature for the design of security systems for observing the network and identifying the interruption, frequency based technique uses the typical that network attacker's commonly use brute-force method of running pre-written scripts to mechanize the procedure of creation fake connections or forged packets.

The best imperative constituent module of a bulge model is the processor module which apparatuses the part of the given node in the network, which is coded into a 'Process Model. The plan and employment of distinguished service routers in OPNET and revision of UDP performance over Diffserv in a large scale network. An important advantage of distributed simulations is that, compared to stand-alone scenario, it provides opportunities for speed-up. In the Stand-Alone Mode, all the aspects of simulated entities are computed and managed within the OPNET modeler system. This approach suffers from the artificial need for OPNET to capture the behaviors of many non-network related components such as mobile nodes etc. To achieve precision in such modeling is a computation intensive and costly task. This task could better be distributed to other specialized and already-existing simulators. Therefore to improve such simulations, a distributed approach where OPNET is engaged to completely model scenario communications conduct while external simulations collaboratively accomplish non networking matters. This kind of methodology reduces the load of determining communication effects and recovers the excellence of general results. This however requires careful management of interdependent simulations to maintain cooperative operation. This includes taking care of issues such as data format, protocols and time management to maintain synchronization and causal relationships.

From above discussion by taking various application areas where OPNET Simulator is used it is clear that OPNET is becoming one of the most useful ,effective and accurate tool for analyzing and doing simulations of next generation communication networks. Now it is compared with the other available simulators. The network simulators are primarily divided on the basis of methods of simulation and fall into two categories: discrete event or analytical simulation. The OPNET andNS-2 are called hybrid simulators as they syndicate both methodologies to give a sensible speed and, at the same time upholding accuracy in the critical areas. NS-2 is an event driven network simulator primarily used by networking research community. The authors compared two scenarios, one for CBR experiment and another for FTP session. The CBR traffic is characterized by a fixed bandwidth across the network and is typically used by applications such as video and audio while as FTP is intended to share, transfer and transmit information between two computers. For the two simulators, different design parameters were varied to test their impact on the performance. The simulator was tuned for accuracy and their outputs were compared to the output from a live network test bed. One downside of the network simulators was that due to inherent limitation in the degree of closeness possible in modeling, the simulators are always somewhat different from the real testbeds. Compared to NS-2, OPNET provides more diverse statistics modules at different levels, although this also adds to the overhead cost of the software. NS-2, being a freeware is likely to be a choice of many students while as OPNET, due to the requirement of a license, would not be the as attractive. OPNET used for a commercial product has a well-engineered GUI, while as NS-2 models needs consociate with the *tcl* scripting language.

Apart from this comparison, OPNET has an edge as a simulator over other simulators which is described below

1. OPNET is a GUI based tool which makes the modeler very attractive and easy to learn.
2. OPNET is designed for researchers and independent developers and is used by academia, commercial and industrial communities.
3. OPNET provides a development environment for the specification, simulation and analysis of networks from different fields and with different levels of complexity.
4. OPNET IT Guru is a network simulation package which allows a user to use point-and-click to design simple or sophisticated network systems, and analytically study performance of the network.
5. OPNET Modeler additionally allows the user to modify existing system components and create new ones. The engine of the OPNET Modeler is a finite state machine model in combination with an analytical model.
6. OPNET can model protocols, devices and behaviors with about 400 special-purpose modeling functions.
7. The detailed documentation and a large number of study cases are provided in it.
8. The functionality to create multiple scenarios to create different design variations helps to create closely related designs.
9. The functionality and the models available are regularly increasing with the use of forum named, "Contributed Model Depot", where different users/researchers can submit their models for others to use. The OPNET model in its very core consists of C++codes. These codes are compiled and executed just like the C++ program. This enables very detailed control of the model by the user. Moreover, C++ is a popular language.
10. Numerous studies have been conducted, using OPNET for building and testing models of academic and research interest. Such studies give the researchers significant directions and overview of the state-of-art.
11. In OPNET instead of writing everything from beginning it has libraries which have the option of choosing from a large number of the existing components, which is much more reliable and accurate.
12. OPNET has library models which have a large number of functionalities or parameters already implemented which save a lot of time of the

programmers and therefore have a demand in commercial applications and mission-critical applications such as military etc.

CONCLUSION

This paper gives an in depth knowledge of PSTN and VOIP Based systems. This paper summarizes the key challenges and benefits for Service Providers (SPs) that plan to migrate from PSTN to VOIP Based Networks

1. The PSTN transformation is necessary to help SPs lower their overall cost structure. Migrating the PSTN traffic to IP Networks will increase the utilization in NGN and help SPs' lower NGN cost structure.
2. SPs are rolling out major broadband access transformation programs. A strategy that couples the PSTN migration and the access transformation will help minimize the PSTN migration costs.
3. A seamless co-existence of NGN with some consolidated/integrated parts of the PSTN is a more realistic target over the short and intermediate terms. A key limitation in the delivery of hybrid services is the distribution of point codes in NGN.

A staged migration to a target converged NGN architecture is recommended. This paper is a research work of the practical systems of communication. A lot of detailed work has been given on IP Based systems in this paper. Finally, the usage of OPNET Simulator Tool in various scenarios and applications ranging from simple simulations to verification of complex research ideas is proved, the OPNET Tool is compared with NS-2 Simulator Tool for the packet forwarding case of IP Transport network and finally the versatility of the OPNET Tool over other Simulator Tool is proved. The advantages of our work are manifold, ranging from a review of state-of-art in network simulations, to the type of applications where the OPNET Tool can be used by the researchers and the industry professionals. The paper will be of great help for the beginners as well as experienced users to give them a good direction about the versatility of the OPNET Simulator tool. Future work includes use of the OPNET Tool to more number of applications and to compare its results in different areas of communication networks.

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