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Design of Cost-effective Synthetic IP Backbone Network Topology for a Developing Economy

Nathaniel S. Tarkaa[1], Cosmas I. Ani[2]
[1]Department of Electrical and Electronics Engineering, University of Agriculture, Makurdi, Nigeria
[2]Department of Electronic Engineering, University of Nigeria, Nsukka, Nigeria

ABSTRACT: *As Internet Protocol (IP) has become the universal transport for almost all voice, data, video, and emerging communications worldwide, today, it is no longer practical to construct networks by merely connecting many standalone components without careful planning and design. As cost is the most fundamental driving factor behind the network design process,a well-designed IP backbone network should not only be cost-effective; it should also be characterized by consistent running costs. In the case of developing economies, since there is little or no existing credible legacy infrastructure to transform to the Next Generation Network (NGN)in accordance with common practices based on collected network maps and measurements, the problem is often the initiation of almost entirely new telecommunications development schemes. In such circumstances, the building of synthetic network topologies is often the preferred approach. It is believed that synthetic network topologies offer more suitable design model generalizations in such circumstances. In this paper, the IP backbone network topology is synthesized based onvarious qualitative and quantitative criteria and then developed by modelling and simulation in Riverbed Modeler environment. The full laboratory-based approach of modelling and simulation with Riverbed modeler enabled an iterative refinement of the IP backbone network until the technical design was fully refined. The results showed that the topology will meet the primary design goals of cost-effectiveness, reliability and scalability.*

Keywords: *Synthetic,Internet Protocol,Backbone Network, Cost-Effectiveness, Developing Economy, Riverbed Modeler*

I. INTRODUCTION

The aim of telecom operators (fixed and mobile) around the world is to migrate their networks to the Next Generation Network (NGN) based on IP in order to benefit from the rich NGN features as well as to avoid the drawbacks coming from their legacy networks, beside the financial factors such as reducing the capital expenditure (CAPEX) and the operation expenditure (OPEX) [1], [2]. NGN is a generic name for packet-based telecommunication technologies that have been developed for replacement of the Public Switched Telephone Network (PSTN) and having the additional capabilities of providing converged voice, data and video services simultaneously. The focus of this study is the IP backbone layer of the NGN network, which constitutes the least developed telecommunications infrastructure in context of developing countries. Particularly, in Benue State, Nigeria,which is the case study, there are no known IP networks for general public use. Hence, thisstudy entails methodologies for entirely new IP-based network planning and design that are majorly nonconventional starting from traffic demand characterization to the design of the network topology. Normally, the use of synthesized models best suited for new telecom development schemes such as those encountered in developing countries has not been adequately studiedfor the design of IP networks as noticed from related literature. Also the critical requirement of the use of the laboratory for the efficient design of IP networks has not been adequately addressed in previous related works and there has been seen an emergence of new advanced software in recent time. Furthermore, in several previous research works on the design of IP backbone networks, the design methodology is based on laid down generic principles without serious thought to the unique socio-economic, socio-cultural and demographic characteristics of the heterogeneous segments of the society, especially developing countries, which may impact the design of the network.

Generally, cost is considered as the most fundamental driving factor behind the IP network design process and network designs are typically characterized by a trade-off of cost versus performance and availability [9].A well-designed IP network should not only be cost-effective; it should also be characterized by consistent running costs [9].The cost-effective design of IP networks together with the guarantee for delivery of high quality of services can be primarily achieved intwo ways: (i) careful data collection and estimation of traffic parameters [2];and (ii) careful planning and design of the transmission link topology [2]. The methodologies for performance of the former are articulated in [4], for Benue state, Nigeria, while the best tool for performance of the second approach is the laboratory through development of a synthetic simulation model for computer simulation [9]. The laboratory process which forms the basis of this study entailed mainly the use of the state-of the-art Riverbed Modeler for both network configuration and simulation. Firstly, the hierarchical model and partial-mesh topology was adopted in order to ensure simplicity of wiring cost-effectiveness, scalability and manageability of the network. The resources of the IP backbone network which include mainly routers and the high speed transmission links interconnecting them were selected and configured in the Riverbed Modeler environment. Here the routers having the required features for the design (i.e. routing protocols, number of IP serial/Ethernet ports, processing speed, and link speed) were carefully selected. The choice of the transmission links models was also carefully made in line with standard IP backbone network design practices. The VoIP traffic flows were designed by configuring the main cities of Makurdi, Otukpo and Gboko as the destination of most traffic flows from other local government areas using set criteria to reflect the conditions of Benue State. The result of the simulation displayed the graphs of packet queuing delay, bandwidth utilization and packet loss ratio as key performance metrics for the various transmission links.The laboratory-based approach of modelling and simulation with Riverbed modeler enabled an iterative refinement of the IP backbone network until the technical design was fully refined. The results showed thatthe topology will meet the primary design goal of IP backbone network of cost-effectiveness, reliability and scalability.

The rest of the paper is organized as follows: Section 2 describes the IP backbonenetwork architecture. This is followed by a discussion in section 3 about the design of IP networks. In section 4, the IP core network design techniquesare outlined; while section 5 presents the distribution network design considerations. Section 6 is aboutIP network performance metrics. Other sections include section 7, which describes the designing of the Benue State IP backbone network architecture; section 8, which deals with configuration and simulation of the IP backbone network; section 9, which discusses the simulation results; section 10, which is about fine-tuning of the IP backbone network topology; andlastlysection 11 is the conclusion.

II. IP BACKBONE NETWORK ARCHITECTURE

The topology of an IP backbone network typically consists of a set of nodes known as Points of-Presence (PoPs) connected through multiple high capacity links [5]. Each PoP is a collection of routers following a two-level hierarchy, featuring edge or distribution routers, and core routers.Edge routers are normally lower-end routers with high port density, where customers get attached to the network [5], [8]. These routers aggregate the customer traffic and forward it toward the PoP' score routers. The core routers receive the aggregate customer traffic and forward it to other PoPs or the appropriate edge routers inside the same PoP[5]. Public and private peering points, where one Internet Service Provider (ISP) exchanges traffic with other ISPs, are often accommodated by selected core routers inside a PoP.

The capacity of links interconnecting routers inside the PoP depends on their level in the hierarchy, i.e. the level of traffic aggregation they correspond to [5]. For instance, customer links are usually 45 Mbps (T3 or DS3) or greater. The links connecting edge routers to core routers are an order of magnitude larger, reaching 622 Mbps (OC-12). The core routers inside a PoP are densely meshed (not necessarily fully meshed), and interconnected through higher speed links, i.e. OC-12 to OC-48 (2.5 Gbps). The inter-PoP links are long-haul optical fibres with bandwidth of 2.5 Gbps (OC-48) or 10 Gbps (OC-192).

IP cores are engineered for high availability and resilience to multiple link and router failures, while meeting the contracted Service Level Agreements (SLAs) [5], [8]. Traffic is guaranteed to transit the network experiencing bounded edge-to-edge delays. In addition, even in cases when certain IP paths through the network become unavailable, traffic is guaranteed to be delivered at its destination through alternate paths. Consequently, each PoP is designed to connect to multiple other PoPs through multiple high-capacity links (Fig. 1) [5], [8]. Dense connectivity between PoPs guarantees that traffic will go through the network transiting a bounded number of PoPs. In the presence of link failures, there should be other links available to take on the affected load without introducing additional delay or loss. Given that the inter-PoP links cover large distances, thus accounting for the largest part in the edge-to-edge delay, rerouting traffic across PoPs should be employed only as a last resort. For that reason, adjacent PoPs are interconnected through multiple links. These links, when feasible, follow at least two physically disjoint paths, thus offering resilience against fibre cuts which may affect more than one physical links sharing the same fibre path [5].

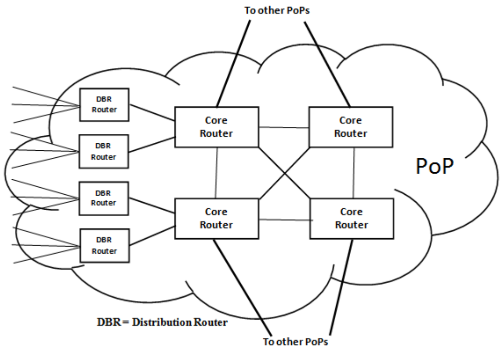


Figure 1: IP backbone architecture showing configuration of point of presence [5]

III. THE DESIGN OF IP NETWORKS

The design of packet switching networks is covered in large part by the generic principles for modeling of telecommunications networks and next generation networks. The three main model elements of system structure, operational strategy, and statistical properties of the traffic are equally applicable to packet switching networks. A number of additional challenges are however posed when it comes to designing and developing packet switched networks that are based on Internet Protocol (IP) [6]. The TCP/IP protocol suite was developed mainly for data applications in the initial days of the Internet [6]. To use the Internet for audio, video, and fax applications, real-time transmission of data is required [6]. The TCP/IP architecture has inherent limitations for supporting real-time transmission. In a TCP/IP network, there is no guarantee that all packets will be received at the destination with constant delay and in sequence. This lack of guarantee of quality of service is unacceptable for applications such as voice, fax and video where real-time transmission is a must. Loss of packets can be minimized through acknowledgements in the TCP layer, but the delay will be very high if a few packets are lost and the destination has to wait until these packets are received [6]. The variable delay (jitter) also causes problems in two-way conversations. Hence, to support real-time applications, using multimedia, a new set of protocols is required [6]. There are a number of other specific issues that need to be considered in the design and development of IP networks. These include addressing [7]; congestion control; support for real-time communication services such as voice/video; internetworking; different computing platforms; mechanisms to control errors and the different speeds of the computers; security; network management, etc. [6]. The totality of network design principles and considerations are specifically employed to achieve the following fundamental design goals.

3.1 Availability

The network should provide a resilient platform for the applications that it supports [8], [9]. A highly specified network might have to meet an availability target of 99% for all applications with a 'zero-downtime' requirement for mission critical applications [8], [9]. Ideally the failure of any one link or networking device along the client to server path should not result in the loss of a client-server session. Automatic failover to an alternate path should occur within a time-interval that is short enough to minimize the effect on existing sessions [8]. This time-interval is called the convergence time, which can be defined as the duration from a network topology change (such as the loss of a link) occurring until each device on the network is aware of the change [8]. Well-designed networks are characterized by consistently low convergence times [8].

3.2 Scalability

A scalable network is capable of adequately supporting growth without having to be radically re-designed [9]. Growth in terms of the number of users, the number of network nodes or sites must be catered for along with the possible addition of new applications and the increased bandwidth consumption that they will entail [9]. The fundamental network topology and the technology employed should not have to be redesigned in order to accommodate growth. New nodes and users can be added to a scalable network in a simple building block approach [9]. The new nodes, for example, should simply entail the addition of a new section or block to an existing structure, which is the core or backbone of the network [9]. Increased bandwidth demands should be accommodated by appropriately augmenting the LAN and WAN bandwidth as necessary [9].

3.3 Security

Security is a feature that must be designed in the network. Planning the location of security devices, filters, and firewall features is critical to safeguarding network resources [8].

3.4 Manageability

No matter how good the initial network design is, the available network staff must be able to manage and support the network. A network that is too complex or difficult to maintain cannot function effectively and efficiently [8], [9].

To meet the four fundamental design goals, the network must be built on a hierarchical network architecture [9]. The hierarchical architecture allows for both flexibility and growth [9]. The hierarchical design is used to group devices into multiple networks. The hierarchical design model has three basic layers: core layer; distribution layer; and access layer [8].

IV. IP CORE NETWORK DESIGN TECHNIQUES

The core layer is also called the backbone network [8]. Routers and switches at the core layer provide high speed connectivity [8]. The core layer may connect multiple buildings or multiple sites, and may provide connectivity to the server farm [8]. The core layer includes one or more links to the devices at the enterprise edge to support Internet, Virtual Private Networks (VPN), extranet, and WAN access. The core layer must be designed with fault tolerance, especially because all users in the network can be affected by a failure at the corelayer [5], [8], [9]. The following technologies and techniques are employed to achieve this fault tolerance requirement [8].

- The use of routers or multilayer switches that combine routing and switching in the same device,
- The use of redundancy and load balancing,
- The use of high-speed and aggregate links,
- The use of routing protocols that scale and converge quickly, such as Enhanced Interior Gateway Routing Protocol (EIGRP) and Open Shortest Path First (OSPF) Protocol,
- Network traffic prioritization,
- Multiprotocol Label Switching (MPLS).

Implementing redundant links at the core layer ensures that network devices can find alternate paths to send data in the event of failure [8]. When layer 3 devices are placed at the core layer, these redundant links can be used for load balancing in addition to providing backup [8].

Most core layers in a network are wired in either a full-mesh or partial-mesh topology [8]. A full-mesh topology is one in which every device has a connection to every other device. Although full-mesh provides the benefit of a fully redundant network, they can be difficult to wire and manage and are more costly [8]. For larger installations, a modified partial-mesh topology is used. In a partial mesh topology, each device is connected to at least two others, creating sufficient redundancy without the complexity of a full mesh. The choice of routing protocol for the core layer is determined by the size of the network and the number of redundant links or paths available. A major factor of choosing a protocol is how quickly it recovers from a link or device failure [10].

Multiprotocol Label Switching (MPLS) enables enterprises and service providers build next generation intelligent networks [11]. MPLS encapsulates packets with an additional header containing “label” information. The labels are used to switch the packets through the MPLS based network. MPLS can be integrated seamlessly over any existing infrastructure, such as IP, Frame Relay, ATM, or Ethernet [8], [9]. MPLS is independent of access technologies. MPLS technology is critical to scalable VPNs and end-to-end QoS. MPLS enables efficient use of existing networks to meet future growth and rapid fault correction of link and node failure [8], [9].

V. DISTRIBUTION NETWORK DESIGN CONSIDERATIONS

The distribution layer is the communication point between the core layer, the access layerand other remote sites, and is associated with routing and filtering [5]. The distribution layer is built using layer 3 devices just like the core layer. Distribution layer networks are usually wired in partial-mesh topology [5]. This topology provides enough redundant paths to ensure that the network can survive a link or device failure. When the distribution layer devices are located in the same wiring closet or data centre, they are interconnected using Gigabits Ethernet (GE) links. When the devices are separated by longer distances, fibre cable is used [5]. Switches that support multiple high-speed fibre connections can be expensive, so careful planning is necessary to ensure that enough fibre ports are available to provide the required bandwidth and redundancy [5].

VI. IP NETWORK PERFORMANCE METRICS

On a general note, quality of service (QoS) may be defined as the collective effect of service performances which determine the degree of satisfaction of a user of the service [14]. Specifically, QoS is defined as the measure of performance for a transmission system that reflects its transmission quality and service availability. On the Internet and in other IP networks, QoS is the idea that the throughput, losses, delays and other network characteristics can be measured, improved, and guaranteed in advance [12], [14].

A functional description of network performance encompasses a description of speed, capacity, and distortion of transactions that are carried across the network. This informal description of what constitutes network performance implies that if one knew the latency, available bandwidth, loss, and jitter rates and packet reorder probability as a profile of network performance between two network end points, as well as the characteristics of the network transaction, it is possible to make a reasonable prediction relating to the performance of the transaction [6], [12], [14].

The end effect at the terminal is also heavily dependent upon issues such as compression algorithms, coding schemes, the presence of protocols for security, data recovery, re-transmission, etc., and the ability of applications to adapt to network congestion [6]. However, network providers need performance metrics that they can agree with service providers buying resources from them with certain performance guarantees. Nevertheless, there are system performance metrics that are considered as the most important in terms of their impact on the end-to-end QoS, as perceived by a user as follows [14]:

6.1 Network/Devices Availability

The fraction of time that network connectivity is available between an ingress point and a specified egress point is defined as network availability.

6.2 Network Throughput

The available user bandwidth between an ingress point of presence (PoP) and an egress PoP. This is the effective data transfer rate measured in bits per second. It is not the same as the maximum capacity of the network, often erroneously called the network's bandwidth. A minimum rate of throughput is usually guaranteed by a service provider (who needs to have a similar guarantee from the network provider).

6.3 Bandwidth Utilization

Bandwidth utilization is the percentage of a network’s bandwidth that is currently being consumed by network traffic [20]. Consistently high bandwidth utilization indicates points of network degradation and a need for changes or upgrades in the network infrastructure. The purpose of knowing bandwidth utilization is to find out whether a link in the network is overloaded. Bandwidth utilization is calculated by the formula: percentage utilization = (data rate x 100)/bandwidth x time interval [20].

6.4 Packet Delay or IP Transfer Delay (IPTD)

IPTD is defined as the finite amount of time it takes a packet to reach the receiving endpoint after being transmitted from the sending endpoint [13]. In the case of voice, this delay is defined as the amount of time it takes for sound to leave the speaker's mouth and be heard in the listener's ear. Voice and video are delay-sensitive applications while most data applications are not. When voice packets are lost or arrive late they are discarded; the results are reduced voice quality.

6.5 Packet Delay Variation (Jitter) or IP Packet Delay Variation (IPDV)

This represents the variability in packet arrival times at the destination; the difference in the end-to-end delay between packets. For example, if one packet required 100 milliseconds (ms) to traverse the network from the source-endpoint to the destination-endpoint and the following packet required 125 ms to make the same trip, then the delay variation would be calculated as 25 ms.

6.6 Packet Loss

A comparative measure of packets faithfully transmitted and received to the total number of packets that were transmitted is known as packet loss. Loss is expressed as the percentage of packets that were dropped. Loss is typically a function of availability. If the network is highly available, then loss (during periods of non-congestion) would essentially be zero. During periods of congestion, however, QoS mechanisms would determine which packets would be suitable to drop. ITU Recommendations Y.1540/1541 gives a formal definition of Packet Loss as IP Packet Loss Ratio (IPLR), which is the ratio of total lost IP packet outcomes to the total transmitted IP packets in a population of interest.

VII. DESIGNING OF THE BENUE STATE IP BACKBONE NETWORK ARCHITECTURE

The hierarchical network structure is employed for grouping of devices into multiple networks providing best scalability for the network. The hierarchical model is composed of the backbone layer, distribution layer, and access layer. In practical networks (normally referred to as Enterprise Networks), the distribution layer is logically configured together with the core layer for aggregation of traffic from the access layer. The distribution layer is also referred to as Provider Edge (PE). Hence in the design of IP networks, the core layer is essentially configured together with the distribution layer for the purpose of testing and performance evaluation of the network.

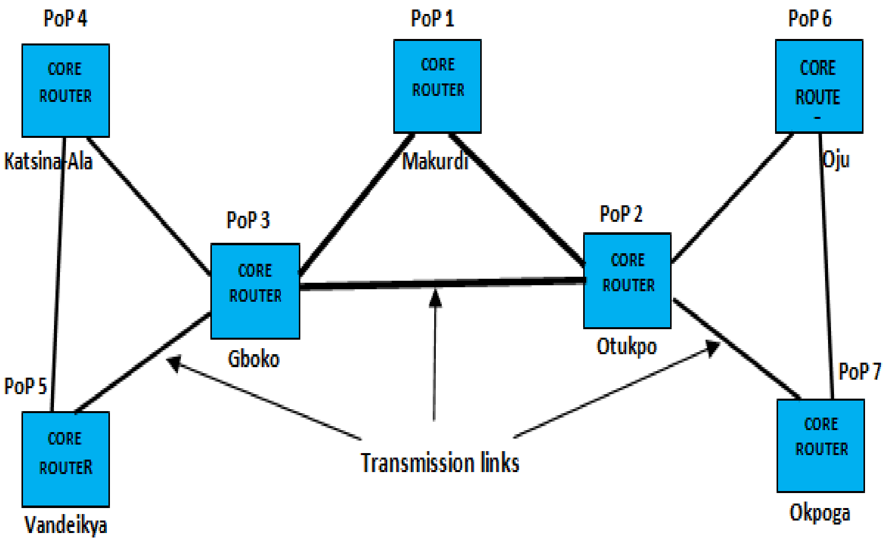


Figure 2: Block Diagram Of Benue State Core Network Showing Pops

Benue has three major ethnic nationalities: Tiv, Idoma, and Igede. Each of these tribes live in disparate regions and close proximities. Within these tribes there are clans also having the same pattern of settlement. With this arrangement, it is obvious that there would be high concentration of traffic locally in these regions. This requires the need, after careful planning, to provide a core network with 7 Points of Presence (PoPs).

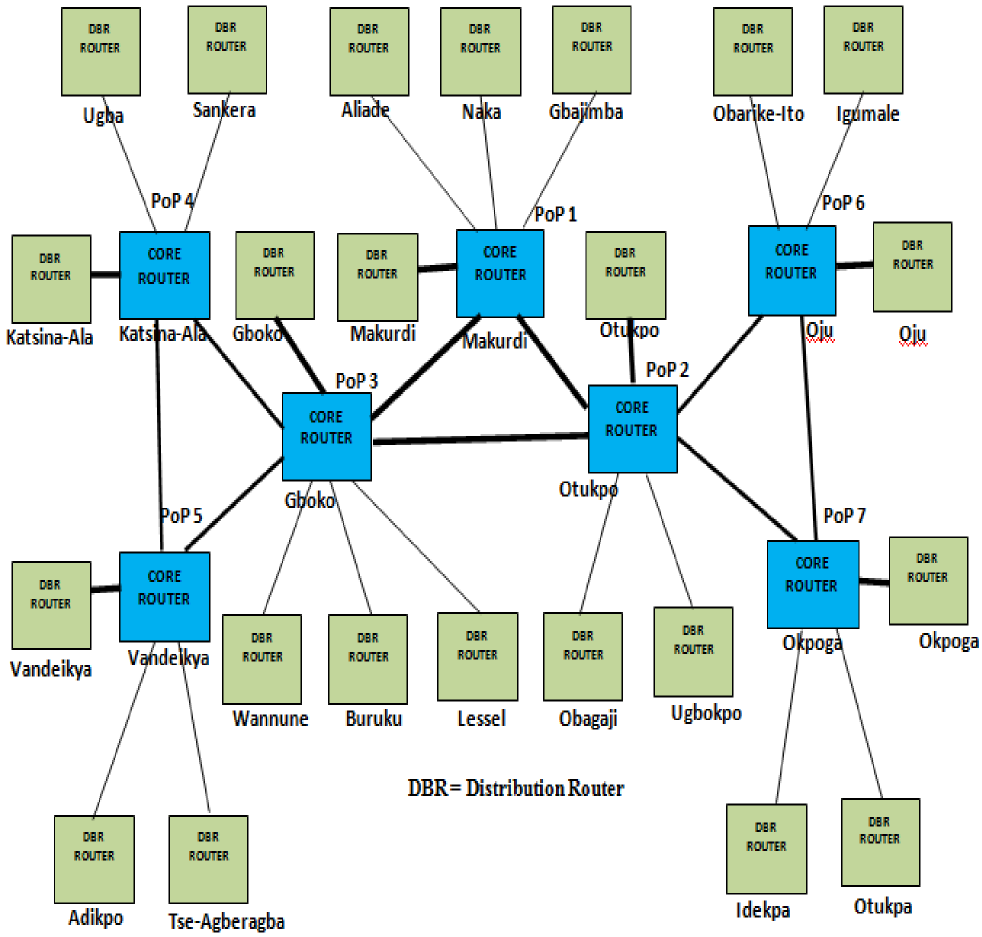
The locations of the PoPs were carefully chosen to ensure cost effectiveness and simplicity of installation. To this end, economic activities, main roads connectivity, proximity, and population of the towns were used as main indices. The block diagram of the core network depicting the PoPs is shown in Fig. 2. The design adopts partial mesh topology, instead of full mesh topology, in view of the few roads connecting the main cities, and to maintain simplicity and cost-effectiveness.

In the case of the distribution network, one router is located at each Local Government Headquarters. The initial provision of one router is done without prejudice to the provision of redundancy essential for implemented networks. The same factors used in the creation of the PoPs at the core layer are also applied here. Fig. 3 shows the complete block diagram of the IP network architecture with edge routers.

VIII. CONFIGURATION AND SIMULATION OF THE IP BACKBONE NETWORK

With the design of the network architecture now completed, the stage is now set for its configuration and simulation. Riverbed (or OPNET) Modeler was used for the configuration, simulation and other performance evaluation tasks.

Riverbed Modeler Academic Edition incorporates tools for all phases of a project, including model design, simulation, data collection, and data analysis [15]. Simulations in the Modeler are run by representing real world devices as nodes and links. The Modeler provides an environment on which attributes of these nodes and links can be configured and used as inputs in the simulation run, after which results are analyzed. The configuration process entailed selection of right types of router for the design. For instance, the router must have sufficient port density to accommodate serial and Ethernet transmission links as specified by the network blue print and make provision for redundancy. In addition, the router must be IP router and support a wide variety of other technologies and protocols to make provision for reliability and scalability of the network. In view of this, after careful and long search, the Juniper Networks' universal T640 Router was selected for both the core and distribution nodes.



DBR = Distribution Router

Figure 3: Block Diagram of Benue State IP Backbone Network Architecture

The T640 Core Router is a complete routing system that provides Gigabit Ethernet, SONET/SDH, and other high-speed interfaces for large networks and network applications [16]. In a standalone configuration, the router's maximum aggregate throughput is 320 Gbps, full duplex. The T640 also supports lower speed interfaces for combining high-speed core routing with dedicated access aggregation in a single platform. It can be upgraded in-service to any T Series platform [16]. The router is a modular, rack-mountable system. Two routers can be installed in one standard, 78-inch rack [16].The T640 Router model represents a specific configuration of an IP-based router model. IP packets arriving on an IP interface are routed to the appropriate output interface based on their destination IP address [15]. The Routing Information Protocol (RIP), Open Shortest Path First (OSPF) protocol, Border Gateway Protocol (BGP), may be used to automatically and dynamically to create the routing tables and select routes in an adaptive manner [14].

To create the traffic matrix used for this simulation, it is assumed that the aggregated traffic by each node is proportional to its population. This particular assumption can be found in [17]. Assuming that 100% of theinhabitants own mobile phones. The projected 2015 population figures and values of traffic intensities estimated as described in [4] were used to calculate the total voice/SMS traffic intensities at each network node. Voice traffic is used for simulating the network because it is not only the dominant service in Benue State but it is also the most important and time critical service of IP based communication networks [18]. Besides, IP-based internet applications, such as email and unified messaging, may be seamlessly integrated with voice applications [19].The values of the traffic intensities, call holding times estimated in the previous work [4] together with the traffic prioritization scheme were used to create VoIP traffic at the various network nodes. The traffic prioritization scheme is normally employed for bandwidth reservation, to enhance performance. The traffic prioritization is characterized by Type of Service (ToS) and have values as Best effort (0); Background (1); Standard (2); Excellent Effort (3); Streaming Multimedia (4), Interactive multimedia (5); Interactive voice (6); and Reserved (7). The simulation input parameters are given in Table 2.

Table II: Parameters for Creation of VoIP Traffic

S/N	Network Node	Total Traffic Intensity in Erlangs	Call Holding Times in Seconds	Type of Service (ToS)
1	Igumale	1059	135	3
2	Obagaji	766	168	2
3	Ugbokpo	724	174	2
4	Buruku	1528	113	3
5	Gboko	7516	192	6
6	Gbajimba	1287	168	3
7	Aliade	1316	174	4
8	Naka	747	174	2
9	Katsina-Ala	2726	153	5
10	Tse-Agberagba	913	109	3
11	Adikpo	1099	115	4
12	Ugba	1367	186	3
13	Makurdi	7732	194	6
14	Obarike-Ito	603	174	2
15	Otukpa	851	174	2
16	Idekpa	294	121	1
17	Oju	2891	192	5
18	Okpoga	936	146	4
19	Otukpo	5888	205	6
20	Wannune	309	103	2
21	Sankera	1354	180	3
22	Lessel	549	79	3
23	Vandeikya	1037	107	4

Fig. 4 shows the network domain showing the originating and terminating VoIP traffic flows on all the network nodes together with graphs of the stochastic traffic patterns. In Fig. 5, the VoIP traffic flows are depicted in circle diagram view showing the generated traffic volume on each flow. This is a clear indication that all the links are consistent and the network is ready for simulation.

However, the network of Figs. 4and 5 cannot provide an economically viable solution for Benue State. Benue State, typical of every developing economy, is not bustling with high economic, social and industrial activities that would justify such near one hundred percent provision. Rather, the State economy is mainly subsistence agrarian and small/medium scale business. Further, the State is highly social and ethnically diverse so that social and business interactions are confined to individual tribes/clans and villages rather than largely across the state. Hence, there is the need to configure the network traffic to reflect these conditions. Fig. 6 shows the network domain for the required customized solution. Here, the traffic flows from areas considered to have no tangible traffic generating ties between them have been eliminated. The remaining ones thus constitute the major traffic contributors in Benue State (see Table 3). It can be seen in Table 3 and Fig. 6 that majority of the traffic flows are towards Makurdi the main commercial and political capital of Benue State originating from all parts of the State. Then most of the calls generated in the region occupied by the Tiv people are directed towards Gboko town, their provincial capital city. Similarly, the calls generated in the region occupied by the Idoma and Igede are mostly directed towards Otukpo. Similar trends can be noticed around a few other towns. In Fig. 7, the customized IP backbone network is depicted in circle diagram view. Fig. 8 shows the IP backbone network as configured and simulated in Riverbed Modeler environment. The performance metrics chosen for the simulation are: Packet Queuing Delay, Average Bandwidth Utilization and Packet Loss Ratio. They represent conventional metrics used for performance evaluation of IP networks [12],[14].

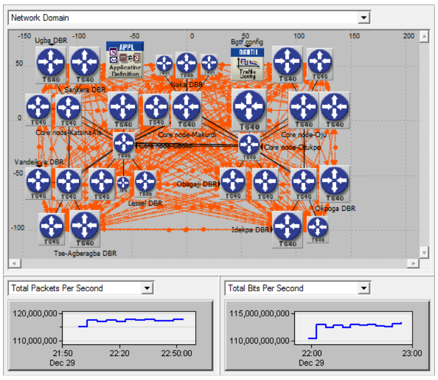


Figure 4: IP network domain showing traffic flows to all nodes

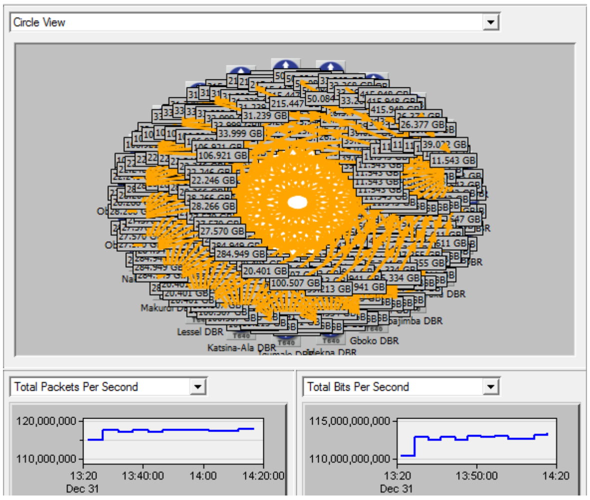


Figure 5: Circle Diagram View of VoIP Traffic Flows in IP backbone Network To All Nodes

Table III: Customized Ideal VoIP Traffic Flows for Benue State

S/No.	*From Source*	*To Destination*
1	Igumale	Makurdi, Otukpo
2	Obagaji	Makurdi, Otukpo
3	Ugbokpo	Makurdi, Otukpo
4	Buruku	Makurdi, Gboko, Katsina-Ala,
5	Gboko	Adikpo, Aliade, Buruku, Katsina-Ala, Lessel, Makurdi, Sankera, Tse-Agberagba, Ugba, Vandeikya, Wannune
6	Gbajimba	Makurdi
7	Aliade	Makurdi, Gboko
8	Naka	Makurdi
9	Katsina-Ala	Makurdi, Gboko, Sankera, Ugba, Buruku
10	Tse-Agberagba	Makurdi, Gboko
11	Adikpo	Makurdi, Gboko
12	Ugba	Makurdi, Gboko, Katsina-Ala
13	Makurdi	Adikpo, Aliade, Buruku, Katsina-Ala, Lessel, Otukpo, Gboko, Sankera, Tse-Agberagba, Ugba, Vandeikya, Wannune, Idekpa, Igumale, Naka, Obagaji, Gbajimba, Obarike-Ito, Oju, Okpoga, Otukpa, Ugbokpo
14	Obarike-Ito	Makurdi, Otukpo
15	Otukpa	Makurdi, Otukpo
16	Idekpa	Makurdi, Otukpo
17	Oju	Makurdi, Otukpo
18	Okpoga	Makurdi, Otukpo
19	Otukpo	Makurdi, Idekpa, Igumale, Obagaji, Obarike-Ito, Oju, Okpoga, Otukpa, Ugbokpo
20	Wannune	Makurdi, Gboko
21	Sankera	Makurdi, Gboko
22	Lessel	Makurdi, Gboko
23	Vandeikya	Makurdi, Gboko

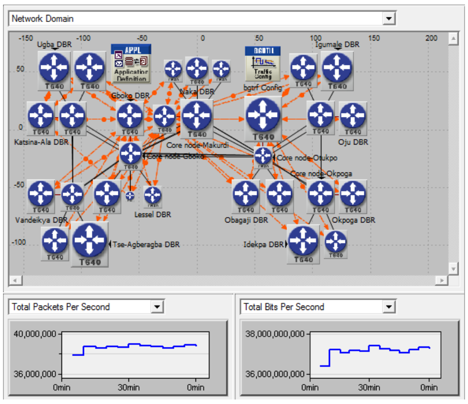


Figure 6: VoIP traffic flows customized for Benue State IP network

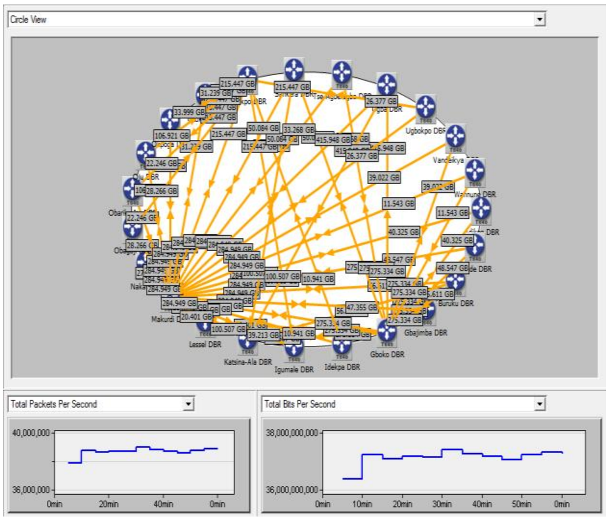


Figure 7: Circle Diagram View of Customized VoIP Traffic Flows in IP Backbone Network

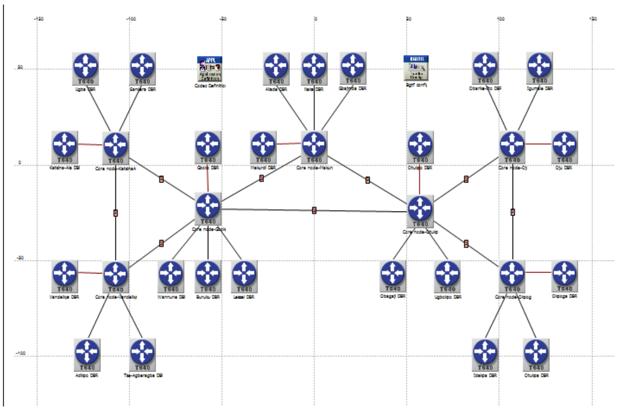


Figure 8: Topology of Benue State IP Backbone Network as Configured and Simulated In OPNET Modeler

IX. SIMULATION RESULTS

A series of graphs were generated for the results of simulation of the Benue State IP network.The analyses of the results were carried out for every link in the system by checking their status against the performance metrics and set benchmarks: packet point-to-point queuing delay: minimum = 100ms; Average Bandwidth Utilization = 100 [14]. The parameters are used for the evaluation of the transmission links which are the resources mostly affecting network cost and reliability [8], [14].The results showed the initial Benue State to be composed of 44 transmission links. Out of this number, the results showed that 5 links were congested, 7 links were underutilized, 3 were highly loaded, and 29 were optimal.

X. FINE-TUNING OF THE IP BACKBONE NETWORK TOPOLOGY

After analysis of the simulation results and taking note of the problems of congestion, high loading, and underutilization, it became necessary to fine-tune the network topology. Towards this end, the capacity of the highly loaded and congested links was increased and that of the underutilized links was reduced or cancelled outright. For instance on the Katsina-Ala – Vandeikya route, one of the two SONET OC 48 links was cancelled and the capacity of the remaining one reduced to E1 (2 Mbps). Other routes that had their capacities similarly reduced to E1 were Gboko – Otukpo, and Oju – Okpoga. The resulting topology was configured and simulated. The results of the affected links were checked and were found to be satisfactory, except for the three routes that had their capacities reduced to E1 which results still showed underutilization like in the previous simulation. It was however better to leave it like that to cater for sporadic traffic on those routes.

XI. CONCLUSION

The main problem addressed in this study is developing asynthetic cost-effective IP based network layer of soft switch based telecommunications network, the so called next generation network or NGN. Three major steps are involved in this challenging task: modelling of the traffic characteristics, design of the network topology, and simulation. The difficult aspect of the problem is the target area of coverage which is in a developing economy for which Benue State of Nigeria is the case study. In this case, existing telecom infrastructure and historical data, which usually form the basis of such designs, are completely nonexistent making the use of validation models based on collected network maps and measurements unreliable. Additionally, the developing nature of the area means the possibility of existence of new deserved areas extending the problembeyond mere migration studies. The diverse socio-cultural, economic, demographic and ethnic peculiarities associated with developing countries could not also be neglected in the network topology design and simulation that entails the use of the exclusive probabilistic traffic model [4], and the state-of-the-art Riverbed Modeler.This laboratory-based process facilitated the iterative refinement of the IP network until the technical design was fully refined.

The summary of the results of the initial simulation showed that the Benue State IP backbone network is made up of a total of 44 transmission links. Out of this number, the results showed that 5 links were congested, 7 links were underutilized, 3 were highly loaded and 29 were optimal. This called for the need to fine-tune the network topology by increasing the capacity of the congested and highly loaded links, and reduction in capacity of the underutilized ones. A second simulation was carried out and the results showed that all the affected links performed optimally. This shows that the laboratory-based approach of modelling and simulation with Riverbed Modeler may be effectively employed for the design of cost-effective and efficient IP backbone topology.

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