Dimensioning of NGN Main Components with Improved and Guaranteed Quality of Service

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Abstract - **In this paper, we propose a set of dimensioning rules, which deliver high quality session-based services over an NGN networks based IP/MPLS transport infrastructure. In particular, we develop a detailed dimensioning methodology for achieving a target QoS requirement, expressed in terms of per-service traffic flow throughput and delay. One of the benefits of the proposed methodology lies in outlining an optimal equipment allocation strategy for a requested capacity. We explore the benefits of operating a network under the paradigm of generous dimensioning, for converged multiservice traffic flows which include target QoS guarantee, scalability, and resilience of the transport network, while maintaining high resource management. We also present and discuss experimental results which illustrate a practical implementation of the proposed dimensioning strategy and its benefits.**

Index Terms - **NGN QoS, multiservice traffic surveying, Signaling traffic, NGN dimensioning, IP/MPLS.**

I. INTRODUCTION

Despite their advantages, NGNs are not without limitations and further research, development, and fine-tuning may be required when they become in service. The natural next step once NGNs are deployed is to monitor and manage them so that the promised QoS can be delivered. Several international Standard Development Organizations and industries are dedicated to standardizing and improving NGNs and have achieved significant progress and development, such as 3GPP, ETSI, MSF, PacketCable and ITU-T. 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 for multiservice traffic. Current IP/MPLS-based NGN dimensioning methods are limited and are still at the development stage. For instance, a dimensioning method for a Media Gateway (MG) voice services was proposed in [1]. This method was limited to defining dimensioning rules related to an MG carrying voice traffic. Also, in [2], various optimizing techniques for dimensioning the MPLS networks as the underlying transport technology for NGNs are outlined. However, these techniques were proposed for a specific category of networks and lack several functionalities, which are necessary for a standard NGNs architecture dimensioning. An optical network dimensioning method, which allocates appropriate capacities to links based on the network topology and traffic requirements, is proposed and studied through simulations in [3]. This method estimates the grade of service of an optical network based on the absorption probability. However, as it is known, this requires large computing resources, and is typically applicable for small networks. Besides, in [4], a method limited to defining dimensioning rules related to future transport networks for real time telephony application is proposed. Several other network dimensioning methods were published in the literature [5], [6] and [7].

This work is a continuation of our previous work as referenced in [8]. In previous work, we addressed the dimensioning of the Call Server (CS) and Media Gateway (MG). In this work, we address the additional dimensioning issues. First, we introduced an NGN traffic flow model. Second, we added the Signaling Gateway (SG) dimensioning rules. Finally, we improved the overall dimensioning methodology. Also, in this work, we build on previous contributions, as in [1], [2], [3], [4], [5], [6], [7], [8] and [9], in particular, we focus on proposing an exhaustive methodology for improving and guaranteeing NGNs QoS through dimensioning. The methodology takes several additional factors into considerations and provides the desired improvement in order to satisfy requirements in terms of QoS. We also propose additional rules for dimensioning the main NGN components. Especially, we analytically derive solutions for dimensioning the CS, MG and SG. We study and analyze, in details, the signaling, voice, and data traffic for an NGN based IP/MPLS transport network.

For practical purposes, we will also present and discuss experimental results which illustrate the practical implementation and advantages of the proposed dimensioning strategy in terms of performance gain and entities dimensioning benefits. In particular, we shall investigate the performance of the MG, SG and CS dimensioning strategies by experimentally assessing the relationship between the MG, SG and CS capacity on the generated traffic, respectively. In practice, the selection of adequate range of equipments, which meet the required capacity, is determined based on the dependence of the capacity on the generated traffic.

The organization of this paper follows a standard format consisting of four sections. The first section is this introduction. In the second section, we first discuss the NGN multiservice traffic modeling, before we address dimensioning of the NGN main components and the proposed QoS improvement. Some experimental results and performance analysis are presented and discussed in the third section. Brief summary and concluding remarks are presented in the last section.

doi:10.4304/jnw.5.7.782-791

II. NGN Dimensioning Method

The CS, MG and SG are three NGN functional entities that may be treated as independent elements. This is one of the major advantages brought by the de-correlation of the control and transport layers. In this work, we assume that each of these components represents an independent and separate entity. Physically, the MG and the SG are implemented in a distributed way and represent the concentration points of the user and the associated signaling traffic, respectively. The CS is typically implemented on central sites in order to handle grouped signaling traffic. These entities communicate via the transport network by means of control protocols. The MG dimensioning is implemented according to the user traffic handling volume, while the CS is dimensioned according to the call handling capacity and corresponding signaling processing load. The SG is dimensioned according to the signaling traffic conversion and adaptation.

In this section, we will first discuss the NGN traffic modeling, path selection, and traffic matrix before we address dimensioning of the NGN main components. We begin by outlining the general assumptions and hypotheses.

A. *Assumptions and Hypotheses*

In this work, it is assumed that the average throughput represents the key parameter for network dimensioning. Also, the Originating A_{ori}, Terminating A_{ter}, Outgoing A_{out}, Incoming A_{in}, Internal A_{intra} and Transit A_{tr} traffic are assumed to model the distributions of traffic in an MG, as detailed in [1]. We also assume that the originating traffic (A_{ori}), internal traffic proportion (P_{intra}) and transit traffic proportion (P_{tr}) are known, and the terminating traffic (A_{ter}) is as large as the originating traffic (A_{ori}) in the MG access side. In order to apply the theory, we selected a set of typical connecting interfaces for illustrative purposes. In the following sections, the modeling theory and application are outlined.

B. *NGN Traffic Modeling*

In a telecommunication network, users transmit and receive various forms of data including speech, images and text. The analysis of these various natures and forms of demands of data transmission is conducted at two different time-scales: *packet* and *flow* [2]. The packet traffic between any pair of nodes may be modeled in terms of flows or aggregates [2]. For the flow traffic, it is typically modeled as a Poisson process [3], [6], [10] and [11]. Practically, the user *session* traffic consists of successions of high-volume flow interrupted by periods of inactivity. The use of average traffic, which may be interpreted as the *demand*, allows us to better estimate the required dimensioning model. A vector may be used to denote the entering demand:

$$D = \left(d_1 \cdots d_Q\right)^T \tag{1}$$

Where the superscript T stands for the vector transpose operation, and each entry d_q represents the traffic value required for each source-destination pair $q = (m, n)$, for $q=1, 2\ldots Q$, and (m, n) is the node pairs, with $m \neq n$ and $Q = N(N - 1)$.

B.1. *Network Graph Layout*

The NGN architecture and various types of NGN traffic flows cases are illustrated in Fig. 1.

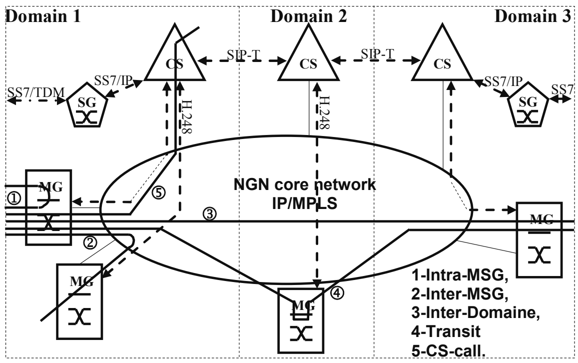

Figure 1. NGN architecture and traffic cases and distributions

The underlying physical network provides transmission lines between different nodes. In this work, we identify nodes with Call Server (CS), Media Gateway (MG), Signaling Gateway (SG) and IP/MPLS routers as well as traffic with voice, data and signaling traffic. The Label Switched Path (LSP) defines the channel for carrying traffic. Typically, the physical network is modeled as a directed graph $G = \{V, E\}$ where V is the set of nodes indexed, $i = 1, 2\ldots n$ and E is the set of directed links indexed, $j = 1, 2\ldots m$. Suppose that, link l corresponds to a transmission line with capacity c_l, and C denotes the m-dimensional capacity vector:

$$C = \left(c_1 \cdots c_m\right)^T \tag{2}$$

Generally, an NGN network considers multiple paths connecting a pair of nodes. Let the node pair (m, n), with $m \neq n$, be indexed by $1, 2\ldots Q$. The total number of node pairs may be written as follows:

$$\mathrm{Q} = \mathrm{N}(\mathrm{N} - 1) = \sum_{i=1}^{n-1} \sum_{j=i+1}^{n} q_{ij} \tag{3}$$

Where N represents the set of the graph nodes.

Moreover, let the total traffic-path matrix denoted by A represents the different acyclic traffic-paths P_{ij} indexed by $a_{ij}^1, a_{ij}^2, \ldots, a_{ij}^{P_{ij}} = a_q^1, a_q^2, \ldots, a_q^{P_q}$ and implemented for connecting a source entity i, to a destination entity j, for $i=1, 2\ldots m$ and $j = 1, 2\ldots n$, with $i \neq j$ and $m \neq n$. The source and destination entities could be one of the following types: MG, CS, SG or core node. This total traffic-path matrix A corresponding to the different acyclic paths linking a node i to a node j over the graph, can be represented as follows:

$$A = \left(\begin{array}{ccc|c|ccc} a_{11}^1 & \cdots & a_{1n}^1 & \cdots & a_{m1}^1 & \cdots & a_{mn}^1 \\ \vdots & \cdots & \vdots & \cdots & \vdots & \cdots & \vdots \\ a_{11}^{P_{11}} & \cdots & a_{1n}^{P_{1n}} & \cdots & a_{m1}^{P_{m1}} & \cdots & a_{mn}^{P_{mn}} \end{array}\right) \tag{4}$$

This total traffic-path matrix A corresponding to the different acyclic paths P_q linking the node pairs q over the graph, can be also defined as follows:

$$A = \begin{pmatrix} a_{1,1} & \cdots & a_{1,P_1} \\ \vdots & \cdots & \vdots \\ a_{Q,1} & \cdots & a_{Q,P_Q} \end{pmatrix} \quad (5)$$

Where, $a_{ij}^{P_{ij}}$ is a traffic-path value identified for each path linking a source entity i to a destination entity j. The matrix A represents the total path-traffic matrix corresponding to the total number of node pairs. Note that the diagonal elements of A are non-zero since we consider the intra-site traffic [6], [12]. The corresponding elements of the traffic-path matrix A identified during a time duration T subdivided into n sampling periods with step size T_0 can be determined as follows:

$$a_{ij\,day} = \underset{k=0}{\overset{n}{Max}}\left[a_{ij}\left(T_k\right)\right] \quad (6)$$

Where $n = T/T_0$ and T_k is the levy times during one day.

The time period, T, typically corresponds to rush hours over several typically loaded days. Based on maximal values of $a_{ij\,\mathrm{day}}$ obtained for each day, the traffic value a_{ij} obtained from observations corresponding to m days for each path binding a node pair (i, j) can be written as:

$$a_{ij} = \underset{g=1}{\overset{m}{Max}}\left(\underset{k=0}{\overset{n}{Max}}\left[a_{ij}(T_{k,g})\right]\right) \quad (7)$$

Where $T_{k,g}$ is the different levy times during m days.

The use of the mean, as computed from the busiest working days during rush hour periods, provides better estimate for the average maximum capacity.

Note that the superscript is not an exponent, but it is an index of the acyclic paths linking the node pairs (i, j).

B.2. Paths' Bandwidth Allocation

Generally, the NGN QoS guarantee should be satisfied if we ensure optimal network components dimensioning and assignment, and traffic path allocation and routing. In this subsection, we propose an optimal flow allocation strategy subject to the constraint that no path may exceed its flow allocation. That is, the total amount of flow allocated to the set of paths P_q connecting any given source-destination pair of nodes q, where each path is allocated a proportion of traffic, must equal to the total demand d_q for that node pair. Thus, we must ensure that sufficient BW is allocated to each set of paths, in order to guarantee the requested throughput [5]. Therefore, for each node pair $q=1, 2\ldots Q$, the BW guarantee is satisfied by requiring that:

$$d_q = \sum_{p=1}^{P_q} a_q^p \quad (8)$$

In addition, for each link, $l=1, 2\ldots m,$ traversed by paths connecting node pairs, the total of flows allocated to these paths must not exceed the link capacity C_l. That is:

$$c_l \geq \sum_{q=1}^{Q}\sum_{p=1}^{P_q} a_{q,l}^p \quad (9)$$

This may notably improve the network performance and remarkably reduce or even avoid congestion problems.

The occupied bandwidth ς_l for each link l may be formulated as:

$$\varsigma_l = \sum_{q=1}^{Q}\sum_{p=1}^{P_q} x_{l(p,q)} \quad (10)$$

Where $x_{l(p,q)}$ is the volume of traffic for path p of node pair q that flows through link $l.$ Note that, the optimal allocation of the set of paths P_q for different commodities d_q, may be reached if the consumed bandwidth ς_l on each link l is shared by a set of shortest paths. This minimizes the global network end-to-end delay and improves the global network bandwidth reserve and provision.

The resulting load ρ from the consumed bandwidth ς and the total path capacity C is determined as following:

$$\rho = \frac{\varsigma}{C} \quad (11)$$

B.3. Paths' End-to-End Delay

Any given path within the set of paths, $p = 1, 2\ldots P_q$, connecting a given node pair q, which has an allocated path flow cannot exceed a fixed end-to-end allowed delay T_q. This requirement must be given to any traffic-path of the aggregate d_q traversing any path connecting the node pair q. In addition, as discussed in [2], the path delay for a given traffic also depends on all other traffic flows sharing the same links. Thus, we have:

$$\sum_{l=1}^{L}\frac{a_{q,l}^p}{c_l - \varsigma_l} \leq T_q \quad (12)$$

Note that, if the consumed BW reaches the available capacity, the admission control will react and reject any new entering traffic.

Furthermore, since several paths may exist between any pair of nodes, the path selection priority for traffic sharing varies according to the fixed path length. The objective is then to minimize the weight associated with each path. Setting weights on a hop-count basis ensures higher priority of shorter paths over longer ones. This minimizes the global network end-to-end delay, which is given by:

$$w_q^p = \sum_{l=1}^{L} a_{q,l}^p \quad (13)$$

Note that a good reference which presents methods for determining the consumed bandwidth on each link may be found in [13].

C. Media Gateway Dimensioning Method

The MG corresponding to the connectivity level in the NGN architecture ensures the aggregation and the commutation of diverse access traffic towards the transport level. For dimensioning this entity, we shall determine its call processing capacity as well as the number and type of necessary interfaces in the access and core sides for voice and data traffic. These tasks depend on the voice and data traffic distribution, the number of per-second switched voice calls, and the total volume of voice and data traffic processing and forwarding, in terms of per-second BW.

C.1. Voice and Data Traffic in the Access Level

Voice Traffic

In this section, it is assumed that each voice access service k indexed by *1, 2...n*, is characterized by a mean holding time S_k and a bearer occupancy ratio Z_k (the result of the Grade of Service criteria), for k = *PSTN, ISDN...IN*, as modeled in [1]. Based on this model, closed-form expressions may be derived for several traffic parameters, as follows:

a) The voice traffic A_k, corresponding to voice access service k, can be written as:

$$A_k = Y_k \times S_k \quad (14)$$

Where Y_k is the number of calls per-second of the voice access service type-k. Thus, the total voice traffic A_v is the sum of A_k:

$$A_v = \sum_{k=1}^{n} Y_k S_k \quad (15)$$

We can also write this total voice traffic as:

$$A_v = \sum_{k=1}^{n} \alpha_k a_k N_k \quad (16)$$

Where:

N_k: is the number of subscribers using the type-k service.

a_k: is the average fraction use of a 64 kbps bearer belonging to a type-k service.

α_k: is the number of circuits corresponding to a type-k service (e.g. 30 for $ISDN_{PRA}$, 2 for $ISDN_{BA}$, etc.).

b) The *MG* busy hour call attempts (BHCA) can be written as:

$$BHCA_{MG} = 3600 \sum_{k=1}^{n} \frac{A_k}{S_k} = 3600 \sum_{k=1}^{n} \frac{\alpha_k a_k N_k}{S_k} \quad (17)$$

c) The number of *E1* interfaces for different voice access services in an *MG* access side, may be found as:

$$N_{E1,v} = \sum_{k=1}^{n} \frac{A_k}{30 z_k} = \sum_{k=1}^{n} \frac{Y_k S_k}{30 z_k} \quad (18)$$

Where an *E1* interface is part of a higher order Line Interface Board (LIB). It is composed of 30 voice channels 01 signaling, and 01 synchronization channel.

d) When the traffic generated by different access networks is converted in Mbps, the number of interfaces, *E1*, can be re-written as follows:

$$N_{E1,v} = \sum_{k=1}^{n} \frac{A_k}{2.048} \quad (19)$$

e) When carrying voice traffic, the *effective* bandwidth, $BW_{e,v}$, is generally less than the *absolute* bandwidth BW_a offered by a such interface. In practice, $BW_{e,v}$ is typically set to 85 % of the BW_a. The extra bandwidth ($BW_r = BW_a - BW_{e,v}$) may be used for carrying data traffic. Moreover, an $STM1_v$ interface is approximately equal to $\delta \times E1_v$. In the worst case scenario, $\delta \approx 63$, as in [1]. Accordingly, the number of *STM1* corresponding to voice traffic is given by:

$$N_{STM1,v} = \frac{BW_v}{BW_{e,v}} \approx \frac{BW_v}{127.5} \approx \frac{N_{E1,v}}{\delta} \quad (20)$$

Note that, an *STM1* is a higher order line interface board, and one STM1 interface has 63 logical E1 interfaces. The available bandwidth of an *STM1* interface is 155.52 Mbps. In this available bandwidth we find that 9/270 is a hierarchy SDH frame overhead and 1/270 is used as a pointer, so the effective bandwidth which corresponds to the transferred payload will be 155.52 × 260/270 = 150 Mbps. If the *STM1* interface is used for carrying voice traffic then the effective bandwidth is set to 85 % of 150 Mbps, which equals to 127.5 Mbps. Indeed, the usage of the remaining extra bandwidth (22.5 Mbps) for carrying data and signaling traffic avoids wasting resources.

Data Traffic

In this section, we derive the number of *E1* interfaces for different data access services.

a) Firstly, the data traffic can be expressed as the sum of the products of the number of subscribers, the mean traffic and the Simultaneous Attached Users' ratio (*SAU*), as follows:

$$A_d = \sum_{k=1}^{n} N_k d_k SAU_k \quad (21)$$

Where d_k is the average throughput per user. The Simultaneous Attached Users ratio represents the number of subscribers, which simultaneously request the same service regarding the total number of subscribers registered in the indicated service.

b) Secondly, the number of *E1* interfaces for the different data accesses can be expressed in terms of the data traffic as follows:

$$N_{E1,d} = \sum_{k=1}^{n} \frac{A_d}{2.048} = \sum_{k=1}^{n} \frac{N_k d_k SAU_k}{2.048} \quad (22)$$

c) In *SDH*, 10/270 bytes represent the overhead and pointer. Thus, the BW_e for an *STM1* is in the order of:

$$BW_{e,d} = BW_{a,d} \times 260/270 \approx 150 \text{ Mbps}. \quad (23)$$

Accordingly, the required number of *STM1* needed for carrying data traffic, may be expressed as:

$$N_{STM1,d} = \frac{BW_d - N_{STM1,v} BW_r}{BW_{e,d}} \quad (24)$$

d) In the access side, the total number of *E1* and *STM1* interfaces, are given by:

$$N_{E1,tot} = N_{E1,d} + N_{E1,v}, \text{ and}$$
$$N_{STM1,tot} = N_{STM1,d} + N_{STM1,v} \quad (25)$$

e) Finally, the total *BW* in the access side for carrying voice and data traffics is given by the sum of the voice and data traffic:

$$BW_{tot\_access} = A_v + A_d \quad (26)$$

This total voice and data *BW* permits to determine the number and the type of necessary interfaces in the access side of a Media Gateway (MG).

f) Here we adress the derivation of the Simultaneous Attached Users (SAU) for the data sessions, and the simultaneously active connections Y_k for the voice sessions. First, suppose that there are N customers connected to the MG access level, and let t_i denote the total time interval during which i links are simultanouesly occupied by active connections. Now consider an observation period, of length T, during which the occupation state of the access links group is recorded. This observation period is represented as a succession of short examination time intervals of length T_0, indexed by $j=1,2,\ldots n$, with $n=T/T_0$. Let t_{ij} defines an examination time interval j at which i links are simultanousely occupied. Then, t_i may be written as:

$$t_i = \sum_{j=1}^{n} t_{ij} \tag{27}$$

Thus, the total time of the links group occupation noted by τ, my be defined as follows:

$$\tau = \sum_{i=1}^{N} \sum_{j=1}^{n} it_{ij} = \sum_{i=1}^{N} it_i \tag{28}$$

In view of the above derivation, the average number of the occupied links noted by Y, from the N physical connected links, may be defined as:

$$Y = SAU = \frac{1}{nT_0} \sum_{i=1}^{N} \sum_{j=1}^{n} it_{ij} = \sum_{i=1}^{N} \frac{it_i}{nT_0} = \sum_{i=1}^{N} \frac{it_i}{T} \tag{29}$$

The term t_i/T in the last formula indicates the rate of time during which excatly i connections are simultaneously active during the monitoring interval T. Therefore, Y represents the average number of occupied links (active connections) among N available links. Note that the observation period, T, corresponds to rush hours over several typically loaded days.

C.2. Voice and Data Traffic in the Core Level

Voice Traffic

In this dimensioning work, we followed the RTP/UDP/IP/MPLS protocols for carrying voice traffic. As such, voice traffic is first converted by the MG from circuit to packet switching traffic and then encapsulated within the IP packets and carried thereafter on the IP/MPLS transport level. Accordingly, an RTP header, with size of 12 bytes and an UDP header with size of 8 bytes are also added, as overhead. Additional 20 and 4 bytes of overhead are also required for the IP packaging and the MPLS labeling, respectively. Thus, the total length of these headers is 44 bytes, which is transmitted every time a packet containing voice samples is sent. Besides, the length of the RTP payload used for carrying the voice samples depends directly on the type of CODEC used. Also, the packet transmission frequency, per second, is the inverse of the time-duration required for packing the voice samples within a packet. The selection of this payload duration is a trade-off between the bandwidth availability and the QoS requirement. It has been shown that the G.711 codec generally yields better voice quality, resulting in an improvement in the QoS offered to customers. For these reasons, we chose the G.711 codec with a rate of 64 Kbps for the applications of interest. As discussed in [15], we also assume that voice samples in each packet have a 20ms-duration, which means that 50 packets are sent every second. In view of the above description, the RTP payload has a size of 160 bytes, as illustrated in Fig. 2.

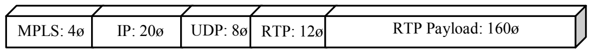

Figure 2. Voice traffic encapsulation

a) In view of the above description, the number of circuits that carry access voice traffic towards the core level is determined by:
   $ch = ch_{access} \times (1 - P_{intra}) = N_{E1,v} \times 30 \times (1 - P_{intra})$ (30)
   Where ch_{access} and P_{intra} represent the number of used 64 kbps channels in the MG access side and the proportion of intra-MG traffic, respectively.
b) In order to compute the required Bandwidth (BW) corresponding to an *E1* circuit emulation of the access side in the IP/MPLS core side, it is assumed that a voice channel with 64 Kbps requires:
   $$64 \times (204/160) = 50 \times 204 \times 8.10^{-3} = 81.6 \text{ Kbps.} \tag{31}$$
c) Thus, the total BW requirement in the IP/MPLS core network side, for carrying the total voice traffic can be calculated as follows:
   $$\begin{aligned} BW_v &= ch \times 64 \times (204/160) = ch \times 50 \times 204 \times 8.10^{-3} \\ &= ch \times 81.6 \text{ Kbps} = ch \times 0.081 \text{ Mbps.} \end{aligned} \tag{32}$$

Note that the factor 8.10^{-3} is needed to calculate in Kbps.

Data Traffic

As illustrated in Fig. 3, Data traffic is carried over packet-based IP/MPLS Tunnels or LSPs established through the core network, with a header size equal to 24 bytes and an IP data payload "*D*".

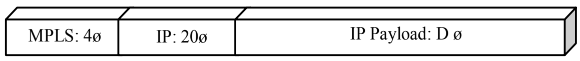

Figure 3. Data traffic encapsulation

a) The total BW reserved for the data traffic volume can be expressed in terms of data traffic, as follows
   $$BW_d = Data \times (24+D)/D. \tag{33}$$
   $$\text{Where: } Data = Data_{access} \times (1 - P_{intra}) \tag{34}$$
   Where $Data_{access}$ and P_{intra} represent the volume of data traffic in the MG access side and the proportion of intra-MG traffic, respectively.
b) Knowing the originating traffic (A_{ori}), internal traffic proportion (P_{intra}), transit traffic proportion (P_{tr}) and assuming for simplicity, that the terminating traffic (A_{ter}) is as large as the A_{ori} traffic in the MG access side; all traffic distributions can be derived. Indeed, the $BW_{transit}$, that represents the transit traffic received from one node and forwarded through the MG core side towards another node, can be written as follows:
   $$BW_{transit} = A_{tr} = A_{in}\, P_{tr} = \frac{A_{ori}\, P_{tr}\, (1\text{-}P_{intra})}{(1\text{-}P_{tr})} \tag{35}$$
   Practically, this traffic is typically composed of users' voice and data, transit and control signaling traffic.

Control Signaling Traffic

The control signaling traffic, exchanged between the CS and the MG, is carried on packet switching-based H.248/SCTP/IP/MPLS Tunnels or LSPs established through the core network. The choice of SCTP for carrying the control signaling messages is justified by its potentialities and advantages over UDP or TCP, in establishing secure and reliable connections. The SCTP packet is composed of a common header, with size of 12 bytes and one or multiple bundled chunks. Each chunk is formatted with a block header, with size of 16 bytes and a data chunk payload "*D*". Furthermore, the IP packaging and MPLS labeling require an additional 20 and 4 bytes of overhead, respectively. The total length of these headers is then 52 bytes, which is transmitted each time a signaling packet is sent. As illustrated in Fig. 4, in the simple case, we assume that an SCTP packet consists of a header followed by only one data chunk with fixed "*D*" payload length in bytes. As discussed in [16], the total chunk-length should be a multiple of 4 bytes. If this is not the case, the sender needs to pad the chunk with "*P*" bytes, where *P = 1, 2, or 3*.

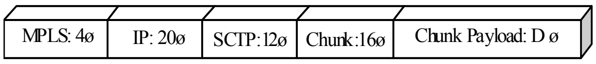

Figure 4. Signaling traffic encapsulation

Thus, the *BW* (in Mbps) reserved for the control signaling traffic volume (e.g. H.248 or MGCP messages) exchanged between the CS and the MG is given by:

$$BW_{sig\_cmd} = \left(\sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y^i_{k,q} Li_{q,i} Ni_{q,i} + \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y^i_{k,q} Lo_{q,i} No_{q,i} \right) \times (52+D+P)/D \text{ x } 8.10^{-6} \quad (36)$$

In view of these transmission needs, the total bandwidth in the IP/MPLS core network side, required for carrying the voice, data and control signaling traffic can be expressed as the sum of the various *BW* requirements, as follows:

$$BW_{tot\_core} = BW_v + BW_d + BW_{sig\_cmd} + BW_{transit} \quad (37)$$

Where:

- The quantities Li_q and Lo_q represent the length of the *q* signaling messages (in bytes/s) sent in the uplink and downlink directions, respectively. Similarly, Ni_q and No_q, represent the number of such messages. The indices *q* indexed by *1, 2...m,* and *i* represent the used control signaling and the call status (success/failure), respectively. Note that the superscript *i* is not an exponent, but it is an index of the call status.
- *P* is used only if the total chunk-length is not a multiple of 4 bytes, and the factor 8×10^{-6} is used to convert the bandwidth to Mbps.

a) The total number of *STM1* and *E1* interfaces in the core network side for a MG can now be derived from the above BW requirements as follows:

$$STM1_{tot\_core} = BW_{tot\_core}/BW_{e,d} \text{ and} \quad (38)$$

$$E1_{tot\_core} = BW_{tot\_core}/2.048 \quad (39)$$

The choice of the type of interfaces depends essentially on the total volume of data traffic in the core side.

Call Handling Capacity

In this section, we evaluate the MG call processing capacity in order to identify its performance. This depends on the voice calls' distribution and the number of per-second switched voice calls, and outlined, as follows:

a) Let Y_k denote the number of calls per second of access type-*k* and Y_{ori} and Y_{ter} denote the total number of originating and terminating calls per second for the MG access side, respectively. Then we have:

$$Y_k = \frac{A_k}{S_k} \text{ and } Y_{ori} = Y_{ter} = \sum_{k=1}^{n}\frac{Y_k}{2} = \sum_{k=1}^{n}\frac{A_k}{2S_k} \quad (40)$$

b) The number of outgoing, Y_{out} and incoming, Y_{in} calls, per second of the MG core side can now be derived from the number of originating calls, Y_{ori} the proportion of Intra-MG calls, P_{intra} , as follows:

$$Y_{out} = Y_{ori}\left(1 - P_{intra}\right) + Y_{tr} = \left(1\text{-}P_{intra}\right)\sum_{k=1}^{n}\frac{Y_k}{2} + Y_{tr} \quad (41)$$

$$Y_{in} = Y_{ori}\left(1 - P_{intra}\right)\left(1\text{-}P_{tr}\right) = \left(1\text{-}P_{intra}\right)\left(1\text{-}P_{tr}\right)\sum_{k=1}^{n}\frac{Y_k}{2} \quad (42)$$

Where the number of MG transit calls, Y_{tr}, is given by:

$$Y_{tr} = Y_{ori}\, P_{tr} \frac{\left(1\text{-}P_{intra}\right)}{\left(1\text{-}P_{tr}\right)} \quad (43)$$

c) The total intra and inter-MG calls of different access types per second, is given by:

$$Y_{intra\text{-}MG} = \sum_{k=1}^{n}\frac{A_k}{2S_k} P_{intra} \text{ and } Y_{inter\text{-}MG} = P_{inter\text{-}MG}\sum_{k=1}^{n}\frac{A_k}{2S_k} \quad (44)$$

d) The total inter-domain calls of different access types per second, is given by:

$$Y_{inter\text{-}domain} = P_{inter\text{-}domain}\sum_{k=1}^{n}\frac{A_k}{S_k} \quad (45)$$

Where P_{intra}, $P_{inter\text{-}MG}$, $P_{inter\text{-}domain,}$ P_{tr} and $P_{CS\text{-}call}$ define the proportions of intra and inter-MG calls, inter-domain calls, transit calls and calls originating or terminating in the CS, respectively.

e) The number of calls originating or terminating in the CS is given by:

$$Y_{CS\_call} = P_{CS\_call}\sum_{k=1}^{n}\frac{A_k}{S_k} \quad (46)$$

f) Finally, the number of forwarded calls per second corresponding to access type-*k* is given by:

$$Y_k = Y^k_{Intra\text{-}MG} + Y^k_{Inter\text{-}MG} + Y^k_{Inter\text{-}domain} + Y^k_{Transit} + Y^k_{CS\text{-}call} \quad (47)$$

Accordingly, the total number of forwarded calls/s corresponding to each MG can be written as follows:

$$Y_{MG} = \sum_{k=1}^{n}Y^k_{Intra\text{-}MG} + \sum_{k=1}^{n}Y^k_{Inter\text{-}MG} + \sum_{k=1}^{n}Y^k_{Inter\text{-}domain} + \sum_{k=1}^{n}Y^k_{Transit} + \sum_{k=1}^{n}Y^k_{CS\text{-}call} \quad (48)$$

In this section, we outlined the rules for dimensioning an MG. These rules can help operators determine adequate equipments for satisfying the users QoS needs.

D. Signaling Gateway Dimensioning Method

The SG converts the signaling exchanged between the NGN networks and the external connected networks. Notably, it ensures the signaling adaptation between the TDM access networks and the packet-based IP/MPLS transport network. The dimensioning of the SG is realized by determining the volume and capacity of the signaling traffic conversion and adaptation as well as the number and type of interfaces in the SG access and core sides.

In this section, the capacity of the SG in signaling traffic conversion and adaptation is determined in order to assess its performance.

D.1. Signaling Traffic at the Access Level

For the signaling traffic, each access type-k may use one of its corresponding q_k signaling (*ISUP*, *TUP*, etc.). For each call, the number and length of used signaling exchange messages (in bytes/s) depend on the direction (uplink/downlink) and the call status (succeeded/failed), as shown in [1, 14]. In fact, the signaling traffic and its corresponding *E1* may be calculated as follows:

a) First, the number of calls per second, which use different types of q-signaling is:

$$Y_{sig} = \sum_{k=1}^{n} \sum_{q=1}^{m_k} Y_{k,q} \quad (49)$$

Where, q represents the used signaling (e.g. *ISUP*, *TUP*, *INAP*, etc) per voice access service or Intelligent Network (*IN*) service.

b) Then, the required *BW* for the signaling messages traffic handled by the *SG* is:

$$A_{sig} = \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y_{k,q}^{i} Li_{q,i} Ni_{q,i} + \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y_{k,q}^{i} Lo_{q,i} No_{q,i} \quad (50)$$

c) Thus, the corresponding *E1* may be derived from the above signaling messages traffic *BW* as follows:

$$N_{E1,sig,acc} = \left(A_{sig,\, \phi/s} \times 8.10^{-6}\right)/2.048 \quad (51)$$

Where, the factor 8.10^{-6} is used to convert results to Mbps.

After determining the total signaling messages traffic BW and the corresponding required interfaces in the SG access side, next we address the same issue at the SG core level.

D.2. Signaling Traffic at the Core Level

After conversion, the signaling traffic is carried on packet switching-based SS7/SCTP/IP/MPLS Tunnels or LSPs established through the core network towards the CS. As discussed in C.2, the SCTP packet is composed of a common header, with size of 12 bytes and one or multiple bundled chunks. Each chunk is formatted with a block header, with size of 16 bytes and a variable data chunk payload "*D*". Furthermore, the IP packaging and MPLS labeling require an additional 20 and 4 bytes of overhead, respectively. The total length of these headers is then 52 bytes, which is transmitted each time a signaling packet is sent. As illustrated in Fig. 4, in the simple case, we assume that an SCTP packet consists of a header followed by only one data chunk with fixed "*D*" payload length in bytes. As discussed in [16], the total chunk-length should be a multiple of 4 bytes. If this is not the case, the sender needs to pad the chunk with "*P*" bytes, where *P = 1, 2, or 3*. In view of this discussion, the required BW corresponding to the E1 circuit emulation of the access side at the IP/MPLS core side can be computed using the following principle:

a) First, a voice signaling channel with 64 Kbps requires a bandwidth (BW) equal to:

$$64 \times (52+D+P)/D, \text{ in Kbps} \quad (52)$$

Where D is the data chunk payload and P is the number of added padding bytes.

b) Thus, the total *BW* requirement (in Mbps), at the IP/MPLS core level, needed for conveying the total signaling messages traffic is given by:

$$BW_{sig\_core} = A_{sig,Ø/s} \times 8.10^{-6} \times (52+D+P)/D \text{ Mbps}$$
$$= N_{E1,sig,acc} \times 2.048 \times (52+D+P)/D \text{ Mbps} \quad (53)$$

c) Accrdingly, the corresponding *E1* can be derived from the above *BW* as follows:

$$N_{E1,sig,core} = \frac{BW_{sig\_core}}{2.048} \quad (54)$$

Note that "*P*" may be used only if the total chunk-length is not a multiple of 4 bytes.

E. Call Server Dimensioning Method

In the NGN architecture, the CS is the central component that represents the control layer and the intelligence functions. In order to address the dimensioning of this entity, we shall first determine its Call Handling Capacity (CHC) and the interfaces required for call and control signaling exchanged with SGs, MGs and other CSs entities. This CS dimensioning depends basically on the number of MGs and SGs under its control, the signaling traffic distribution, the signaling traffic processing volume in terms of BW/s, and the number of handled calls/s. Practically, the CHC of a CS may be derived from the sum of the CHC of MGs under its control.

Voice Traffic

The control signaling messages relative to the voice and IN calls are routed through Permanent Virtual Connection on the SCTP/IP/MPLS core network towards the CS. The total volume of these messages is practically dependent on the number and type of the requested services. Therefore, the session signaling traffic can be estimated as follows:

a) The CS capacity, in terms of call processing (per second) is given by:

$$Y_{CS} = \sum_{k=1}^{n} Y_{Intra\_MG}^{k} + \sum_{k=1}^{n} Y_{Inter\_MG}^{k} + \sum_{k=1}^{n} \frac{Y_{Inter\_domain}^{k}}{2} + \sum_{k=1}^{n} \frac{Y_{CS\text{-}call}^{k}}{2} + \sum_{k=1}^{n} Y_{IN}^{k} \quad (55)$$

Where, $Y_{IN}{}^{k}$ is the number of calls (per second) of the IN type-k service. Note that the superscript k is not an exponent, but it is an index of the service type-k.

b) The number of calls (per second), using different q signaling is given by:

$$Y_{sig} = \sum_{k=1}^{n} \sum_{q=1}^{m_k} Y_{k,q} \quad (56)$$

Here, q represents the call signaling (e.g. *ISUP*, *TUP*), the *IN* signaling (e.g. *INAP*) or the control signaling (e.g. *SIGTRAN*, *H.248, MGCP, SIP*...).

c) The total *calls signaling traffic* handled (in bytes per second) by the *CS* is:

$$A_{sig} = \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y_{k,q}^{i} Li_{q,i} Ni_{q,i} + \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y_{k,q}^{i} Lo_{q,i} No_{q,i} \quad (57)$$

d) The total *control signaling traffic* (in bytes per second) exchanged with all nodes controlled by the *CS* is:

$$A_{sig\text{-}ctrl} = \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y_{k,q}^{i} Li_{q,i} Ni_{q,i} + \sum_{i=0}^{1}\sum_{k=1}^{n}\sum_{q=1}^{m_k} Y_{k,q}^{i} Lo_{q,i} No_{q,i} \quad (58)$$

e) Recall that, for the *IN* signaling traffic, each type-*k* service may use the *INAP* signaling. This type of signaling exchanges messages depending, in number and length, on the service type (*k*), the direction (uplink/downlink) and the call status (success/failure). In particular, the volume of *IN* signaling traffic treated by a CS (in bytes/s) can be estimated as follows:

$$A_{IN} = \sum_{i=0}^{1}\sum_{k=1}^{n} Y_{k}^{i} Li_{k,i} Ni_{k,i} + \sum_{i=0}^{1}\sum_{k=1}^{n} Y_{k}^{i} Lo_{k,i} No_{k,i} \quad (59)$$

f) The total *BW* of various signaling traffic in bytes per second and Mbps, respectively, is given by:

$$A_{CS\_sig,Ø/s} = A_{sig} + A_{sig\text{-}ctrl} + A_{IN} \quad (60)$$

$$BW_{CS,tot} = A_{CS\_sig,Ø/s} \times 8.10^{-6} \times (52+D+P)/D \text{ Mbps} \quad (61)$$

g) Finally, the total number of *E1* interfaces in the *inbound* and *outbound* sides of the *CS* is given by:

$$N_{E1,sig} = 2 \times \frac{BW_{CS,tot}}{2.048} \quad (62)$$

In this section, we outlined rules for dimensioning a *CS*. Operators can then select adequate equipments for satisfying the users' *QoS* requirements, according to these rules.

Data Traffic

The data traffic uses connections established through the IP/MPLS core network. Once the data session is established, it is managed by the control signaling and routing protocols of the transport network. In this work, we assume that voice is more important, requires more central CS processing, and occupies most of BW and slots. It is also assumed that, the unused BW and slots are allocated for data traffic, which is assumed to have negligible effects on the CS central processor load.

III. EXPERIMENTAL RESULTS AND DISCUSSION

In this section, we present and discuss experimental results which illustrate the practical implementation and advantages of the proposed dimensioning strategy in terms of performance gain and entities dimensioning benefits. To generate the experimental results, we developed an NGN entities' dimensioning software tool, which was used for dimensioning the different NGN components based on the different access network traffic. For our dimensioning purposes, we implemented an NGN architecture based on a transport network composed of several IP/MPLS nodes and four MGs and a control level consisting of two SGs and one CS.

It should be noted that the selected values of all input parameters used in this simulation are not absolute fixed values. They are only suitable and practical values used for our case study and practical purpose. Also, for simplicity, it is assumed that the message number and length in the uplink and downlink directions for various signaling traffic flows are the same. As well, the selected services may vary from one entity to another.

A. Media Gateway Dimensioning Implementation

First, the different connected voice and data services were fixed, for each MG. We selected the RTPC, RNIS-BA, RNIS-PRA and DIAL-UP as the voice service accesses connected to the MG. As listed in Table 1, for each voice service access, we specified the number of calls per second, the bearer occupancy ratio and the Mean Holding Time (MHT).

TABLE I. VOICE TRAFFIC SERVICE FEATURES

Voice access type	Call number	Occupancy ratio (Erl/call)	MHT (s)
RTPC	5 000	0.12	120
RNIS-BA	3 000	0.3	140
RNIS-PRA	2 000	0.6	140
DIAL-UP	800	0.6	600

Secondly, we also selected the ADSL, ATM, FR and IP as the data service accesses connected to the MG. As listed in Table 2, for each data service access we specified the number of calls per second, average throughput per call and the simultaneous attached users' ratio (SAU).

TABLE II. DATA TRAFFIC SERVICES FEATURES

Data access type	Call number	Average throughput (Mbps)	SAU
ADSL	1 200	0.256	0.25
ATM	40	34	0.5
FR	300	2	0.5
IP	100	4	0.35

Thirdly, as illustrated in Table 3, we specified the number and length of the control signaling messages (H.248/SCTP/IP/MPLS) exchanged with the CS, in the uplink and downlink directions for the cases of success and failure calls.

TABLE III. CONTROL SIGNALING FEATURES

Control Signaling with the CS	MSU length in ϕ	Number of MSU per success call	Number of MSU per failure call
H.248/SCTP/ IP/MPLS	52 + 160	12	3

Finally, the traffic distribution proportions are listed in Table 4.

TABLE IV. TRAFFIC DISTRIBUTION

Type of distribution	Proportion
INTRA-MSG	0.2
INTER-MSG	0.45
INTER-DOMAINE	0.30
CS-CALL	0.05
TRANSIT	0.2

The above input parameters were entered into our simulation tool and the corresponding dimensioning results are illustrated in Table 5.

TABLEAU V. MG CHC AND INTERFACES

CHC/s	MSG Access Side		MSG Core Side	
21	Voice traffic in Erlang	3180	Voice traffic in Mbps	659
	Data traffic in Mbps	1196	Data traffic in Mbps	1469
	The choice of interfaces (e.g. E1, STM1...) in the MSG access and core sides is essentially related to the volume of respectively calculated traffic, and the possibility of mixing various interfaces in the same equipment.			

Performance evaluation: We varied the traffic volume generated by an access network connected to the MG. Fig. 5 illustrates the variation of the MG capacity in calls' handling and traffic commuting according to the inbound generated traffic variation. This figure shows that the MG call handling capacity and performance variation strongly depends on the overall active connections in the access network level.

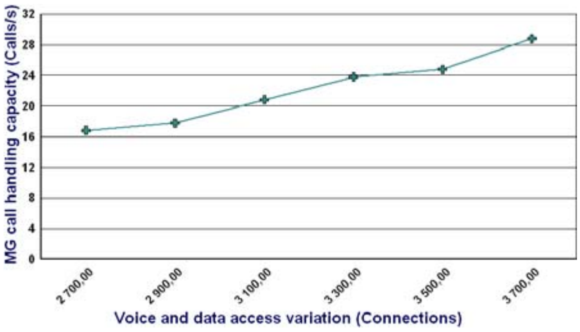

Figure 5. MG performance evaluation as a function of access users' traffic variation

B. Call Server Dimensioning Implementation

As listed in Table 6, for the CS dimensioning, we specified the number of IN services' calls per second, the number and length of the service message in the uplink and downlink directions for the cases of success and failure service calls. Also, Table 7 lists the selected number and length of each voice service call signaling message in the uplink and downlink directions for the cases of success and failure calls. Moreover, Table 8 illustrates the chosen number and length of the control signaling messages (H.248/SCTP/IP/MPLS) and (SS7/SCTP/IP/MPLS) exchanged with the MG and SG, respectively, in the uplink and downlink directions for the cases of success and failure calls. Finally, for all practical purposes, the proportions of failed and successful calls are set to be 0.2 and 0.8, respectively.

TABLE VI. IN SERVICE CALLS' SIGNALING FEATURES

IN access type	Call number	MSU length in octet (ϕ)	Nb. of MSU per success call	Nb. of MSU per failure call
Prepaid	20	500	2	1
Free phone service	30	430	2	1
Premium rate service	5	400	2	1

TABLE VII. VOICE SERVICE CALLS' SIGNALING FEATURES

Voice access type	MSU length in ϕ	Number of MSU per success call	Number of MSU per failure call
RTPC	15	3	2
ISDN-BA	30	3	2
ISDN-PRA	30	3	2
DIAL-UP	50	3	2

TABLE VIII. CONTROL SIGNALING FEATURES

Control Signaling with the MG and SG	MSU length in ϕ	Number of MSU per success call	Number of MSU per failure call
H.248/SCTP/ IP/MPLS	52 + 160	12	3
SS7/SCTP/IP/MPLS	52 + 45	5	3

Once the parameters concerning the IN calls' signaling and control signaling have been appropriately specified, we have completed the dimensioning stage. Table 9 illustrates the CS call handling capacity (CHC) and the necessary interfaces for the inbound and outbound sides.

TABLE IX. THE CS CHC AND INTERFACES

CHC per second	Signaling traffic in ϕ	E1 Interfaces per side
85	28 973	01

Performance evaluation: We varied the values of the voice access types connected to the MGs in order to assess the variation of the CS capacity as a function of generated traffic. Fig. 6 illustrates the dependence of the CS capacity, defined in terms of call handling, and signaling traffic processing and commuting, on generated voice traffic evolution. Note that the CS call handling capacity is approximately equal to the sum of calls handled by all MGs under its control (Four MGs), as illustrate Fig. 5 and Fig. 6. That is, the CS performance variation is closely related to the overall controled MGs performance.

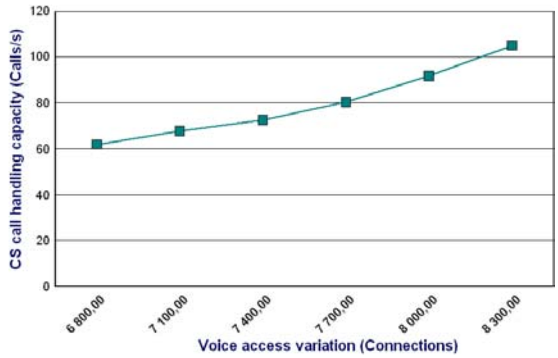

Figure 6. CS performance evaluation as a function of various signaling traffic variation

C. Signaling Gateway Dimensioning Implementation

For the SG dimensioning, we used the same IN Service Calls' Signaling Features and Voice Calls' Signaling Features as listed in Table 6 and Table 7, respectively. Also, the number and length of the call control signaling messages (SS7/SCTP/IP/MPLS) exchanged with the CS, in the uplink and downlink directions for the cases of success and failure calls are illustrated in Table 8.

The output of the SG dimensioning, consisting of the call signaling processing (traffic conversion and adaptation) and corresponding interfaces, is illustrated in Table 10.

TABLE X. SG CALL SIGNALING CONVERSION AND INTERFACES

Call signaling /s	Signaling traffic in ϕ	E1 Interfaces per side
48	28 973	01

Performance evaluation: At this stage, we varied the value of inbound traffic volume generated by various access networks connected to the Media Gateways in order to assess the variation of the SG capacity as a function of generated signaling traffic. Fig. 7 illustrates the variation of the SG capacity in call signaling traffic conversion and adaptation according to the generated traffic variation. This figure clearly shows that the SG processing capacity and performance variation strongly depend on the overall active voice and IN connections in the access network level.

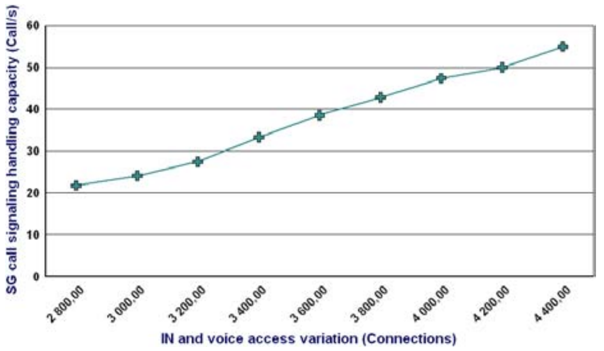

Figure 7. SG performance evaluation as a function of the IN and call signaling variation.

In this section we illustrated the simulation results corresponding of the dimensioning of the three main NGN components (MG, SG and CS). In practice, assessing the dependence of the capacity of each NGN component on the traffic volume generated by different access network types can provide us with a strategy for selecting the adequate range of equipments and meeting the required capacity. Some of the resulting benefits include avoiding under and over dimensioning, minimizing incoming traffic blockage and congestion, and enhancing the required QoS.

IV. CONCLUSION

In this work, we investigated the fundamental multi-service traffic relationships between the performance, capacity and QoS requirements for high-speed NGN networks. Dimensioning NGN networks components involves exploiting available related traffic flows in order to estimate the necessary capacity. This capacity may be in terms of physical devices performance. The call handling capacity is almost always the limiting factor.

The proposed dimensioning strategy ensures that NGN networks are robust, resilient towards traffic variations, and protected from under or over dimensioning problems. More importantly, the proposed NGN dimensioning also ensures that QoS guarantees offered to subscribers are always realized, and allocates resources in an accurate and reliable manner. It also improves the control and transport network scalability and resilience, while maintaining fair and high resource utilization.

We also presented and discussed experimental results that illustrate the practical implementation and advantages of the proposed dimensioning strategy in terms of performance gain and entities dimensioning benefits. In particular, we investigated the performance of the MG, SG and CS dimensioning strategies by experimentally assessing the relationship between the MG, SG and CS capacity on the generated traffic, respectively. In practice, the selection of adequate range of equipments, which meet the required capacity, is determined based on the dependence of the capacity on the generated traffic.

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