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Analysis and Study of IP Telephony Traffic Characteristics over Next Generation Network with NGN Carrier's Case Study

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NGN (Next Generation Network) refers to a network developed to be suitable for an environment in which there is a convergence between wired, wireless communications and broadcasting. Most telecommunication providers have a plan or are conducting a migration of their network to NGN. One of the most important issues for constructing NGN is to provide end-to-end QoS (Quality of Service). This paper aims to propose carriers, who begin to evolve their network to NGN, a construction strategy in the interests of QoS by introducing an example of nationwide NGN in Korea. In viewing the strategy, we develop converged services, and define services provided through NGN first. Next, we define our own standard of service performance metrics, network performance metrics, and quality of service policy from the transport stratum view point. Then we design and construct NGN. Finally, we verify end-to-end performance objectives comparing predefined metrics with collected measurement data on the NGN and derive improvements. This paper deals with voice and video telephony data to analyze the traffic characteristics statistically. It should be noted that voice and video are real time data and reflect the absolute need for QoS in the network.

1. Introduction

Until recently, telecommunication networks have been broadly categorized into wired and wireless networks, and their respective business areas have been firmly established. Legacy communication services are provided through their unique network infrastructures. However, the existing fixed boundaries between services are in the process of being dismantled and are evolving into a unified IP network. Telecommunication service providers are constructing or have a plan to construct a NGN which is an IP Multimedia Subsystem (IMS) based on a unified IP packet network. It supports the migrate of the Public Switched Telephone Network (PSTN) service to IP telephony as well as to provide various multimedia services by integrating the heterogeneous networks. Consequently NGN supports both legacy and new emerging services concurrently. The examples of service types supported by NGN are multimedia services, PSTN/ISDN emulation services, PSTN/ISDN simulation services, various data services, Internet access and public services according to ITU-T [1)]. **Figure 1** describes a network evolution to NGN. As PSTN, Cellular, Internet, and TV network are integrated to the IP packet network, service and control functions are integrated.

The standard of NGN architecture has been defined by ETSI and ITU-T mainly, but based on 3GPP's IMS architecture [4)]. NGN functions consist of service and transport strata [2),3)]. The service stratum supports session and non-session based services, and the transport stratum provides IP connectivity for the corresponding service stratum's request. Multi-Protocol Label Switching (MPLS) has become the selection of a packet core transport technology to support converged services in the NGN [7)]. Telecommunication service providers have adopted a technology, which is Differentiated Service (DiffServ) over an MPLS network, for providing the quality of service of the classified traffic flows [8)]. In wired-line subscriber's networks, FTTH technologies are adopted as a transmission solution in NGN [10),11)].

This paper proposes an NGN construction strategy from a quality of service point of view by seeking an example of nationwide NGN construction in Korea. The paper analyzes IP telephony traffic characteristics which require real time transmission and urgency of QoS by concentrating on transport stratum. As measurement of transport stratum performance is quite similar to the transmission performance of the general IP network, Performance objectives of IP network are used. It is dealt with an end to end connection as a section of measurement. The paper introduces carrier 'A' who has already constructed a nationwide NGN and carrier 'B' who has started to construct an NGN from the regional areas and tries to expand to nationwide. These carriers have different approaches of NGN construction because they have different network resources and markets.

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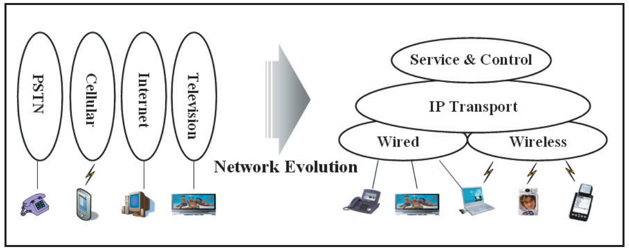

Fig. 1 Evolution of heterogeneous networks to NGN.

We describe NGN architectures of carrier 'A' and 'B' and present the analysis of IP telephony traffic characteristics of nationwide NGN carrier 'A'. We believe that our proposed strategy provides proper guidelines for those who begin to evolve their network to NGN. The rest of the paper is organized as follows. Section 2 suggests our own standards of performance objectives for the NGN and describes NGN architecture of carriers. Section 3 describes traffic measurement and Section 4 presents an analysis of IP telephony traffic characteristics and verification of performance by comparing them with service performance objectives. The paper concludes in Section 5.

2. Network Architectures of NGN Carriers

This section presents a definition of services as a first step, a definition of service and network performance objectives as a second step, and the design of a network which contains a quality of service policy as the third step of the strategy.

2.1 Definition of Services

We have developed converged services as a deriver of NGN construction. **Table 1** shows the developed services. The service models in Table 1 are categorized from the provider's view point. Voice and video IP telephony, IPTV are representative services. Carrier 'A' defines services provided through NGN from pre-existing ones and new converged services before construction. Defined services are voice IP telephony migrated from PSTN, Multi-Media over IP (MMoIP) contains video IP telephony, WiBro (Mobile WiMAX) and VPN for enterprises. Internet access service is excluded except WiBro. Carrier 'A' approaches to construct a new core network which is independent of the pre-existing Internet core.

Table 1 Developed services.

Service Model	Applications of Service Model
Voice telephony	Voice IP telephony (PSTN quality VoIP)
Video telephony (MMoIP)	Video IP telephony Video conference Multimedia caller ID & ring-back tone Video mail box Presence information
Service based on open API (MMoIP)	Joint watching movie over video phone Guerilla vote over video phone Information for living over video phone Interworking with groupware applications
Video telephone interworking	Video IP telephony and Video conference between wired and wireless phone
IPTV	SD & HD quality VoD Near VoD and Channel Broadcast Interactive data broadcast

We expect that the new core will replace the Internet core and absorb the internet access services for the future. On the other hand carrier 'B' approaches to upgrade the pre-existing Internet core for providing VoIP, enterprise VPN, and Internet access services and plans to construct a new network for IPTV multicast service only. Carrier 'A' has as a main object PSTN to IP telephony migration as a leading telecommunication service provider who has about 90% market share of PSTN service. This carrier approaches the provision of an IPTV service aggressively also. Carrier 'B' approaches to enter the voice telephony market through VoIP service and has a different approach for IPTV service by using many kinds of HD quality contents.

2.2 Performance Objectives of Service and Network

Metrics are used for verifying quality of service and customer satisfaction. ITU-T Y.1541 recommendation provides classes of network and quality of service with objectives for IP network performance parameters[20]. In terms of VoIP, R-value of E-model implies the transmission rating model recommended

Table 2 Service performance objectives.

Service Cat.	Grade	Metrics: Call Success Rate	R-Value	Loss Rate	One-Way Delay	Jitter
Inter-active voice	S+	>99%	>90	10^{-5}	<20ms	<50ms
	S	>97%	>80, <90	10^{-4}	<50ms	<50ms
	A	>95%	>70, <80	10^{-3}	<100ms	<50ms
On-demand voice	S+	>99%		10^{-5}	<20ms	<100ms
	S	>97%		10^{-4}	<50ms	<100ms
	A	>95%		10^{-3}	<100ms	<100ms
Inter-active video	S+	>99%		10^{-5}	<20ms	<50ms
	S	>97%		10^{-4}	<50ms	<50ms
	A	>95%		10^{-3}	<100ms	<50ms
On-demand video	S+	>99%		10^{-5}	<20ms	<100ms
	S	>97%		10^{-4}	<50ms	<100ms
	A	>95%		10^{-3}	<100ms	<100ms

Service Cat.	Grade	Metrics: Loss Rate	One-Way Delay	Jitter	Down-load Speed	Avail-ability
Data (Internet access)	S+	10^{-5}	<100ms		>20Mbps	
	S	10^{-4}	<100ms		>10Mbps	
	A	10^{-3}	<100ms		>5Mbps	
Lease Line (VPN)	S+	10^{-5}	<100ms	<50ms		>99.99%
	S	10^{-4}	<100ms	<50ms		>99.7%
	A	10^{-3}	<100ms			>99.5%

by ITU-T G.107 [19]. R-value indicates the level of user satisfaction for an end-to-end transmission. We defined metrics and objectives of service and network performance through an actual proof of field test-bed before nationwide NGN construction. The different grades of quality of services are defined as A, S, S+ (as shown in **Table 2**). Services are defined as interactive voice, on-demand voice, interactive video, on-demand video, Internet access and VPN. **Table 3** shows three different grades of network performance objectives. The expected

Table 3 Network performance objectives.

Grade	Metrics: Loss Rate	One-Way Delay	Jitter	Avail-ability	MTBF	MTRS
S+	10^{-5}	<10ms	<50ms	>99.99%	8640 hour	<1 hour
S	10^{-4}	<30ms	<50ms	>99.7%	4320 hour	<2 hour
A	10^{-3}	<80ms	<50ms	>99.5%	2880 hour	<3 hour

grade of general IP network is 'A' grade of the suggested performance objectives.

2.3 Implementation of Service Stratum

In the short term, 'A' carrier aims to lure 5.5 million IP telephony, 3 million IPTV and 2 million mobile WiMAX subscribers onto its NGN by the year 2010. To accomplish the design of service stratum, we need to estimate the amount of subscribers per service, region and year. Therefore the decision of the control systems capacities, control links and reliability of the control network is driven based on estimated demands [12]. Eq. (1) describes the model of capacity and the expense calculation process of control systems.

$$Q = \sum_{r=1}^{R} PC_r / PC_u, \qquad PC = \sum_{i=1}^{SV} SB_i \cdot BHCA_i \tag{1}$$

Q is the quantity of control systems, PC_r is the required processing capacity of a control system at a coverage r, PC_u is the unit processing capacity of one control system. R is the number of service coverage. SB is the number of subscribers for the service, $BHCA$ is the busy hour call attempt per service, and SV is the number of services which need to be processed by control systems. Eq. (2) shows the model that explains capacity and the expense calculation process of control links.

$$LC = LC_T/_w C_2 \tag{2}$$

$$LC_T = \sum_{i=1}^{SV} SB_i \cdot BHCA_i \cdot TR_i \cdot ML_i / 3600$$

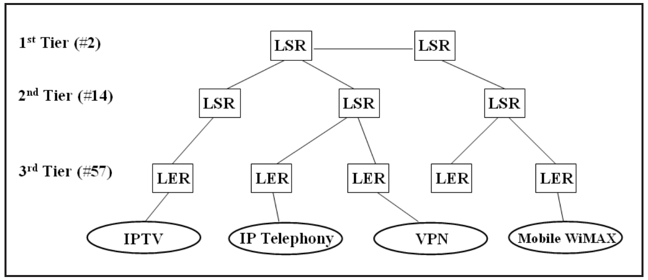

Fig. 2 Network architecture of NGN carrier 'A'.

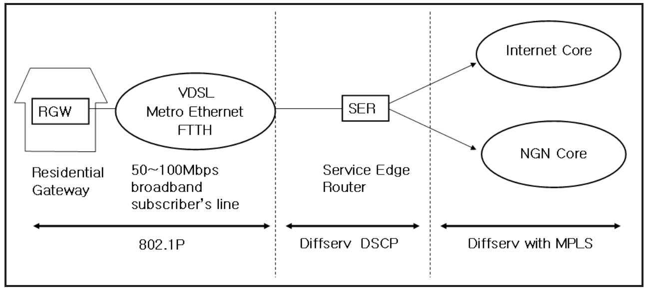

Fig. 3 QoS policy of NGN carrier 'A'.

LC is the capacity of the per control link, LC_T is the total amount of control traffic between two control systems. w is the total number of session controllers in the control network. TR is the number of required transactions and ML is the length of control message for service provisioning.

Carrier 'A' has implemented control systems which have capacities of 2 million subscribers based on voice and video IP telephony services at present and will expand the capacities corresponding to increase in subscribers. The call control of IP telephony services is considered so far. However if any application over IPTV or mobile WiMAX needs call session control, the more capacity of control systems should be implemented. Carrier 'B' has implemented 1.2 million subscriber capacities control systems for voice only IP telephony and constructed a trial control system for video IP telephony. The capacity of control system affects mainly call success ratio of the service performance objectives.

2.4 Implementation of Transport Stratum

Since IP packet technology is adopted as the transport stratum in NGN, Resource and Admission Control (RAC) as a traffic control technology is considered to able to accommodate the quality of service demands of the service stratum [5),6)]. Both carrier 'A' and 'B' are still developing RAC function, therefore the function is not applied yet. As a result, quality of service depends on the transmission capabilities of transport stratum. Carrier 'A' has constructed IP/MPLS core network in three tiers as shown in **Fig. 2**. Two Label Switched Router (LSR) nodes in the first tier, 14 LSR nodes in the second tier, and 57 Label Edge Router (LER) nodes in the third tier exist at present. The link capacity between each node is 40 G × N or 2.5 G to 10 G × N.

The new core network is independent of pre-existing broadband Internet core as shown in **Fig. 3**. Subscriber's networks which consist of VDSL, Metro Ethernet and FTTH support over 50 Mbps have been constructed. Carrier 'A' will expand Metro Ethernet and FTTH support 100 Mbps up to 90 percentile ratios of its subscribers by 2010. Service Edge Router (SER) is used for classifying the traffic and forwarding NGN core or Internet core network. As a result, core networks are selected according to service types and subscribers' networks are shared for both NGN and non-NGN traffics.

From a Residential Gate-Way (RGW) to a subscriber's network termination point uses 802.1p, a metro region uses DSCP (DiffServ Code Point) for traffic classification and it uses an MPLS experiment field of DiffServ aware IP/MPLS in NGN core for supporting quality of service. All sections have no less than four classes. In queue management, PQ (Priority Queuing) and WFQ (Weight Fair Queuing) based on CBQ (Class based Queuing) are used. RED (Random Early Detection) is used for congestion avoidance [9)]. There is no adoption of a RED in class A. Class B, C and BE applying RED drop preference from 50%, 30% and 20% of allocated bandwidth respectively. A traffic conditioning such as traffic shaping and policing is also used. **Table 4** and **Table 5** show the queue policy and supported services per class.

Carrier 'B' has upgraded eight nodes of pre-existing Internet core by applying IP/MPLS technology and will expand to upgrade. The link capacity of inter nodes is 40 G × N or 10 G × N. There are mainly HFC and Metro Ethernet in

Table 4 Subscriber's network queue policy of carrier 'A'.

Class	DSCP	Queue Policy	Services
Class A	CS 6	PQ	VoIP, MMoIP, IPTV multicast
Class B	CS 4	WRR/WFQ	IPTV unicast (VoD)
Class C	CS 2	WRR/WFQ	Reserved
Class BE	CS 0	Best Effort	Internet access etc.

Table 5 NGN core network queue policy of carrier 'A'.

Class	Exp	Queue Policy	Services
class A	7	-	Reserved
	6	PQ	VoIP, MMoIP, Signaling
class B	5	WRR/WFQ, RED	Reserved
	4		IPTV multicast
class C	3	WRR/WFQ, RED	IPTV unicast (VoD)
	2		VPN
Class BE	1	RED	Reserved
	0		Mobile WiMAX

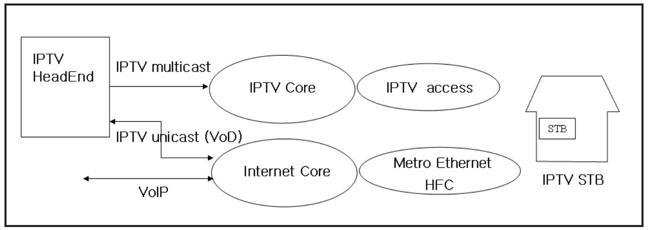

Fig. 4 Network architecture plan of NGN carrier 'B'.

subscriber networks. The QoS policy of subscriber networks is to use 4 classes and is similar to Table 4. However the allocated services are different from carrier 'A'. VoIP and MMoIP are in class A, IPTV unicast (VoD) and multicast are in class B, Internet access is in class BE and class C is reserved. There are 4 classes in the core network and the allocated services are the same as those of subscriber networks except for IPTV multicast. Carrier 'B' plans to construct new core network only for IPTV multicast that is real time broadcast by using Ethernet over SDH/SONET technology. Carrier 'B' runs parallel to upgrade Internet core, HFC and to construct Metro Ethernet for evolving to NGN. **Figure 4** shows the network architecture plan of carrier 'B'.

3. NGN Traffic Measurement

A traffic measurement as the fourth step of the strategy is described in this section.

3.1 Video IP Telephony User Equipment

'A' carrier provided trial services with IP telephones to about 700 households for verifying the feasibility of commercialization of video IP telephony over NGN. 'B' carrier provided almost the same trial services to about 300 households as well. Two kinds of IP telephones were distributed to both subscribers. These equipments support VoIP and MMoIP with video IP telephony and bandwidth from 64 kbps to 4 Mbps. The IP telephone has each voice and video de-jitter buffer and operates for alleviation of effect by jitter.

3.2 Implementation of Metric Measurement

We developed the metric measurement program as described in Ref. 14). And we deployed it in the IP phones which are provided to trial subscribers. Metric data were collected for a month in the trial service period. In terms of the measurement itself, the measurement pattern is IP telephone – IP network – IP telephone and the measurement type is passive because of one point measurement which is performed by real user equipment. It is regarded as a standard condition of the end-to-end reference system [13]. The program measures call success ratio, packet loss, one-way delay (maximum, average and minimum value), jitter, R-value, MOS (Mean Opinion Score), required traffic bandwidth (bps), and image resolution [15)–17)]. The measured data of the phone are sent and corrected through RTCP-XR (Real-time Transport Control Protocol Extended Report) messages [18]. A flow chart of the algorithm is shown in **Fig. 5**. Information contained in SIP (Session Initiation Protocol) and RTP (Real-time Transport Protocol) are corrected by capturing and filtering when packets are received. If a packet is SIP Invite message, a session table and a RTP status table are created. Parsed information is stored in the session table also.

The program extracts the resolution as well at this time. If it is receiving a SIP

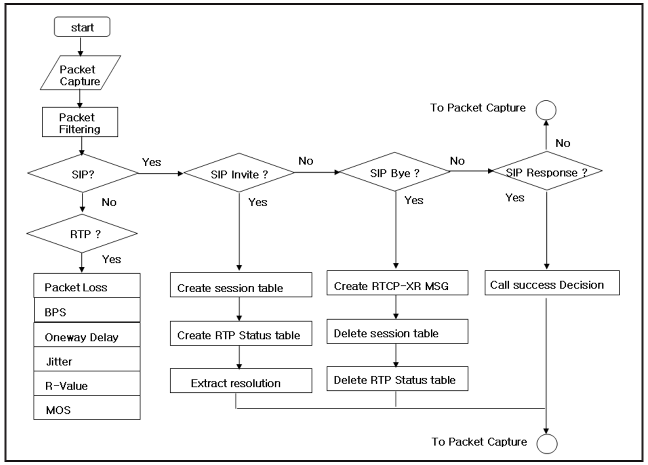

Fig. 5 Algorithm of measurement program.

Bye message, a RTCP-XR message is created by using information of the session table and the RTP status table. These tables are deleted after the RTCP-XR message is reported. If a packet is SIP Response message, the program has to decide whether the call is successful or not. The value of the status code is checked, 2xx and 3xx are regarded as success, and 4xx, 5xx, and 6xx as fail. If a packet is RTP message, bandwidth, packet loss, one-way delay, jitter, R-value and MOS are calculated and stored in the RTP status table. The packet loss is acquired from received RTP packet counts in real and estimated sending RTP packet counts driven by RTP sequence number. One-way delay is the half of RTT (Round Trip Time) which is driven from information of a RTCP Receiver Report message. Jitter is calculated from time of arrival and RTP time stamp. R-value is driven from the equation recommended by G.107 applying default values. MOS is calculated from R-value [17),19)].

4. Analysis and Study of Traffic Characteristics

As the final step of the strategy, we analyze the traffic characteristics and verify the performance of the NGN using statistical methods in this section. Voice and video call traffics are dealt with in the various aspects.

Table 6 The number of observations required for measure of time.

s/mean(x)	observations
<0.1	100
0.1 ~ 0.3	1,000
>0.3 ~ 0.5	2,500
>0.5 ~ 0.7	5,000
>0.7 ~ 0.9	7,500
>0.9	10,000

4.1 The Number of Samples for Traffic Analysis

If k unsuccessful calls are observed out of N call attempts, then the true value of the unsuccessful call ratio lies between $k/N - \Delta$ and $k/N + \Delta$ with a confidence level $1-\alpha$, Δ being approximated by Eq. (3), on account of the large value of N. p is expected unsuccessful call ratio and $\sigma(\alpha)$ is the $(1 - \alpha/2) \times 100$ percentile of the normal distribution with mean 0 and standard deviation 1 ($N(0,1)$). Therefore we derive the number of samples if there is required accuracy for p. If required accuracy for p is equal to or less than 0.01, Δ $(= \Delta p)$ is set to 0.001 or if that is more than 0.01, Δ $(= \Delta p)$ is set to $0.1p$ [21)]. As Eq. (3) shows the number of samples for verifying loss ratio, we apply the equation for the exactness of unsuccessful call ratio as well as packet loss ratio in the paper.

$$\Delta \approx \sigma(\alpha) \times \sqrt{\frac{p \cdot (1-p)}{N}} \Rightarrow N = \frac{\sigma(\alpha)^2 \cdot p(1-p)}{\Delta^2} \tag{3}$$

The number of samples related with measurement of time is derived from Eq. (4), where $z_{1-\alpha/2}$ is the $1-\alpha/2$ percentile of the standard normal distribution. S is the expected standard deviation of the time for measurement. $mean(x)$ is the expected mean value of the time, and a is the relative accuracy. Even though there is no standard deviation, an estimate should be available for use with this formula. If there is $z_{1-\alpha/2} = 1.96$ for a confidence level 95%, and $a = 2\%$, the number of observations are derived from **Table 6** [21)]. We apply Table 6 for the exactness of delay and jitter in the paper. In relation to exactness of R-value, we consider both Eq. (3) and Table 6 because R-value is derived from delay, jitter and packet loss.

Table 7 Descriptive statistics of voice call traffics.

Statistics	N	Mean	95% mean interval	StDev
R-value	13101	92.988	92.9622-93.0136	1.501
Delay(ave)	13101	7.835	7.568-8.101	15.555
Delay(max)	13101	11.547	11.218-11.875	19.184
Jitter	13101	0.4017	0.3873-0.4160	0.8395
Packet loss	13101	0.00118	0.000937-0.001425	0.01424
Bandwidth	13101	72.558	72.3099-72.8062	14.489

$$n = \frac{(z_{1-\alpha/2})^2}{a^2} \times \left[\frac{s}{mean(x)}\right]^2 \quad (4)$$

4.2 Voice Call Traffic Characteristics

(1) General Characteristics of Voice Call Traffics

Approximately 13,000 Voice calls are attempted for a month in the NGN of carrier 'A'. In **Table 7**, R-value, average delay, maximum delay, jitter, packet loss and required bandwidth of every call are presented. The mean values of each metric are estimated from one-sample T test in which interval estimation is performed with 95% of confidence level. The mean value of R-value is 92.988 and the interval of the metric is from 92.9622 to 93.0136. The average delay is 7.835 ms and exists from 7.568 to 8.101 ms. The maximum delay is 11.547 ms and exists from 11.218 to 11.875 ms. Jitter is 0.4017 ms and exists from 0.3873 to 0.4160 ms. Packet loss is 0.00118 and exists from 0.000937 to 0.001425. Required bandwidth is 72.558 kbps and exists from 72.3099 to 72.8062 kbps. Call success ratio is 100% in all the attempted calls. About the delay and jitter, it is verified that more than 10,000 observations are required for the exactness of analysis results because of $s/mean(x)$ is more than 0.9. According to the Eq. (3), 13,000 observations are required for the exactness of an analysis result if it is verified based on 3% loss ratio, and 39,600 observations are required if it is verified based on 1% loss ratio about the packet loss. Approximately 4 million samples are required if it is verified based on more than S grade of Table 2. Therefore 13,000 samples are enough for the measure of time, and it is not certain that the result of packet loss analysis is exact. However it is possible to check the tendency of packet loss because the number of samples is per voice call basis and lots of packets were transferred per voice call. We limit the meaning of R-value analysis to checking the tendency rather than the exactness.

Table 8 Comparison of voice call traffic characteristics and service performance objectives.

One-sample Z	N	Mean	H_0	H_1	P-value
R-value	13101	92.9879	$\mu = 90$	$\mu > 90$	0.000
Delay(ave)	13101	7.835	$\mu = 20$	< 20	0.000
Delay(max)	13101	11.547	$\mu = 20$	< 20	0.000
Jitter	13101	0.40166	$\mu = 50$	< 50	0.000
Packet loss	13101	0.001181	$\mu = 0.001$	$\mu < 0.001$	0.927
			$\mu = 0.01$	$\mu < 0.01$	0.000

Table 8 shows that comparison between the statistics of the samples in Table 7 and service performance objectives defined in Table 2. We perform one-sample Z test with 95% confidence level for checking the voice call service performance. H_0 is null hypothesis and H_1 is alternative hypothesis. If P-value is higher than significance value, the result of the analysis has a tendency of H_0, otherwise it means H_1. The significance value is 0.05 when the confidence level is 95%. R-value is estimated over 90, both average delay and maximum delay are estimated less than 20 ms, and jitter is estimated less than 50 ms. Packet loss has a tendency of about 10^{-3} loss rate. Call success rate is 100% of course. Call success rate, R-value, delay, and jitter are estimated S+ grade of service performance objectives. Packet loss is estimated A grade therefore it is necessary to check and improve the decreasing of the packet loss ratio.

(2) Correlation of Metrics

We perform a correlation analysis of between R-value and the other metrics in $\alpha = 0.05$ that is 5% of significance level. The result shows that all other metrics are correlated with R-value, and the correlation coefficient of average delay, maximum delay, jitter and packet are −0.931, −0.781, −0.060 and −0.058 in each. The correlation coefficient of bandwidth is rarely related with R-value as −0.018. Delay has a relatively high correlation with R-value. Generally we presume the effect of jitter is very high, but the de-jitter buffer of the user equipment may absorb the jitter, and transform to delay. We check the correlation using a three

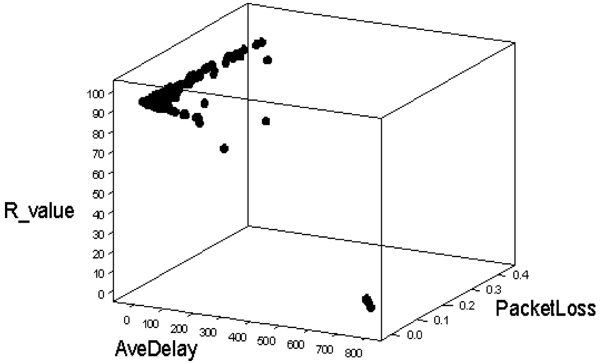

Fig. 6 Three dimensional regression plot.

dimensional regression plot focusing on R-value, average delay and packet loss. The plot is shown in **Fig. 6**. R-value is located in the left upper side of the cube in general. It is found that there is linearity between R-value and average delay, but not between R-value and packet loss. We don't find linearity between R-value and jitter either using a two dimensional plot. Regression equation is driven as Eq. (5) by the use of regression test, in which alpha to remove value equals 5%.

As adjusted coefficient of determination is 86.7%, the regression model has a comparably higher explanation. However we limit the meaning of the equation that checks a tendency of relation average delay with R-value which represents voice IP telephony quality rather than the exactness of carrier 'A'.

$$\text{R-value} = 93.69 - 0.0898 \cdot delay(average) \quad (5)$$

(3) Comparison of Ordinary and Peak Time

The peak times of voice calls are identified in the 17 to 18 time slot. We collected 1,271 voice call data in this time slot for a month. The mean value of R-value, average delay, maximum delay, jitter, packet loss and bandwidth are 93.015, 7.781 ms, 11.388 ms, 0.4219 ms, 0.00042, and 72.5112 kbps respectively. The result is shown in **Table 9**. Analysis of variance (ANOVA) is performed for the purpose of comparison of the R-value and average delay in each time slot. The results of analysis of R-value and average delay are shown in **Figs. 7** and **8** respectively. R-value of each time slot has a tendency of no difference as P-value is 0.515. Average delay has a tendency of difference as P-value is 0.009. We

Table 9 Descriptive statistics of peak time voice calls.

Statistics	N	Mean	95% mean interval	StDev
R-value	1271	93.015	93.0041 - 93.0260	0.198
Delay(ave)	1271	7.781	7.346 - 8.215	7.893
Delay(max)	1271	11.388	10.673 - 12.104	13.007
Jitter	1271	0.4219	0.3723 - 0.4715	0.9011
Packet loss	1271	0.00042	0.000089 - 0.000760	0.00609
Bandwidth	1271	72.5112	71.6707 - 73.3517	15.2733

One-way ANOVA: R_value versus Time

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Analysis of Variance for R_value
Source     DF        SS        MS        F       P
Time       21     45.28      2.16     0.96   0.515
Error   13079  29459.65      2.25
Total   13100  29504.93
                                 Individual 95% CIs For Mean
|                                Based on Pooled StDev
Level       N      Mean     StDev  -----+---------+---------+---------+-
 0         35    93.096     0.119            (----*----)
 1          8    93.059     0.123       (----------*---------)
 2          3    93.200     0.000  (----------------*----------------)
 5          6    92.980     0.074     (-----------*-----------)
 6          9    93.038     0.079       (--------*---------)
 7        105    93.033     0.104             (--*--)
 8        429    93.013     0.163               (*-)
 9        801    92.990     0.648               (*)
10        974    93.016     0.272               (*)
11        920    93.031     0.183               (*)
12        736    93.037     0.187               (*)
13        836    93.032     0.148               (*)
14        906    92.836     3.894             (*)
15        950    93.019     0.197               (*)
16       1129    92.940     2.519               *)
17       1271    93.015     0.198               (*)
18       1135    92.997     0.218               (*)
19       1046    92.886     2.747              (*)
20        777    93.022     0.212               (*)
21        611    93.019     0.154               (*)
22        318    93.027     0.162               (*-)
23         96    93.029     0.175             (--*--)
                                   -----+---------+---------+---------+-
Pooled StDev =    1.501               92.0      93.0      94.0      95.0
```

Fig. 7 ANOVA for R-value in each time slot.

Test for Equal Variances: AveDelay vs Time

One-way ANOVA: AveDelay versus Time

```
Analysis of Variance for AveDelay
Source     DF        SS        MS        F        P
Time       21      9481       451     1.87    0.009
Error   13079   3160269       242
Total   13100   3169750
                                   Individual 95% CIs For Mean
                                   Based on Pooled StDev
Level       N      Mean     StDev  -----+---------+---------+---------+-
 0         35      4.49      4.63                (----*---)
 1          8      6.10      5.19             (--------*--------)
 2          3      0.08      0.14  (--------------*--------------)
 5          6      9.10      2.86              (----------*---------)
 6          9      8.31      3.28               (--------*-------)
 7        105      6.94      4.17                    (--*-)
 8        429      7.69      4.94                      (*-)
 9        801      8.27     12.57                       (*)
10        974      7.60      8.86                       *)
11        920      6.89      5.58                      (*)
12        736      6.84      7.51                      (*)
13        836      6.97      5.14                      (*)
14        906      9.29     36.40                        (*)
15        950      7.21      6.22                      (*)
16       1129      8.30     23.81                       (*)
17       1271      7.78      7.89                       *)
18       1135      8.28      7.59                       (*)
19       1046      9.54     26.88                        (*)
20        777      7.43      8.49                      (*)
21        611      7.35      3.98                      (*)
22        318      7.12      5.42                      (*)
23         96      6.47      5.07                    (-*--)
                                   -----+---------+---------+---------+-
Pooled StDev =    15.54               -12         0        12        24
```

Fig. 8 ANOVA for average delay in each time slot.

limit the meaning of the results to check tendencies of voice IP telephony traffics rather than exactness because of the number of samples in each time slot.

For the purpose of checking the traffic characteristics of peak time and ordinary time in depth, we compare 17 to 18 time slot with the 8 to 9 time slot which is regarded as ordinary time. The 8 to 9 time slot has a similar number of call attempts as 429 to average call attempts as 422 for one month. Analysis of two time slots is shown in **Table 10**. The notation (p) means peak time and (o) means ordinary time. H_0 as null hypothesis means that (p)'s mean value equals (o)'s mean value in each metric. On the other hand H_1 as alternative hypothesis means that two values are not equal. If P-value is higher than the significance value, the result of the analysis has a tendency of H_0, otherwise it means H_1. The significance value is 0.05 because the confidence level is 95%. R-value, average delay, maximum delay, and packet loss have tendencies of no difference. In the case of jitter, it is analyzed that the value of peak time is higher than that of ordinary time. Generally there are no apparent difference tendencies. It is assumed that the traffic is relatively very low in comparison with the transport capacity of the network because of the small number of trial subscribers.

Table 10 Comparison of ordinary and peak time voice calls.

ANOVA	N	Mean	H0	H1	P-value
(p)R value	1271	93.015	(p)= (o)	(p)≠ (o)	0.825
(o)R value	429	93.013			
(p)Delay(ave)	1271	7.781	(p)= (o)	(p)≠ (o)	0.822
(o)Delay(ave)	429	7.690			
(p)Delay(max)	1271	11.39	(p)= (o)	(p)≠ (o)	0.195
(o)Delay(max)	429	10.53			
(p)Jitter	1271	0.4219	(p)= (o)	(p)≠ (o)	0.004
(o)Jitter	429	0.2953			
(p)Packet loss	1271	0.000425	(p)= (o)	(p)≠ (o)	0.722
(o)Packet loss	429	0.000559			
(p)BW	1271	72.51	(p)= (o)	(p)≠ (o)	0.516
(o)BW	429	73.05			

(4) **Comparison of Different Distances**

It is observed that the traffic characteristics are biased by distances. When the distance is relatively close between calling party and called party, both of them share the same second tier LSR as shown in Fig. 2. On the other hand, when the distance is relatively large between calling party and called party, two parties share the same first tier LSR. However the second tier LSR of each party is different. We have used notations of S and L to indicate near and far distances respectively. The collected data of S is about 8,146 and that of L is about 551. We limit the meaning of the analysis results to as a check on a tendency of distance because the number of observations cannot provide exactness. **Table 11** presents the result of ANOVA. R-value, the representative quality metric, is not different between two distances. Jitter has difference but it is not apparent because P-value is very close to significance value as 0.042. Packet loss has a tendency of difference.

Table 11 Comparison of different distances.

ANOVA	N	Mean	H0	H1	P-value
(S)R value	8146	92.964	(p)=	(p)≠	0.588
(L)R value	551	93.007	(o)	(o)	
(S)Delay(ave)	8146	8.50	(p)=	(p)≠	0.424
(L)Delay(ave)	551	7.88	(o)	(o)	
(S)Delay(max)	8146	11.56	(p)=	(p)≠	0.887
(L)Delay(max)	551	11.44	(o)	(o)	
(S)Jitter	8146	0.3173	(p)=	(p)≠	0.042
(L)Jitter	551	0.2961	(o)	(o)	
(S)Packet loss	8146	0.00067	(p)=	(p)≠	0.008
(L)Packet loss	551	0.00191	(o)	(o)	
(S)BW	8146	72.65	(p)=	(p)≠	0.730
(L)BW	551	72.86	(o)	(o)	

Table 12 Descriptive statistics of video call traffics.

Statistics	N	Mean	95% mean interval	StDev
Delay(ave)	1122	11.6584	10.888 - 12.429	13.1520
Delay(max)	1122	24.6847	22.995 - 26.375	28.8474
Jitter	1122	3.0698	2.9445 - 3.1952	2.1399
Packet loss	1122	0.00685	0.004586 - 0.009122	0.03872
Bandwidth	1122	334.135	325.489 - 342.781	147.608

4.3 Video Call Traffic Characteristics

(1) General Characteristics of Video Call Traffics

Video calls are relatively very little attempted. 1,122 video calls are attempted during a month in carrier 'A'. The distributions of average delay, maximum delay, jitter, packet loss and required bandwidth of video calls are shown in **Table 12**. The mean value estimation of each metric is derived from one sample T test which has 95% confidence interval. The mean value of average delay is 11.6584 ms and the interval of the metric is from 10.888 to 12.429. Maximum delay is 24.6847 ms and the interval of the metric is from 22.995 to 26.375 ms. Jitter is 3.0698 ms and exists from 2.9445 to 3.1952 ms. Packet loss is 0.00685 and exists from 0.004586 to 0.009122. Required bandwidth is 334.135 kbps and exists from 325.489 to 342.781 kbps. Call success ratio is 100% in all the attempted calls. We limit the meaning of video call analysis to check the tendency rather than the exactness of carrier 'A' because of the number of observations. Average delay and jitter are S+ grades of service performance objectives. Packet loss is 'A' grade. The delay of service performance objectives represents average value. But if we consider the need to control maximum delay, the value is regarded as S grade.

(2) Comparison of Voice Call and Video Call Traffics

It is observed that the traffic characteristics are biased by the required bandwidth. Hence voice call and video call traffics are compared. The required bandwidth of voice call is 72.558 kbps as mean value and 14.489 kbps as standard deviation shown in Table 7. The median value is 76.22 kbps. That of video call is 334.135 kbps as mean value and 147.608 kbps as standard deviation shown in Table 12. The median value is 387.57 kbps. We perform ANOVA which represents average delay, maximum delay, jitter, and packet loss biased by bandwidth. The result is shown in **Fig. 9**. Average delay, maximum delay, jitter, and packet loss have tendencies for apparent differences. As required bandwidth increases, all metrics are increased. This result presents required bandwidth as a fixed factor has a tendency of influence on delay, jitter, and packet loss as response. More bandwidth tends to introduce more delay, jitter and packet loss. In QoS policy of network, the same queue is supported to VoIP and MMoIP. However the user equipments for trial services provide each buffer for processing voice and video respectively. And the RTP packet sizes of voice and video are different. It is presumed that the differences of voice and video call's metrics are derived owing to these reasons.

4.4 Comparison of Two Carriers' NGN Performances

The NGN traffic characteristics of 'A' and 'B' carriers are compared based on voice call. The IP telephony traffic characteristics of 'B' carrier's NGN is estimated with 95% confidence level. The result shows that R-value, average delay, maximum delay, jitter, and packet loss exist from 92.9741 to 93.0663, from 3.22707 to 3.39275 ms, from 9.4537 to 11.2612 ms, and from 0.16805 to 0.21614 ms, and 0.0024 to 0.008827 respectively. We perform a correlation analysis of between R-value and the other metrics. Regression equation is driven according to Eq. (6).

```
One-way ANOVA: AveDelay versus CodeAV

Analysis of Variance for AveDelay
Source     DF        SS        MS       F       P
CodeAV      1     15111     15111   63.89   0.000
Error   14221   3363655       237
Total   14222   3378766
                                   Individual 95% CIs For Mean
                                   Based on Pooled StDev
Level       N      Mean     StDev  ----------+---------+---------+------
A       13101      7.83     15.56  (-*-)
V        1122     11.66     13.15                      (-----*-----)
                                   ----------+---------+---------+------
Pooled StDev =    15.38                    9.0      10.5      12.0

One-way ANOVA: MaxDelay versus CodeAV

Analysis of Variance for MaxDelay
Source     DF        SS        MS       F       P
CodeAV      1    178381    178381  440.86   0.000
Error   14221   5754177       405
Total   14222   5932558
                                   Individual 95% CIs For Mean
                                   Based on Pooled StDev
Level       N      Mean     StDev  --+---------+---------+---------+----
A       13101     11.55     19.18  (*)
V        1122     24.68     28.85                               (--*--)
                                   --+---------+---------+---------+----
Pooled StDev =    20.12           12.0      16.0      20.0      24.0

One-way ANOVA: Jitter versus CodeAV

Analysis of Variance for Jitter
Source     DF        SS        MS       F       P
CodeAV      1   7357.83   7357.83 7284.28   0.000
Error   14221  14364.59      1.01
Total   14222  21722.42
                                   Individual 95% CIs For Mean
                                   Based on Pooled StDev
Level       N      Mean     StDev  ------+---------+---------+---------+
A       13101     0.402     0.839   *
V        1122     3.070     2.140                                     *)
                                   ------+---------+---------+---------+
Pooled StDev =    1.005                0.80      1.60      2.40      3.20

One-way ANOVA: PacketLoss versus CodeAV

Analysis of Variance for PacketLo
Source     DF        SS        MS       F       P
CodeAV      1  0.033261  0.033261  109.06   0.000
Error   14221  4.337027  0.000305
Total   14222  4.370287
                                   Individual 95% CIs For Mean
                                   Based on Pooled StDev
Level       N      Mean     StDev  ------+---------+---------+---------+
A       13101   0.00118   0.01424  (-*)
V        1122   0.00685   0.03872                        (----*----)
                                   ------+---------+---------+---------+
Pooled StDev =  0.01746               0.0020    0.0040    0.0060    0.0080
```

Fig. 9 Comparison of voice and video call traffics.

Table 13 Comparison of two carriers' voice call traffics.

ANOVA	Mean	StDev	H_0	H_1	P-value
(A)R-value	92.988	1.501	(A)=	(A)≠	0.745
(B)R-value	93.020	0.354	(B)	(B)	
(A)Delay(ave)	7.835	15.555	(A)=	(A)≠	0.000
(B)Delay(ave)	3.3099	0.6362	(B)	(B)	
(A)Delay(max)	11.547	19.184	(A)=	(A)≠	0.349
(B)Delay(max)	10.357	6.941	(B)	(B)	
(A)Jitter	0.4017	0.8395	(A)=	(A)≠	0.000
(B)Jitter	0.1921	0.1847	(B)	(B)	
(A)Packet loss	0.00118	0.01424	(A)=	(A)≠	0.045
(B)Packet loss	0.00321	0.04315	(B)	(B)	

$$\text{R-value} = 93.21 - 0.0496 \cdot delay(average) - 8.19 \cdot packetloss \qquad (6)$$

As the adjusted coefficient of determination is 99.9%, the regression model has a much higher explanation. However we limit the meaning of the equation as checking a tendency of relation R-value and other metrics rather than the exactness of carrier 'B' because of the relatively small observations.

Table 13 shows ANOVA results between 'A' and 'B'. However we limit the meaning of this analysis because the analysis does not have exactness owing to the number of observations of carrier 'B'. The notation (A) means carrier 'A' and (B) means carrier 'B'. H_0 as null hypothesis means that (A)'s mean value equals (B)'s mean value in each metric. On the other hand H_1 as alternative hypothesis means that two values are not equal. Call success ratios of both carriers are 100% in the all attempted calls.

5. Conclusions

In this paper, we propose a strategy for NGN construction by introducing a case of nationwide NGN. The strategy consists of definition of service, standardization of network and service performance objectives, design of network, measurement of metrics and evaluation of performance by steps. This paper presents estimations of service performances of a nationwide NGN carrier. The objectives of analysis are voice IP telephony and video IP telephony services. These services correspond to interactive voice and interactive video respectively. We present the traffic characteristic tendencies of IP telephony service over NGN of carrier 'A' from the collected observations. It is estimated that voice and video calls are satisfied with S+ grade of service performance objectives for NGN except packet loss. Packet loss is estimated 'A' grade. It is necessary to consider the QoS policy as well as the processing of user equipment. As shown in Section 4.3, required bandwidths of voice call and video call are different and delay, jitter, and packet loss of video call are higher than those of voice call. The required bandwidth of IPTV is much higher than video IP telephony. It is possible to estimate that larger delay, jitter and packet loss may occur. The processing of traffics for providing end-to-end QoS will be more important in the consideration of real time IPTV multicast services. This paper introduces the different NGN construction approaches of two carriers. Carrier 'A' has constructed nationwide NGN through new NGN core and upgraded subscriber network such as FTTH and Metro Ethernet. Carrier 'B' plans to construct a new transport network only for IPTV multicast and runs parallel to upgrade Internet core and HFC subscriber networks for evolving to NGN. The differences between the approaches are caused by their different network resources and markets.

We need further studies about representative metrics of video IP telephony and IPTV such as R-value of VoIP, a practical approach of adopting resource and admission control scheme for quality of service, and a methodology of verification and improvement of NGN performance like six sigma used in other fields.

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References

1) Carugi, M., Hirschman, B. and Narita, A.: Introduction to the ITU-T NGN focus group release: Target environment, services, and capabilities, *IEEE Communication Magazine*, pp.42–48 (Oct. 2005).
2) Sun, D., Joseph, J.-P., Magee, Jr., F.R., Mukhopadhyay, A. and Tang, B.: A SIP-enabled all-IP architecture for converged next-generation networks, *Bell Labs Technical Journal*, Vol.9, Issue 3, pp.15–37 (2004).
3) Knightson, K., Morita, N. and Towie, T.: NGN architecture: Generic principles, functional architecture, and implementation, *IEEE Communication Magazine*, pp.49–56 (Oct. 2005).
4) Kovacikova, T. and Segec, P.: NGN standard activities in ETSI, *Proc. 6th International Conf. on Networking* (*ICN'07*), p.76 (2007).
5) Anderson, T.W., Busschbach, P., Faynberg, I., Lu, H.-L. and Sun, D.: The emerging resource and admission control function standards and their application to the new triple-play services, *Bell Labs Technical Journal*, Vol.12, Issue 1, pp.5–21 (2007).
6) Song, J., Chang, M.Y. and Lee, S.S.: Overview of ITU-T NGN QoS control, *IEEE Communication Magazine*, pp.116–123 (Sep. 2007).
7) Tang, B., Akyamac, A.A., Hung, C., Chu, K. and Nagarajan, R.: MPLS network requirements and design for carriers: Wireline and wireless case studies, *Telecommunications Network Strategy and Planning Symposium 2006, Networks 2006, 12th International*, pp.1–6 (2006).
8) Le Faucheur, F., Wu, L., Davie, B., Davari, S., Vaananen, P., Krishnan, R., Cheval, P. and Heinanen, J.: Multi-protocol label switching (MPLS) support of differentiated services, RFC 3270 (2002).
9) Floyd, S. and Jacobson, V.: Random early detection gateways for congestion avoidance, *IEEE/ACM Trans. Networking*, Vol.1, Issue 4, pp.397–413 (1993).
10) McClelland, S. and Budwey, J.N.: Japan's 21st century network, *Telecommunications-International Edition*, Vol.40, Issue 6, pp.36–45 (2006).
11) NGN's potential: This is where NGN differs from the Internet, NIKKEI Communications, pp.45–54 (Nov. 2007).
12) Choi, J.Y., Chae, T., Lee, J. and Shim, B.K.: Introduction to KT-BcN: Network design concept and a case study, *Proc. Broadband Convergence Networks 2007* (*BcN'07*), *2nd IFIP/IEEE International Workshop*, pp.1–9 (2007).
13) Guide line for VoIP quality measurement, 3rd edition, Telecom Services Association Japan (June 2006).
14) Kim, C.C., Shin, S.C., Ha, S.Y., Han, S.Y. and Kim, Y.J.: End-to-end qos monitoring tool development and performance analysis for NGN, *APNOMS 2006*, LNCS 4238, pp.332–341 (2006).
15) Almes, G., Kalidindi, S. and Zekauskas, M.: A one-way packet loss metric for IPPM, RFC 2680 (1999).
16) Almes, G., Kalidindi, S. and Zekauskas, M.: A one-way delay metric for IPPM, RFC 2679 (1999).
17) Schulzrinne, H., Casner, S., Frederick, R. and Jacobson, V.: RTP: A transport protocol for real-time applications, RFC 3550 (2003).
18) Friedman, T., Caceres, R. and Clark, A.: RTP control protocol extended reports (RTCP-XR), RFC 3611 (2003).
19) The E-model, a computation model for use in transmission planning, ITU-T G.107

(2000).
20) Network performance objectives for IP-based services, ITU-T Y.1541 (2006).
21) Speech Processing, Transmission & Quality Aspects (STQ); QoS parameter definitions and measurements; Parmeters for voice telephony service required under the ONP Voice Telephony Directive 98/10/EC, ETSI EG 201 769 v1.1.2 (2000-10).

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