Analysis of Connectivity Model and Encoding Standards on IP Interconnection Implementation in Indonesia

(Case Study: Low Data Rate up to 72 Mbps)

Analisis Model Keterhubungan dan Standar Pengkodean pada Implementasi Interkoneksi Berbasis IP di Indonesia

(Studi Kasus: Laju Data Rendah hingga 72 Mbps)

Siska Riantini Arif[1], **Doan Perdana**[2], **Taufik Hasan**[3], **Imam Nashiruddin**[4]
[1,2] *Master of Electrical Engineering, Telkom University*
[1,2] *Jl. Telekomunikasi No. 1, Dayeuh kolot 40257, Bandung*
[3, 4] *Badan Regulasi Telekomunikasi Indonesia*
[3, 4] *Menara Ravindo 11st Floor, Jl. Kebon Sirih Kav 75 Jakarta*
email: [1]siskariantini@telkomuniversity.ac.id , [2]doanperdana@telkomuniversity.ac.id, [3]taufikhasan@brti.or.id, [4]imam@brti.or.id

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A B S T R A K

Saat ini Indonesia dihadapkan pada permasalahan dimana lalu lintas data, termasuk OTT di dalamnya, mendominasi layanan telekomunikasi yang menyebabkan pendapatan interkoneksi semakin menurun. Padahal, biaya pemeliharaan jaringan cenderung naik. Kemunculan teknologi IP dapat memberikan keuntungan, baik terhadap Operator dalam scissor effect *maupun menaikkan tingkat loyalitas pelanggannya. Namun, saat ini regulasi Interkoneksi di Indonesia masih menggunakan Time Division Multiplexing (TDM). Oleh karena itu, diperlukan suatu rekomendasi mengenai standarisasi pengkodean dan model interkoneksi IP. Dalam penelitian ini, aspek teknis dari model interkoneksi IP dianalisis dengan menggunakan perbandingan model, yaitu Peering dan Hubbing dengan metode* no-transcoding *pada 6 jenis codec(G.711a, G.711u, GSM, G.723, G.729, dan G.722) dengan pemberian berbagai beban trafik, (0 Mbps, 15 Mbps, 40 Mbps, dan 72 Mbps). Hasil performansi QoS berupa* delay, Mean Opinion Score, packet loss, *dan* throughput *yang diperoleh dari hasil simulasi masing-masing model dan kombinasi* codec *dianalisis dengan menggunakan server* VOIP Asterisk 11 *dan* Microsip 3.17.3 *untuk SIP phone juga* Wireshark 2.2.4 *dianalisis untuk mengetahui performansinya. Nilai* one way delay QoS *mengacu pada standar nilai pada* ITU-T G.1010. *Dari hasil simulasi diperoleh bahwa secara keseluruhan dengan beban trafik sampai 72 Mbps, model Peering merupakan alternatif model interkoneksi IP yang terbaik. Selain itu, penggunaan* codec G729 *menghasilkan performansi paling baik dengan nilai delay paling minimum dan MOS paling besar, sehingga paling direkomendasikan untuk digunakan dalam implementasi interkoneksi IP.*

Keywords :
IP interconnection
Codec
Peering

A B S T R A C T

Currently, Indonesia is faced with problems where data traffic including OTT dominates the telecommunications services lead to interconnection revenue declining. In the other hand, the cost of network maintenance tend to increase. The emergence of IP technology may provide benefit to the operators in handling the scissor effect and improving the level of customer's loyalty. However, the current interconnection regulations in

[1] Email : siskariantini@telkomuniversity.ac.id

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Indonesia are still using TDM. Therefore, a recommendation on standardization of IP encoding and interconnection model is required. In this research, technical aspect analysis of IP interconnect model is analyzed using comparison model, that is Peering and Hubbing with no-transcoding method on 6 types of codec (G.711a, G.711u, GSM, G.723, G.729, G.722) and loading of various traffic loads (0 Mbps, 15 Mbps, 40 Mbps, 72 Mbps). The results of QoS performance (delay, Mean Opinion Score, packet loss, throughput) obtained from the simulation results of each model and combination of codec are analyzed using VOIP server Asterisk 11 and Microsip 3.17.3 for SIP phone also Wireshark 2.2.4 to assess the performance. One-way delay QoS value refers to the standard in ITU-T G.1010. From the simulation results, it is obtained that for overall traffic load up to 72 Mbps, Peering model is the best alternative IP interconnect model. The usage of G.729 codec was the best performance codec with the minimum delay value and the biggest MOS, thus it was the most recommended for used in the IP interconnection implementation.

1. Introduction

1.1 Background

By entering a new ecosystem, convergence, where the integration of technologies covering hardware/ terminals, software, content, networks and services toward digital Internet Protocol or what we call the Next Generation Network (NGN), will affect the policy industry today, including interconnection. With the introduction of IP technology that provides IP-based communications, the birth of IP technology such as Rich Communications Suite (RCS), Voice/Video over LTE (VoLTE/ViLTE) and Voice over WiFi (VoWiFi), consumers will benefit from call setup faster with higher video sound quality, and operators will benefit from efficiency in operating and maintenance costs resulting in service optimization and network management to produce real-time analysis and diagnostics, making reliable data connectivity (Kim, 2016; Vizzarri, 2014). Therefore, IP technology in interconnection will be able to open new business opportunities for operators to compete with each other and improve customer loyalty due to satisfaction of the quality of the network, as well as with new industry players called Over The Top (OTT) so as to achieve ”any to any connectivity” to reach mobile and fixed customers worldwide. Also, with this technology, interconnection is able to anticipate the evolving data services along with the emergence of IoT services with the enrichment of interconnection services, for example, the IoT system that uses one operator can be accessed by other mobile operators. Indonesia’s telecommunication industry is currently growing, as can be seen in Table 1.

Table 1. The growth of telecommunication industry

Year	2011	2012	2013	2014	2015	2016
Trend increase of mobile user (million)	250	282	313	326	339	389

Source: (Kementerian Komunikasi Dan Informatika Republik Indonesia : Direktorat Jenderal Penyelenggaraan Pos Dan Informatika, 2017)

Based on Table 1, although the number of subscribers is increasing, the income of Time Division Multiplexing (TDM) based voice interconnection operators is declining as data traffic dominates services.

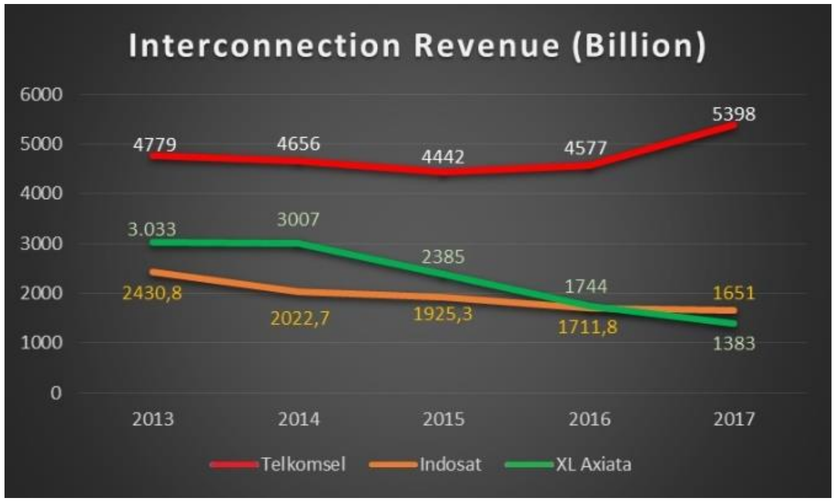

Figure 1. Interconnection revenue of operators (PT Indosat, 2013, 2014, 2016, 2017), (PT Telekomunikasi Seluler, 2013, 2014, 2016, 2017), (PT XL Axiata Tbk, 2013, 2014, 2016, 2017)

Figure 1 shows revenue from interconnection services is decreasing (although contribution to total operating income is still considerably high), while revenue from data services is increasing from year to year. However, even with data traffic growing exponentially as new networks continue to build, they do not provide significant linear revenue, known as the-so-called the scissor effect. This is due to the use of voice is decreasing but the maintenance cost of the traffic quality, which still uses old technology (PDH, TDM based), are still relatively fixed lead to inefficient expenses incurred compared to income. This infrastructure plays a vitalrole in the deployment of service coverage, effectiveness in the use of frequency spectrum, and improvement of service quality. Therefore it is time for renewal of TDM-based voice interconnection technology to switch to a more efficient, IP-based interconnection and based on a clear regulatory analysis.

Initial regulations on interconnection in Indonesia are contained in the Law of the Republic of Indonesia no. 36 of 1999. The birth of Law no. 36 in 1999 on telecommunication followed by the birth of circuit switch ecosystem where its basic technology uses TDM and its service mainly covers voice interconnection and short message services with termination, origination and transit. TDM technology is reinforced in the explanation contained in Government Regulation no. 52 of 2000 article 10 paragraph (1) on the telecommunication operation that the basic telephony service is a telephone service using circuit switch technology, and also in Fundamental Technical Plan 2004. However, regulation above does not include about IP interconnection in Indonesia, there are some things to consider in the implementation of IP interconnection, including :

a) Technical aspects review signaling, interconnection model, coding, Quality of service (QoS), numbering and addressing standards.
b) Business Aspects review the types of interconnection services, tariff, charging and billing models.
c) Economic and social aspects address the compatibility of services and industry and community readiness in the application of IP interconnection.
d) The regulatory element reviews implementation policies both at transition time and implementation time of all IP.

In this study is devoted to performing analysis of technically suitable IP interconnection model implemented that is Peering or Hubbing, as well as its coding standard by considering the bandwidth management that can affect the Opex, through its QoS analysis from the simulation results.

Previous research conducted simulation and codec analysis using VoLTE network with Peering model end-to-end shows that G.711 and GSM EFR codecs can guarantee better performance concerning MOS and G.729A. Moreover, GSM G711 provide good performance regarding sent/received voice traffic (Vizzarri, 2014).

Meanwhile, other paper mentioned that default codec G.711 still gave the best performance in IP transmission, either on wireline or mobile switch, specifically for operators that do not care about transmission efficiency issues. Besides that, codec G.729 has better bandwidth characteristics that can be introduced on the interconnection to reduce operational expenditure (opex) although transcoding in a mobile switch is required when entering MSS side (Baldwin, Ewer, & Yamen, 2010). Meanwhile, there has not been found specific research using interconnect IP Hubbing model.

In Europe, 13 out of 32 countries (41%) are already planning to migrate from TDM to IP-based. Audio codecs used are G.711, G.729, G.729, G.72242, DTMF, RFC 2833, RFC 47334, EFR, AMR-NB, and AMR-WB codecs. The QoS results are the delay of < 150 ms and jitter 0, 99.5% (BEREC, 2015). In India, an evaluation is required in telecommunication industry to regulate the switch of conventional circuits (TDM) to Internet Protocol (IP), given the use of voice codecs is G.711, G.729, and AMR (Meena, R. Saji Kumar, n.d.).

Therefore, this study recommends an interconnection model through comparative evaluation and technical analysis between Peering and Hubbing system model with the test method of no-transcoding (using the same codec in both users), dynamic traffic routing traffic, and using traffic loads (0 Mbps, 15, 40, 75 Mbps). Each with QoS benchmarks based on ITU-T standard, i.e delay < 150ms, packet loss < 3%, MOS > 4, and throughput > 75%.

1.2 Research Objective

Current operators interconnect TDM by using Peering and transit model where only one operator can just do interconnection Therefore, this research will be the breakthrough interconnection regulation which is studied academically, which become the proper necessary recommendation of model IP interconnection and coding standard. The research includes testing and IP to IP link analysis by a new managed Hubbing model using six codecs and compare with Peering in four different traffic condition, that is providing backgroud traffic to approach the actual situation (0, 15, 40, 72 Mbps). QoS performances benchmarks with QoS (delay, MOS, packet loss, and throughput) standard based on ITU-T G.1010 that have good value and bandwith efficiency.

With this system, the operator is expected to focus on improving the quality of customer voice service and able to compete and become better than OTT service, so that customers will use this interconnection IP as a means of interconnection, so can increase customer loyalty due to the satisfaction of the quality of the network.

1.3 Assumption and Problem Limitation

The assumptions and problem limitations to support business model research on the IP-based interconnection are as follows:

a) The discussion is limited to infrastructure with IP-based technology.
b) Literature and benchmarks are drawn from countries that already have IP interconnection in the world, i.e. EU and India countries.
c) Aspects of interconnection research only on technical areas that become legal recommendations
d) In this simulation, research using the simulation of IP to IP link using IP Phone.
e) The QoS parameters tested were the delay, packet loss, MOS, throughput.
f) The research parameters use a combination of six codecs both with transcoding and no-transcoding methods, i.e. G.711a, G711u, G.722, G.729, G.723, GSM.
g) The research use traffic engineering arrangement with dynamic routing method on both links (primary and standby).
h) The research use four traffic loads, i.e. 0 Mbps, 15 Mbps, 40 Mbps and 72 Mbps.
i) The research use Mikrotik channeling hub up to 43 Mbps and SIP from Asterisk as Voip server.

2. Literature Review

2.1. Interconnection between an IP-based Managed Network

There are three types of interconnection networks in the IP managed network, which are Hubbing, Peering and Transit model.

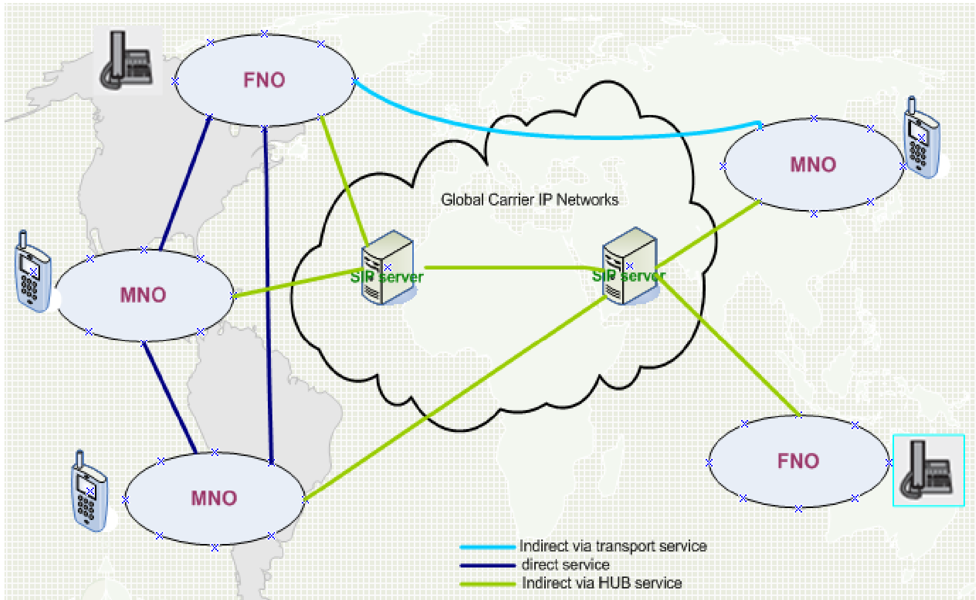

Figure 2. Interconnection between an IP-based Managed Network (Baldwin et al., 2010)

2.1.1. IP eXchange (IPX) or Hubbing Model

The interconnection hub is a telecommunications interconnection model for exchange IP-based traffic between customers from separate mobile and fixed operators and other types of service providers through an IP-to-Network network interface. The Interconnection Hub was developed by GSMA Association, called IP eXchange (IPX). According to GSMA, IP eXchange (IPX) is an interconnection service with mutually agreed technical specifications. IPX can offer the flexibility to apply the appropriate level of quality as required by each different service class (GSMA, n.d., 2016)

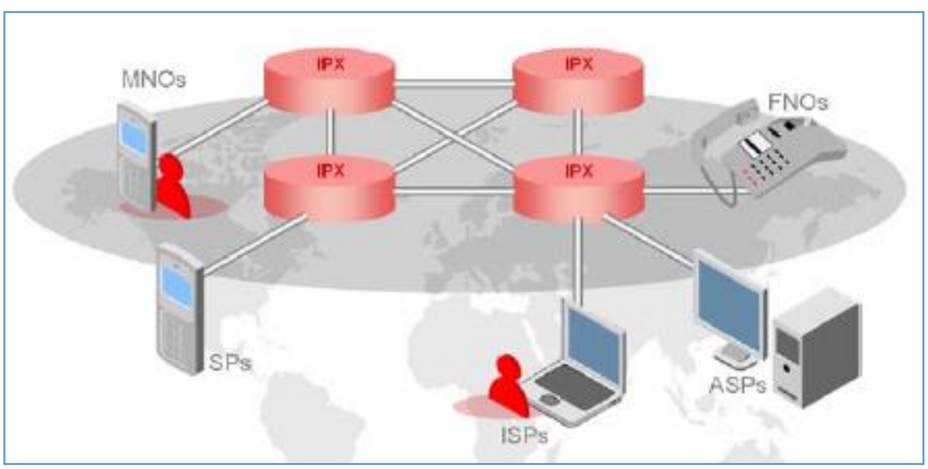

Figure 3. HUB Interconnection according to GSMA (GSMA, 2016)

In addition, with the construction of an IPX network, it will bring good benefits to IPX providers themselves and to IPX (Operator) users. As in the following explanation (GSMA, 2016) :

a) IPX benefits for IPX provider are below :
   1) Provide additional services, such as interworking between fixed cellular providers, by providing interworking signaling and transcoding media;
   2) Provide various services and multilateral interconnection through interconnection scheme with SLA and quality assurance;
   3) Allows secure and seamless interconnection between network providers and services/content;
   4) Differentiate new services such as VoLTE that is in between IMS networks;
   5) Provide interconnection services based on non-minute minutes, by providing interconnects for IP-based multimedia such as VoIP, video telephony, instant messaging, presence, image, and video transfer;
b) IPX benefits for Operator are below :
   1) Allows operators to compete more effectively with Over the Top by providing quality assurance services;
   2) Provide an inter-operability guarantee both technically and commercially;
   3) Facilitate technical, commercial, and administrative issues for bilateral and multilateral cooperation using a single contract with IPX providers;
   4) Reduce OPEX to connect with other players;
   5) Expanding interconnection services with several telecommunication providers communities overseas;
   6) Accelerate time-to-market to provide new services with QoS guarantees and service level agreement.

2.1.2. Peering (Direct Interconnection)

Figure 3 shows network 1 connected with network 2 via a dedicated link using the gateway router. This model is usually used for national or domestic traffic between two operators (Baldwin et al., 2010).

2.1.3. Transit (Indirect interconnection)

A Carrier network (third party) provides a pipe between Operators (bilateral) (Baldwin et al., 2010)

2.2 Coding-Decoding (CODEC)

Coder/decoder (codec) function is to convert the digital data format from the original data into data of smaller size without significantly reducing the sound quality. Each operator, if it has more than one usable codec, can choose a higher priority codec to choose from to provide the best possible sound transmission. Codecs should be chosen, taking into account various parameters such as bandwidth, sample intervals, bit rates, Mean Opinion Scores (MOS), and so on ((ECC), 2017; Fakrudeen, M., Yousef, S., Tapaswi, S., Patnaik, K. K., and Cole, n.d.; Telecommunication Standardization Sector of ITU, 2003). A summary of CODEC is given in Table 2.

Table 2. Table of Codec Standard

CODEC	EXPLANATION	PACKETIZATION TIME / DEFAULT VALUES IN MS	EFFECTIVE DATA RATE (KBPS)
G.711	G.711 is the dafault codec for fixed and mobile applications and uses pulsa-code modulation (PCM). It provides the best speech quality for narrowband codecs with user data rate of 64K.	0.125	64 (over TDM)
G.729	G.729 is one of the most popular narrowband codecs for IP-based fixed networks based on code-excited linear prediction	20	23 (over IP)

CODEC	EXPLANATION	PACKETIZATION TIME / DEFAULT VALUES IN MS	EFFECTIVE DATA RATE (KBPS)
	(CELP). It provides similar speech quality characteristics to AMR 7.4 with a user data rate 8K		
G.722.O	G.722.O is the fixed rate 7KHz wideband audio codec with a user data rate of 64K. Due to extended audio spectrum. It provides excellent audio quality with volume and clarity.	20	54 (over IP)
AMR-NB	Adaptive multi-rate (AMR) is the most popular narrowband codec for wireless networks. It adopts to radio condition and capacity requirements with user data rates between 12.2K down to 4.75K.	20	23 (over IP)
AMR-WB	Adaptive multi-rate wideband (AMR-WB) is the 7KHz wideband audio codec with similar speech quality to G.722.O. It adopts to radio condition and capacity requirements with user data rates between 12.65K down to 6.60K.	20	23 (over IP)

Source:((ECC), 2017)

2.3 Session Initiation Protocol (SIP)

SIP is a protocol used to build communication between networks. SIP is a protocol that works in signaling and control on the application layer to build, modify, and terminate a session for various types of multimedia services with two or more participants. there are three kinds of SIP built in the network, the IETF SIP, SIP IMS, SIP-I (ITU-T) (BEREC, 2015) :

a) SIP IETF (RFC 3261)

The Session Initiation Protocol (SIP) defined by the IETF standard (called RFC) provides space for network operators about how to use SIP, i.e., on the one hand, flexibility for network operators but, on the other hand, further specifications may be required to ensure operation at between different networks (Stalling, 2003).

b) SIP (IETF+3GPP)

SIP (SIP (IETF + 3GPP) is also called SIP IMS, ie SIP that meets 3GPP specification is based on the use of so-called IP Multimedia Subsystem (IMS) defined by 3GPP. SIP signaling is the primary method used for registration sessions and user controls in the IMS architecture.

c) SIP-I (ITU-T)

SIP-I SIP-I is a hybrid signaling protocol defined by the IETF but is used in a slightly specific way by ITU-T where traditional signal protocols (TDM-based) are fed into the ”new” signal SIP protocol. Therefore, this can also be seen as an intermediary step between traditional signaling protocols (ISUPs) and the new ”IP” IP-based SIP protocols. Mobile network standards have separated call controls from transport-based packages over the years and suggest using SIP-I (or different signaling protocols) within mobile networks (BEREC, 2015).

2.4 Quality of Service

With the implementation of VoIP, customers are expected to continue receiving the same voice transmission quality as a traditional phone service. This implies that the resulting sound transmission must remain consistently of high quality. Like other real-time applications, VoIP is very sensitive to ”bandwidth and delay”. In order for VoIP transmissions to be understood by the receiver, voice packets should not be dropped, pending, or jitter. VoIP can guarantee high quality voice transmission only if voice packets, both for signal and audio channels, are given priority over other types of network traffic. In order for VoIP to be used so that customers can receive acceptable levels of voice quality, VoIP traffic must be guaranteed with certain compensation, latency and jitter requirements.

Transmission in the IP network, the relevant QoS parameters as an IP network performance testers are ((ECC), 2017):

a) Delay

Delay manifests itself in a number of ways, including the time taken to establish a particular service from the initial user request and the time to receive specific information once the service is established. Standard one-way delay value which was confirmed by ITU-T G.1010 is < 150 ms and limit < 400 ms (Telecommunication Standardization Sector of ITU, 2001).

$$Delay = \frac{Packet\ Length\ (bit)}{Link\ Bandwith\ (\frac{bit}{s})} \text{...} (1)$$

b) Packet loss

Packet Loss is the number of packets that fail to reach the destination when packet delivery. Standard information loss value which was confirmed by ITU-T G.1010 is < 3% called packet loss ratio (Telecommunication Standardization Sector of ITU, 2001).

$$Packet\ Loss = \frac{Packet\ Sent - Packet\ Received}{Packet\ Sent}\ x\ 100\% \text{ ....} (2)$$

c) Throughput

Throughput is the actual bandwidth were measured in a particular time and in a certain network conditions that are used to transfer files of certain size.

$$Throughput = \frac{\sum Sent\ Data\ (bit)}{Time\ Data\ Delivery\ \ (s)} \text{ ....} (3)$$

d) MOS (R-Factor / E-Model)

The MOS parameter is used in QoS parameters. MOS (Mean Opinion Score) is a method used to measure voice quality on IP networks, another method is the E-Model method based on ITU-T G.107, the final value of E-Model estimation is called R factor (R). R factor is defined as the transmission quality factor that is influenced by several parameters such as signal to noise ratio, device echo, compression codec, packet loss and delay (Telecommunication Standardization Sector of ITU, 2011). Thus, the final value of E-Model estimation is called R Factor. R factor is defined by the equation :

$$R = \text{R} = \text{R0} - \text{Id} - \text{Ief}$$

$$R = 94.2 - [0.024 + 0.11\ (d - 1773)\ H\ (d - 177.3)] - [7 + 30\ ln\ (1 + 15\ e)] \text{ ..} (5)$$

where :

Id = The factor of quality degradation caused by the influence of one-way delay :

$$Id = (0.024) + 0.11\ (d - 1773) H\ (d - 177.3) \text{...} (6)$$

Ief = The quality degradation factor caused by the compression technique and packet loss occurs

$$Ief = (7 + 30 \ln (1+15 e)) \text{...} (7)$$

d = delay values (ms)

H = Ladder function : H(X) = 0 if x < 0, others H(x) = 1 if x > 0 e = Packet loss

2.5 Benchmark

Benchmarks taken in this study came from European Union countries and India. The following are technical characteristics for the application of IP Interconnection in each country in European (BEREC, 2015).

2.5.1 European Union:

QoS parameters in European Union are based on ITU standards, as given in Table 3.

Table 3. QoS Parameters in Europe

		Italia (IT)	Jerman (DE)	Bulgaria (BG)	Perancis (FR)
Audio codec supported by IPvIC		G.711 A-law, G.729, DTMF	G.711 A-law	G.711 A-law	G.711 A-law
Signalling Protocol type (SIP) at POI IPvIC		SIP (IETF), SIP-I (ITU-T)	SIP (IETF+3GPP)	SIP (IETF)	SIP-I (ITU-T)
QoS measurenment	characteristic QoS	yes	yes	yes	yes (but to QoS subjective)
	Delay one-way delay	N/A	< 150 ms	< 150 ms	N/A
	Jitter	N/A	N/A	yes	N/A
	Packe loss ratio	N/A	N/A	yes	N/A
	Network Effectiveness Ratio		99.5%	95%	99.3% (bouyeges telecom)
	Answer Seizure Ratio			50%	65% (bouyeges telecom)
	Availability of interconnection		99.5%		
Traffic types in the network	Termination	yes	yes	no obligation offer RIO	yes
	Origination	yes	yes		
	Transit	yes	yes		
	Access to Services	yes	yes		

Source: (BEREC, 2015)

2.5.2 India

Minimum Requirements to be met at the IP interconnect (Meena, R. Saji Kumar, n.d.) in India are as follows:

a) The device can support SIP, SIP-I and SIGTRAN protocols for signaling and data transfer

b) Transcoding :

1) It is recommended not to transcode due to impact on quality and delay,
2) Perform transcoding by a service provider when there is IP to TDM conversion,
3) Make transcoding only once during conversion,
4) Satellite links are serving mobile. Then, we recommend using mobile codecs.

2.5.3 Regulation Mindmap for interconnection

Regulation mindmap for interconnection illustration is given in Figure 4.

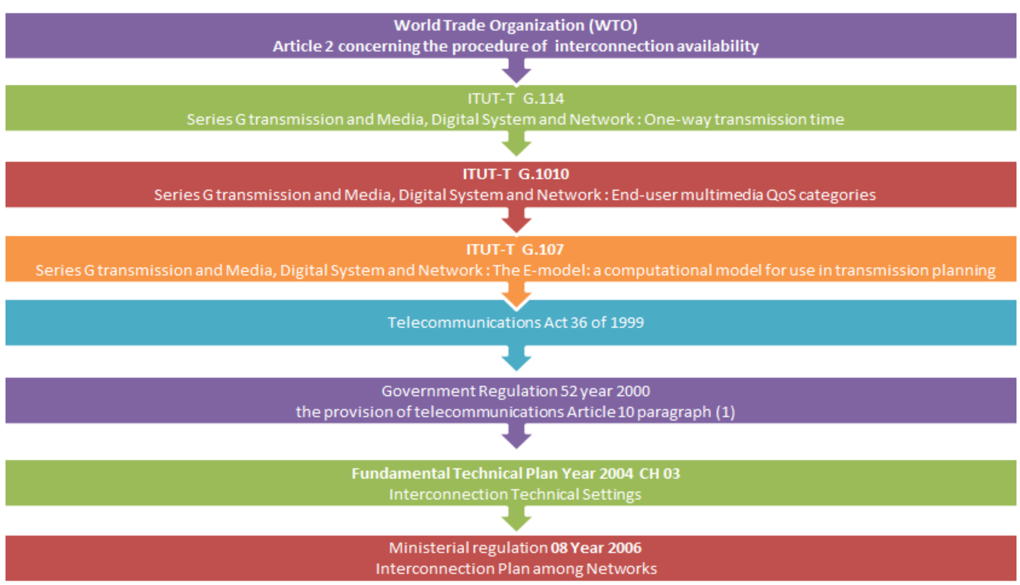

Figure 4. Regulation mindmap for IP Interconnection research

3. Method

In this study, research method based on simulation using software and hardware to get the level of QoS in each type of codec. The research output is to obtain compatible codec for IP interconnection in Indonesia.

3.1 Research supportive facilities

This study used a variety of supporting facilities that are broadly divided into two, namely hardware and software.

3.1.1 Hardware

The hardware in question is a physical supporting device. The hardware used in this study includes one server computer and three clients (two computers and one laptop). The specification is given in Table 4.

Table 4. Hardware for supporting research

Device	Type
Processor	Intel® Core ™ I5-3470 CPU @ 3.20 GHz
RAM	12 GB DDR3 12800
NIC 1 (Management)	D-Link DGE-528T
NIC 2 & 3	Atheros GE AR8161
Storage	1 TB
Connector	UTP cable and RJ45 connector
Client PC-1	Switch Gbps D-link
	Asus All-in-One
Client PC-2	Dell Inspiron All-in-One
Client PC-3	Dell Inspiron All-in-One
Router	Router Mikrotik CHR (4 pcs) V. 6.40.4 1 Gbps

3.1.2 Software

The software in question is a supporting device in the form of software/program/application. The software used in this study is given in Table 5.

Table 5. Software for supporting research

Device	Type
Operation system	VMKernel Release Build 3620759
	Windows 10 Enterprise
	Windows 7 Ultimate
	Ubuntu 14.04
Virtual Router	Cloud Hosted Router (CHR) ver 6.40.4
Machine Virtualization	VMware ESXi 6.0.0
	VirtualBox
Voice Server	Asterisk 11
Softphone	Microsip 3.17.3
Network analyzer	Wireshark 2.2.4
Traffic Generator	Iperf 2.0.5 win 32
NTP Server	Ntpdate

3.2 System Topology

The study focused on simulations on the core, where there are 3 FNO IP clients (client A numbered 1001, Client B numbered 2001, and client C numbered 1002 are interconnected to interconnect). Each has a different network of each other. In this research, Hubbing interconnection model is an interconnection model between two operators via third-party cloud that has Peering connection among routers connection (R-1 to R2 as a main route for user A and C ahead to user B, and R3 to R4 as an alternative route for user B forward to user A and C) and same server at IN also OUT which support no-transcoding and transcoding codec, without user A or B or C must have and manage their POI respectively. Then, the sample in this scenario is described as Hubbing interconnection model between user A to user B which use no-transocoding method. Meanwhile, Peering interconnection conducted between user A with C which only supports to no-transcoding codec because it has only one OPI for each connection which must be managed by each user. This study has conducted two types of simulation, which are a no-transcoding method using Peering and Hubbing model. The research topology can be seen in Figure 5.

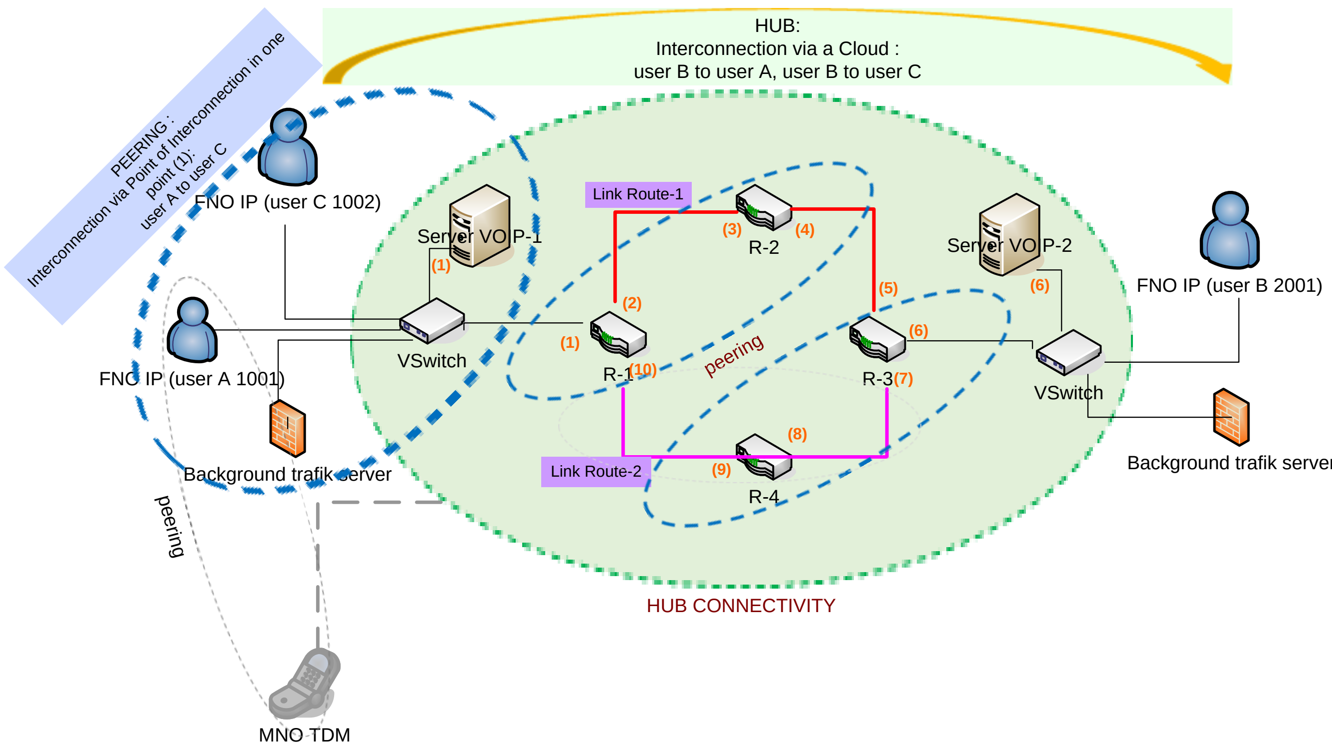

Figure 5. Managed Hubbing dan Peering Topology

3.3 Peering and Hubbing for non transcoding method parameters

The simulation using parameters as given in Table 6.

Table 6. Table Peering and Hubbing parameters

Parameters	Description
Signalling	SIP IETF from Asterisk
Codec	G.11a, G.711u, G.722 16 Khz, GSM 8 Khz, G.723, G.729
Traffic (Mbps)	0, 15, 40, 72
QoS	Delay. MOS, Packet loss. throughput
Traffic	0 Mbps, 15 Mbps, 40 Mbps, 72 Mbps
Literatio numbers per codec per each of traffic load	10 literation
Duration per literation	60 seconds

3.4 Simulated Scenario and Flowchart for Peering and Hubbing for non transcoding method

The model design of the study merge two models of transit and Peering into a new model recommendation, namely Hubbing. This configuration is chosen with the consideration of addressing the deficiencies in Peering and transit models, where Peering and transit when interconnecting to other operators must rebuild new and relatively inefficient, complex network and device configurations in cost. Thus, this Hubbing model can communicate with various organizers by using the same system architecture with high-quality assurance and higher efficiency. However, there is also a simulation for Peering where models are currently used in legacy networks. Therefore, this simulation will be compared with Peering model. One codec simulates four traffic loads, each using hubbing and peering models. The total of one codec simulates 8 times, with each simulation done in 10 times literation. Meanwhile, from six codec, there are a total of 48 simulations performed on each model Peering and Hubbing as given in Table 7.

Table 7. Table Peering and Hubbing scenario

No.	Codec	Traffic Loads	Interconnection Model	No	Codec	Traffic Loads	Interconnection Model
1	G.722 to G.722	0 Mbps	Hubbing	25	GSM to GSM	0 Mbps	Hubbing
2		0 Mbps	Peering	26		0 Mbps	Peering
3		15 Mbps	Hubbing	27		15 Mbps	Hubbing
4		15 Mbps	Peering	28		15 Mbps	Peering
5		40 Mbps	Hubbing	29		40 Mbps	Peering
6		40 Mbps	Peering	30		40 Mbps	Hubbing
7		72 Mbps	Hubbing	31		72 Mbps	Hubbing
8		72 Mbps	Peering	32		72 Mbps	Peering
9	G.711a to G.711a	0 Mbps	Hubbing	33	G.723 to G.723	0 Mbps	Hubbing
10		0 Mbps	Peering	34		0 Mbps	Peering
11		15 Mbps	Hubbing	35		15 Mbps	Hubbing
12		15 Mbps	Peering	36		15 Mbps	Peering
13		40 Mbps	Hubbing	37		40 Mbps	Hubbing
14		40 Mbps	Peering	38		40 Mbps	Peering
15		72 Mbps	Hubbing	39		72 Mbps	Hubbing
16		72 Mbps	Peering	40		72 Mbps	Peering
17	G.711u to G.711u	0 Mbps	Hubbing	41	G.729 to G.729	0 Mbps	Hubbing
18		0 Mbps	Peering	42		0 Mbps	Peering
19		15 Mbps	Hubbing	43		15 Mbps	Hubbing
20		15 Mbps	Peering	44		15 Mbps	Peering
21		40 Mbps	Hubbing	45		40 Mbps	Hubbing
22		40 Mbps	Peering	46		40 Mbps	Peering
23		72 Mbps	Hubbing	47		72 Mbps	Hubbing
24		72 Mbps	Peering	48		72 Mbps	Peering

The work flow diagram and the simulation process are given in Figure 6.

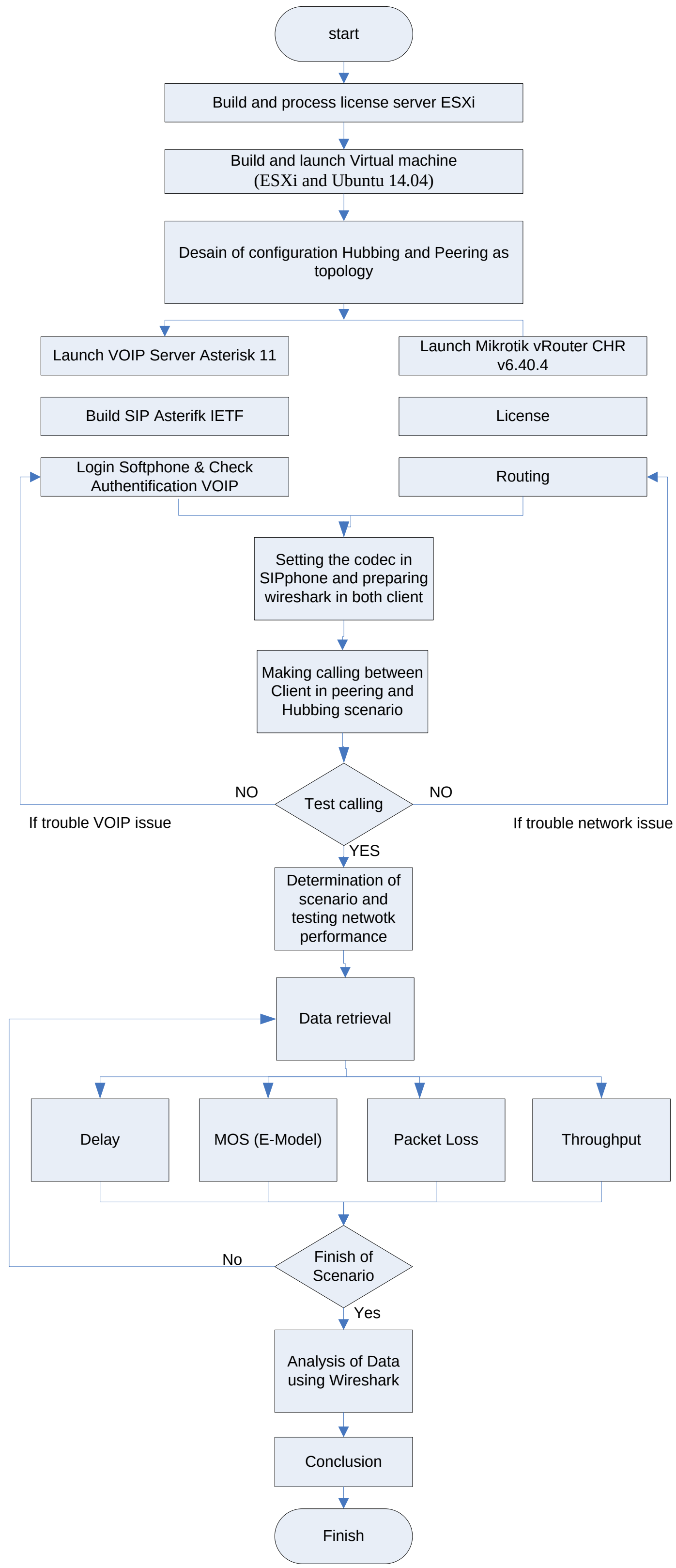

Figure 6. Simulation Flow Chart

4. Result and Discussion

Network performance is evaluated by one-way delay value, throughput, packet loss, and Mean Opinion Score (MOS). Network protocol analyzer (Wireshark 2.2.4) is used to obtain data that capture every passing data packet on the network with time unit. Standardization used as the references in QoS test is given in Table 8.

Table 8. QoS standardization based on ITU-T

	Performance Parameters			
	One Way Delay	**Packet loss**	**MOS**	**Throughput**
ITUT-T Standard	Preferred <150 ms; Acceptable <400 ms	< 3%	> 4	>75 %

Source: (Telecommunication Standardization Sector of ITU, 2001)

Bandwidth size of each codec used is determined by calculating the addition of header for layer 2 (Ethernet) of 14 bytes, thus the total header: Layer 2 (14 bytes) + (IP (20 bytes) + UDP (8 bytes) + RTP (12 bytes)) = 54 bytes. Table 9 shows bit rate types for each codec.

Table 9. Standardized codec based on ITU-T

Codec	Bitrate (Kbps)
G.722	64
GSM	13.2
G.711a	64
G.711u	64
G.729	8
G.723	6.3

Source: (Telecommunication Standardization Sector of ITU, 2003)

Generally, one-way total delay in IP communication is influenced by several factors, i.e. delay process which consists of delay coder, delay of packetization, and delay queue. It is also affected by delay switching, transmission delay, and propagation delay. Meanwhile, the throughput is influenced by the amount of payload and bitrate of speech codec used. The higher the bitrate used the higher the throughput generated. The analysis obtained for QoS performance for each codec and traffic are given in the below section :

4.1 Simulation result of Peering and Hubbing for One Way Delay

From the measurements made on the interconnection service using Peering and Hubbing, it can be seen that there is an effect of traffic and interconnection model changes, on G.722, G.711a, G.711u, GSM, G.723 and G.729 codec. Moreover, from the measurement results, each codec has a feature, the algorithm itself to generate a package, and have different methods with each other. Thus, the codec affects in terms of the system delay. From Table 10, it can be observed that the greater the traffic load given, the greater the value of the delay. This means, ideally, the delay value is directly proportional to the addition of traffic. While, the different results are given for G.722, G.711u and GSM codecs, where the traffic load at 15 Mbps is greater than at 40 Mbps due to different situations and conditions at the time the simulation taken.

In Peering and Hubbing scenarios, ideally, the delay value increases when the total traffic load increases. For G.722 codec seen at 0 Mbps traffic, for Hubbing model, the delay obtained is 39.985 ms, while in Peering model, the delay value obtained equal to 39.986 ms. Whereas at 72 Mbps traffic, the delay increase, where Hubbing model obtained the delay of 40.107 ms and the Peering model obtained the delay value of 40.037 ms.

Th results are similar to other codecs in both peering and hubbing models. However, the overall delay in the codec G.722, G.711a, G.711u, GSM, G.723 and G.729 shows good results with standard limits of not more than 150 ms, according to ITU-T standard. The measurement results in detail can be seen in the Table 10.

From Table 10, it can be seen that G.711u, G711a, G.722 codecs have more delay than G.729 codec. One of the main reasons is due to the sample size of this voice codec which equals to 80 bytes, excluding the 40 bytes of the IP/UDP/RTP headers. This codec sample size is around 8 times the size of other codecs. Moreover, the G.711 codec utilizes the A-law/μ-law algorithm which has low compression rate and accommodates larger link bandwidth fraction compared to the other codecs. By forming several small packet sizes with no compression technique, which means there is no delay for compression and decompression time but packets sent are still large and intact. In contrast, with the same packet size, the G.729 codec with 8 Kbps bit rate (bit rate low) has the ability to compress before compression encoding and decompression after excellent decoding compression, which in turn watu delay to the user tends to be less. Therefore, codecs with compression techniques have a higher delay value than codecs without compression technique.

From Table 10, it can be concluded that by using Peering and Hubbing interconnection models with G.729 codec on traffic load of 0 Mbps, 15 Mbps, 40 Mbps, and 72 Mbps resulted the least delay value compared to other codecs. Thus, from the measurement result on G.729 codec, it is found that for up to 72 Mbps traffic load, the Peering model has less delay compared to Hubbing model.

Table 10. Simulation Result of Peering and Hubbing for One Way Delay

No.	Codec	Traffic Loads	Interconnection Model	ITU-T std < 150 ms		No.	Codec	Traffic Loads	Interconnection Model	ITU-T std < 150 ms	
				Delay (ms) total	**Delay (ms) total**					**Delay (ms) total**	**rank of best value**
1	G.722 to G.722	0 Mbps	Hubbing	39,98458211	Hubbing	25	GSM to GSM	0 Mbps	Hubbing	39,98447475	Hubbing
2		0 Mbps	Peering	39,98570868		26		0 Mbps	Peering	39,98651658	
3		15 Mbps	Hubbing	40,08407	Peering	27		15 Mbps	Hubbing	40,01188093	Hubbing
4		15 Mbps	Peering	40,0356073		28		15 Mbps	Peering	40,04827395	
5		40 Mbps	Hubbing	39,98491484	Hubbing	29		40 Mbps	Peering	39,99532871	Hubbing
6		40 Mbps	Peering	40,37246625		30		40 Mbps	Hubbing	39,98452339	
7		72 Mbps	Hubbing	40,1067925	Peering	31		72 Mbps	Hubbing	40,09161304	Peering
8		72 Mbps	Peering	40,03729667		32		72 Mbps	Peering	40,04882188	
9	G.711a to G.711a	0 Mbps	Hubbing	39,98428286	Hubbing	33	G.723 to G.723	0 Mbps	Hubbing	59,89611415	Peering
10		0 Mbps	Peering	39,98606649		34		0 Mbps	Peering	59,89369778	
11		15 Mbps	Hubbing	39,9936963	Hubbing	35		15 Mbps	Hubbing	59,89666074	Peering
12		15 Mbps	Peering	40,10401785		36		15 Mbps	Peering	59,89603231	
13		40 Mbps	Hubbing	39,9845839	Hubbing	37		40 Mbps	Hubbing	59,91099312	Peering
14		40 Mbps	Peering	39,98687648		38		40 Mbps	Peering	59,89515223	
15		72 Mbps	Hubbing	40,04691602	Hubbing	39		72 Mbps	Hubbing	59,95579285	Peering
16		72 Mbps	Peering	40,10494668		40		72 Mbps	Peering	59,89810037	
17	G.711u to G.711u	0 Mbps	Hubbing	39,98450329	Hubbing	41	G.729 to G.729	0 Mbps	Hubbing	39,9596326	Peering
18		0 Mbps	Peering	39,98634811		42		0 Mbps	**Peering**	**39,95457171**	
19		15 Mbps	Hubbing	40,05538291	Hubbing	43		15 Mbps	Hubbing	40,04827395	Peering
20		15 Mbps	Peering	40,16572815		44		15 Mbps	**Peering**	**39,96016434**	
21		40 Mbps	Hubbing	39,98458483	Hubbing	45		40 Mbps	Hubbing	39,95999545	Peering
22		40 Mbps	Peering	39,98643148		46		40 Mbps	**Peering**	**39,95559987**	
23		72 Mbps	Hubbing	40,06947118	Hubbing	47		72 Mbps	Hubbing	40,01508887	Peering
24		72 Mbps	Peering	40,16670392		48		72 Mbps	**Peering**	**39,95838793**	

4.2 Simulation result of Peering and Hubbing for Mean Opinion Score (MOS by R-Factor / E - Model)

The objective approach is used to determine the quality of service based on the cause of the declining quality of service in the network modeled by the ITU-T G.107 E-Model. Based on Table 11, MOS calculation using E-Model on Peering and Hubbing models of all traffic load conditions shows "Good" VoIP quality, except GSM codec and G.723 using Hubbing model on 72 Mbps traffic load, has less value than "4". The result from Quality of Service to MOS calculation parameter is delay and packet loss, it means that the MOS calculation will have similarity of analysis with delay and packet loss.If the delay or packet loss is small then the MOS in the condition is good and if delay or packet loss is large then the MOS on the condition is not good.

Fom the test results and the R-factor calculation, on the table 11 it can be concluded that Peering model has QoS tend to be slightly better than Hubbing model. For the highest MOS values there is a G.729 codec that uses Peering model in all load traffic (0 Mbps, 15 Mbps, 40 Mbps, 72 Mbps) is about 4.04.

Table 11. Simulation result of Peering and Hubbing for MOS (E-Model)

No.	Codec	Traffic Loads	Interconnection Model	ITU-T std > 4 MOS (E-Model)	ITU-T std > 4 rank of best value
1	G.722 to G.722	0 Mbps	Hubbing	4,043686996	Hubbing
2		0 Mbps	Peering	4,043686073	
3		15 Mbps	Hubbing	4,043605512	Peering
4		15 Mbps	Peering	4,043645204	
5		40 Mbps	Hubbing	4,043686723	Hubbing
6		40 Mbps	Peering	4,0433693	
7		72 Mbps	Hubbing	4,025638311	Peering
8		72 Mbps	Peering	4,042450003	
9	G.711a to G.711a	0 Mbps	Hubbing	4,043687241	Hubbing
10		0 Mbps	Peering	4,04368578	
11		15 Mbps	Hubbing	4,043679531	Hubbing
12		15 Mbps	Peering	4,043589173	
13		40 Mbps	Hubbing	4,043686994	Hubbing
14		40 Mbps	Peering	4,043685117	
15		72 Mbps	Hubbing	4,000585685	Peering
16		72 Mbps	Peering	4,043076624	
17	G.711u to G.711u	0 Mbps	Hubbing	4,04368706	Hubbing
18		0 Mbps	Peering	4,043685549	
19		15 Mbps	Hubbing	4,043629007	Hubbing
20		15 Mbps	Peering	4,04353863	
21		40 Mbps	Hubbing	4,043686994	Hubbing
22		40 Mbps	Peering	4,043685481	
23		72 Mbps	Hubbing	4,032409629	Peering
24		72 Mbps	Peering	4,043537831	
25	GSM to GSM	0 Mbps	Hubbing	4,043687084	Hubbing
26		0 Mbps	Peering	4,043685411	
27		15 Mbps	Hubbing	4,043664637	Hubbing
28		15 Mbps	Peering	4,04363483	
29		40 Mbps	Peering	4,043678194	Hubbing
30		40 Mbps	Hubbing	4,043687044	
31		72 Mbps	Hubbing	3,978875809	Peering
32		72 Mbps	Peering	4,043634381	
33	G.723 to G.723	0 Mbps	Hubbing	4,027362821	Peering
34		0 Mbps	Peering	4,027364804	
35		15 Mbps	Hubbing	4,027362373	Peering
36		15 Mbps	Peering	4,027362888	
37		40 Mbps	Hubbing	4,027350611	Peering
38		40 Mbps	Peering	4,027363611	
39		72 Mbps	Hubbing	3,931805827	Peering
40		72 Mbps	Peering	4,027361191	
41	G.729 to G.729	0 Mbps	Hubbing	4,04370743	Peering
42		0 Mbps	**Peering**	**4,043711575**	
43		15 Mbps	Hubbing	4,043707357	Peering
44		15 Mbps	**Peering**	**4,043710844**	
45		40 Mbps	Hubbing	4,043707133	Peering
46		40 Mbps	**Peering**	**4,043710733**	
47		72 Mbps	Hubbing	4,009195587	Peering
48		72 Mbps	**Peering**	**4,04370845**	

4.3 Simulation result of Peering and Hubbing for Packet Loss

Looking at the results of packet loss measurement in Table 12 shows that packet loss from no traffic loads to 72 Mbps traffic load is still below 3% in accordance with ITU-T G.1010 standard, it shows that all codecs used for interconnect on Hubbing model and Peering is good. Considering the results of packet loss measurements in Table 12 on voice interconnection services, packet loss values that show 0% distances have excellent network quality, ie in traffic load range 0 to 40 Mbps, whereas at 72 Mbps traffic load there are some losses packets due to in this study the available bandwidth is only able to accommodate up to 43 Mbps.

From the result of measuring 72 Mbps traffic load, it is found that there are three most low packet loss codec (almost 0%), ie G.711u, GSM and G.729 codec. This is because the existing voice service at QCI level 1, which involves IP voice service traffic is performed by a special carrier (guaranteed bits), 'so as to ensure the data bit rate remains constant in delivery, and the loss of packets that can prevent it. The smaller the value of the missing packet, the clearer the quality will be.

Based on table 12, that for both Hubbing and Peering topologies, errors occur using G.722, G.711a, G.711u, GSM, G.723 and G.729 at only 72 Mbps traffic load. Based on the table, it can be concluded that the codec G.711u, GSM, G.723 and G.729 have packet loss and codec G.723 to G.723 with the largest packet loss 0.00646 using Hubbing model.

Table 12. Simulation result of Peering and Hubbing for Packet Loss

No	Codec	Traffic Loads	Interconnection Model	ITU-T std > 0,03	
				Packet loss total	rank of best value
1	G.722 to G.722	0 Mbps	Hubbing	0	Peering/ Hubbing
2		0 Mbps	Peering	0	
3		15 Mbps	Hubbing	0	Peering/ Hubbing
4		15 Mbps	Peering	0	
5		40 Mbps	Hubbing	0	Peering/ Hubbing
6		40 Mbps	Peering	0	
7		72 Mbps	Hubbing	0,001177778	Peering
8		72 Mbps	Peering	7,77778E-05	
9	G.711a to G.711a	0 Mbps	Hubbing	0	Peering/ Hubbing
10		0 Mbps	Peering	0	
11		15 Mbps	Hubbing	0	Peering/ Hubbing
12		15 Mbps	Peering	0	
13		40 Mbps	Hubbing	0	Peering/ Hubbing
14		40 Mbps	Peering	0	
15		72 Mbps	Hubbing	0,002855556	Peering
16		72 Mbps	Peering	3,33333E-05	
17	G.711u to G.711u	0 Mbps	Hubbing	0	Peering/ Hubbing
18		0 Mbps	Peering	0	
19		15 Mbps	Hubbing	0	Peering/ Hubbing
20		15 Mbps	Peering	0	
21		40 Mbps	Hubbing	0	Peering/ Hubbing
22		40 Mbps	Peering	0	
23		**72 Mbps**	**Hubbing**	**0,000733333**	Peering
24		72 Mbps	Peering	0	
25	GSM to GSM	0 Mbps	Hubbing	0	peering/ hubbing
26		0 Mbps	Peering	0	
27		15 Mbps	Hubbing	0	Peering /hubbing
28		15 Mbps	Peering	0	
29		40 Mbps	Peering	0	peering/ hubbing
30		40 Mbps	Hubbing	0	
31		**72 Mbps**	**Hubbing**	**0,004333333**	Peering
32		72 Mbps	Peering	0	
33	G.723 to G.723	0 Mbps	Hubbing	0	peering/ hubbing
34		0 Mbps	Peering	0	
35		15 Mbps	Hubbing	0	peering/ hubbing
36		15 Mbps	Peering	0	
37		40 Mbps	Hubbing	0	peering/ hubbing
38		40 Mbps	Peering	0	
39		**72 Mbps**	**Hubbing**	**0,006466667**	Peering
40		72 Mbps	Peering	0	
41	G.729 to G.729	0 Mbps	Hubbing	0	peering/ hubbing
42		0 Mbps	Peering	0	
43		15 Mbps	Hubbing	0	peering/ hubbing
44		15 Mbps	Peering	0	
45		40 Mbps	Hubbing	0	peering/ hubbing
46		40 Mbps	Peering	0	
47		**72 Mbps**	**Hubbing**	**0,002277778**	Peering
48		72 Mbps	Peering	0	

4.4 Simulation result of Peering and Hubbing for Throughput

Throughput is the actual network speed. The value of throughput is inversely proportional to the amount of traffic, the greater the traffic then the value of throughput will be smaller. Increasing the amount of traffic load can reduce the throughput value, this is caused by the traffic load on the link initially idle to full. From the overall measurement result, the throughput of traffic load at 72 Mbps will be smaller than the throughput of traffic load at 0 Mbps. To calculate the throughput value, calculated by comparing the bandwidth of the appropriate results. Percentage standard value ratio> 75%.

From table 13, it is clear that the G.711u, G.711a and G.722 codecs have higher throughput than other codecs. This is because the bit rate codec G.711 and G.722 is much higher that is 64 Kbps compared to G.729 codec which is only 8 Kbps or 1/8 of the G.711 bit codec speed, then the codec has good throughput that is GSM , and the next codec is G.723, G.711 and G.722. Using the G.711 codec, the largest throughput value obtained based on the result is 80.34 Kbps or 93.85% of the total BW.

On the user side, the greater the throughput value, the more data that is sent. For that, a large throughput value would be better. Overall it can be seen from the measurement that the G.711a, G.711u and G.722 codecs have higher throughput values than other codecs (G.729 codec, GSM and G.723), but when the traffic load starts to increase from 0 Mbps up to 72 Mbps, G.711a, G.711u and G.722 codecs decrease considerable throughput value compared to other codecs. And compared to the other three codecs, namely G.729, GSM and G.723 codecs, the G.729 codec has lowered the rate as the traffic load increases. This shows that the G.729 codec is more stable when faced with a large number of traffic

Table 13. Simulation result of Peering and Hubbing for Throughput

No	Codec	Traffic Loads	Interconnection Model	ITU-T std > 75%		No.	Codec	Traffic Loads	Interconnection Model	ITU-T std > 75 %	
				Throughput (kbps)	BW (%)					Throughput (kbps)	BW (%)
1	G.722 to G.722	0 Mbps	Hubbing	80,20136026	91,974037	25	GSM to GSM	0 Mbps	Hubbing	29,26286349	82,488692
2		0 Mbps	Peering	80,18849093	93,6781436	26		0 Mbps	Peering	29,25523002	82,4671741
3		15 Mbps	Hubbing	80,09879048	93,5733534	27		15 Mbps	Hubbing	29,25188665	82,4577495
4		15 Mbps	Peering	80,13164453	93,6117343	28		15 Mbps	Peering	29,245812	82,4406258
5		40 Mbps	Hubbing	80,27736365	93,7819669	29		40 Mbps	Peering	29,24825366	82,4475086
6		40 Mbps	Peering	80,07885471	93,5500639	30		40 Mbps	Hubbing	29,30686211	82,6127191
7		72 Mbps	Hubbing	80,02905521	93,4918869	31		72 Mbps	Hubbing	29,21761237	82,3611342
8		72 Mbps	Peering	80,14190578	93,6237217	32		72 Mbps	Peering	29,24844289	82,448042
9	G.711a to G.711a	0 Mbps	Hubbing	80,19703391	93,6881237	33	G.723 to G.723	0 Mbps	Hubbing	16,85424664	82,3162229
10		0 Mbps	Peering	80,18623492	93,6755081	34		0 Mbps	Peering	16,8555878	82,3227731
11		15 Mbps	Hubbing	80,18173963	93,6702566	35		15 Mbps	Hubbing	16,85350786	82,3126147
12		15 Mbps	Peering	80,08135806	93,5529884	36		15 Mbps	Peering	16,84978083	82,2944119
13		40 Mbps	Hubbing	80,29927055	93,8075591	37		40 Mbps	Hubbing	16,85445307	82,3172311
14		40 Mbps	Peering	80,14995602	93,6331262	38		40 Mbps	Peering	16,84726024	82,2821013
15		72 Mbps	Hubbing	80,0988739	93,5734508	39		72 Mbps	Hubbing	16,84219139	82,257345
16		72 Mbps	Peering	80,090777	93,5639918	40		72 Mbps	Peering	16,85847838	82,3368907
17	G.711u to G.711u	0 Mbps	Hubbing	80,19970989	93,6912499	41	G.729 to G.729	0 Mbps	Hubbing	23,64458488	89,5628215
18		0 Mbps	Peering	80,18639662	93,675697	42		0 Mbps	Peering	23,64269797	79,8739796
19		15 Mbps	Hubbing	80,12612872	93,6052906	43		15 Mbps	Hubbing	23,63755613	79,8566086
20		15 Mbps	Peering	79,83432612	93,2643997	44		15 Mbps	Peering	23,63570763	79,8503636
21		40 Mbps	Hubbing	80,33521788	93,8495536	45		40 Mbps	Hubbing	23,6486447	79,8940699
22		40 Mbps	Peering	80,18467748	93,6736886	46		40 Mbps	Peering	23,6508322	79,9014601
23		72 Mbps	Hubbing	80,07434252	93,5447927	47		72 Mbps	Hubbing	23,62764395	79,8231215
24		72 Mbps	Peering	80,0321419	93,4954929	48		72 Mbps	Peering	23,65561245	79,9176096

5. Conclusion

For the simulation test result using 0 Mbps traffic, the shortest delay result is obtained using G.729 codec and Peering topology, with the delay value of 39,955 ms. The shortest delay result of 39,960 ms for 15 Mbps traffic is obtained using G.729 codec with Peering topology-. Meanwhile, for 40 Mbps traffic and 72 Mbps traffic using G.729 codec with Peering topology , the shortest delay result of 39,956 ms and 39,958 ms are obtained respectively.

Therefore, it can be inferred, there are several resumes from comparison model between Hubbing and Peering as a regulation recommendations on interconnection using IP in Fundamental Technical Plan and Indonesian Government Regulation No. 52 in 2000. It can be concluded that from the simulation results that for overall traffic load up to 72 Mbps, Peering model was the best alternative IP interconnect model. Thus it was the most recommended for used in the IP interconnection implementation for the most efficient codec that maintained the highest voice quality over the under test IP Interconnection was G.729 codec.

For further research, it is suggested to use a more extensive variety of traffic loads with devices which can support up to more than one Gbps with the Hubbing model if the traffic load is high and using G.729 codec transcoding technique and also make a comparison with AMR –WB codec.

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