IOP Conference Series: Materials Science and Engineering

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Enhanced cloud based mobile core network with network function virtualization

To cite this article: H D Septama *et al* 2020 *IOP Conf. Ser.: Mater. Sci. Eng.* **857** 012022

Enhanced cloud based mobile core network with network function virtualization

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Abstract. The mobile technology is improved by adopt the Internet Protocol (IP) based technology. The IP Multimedia Subsystem or IP Multimedia Core Network Subsystem (IMS) is a technology designed by 3rd Generation Partnership Project (3GPP). This technology used IP to deliver the communication packet and recognized as Next Generation Network (NGN). The emerging IP based technology took a lot of advantages by using IP to deliver a packet. However, some issues i.e. availability, security and reliability still need consideration to meet the telecommunication standard. The cloud based computing become buzzword as a technology for cost reduction and hardware efficiencies. This paper conducted the feasibility to setup mobile core network in cloud based environment and enhanced it with network function virtualization (NFV) feature. The NFV made the network more scalable, efficient and robust network management. The QoS is presented as the results to concluded the feasibilities. The results show that cloud based mobile core network with NFV is feasible since the QoS still full fill the telecommunication standard.

Keywords: core network, NFV, cloud, mobile

1. Introduction

The communication nowadays become human basic need. The mobile communication subscribers increase each year with various demand of communication. The voice communication then become multimedia communication. User use not only voice and messaging but also use video and internet access. The mobile technology also improved by adopt the Internet Protocol (IP) based technology. The IP Multimedia Subsystem or IP Multimedia Core Network Subsystem (IMS) [1] is a technology designed by 3rd Generation Partnership Project (3GPP). This technology used IP to deliver the communication packet and recognized as Next Generation Network (NGN). The signalling is delivered with Session Initiation Protocol (SIP) [2] and for the media is using Real Time Protocol (RTP) [3]. The emerging IP based technology took a lot of advantages by using IP to deliver a packet. However, some issues i.e. availability, security and reliability still need consideration to meet the telecommunication standard.

The deployments of telecommunication technology always using dedicated devices. Since the emerging telecommunication technology also using an IP based, therefore we may deploy the service in the cloud environments. The cloud based computing become a use full technology for cost reduction and hardware efficiencies. The telecommunication companies may reduce capital expenditures

(CAPEX) and operating expenditures (OPEX) since a lot of part is running virtually. The cloud computing is believed as one of future telecommunications infrastructures [4]. However, since IP is not a reliable communication protocol, the deployment should ensure that the QoS of telecommunication standard is fulfilled.

This paper conducted the feasibility to setup mobile core network in cloud based environment enhanced with network function virtualization (NFV) feature. The QoS is presented as the results to concluded the feasibilities.

2. Related works

The works in [5] virtualized network elements based on LTE SAE/EPC core network. The paper deployed network elements in virtualized environments and make some QoS measurements to ensure the feasibilities. The testbed consisted of several virtual machine running networks element service and make a voice call to the VoIP server. Traffic generator also used to simulate the actual traffic condition. The results show the networks element works well in virtualization environment.

The works in [6] explain why we need to setup the IMS mobile core networks over cloud networks. The works show that using cloud computing technology will help to make IMS more scalable and easy to deploy. The paper also proposed the possibility of the IMS using Infrastructure as a Service (IaaS) and Platform as a Service (PaaS) and what their weakness.

The NFV made revolutionary changes on network implementation by made the network features running virtually. The leading telecommunication companies support and implement the NFV and is believe become one of the main key of future network technologies [7]. The use cases and standard landscape of NFV is standardized, the work in [8,9] show several cases of how the NFV could help driving a transition towards future programmable carrier grade networks. Thus, replace the default bridge interface with NFV not only made the infrastructure more scalable and efficient but also easier to managed.

3. Cloud and NFV architecture

Cloud computing is an emerging computing technology. This technology enables computing scalability dynamically by virtualized its hardware resources. Users may access cloud through various devices i.e laptop, computer, smart phones. The advantages using cloud technology is cost savings, hardware efficiencies, high availability and easy scalability [10].

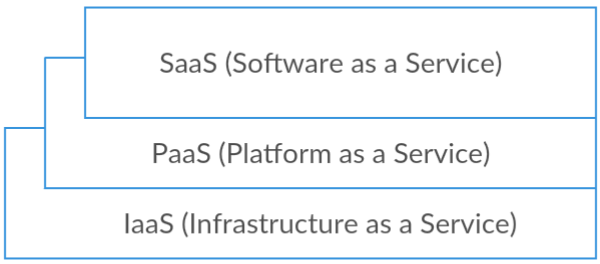

Figure 1. Cloud computing layer

Figure 1 depicts the layer of cloud computing architecture by the services its offered. The top layer is Software as a Service (SaaS), this kind of services is offered an application to be running remotely by the users. Second layer is Platform as a Services (PaaS) that offered a ready to use platform for he users. It means providers should provide not only the hardware resources but also an operating system equipped with software stack to be able users installed or run its own applications. The last layer is Infrastructure as a Services (IaaS), the services that only provide the infrastructure or hardware resources and user manages the infrastructure itself. There are three types of cloud, public cloud that provide its service access through global internet. Private cloud that deployed its cloud on a private networks and combination of both types that called hybrid cloud.

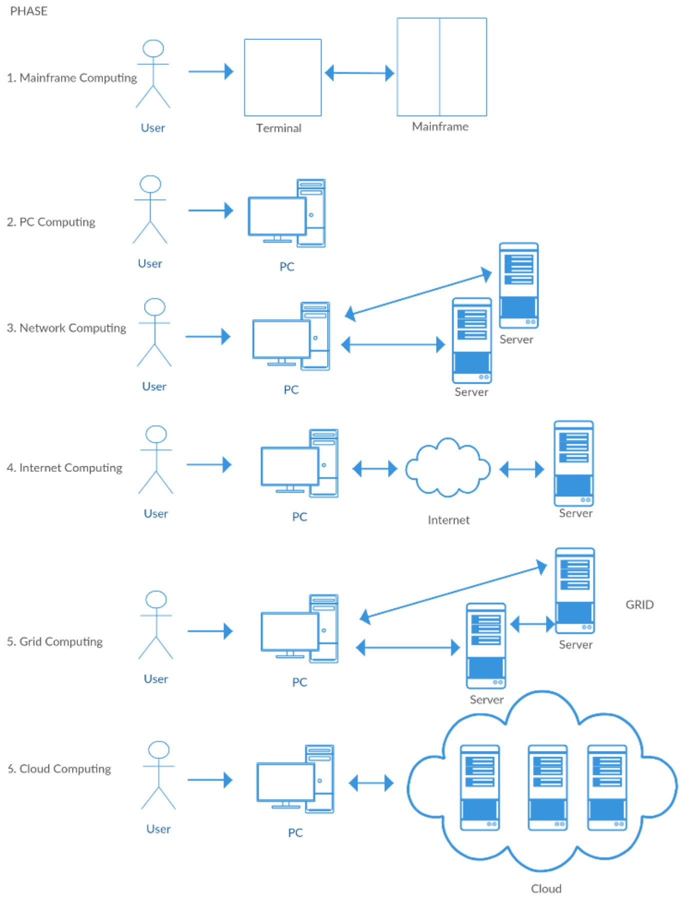

Figure 2. Six computing paradigms [4]

Figure 2 depicts the six phase of computing paradigms. Each phase shows the paradigm or method to access and how allocate the computer resources. The cloud computing method become novel computing architecture. This architecture become widely used since its offers various advantages such as cost reduction and scalable environments. Users only pays that they need or used for the time being, as for the demand is increased the amount of computer resources may be added to the services. However, the cloud also has introducing another problem such are security and latency. A lot of research is done to provide more secure and less latency for the cloud network service.

The ETSI high level NFV framework is depicted on figure 3. The figure shows the relation of hardware resources, virtualization layers/hypervisor and virtualised network function that runs on top of it. NFV management and orchestration is used to manage and control the virtual network functions therefore the management and control process become centralized and easier. The addition, deletion, configuration and another interfaces management and control features of virtual network function is done by single application.

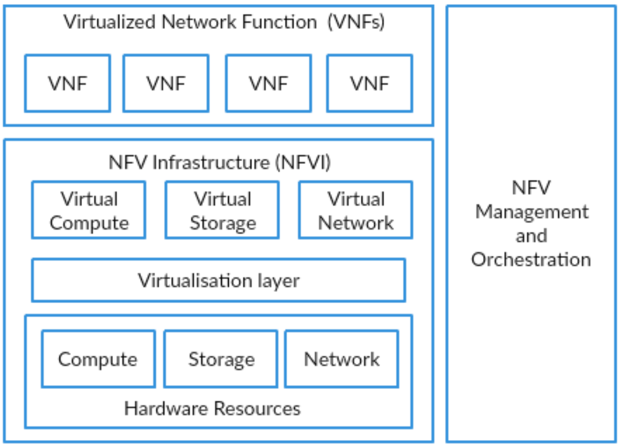

Figure 3. High-level NFV framework [4]

4. Testbed and scenarios

This paper use private cloud infrastructure for the testbed deployments. Figure 4 is the testbed high level architecture model. The mobile core network elements are deployed as a virtual application over operating system in a virtual machine (VM).

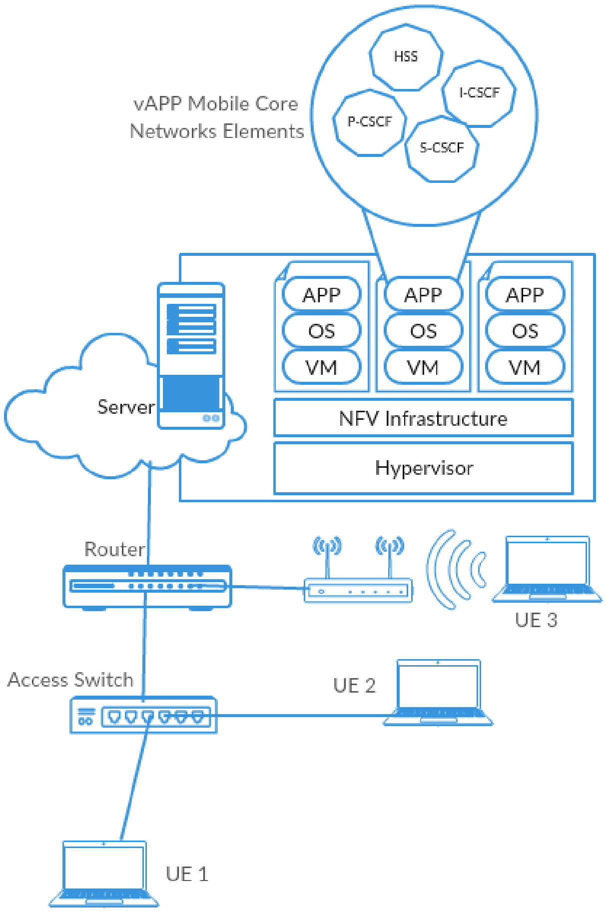

Figure 4. The tesbed environments

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The VM is connected with virtual interface using NFV instead of using its default bridge configuration. The main reason to use the NFVI is to provide fully manage and control to user using centralized managements plane. Thus, two user equipment (UE) is connected via wired connection and another one UE is connected via wireless (Wi-Fi) connection. UE 1 is also equipped with packet sniffing software to capture the packet during call scenarios procedure. The sniffed packet is use to analysis the QoS of the communication to conclude the feasibility using mobile core network in cloud based environments.

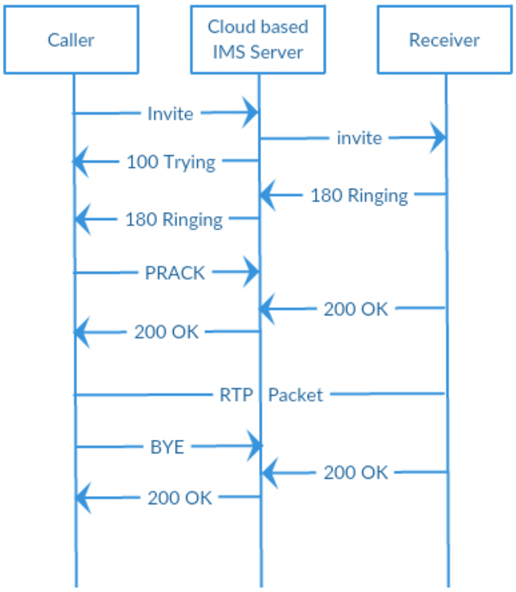

Figure 5. Call scenarios procedure

There are several scenarios conducted to observed the QoS feasibility. Firstly, the voice call quality for wired access connection are observed between UE 1 and UE 2. Secondly, the voice call for wireless communication quality also observed between UE 1 and UE 3 communications. Last, the video call communication quality between UE 1 and UE 2. All call is using SIP signalling protocols and G.711 codec for voice communication. Figure 5 show the basic call procedure for the call test. During the call, RTP packet is sniffed and then the Jitter calculated to present the quality. RTP is a protocol defined by IETF RFC 3550 [3] that provide real time data for delivery service.

2	3	4	8	9	16	32
V	**P**	**X**	**CSRC Count**	**M**	**Payload type**	**Sequence number**
Timestamp						
Synchronization source (SRC)						
Contributing source (CSRC: variable 0 - 15 items, 2 octets each						

Figure 6. RTP packet structure

Figure 6 depicts the RTP packet structure, the timestamp field is used to identify each packet for jitter calculation purposes. This field contains the sampling instant of the first octet in the RTP data packet that derived from a clock and incremented monotonically and linearly in time to allow synchronization and jitter calculations. Jitter of this RTP packet is calculated using jitter calculation described in IETF RFC 3550 [3].

5. Result and Discussion

In this paper, inter arrival jitter calculation is considered to analysed the performance. However, the delay and packet loss for each scenario also presented. The ITU – T standard in table 1 [10] describes parameters for good telephony communication i.e., the average MOS of 4.4, delay < 150 ms, jitter < 20 ms and packet loss < 5 %.

Table-1. ITU-T standard for delay and jitter

Grade	Delay (ms)	Jitter (ms)
Very good	0	0
Good	0 - 150	0 – 20
Average	150 - 30	20 - 5
Poor	>300	>50

Inter arrival jitter (J) and mean deviation of the difference (D) defined for pairs packet, by comparing packets spacing at the receiver and sender. Packet spacing is the difference between the RTP packet timestamp from sender and receiver time when arrived packet also knows as relative transit time.

$$D(i,j) = (Rj - Ri) - (Sj - Si) = (Rj - Sj) - (Ri - Si) \tag{1}$$

Where:
Si,Sj = RTP timestamp from packet i and j,
Ri,Rj = Time of arrival in RTP timestamp units for packet i and j

Equation 1 above showed the difference (D) calculation of two packets i and j. Since the RTP packet arrived continuously at the receiver then Difference (D) from pairs of packets (n packet and n – 1 packet) also calculated continuously. Therefore, the inter arrival jitter could be calculated using equation 2 as follows:

$$J(i) = J(i-1) + \frac{(|D(i-1,i)| - J(i-1))}{16} \tag{2}$$

The Inter arrival jitter algorithm above is the optimal first-order estimator and 1/16 gain parameter used in order to give a good noise reduction ratio while maintaining a reasonable rate of convergence.

The RTP packet sniffed during the call scenarios then calculated. The calculated results is depicted in figure 7 through 8 below. Figure 7 show the inter arrival jitter of voice call for first scenarios procedure using wired access connection. The results show that max jitter is 14,42 ms with mean jitter 13,91 ms. The packet loss is 0% with delta max or latency 40,5 ms. Figure 8 show the jitter of voice call for wireless connection. The results show that that max jitter is 14,26 ms with mean jitter 13,47 ms. The packet loss is 0% with delta max or latency 41,67 ms. As for the video call depicted in Figure 9, the max jitter is 15,86 ms with mean jitter 13,99 ms. The packet loss is 0% with delta max or latency 50,76 ms.

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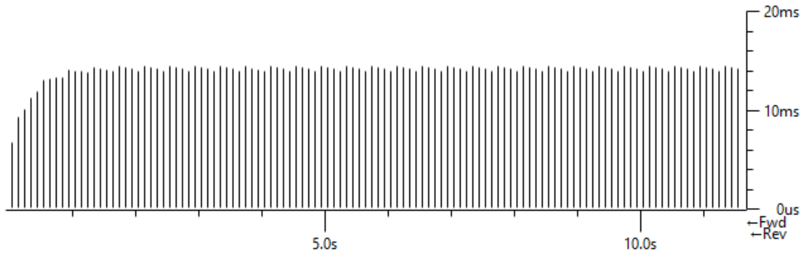

Figure 7. Jitter wired voice call

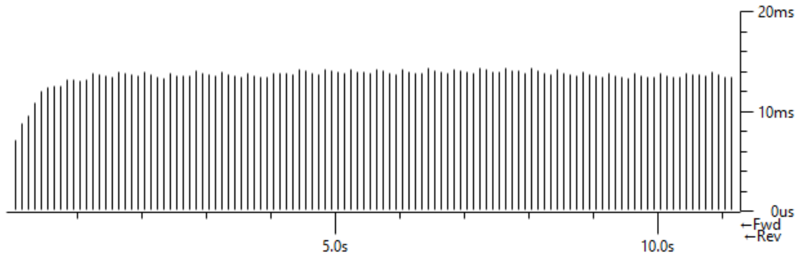

Figure 8. Jitter wireless voice call

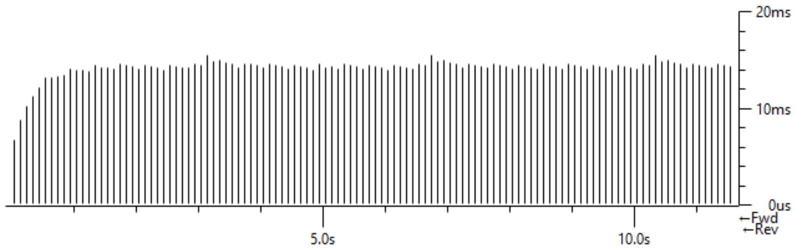

Figure 9. Jitter video call

6. Conclusions

The results show that the QoS of cloud based mobile core network with NFV is feasible. The jitter, packet loss and latency is still fit the ITU.T recommendation in a good grade. For future works the call generator considered to deployed for providing traffic load, so that the simulation results can represent the actual state of the network when operated.

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