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Single Board Computer Applications as Multi-Server VoIP

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Abstract: Telecommunication technology is developing along with information technology and several innovations in several audio and data transmission and reception techniques. Innovation and communication technology are hoped to be able to create efficiencies in regards to time, equipment, and cost. The Public Switched Telephone Network (PSTN) telephone technology has experienced integration towards communication using Internet Protocol (IP) networks, better known as Voice over Internet Protocol (VoIP). VoIP Technology transmits conversations digitally through IP-based networks, such as internet networks, Wide Area Networks (WAN), and Local Area Networks (LAN). However, the VoIP cannot fully replace PSTN due to several weaknesses, such as delay, jitter, packet loss, as well as security and echo. Telephones calls using VoIP technology are executed using terminals in the form of computer devices or existing analogue telephones. The benefit of VoIP is that it can be set in all ethernet and IP addresses. Prefixes can be applied for inter-server placements as inter-building telephone networks without the addition of inefficient new cables on single board computers with Elastix installed. Prefix and non-prefix analysis on servers from single board computers can be tested using QoS for bandwidth, jitter, and packet loss codec. The installation of 6 clients, or 3 simultaneous calls resulted in a packet loss value in the prefix Speex codex of 2.34%. The bandwidth in the prefix PCMU codec has an average value of 82.3Kbps, and a non-prefix value of 79.3Kbps, in accordance to the codec standards in the VoIP. The lowest jitter was found in the non-prefix PCMU codec with an average of 51.05ms, with the highest jitter for the prefix Speex codec being 314.65ms.

Keywords: Single Board Computer, VoIP, prefix, non-prefix, QoS

I. INTRODUCTION

Communication and information technology are technologies that are currently experiencing important developments, aside from basic needs, for people. The industrialization era has given birth to people who require fast, precise, and accurate communication for information media. In the current pandemic condition, the majority of people work in the information and communication sector. Information and communication are further developing alongside with the concept of modern community development that is currently occurring [1]. With the development of the concept of office building and education center layouts that are separate and far from one another, communication technology must be able to adjust with the needs of its users [2]. A system is designed to be able to enable the smooth execution of the sub-system mechanisms that exist in the transmission of information and communication. Currently, the communication system has become the main mediation with the activity of data processing and data transmission being executed according to the people's needs, oriented towards activities in education, business, as well as government [3]. Data is the processing result in the form of information, that then becomes accurate communication used in fulfilling the needs of human interaction. The use of inter-building telephones requires a design, especially in regards to the addition or change of its network [4]. The design of telephone networks requires an estimate of each floor area, and the allocation of rooms required. The commonly uses system from the PT. Telkom Indonesia channel makes use of the Private Automatic Branch Exchange (PABX) facility[5]. The PABX then connects to a Main Distribution Frame (MDF). Through the Distribution Cable (DC), the telephone network is spread to the Junction Box (JB) on each floor of the building. From this box, the telephone network is then forwarded to a telephone. Currently, the development of intelligent buildings with an integrated telephone system fulfills the telephone and computer network integration. Each computer is directly connected to a communication network (telephone), so that the addition of the total number of telephone connections must be provided. Computers in each room are not always used in retrieving data from the outside. The need of modems and telecommunication lines become another costly expense for a company. The utilization of one or several lease lines connected to a server enables storage of information data to become more effective. Furthermore, the use of computer support equipment such as printers and plotters can be done simultaneously in one integrated network, with the existence of internet telephone networks that have developed to become the Voice over Internet Protocol (VoIP), or better known as the IP Telephone [6].

Telephones and Internet Protocol (IP) is an integration between computers, telephones, and televisions, in one communication environment. The VoIP Voice over Internet Protocol (VoIP) is one of the organizational recommendations in the standardization of IP. The basic foundations of unified messages (UM) and unified communication (UC) utilizes VoIP. VoIP is required in the integration of several server programs. Furthermore, VoIP can transcend audio, video, and data traffic that has been transformed in the form of digital codes and grouped into data packets that are sent in the *Internet Protocol* (IP) platform, in real-time. Audio communication, the chief of telephones, can be passed through IP networks in the VoIP technology. With this method, the utilization of bandwidth and the management of telephone connector costs become more efficient. VoIP technology is increasingly developing with the existence of mini computers, better known as the Raspberry Pi3[7]. The Raspberry Pi 3 has similar functions to the average computer. Power use is lower compared to existing computers, allowing the Raspberry Pi to become more practical and cost-effective. Furthermore, the Raspberry Pi3 is a Single Board Computer equipped with a ARM1176JZF-S Core 700MHz processor, 512 MB size RAM, and a Video Core IV GPU. The need of inter-building communication utilizes the IP network with a trunk system to connect all Raspberry Pi3 servers. The use of Internet Asterisk exchange v2 (IAX) is a VoIP protocol to connect between Asterisk Servers. The trunk function uses an IAX forward port, which is more effective since it only requires one port for signaling and media stream, which is the TCP/UDP 4569 port [8]. The IAX is designed to overcome issues in the NAT environment, with one trunk being able to accommodate 300-400 telephone calls. Based on the parameters used in designing a VoIP server with the Raspberry Pi3 server system, tested with the use of prefix, non-prefix, and communication quality from the interconnected multi-server is in line with the system design in this research, The Quality of Service (QoS) parameter analysis in the VoIP network is related to the audio transmission mechanism through internet protocol, which encompasses bandwidth, jitter, and client delay.

II. RESEARCH METHOD

This research is a form of quantitative theory development on the science and technology of Voice over Internet Protocol (VoIP). The system design was tested objectively in order to analyze interconnection between telephones and explained the multi-server multi server Raspberry Pi 3[9]. Interconnection analysis is tested as the IP PBX server from the quality of delay and packet loss. The variables used in this research include a single board VoIP server with the Raspberry Pi3 configurated as a portable server, and an Elastix operation system with the Free PBX GUI configuration display. The wireless data communication media uses a MikroTik Router Board as a local hotspot local network which is integrated with the VoIP server. The VoIP server is connected to the Linephone in an Android smartphone. The softphone application from the client's side eases interconnection between connected clients. The packet loss parameter and bandwidth capacity are used to determine quality during video conference services. The equipment and materials in this research are divided into two:

1) Hardware, consisting of a Raspberry Pi 3 B (Quad Core 1,2GHz Broadcom BCM2837 64bit CPU, 1GB RAM), laptop, Mikrotik RB9561Ui-2nD, MikroTik RB941-2nD, SDcard, and Android smartphone; and
2) Software, consisting of Elastix, FreePBX, Android Linephone, and WinBox.

The system is designed by combining Raspberry Pi 3 as the IP-PBX servers, and executing a trunk setting for the VoIP as the multi-server, with the single board computer. Testing is done on the effect of the codec towards the 3 VoIP servers, in terms of prefix and non-prefix. The steps of this research include the installation of the Raspbian Operating System (OS) using Win32 Disk Imager software, continued with installing Elastix as the VoIP server, as well as the configuration of Elastix to the Raspberry Pi 3. The next step is to install and configure Linephone Softphone on the Android smartphone as a form of video call communication. The final step is to test the telephone using prefix and non-prefix. Fig.1 shows the interconnection between servers.

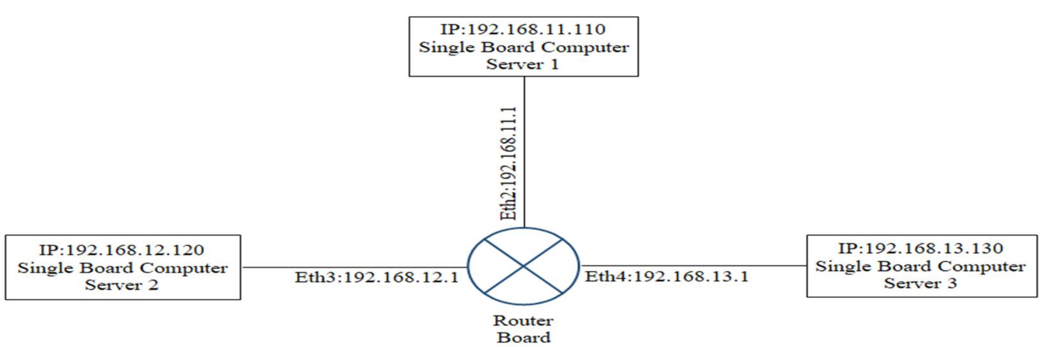

Fig. 1 Server Network Interconection

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The research design is adjusted to the media transmission device in the VoIP service implementation. The Raspberry Pi is used as a VoIP server by utilizing the Elastix operation system, installing it to the SD-Card [10]. The Raspberry Pi configuration acts as a PBX configuration by adding SIP user accounts [11]. A UTP is used for the Raspberry Pi in the router. Once the configuration process is completed, inter-location communication takes place with the use of the local network from the router. The client is connected through the softphone configuration as the media of the video call. The softphone also acts as a linephone. Each client has installed the linephone and owns an SIP user account to enable the clients to communicate without extra spending for a video call. The device used by the client is required to stay connected to the local router network used. The system parameter in the testing with performance of the Quality of Service (QoS) includes the packet loss and bandwidth of the linephone according to the system design. Call data from the Raspberry Pi3 as the VoIP server uses the PCMU, GSM, iLBC, and Speex codec. Call testing involves 1-3 calls. Bandwidth tests were run during voice calls.

Fig. 2 VoIP Server with Raspberry Pi 3

Fig. 2 shows the design of the Raspberry Pi 3 as the VoIP server with the router, connected by a UTP cable. The transmission media is adjusted according to the device used, in order to simplify use between the cable and wireless transmission service. Once the client is connected to the mikrotik router network, they can then make a voice call. The connecting of the VoIP server to other VoIP systems in the sending and receiving of calls in a device uses the trunk configuration [12]. The Inter Asterisk Exchange (IAX) is used to assign names and numbers to the trunk [13,14]. Afterwards, the PEER Detail is filled by filling the destination host, allowed codec, the username sent to the system along with the username used when testing the system's call, and adding the confidential number in the IAX. The assignment of trunks from the PEER Detail with a number of different parameters are done according to the following:

1) Allow = all, meaning that all codec can be used in the call;
2) Type = friend, meaning that the caller and receiver can send or receive calls from the server;
3) Host = 192.168.11.110, as the IP address server;
4) Username = pbx3to1, as the username which will be sent to the long-distance system when attempting and authenticating calls; and
5) Secret, which means giving a password sent through long-distance when attempting and authenticating calls.

The setting of call destinations when outgoing through the trunk is called outbound routes, which is done by entering the PBX > PBX configuration > outbound route menu. The function of the outbound route is to assign names to the routes used in the system. The prefix initiation is filled with the destination area code from the server, while the match pattern is filled by the extension number code of the destination server. The addition of outbound routes is done by filling route names with the same function type, as calls. The trunk sequence for matched routes is added to trunks that match the call device. The prefix method is the initial numbering for the user terminals than can be contacted. The numbering system is related to the user identity to the position terminal or the host/user central address. Fig. 3 shows the trunk configuration from the IAX, added with prefix.

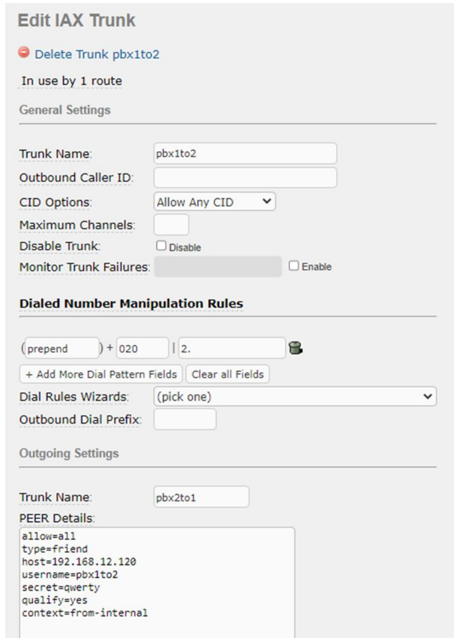

Fig. 3 Trunk Configuration with Prefix

Trunk configuration is done in three servers. Afterwards, outbound route configuration is done to fill the destination extension number by assigning an (X) symbol with the number prefix, 2XX. The outbound route configuration is also done on the three servers. The data analysis for the Raspberry Pi 3 server with the Quality of Service (QoS) performance testing parameters include: packet loss, delay, and bandwidth from the linephone [15]. Testing is done with the GSM, PCMU, Speex, and iLBC audio codec. An example of non-prefix call testing using the G.722 coding is shown in Fig. 4.

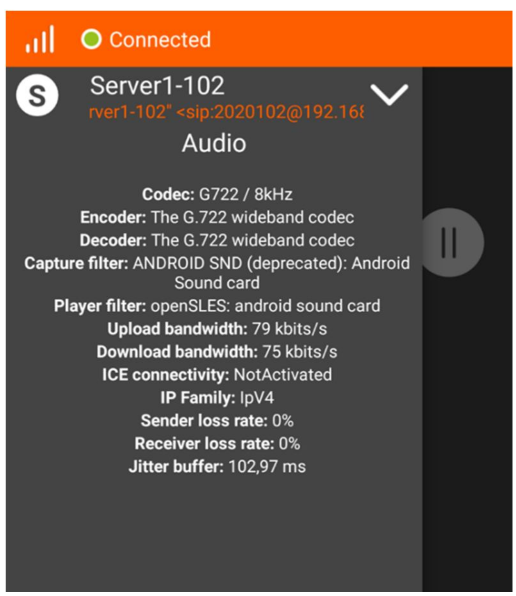

Fig. 4 Non-Prefix Voice Call Testing using G.722 Codec

Voice call testing is done through the extension number. The call static menu shown in the G.722 codec was successfully used as a voice call without prefix. The result is shown by the 79kbps upload bandwidth, and the 75kbps non-prefix download. Voice call testing with prefix in the G.722 codec was done to the extension number from the menu of the 80kbps upload bandwidth data codec, and the 78kbps download bandwidth in the G.722 codec.

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III. RESULTS AND DISCUSSION

This research is conducted based on Raspberry Pi3 application data as a VoIP server with the combination of prefix and non-prefix GSM, Speex, ILBC, and PCMU codec. Using QoS, the following information on data packet loss and bandwidth using 6 clients was retrieved, as shown in Fig. 5.

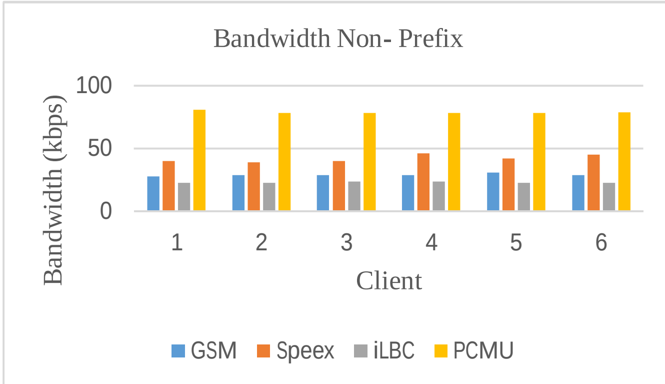

Fig. 5 Non- Prefix Bandwidth

Based on the non-prefix bandwidth audio test, the highest average value of approximately 78.67kpbs is found in PCMU, with the lowest found in iLBC with an average of 23.00kbps. Fig. 6 shows prefix bandwidths for four tests, which include GSM, Speex, iLBC, and PCMU.

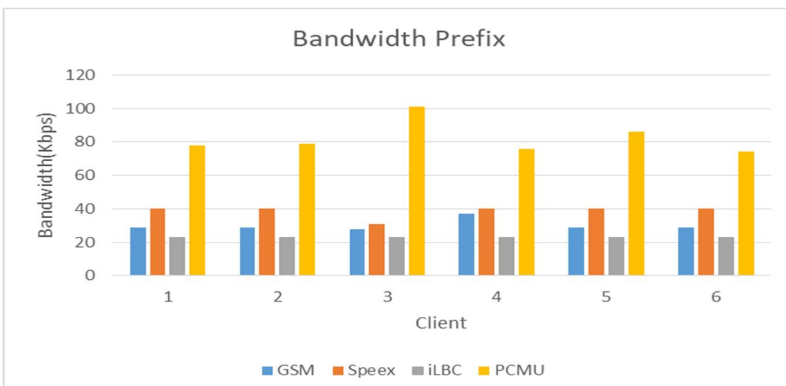

Fig. 6 Prefix Bandwidth

Based on the bandwidth parameter, the best size average is found in PCMU which resulted in clearer audio, due to the more structured order of the PCMU codec. The jitter parameter test on the four devices shows delay or network instability which resulted in data reception delays. Based on the 6-client jitter audio test, the smallest number is shown by the PCMU codec with an average of 95.75ms in non-prefix, 51.05ms in prefix. Meanwhile, the Speex codec jitter averaged at 314.65ms with prefix, and 223.82ms in non-prefix. The use of codec resulted in delays, depending on the time from the processor capacity used during the sending process. Alternatively, this can also be caused by latency from the many variations of delay during the transmission of data in the network. The PCMU/G711 audio jitter test has fulfilled the From the VoIP codec standard of 64kbps for prefix value, with 51.05ms which is still deemed suitable.

IV. CONCLUSIONS

Based on the results of this research, the use of Raspberry Pi 3 as a VoIP server, the IAX2 application trunk and outbound can communicate with one another. The IAX2 device as a trunk for the Elastix communication system server, can connect with other Elastix servers, with an average bandwidth of 82.33Kbps for prefix codec, and 78.67 for non-prefix, according to the VoIP audio standard codec. Meanwhile, the lowest jitter is found in the non-prefix PCMU codec with an average of 51.05ms, and the highest jitter is found in the prefix Speex codec with 314.65ms.

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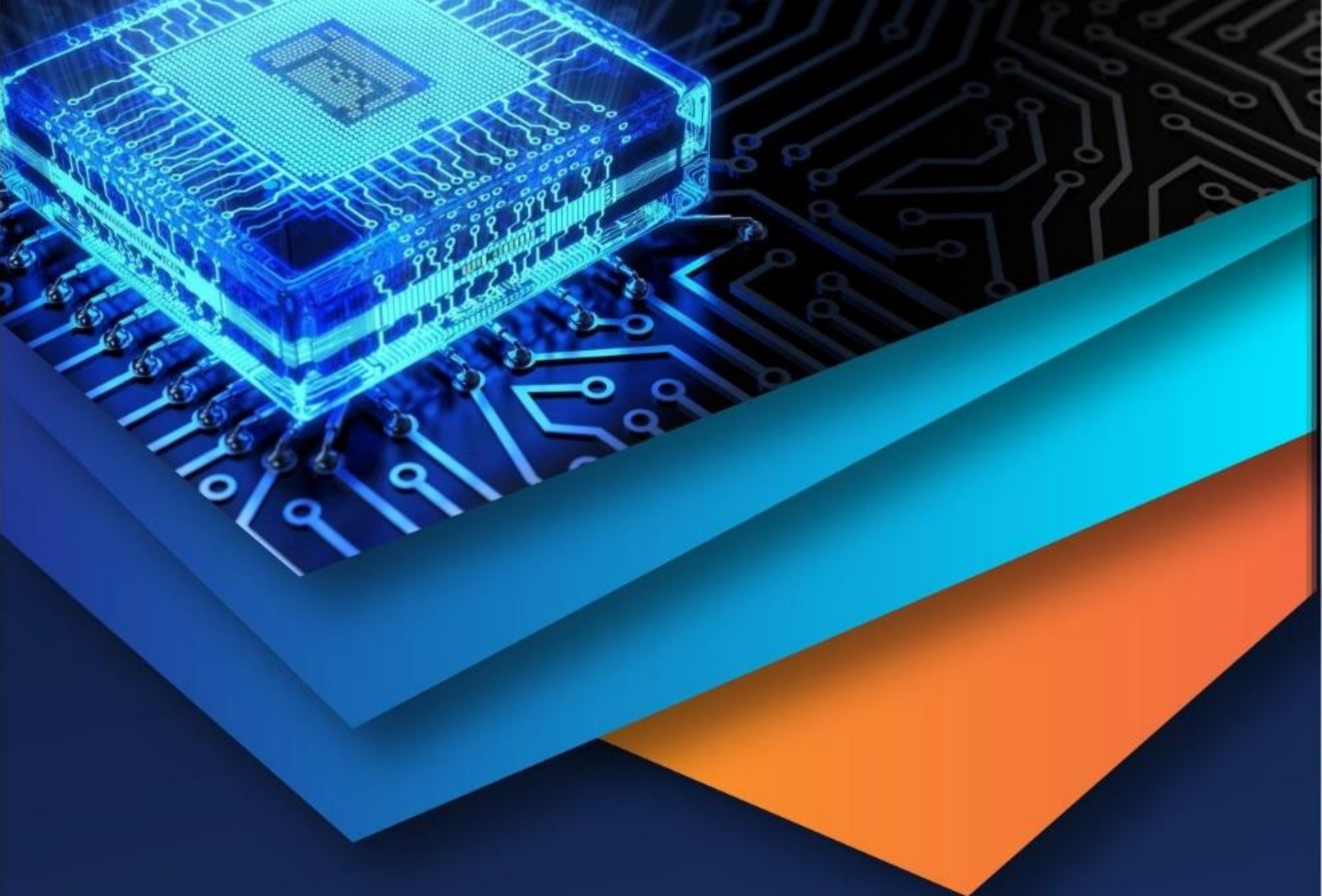

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