Analysis and Design of Voip Server (Voice Internet Protocol) using Asterisk in Statistics and Statistical Informatics Communication of Banten Province using Ppdioo Method

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Abstract: Communication is the most important thing in the success of an activity. Communication that occurs in a computer network does not only process data packet transfers between computers, but computer technology is currently able to transfer data in the form of sound frequencies that are processed into data packets and sent between computers on the internet network. The development of internet technology is not only used for getting information, but also that is widely used. One of them is audio call or VoIP (Voice over Internet Protocol), this technology has been used by several messenger service providers, such as Yahoo Messenger, Goggle Talk, Skype, and others. Voice over Internet Protocol is a technology that makes internet media able to conduct voice communication remotely directly. Analog sound signals, like what you hear when communicating on the telephone are converted to digital data and sent over the network in the form of data packets in real time. Asterisk is a softswitch to operate the proxies, with session initiation protocol (SIP) based. The aim is to build an Asterisk-VoIP based server, so that it can be developed in future research as needed. The methodology in outline consists of two lines, namely literature study and PPDIOO (Plan, Prepare, Design, Implement, Operate, Optimize). It has been implemented in the Banten Province Communication and Information Service, which has been previously established by the internet network. So that VoIP here is functioned for maximizing it.

1. Introduction
In Regional Device Organizations, to facilitate work activities between OPDs, communication carried out still uses two ways, namely direct communication (face to face) and communication using technology such as telephone, fax and e-mail. Which of course is very helpful and accelerates the process of delivering information. For communication by utilizing voice transfer technology (for example telephones) it is usually provided by outsiders by using a network of Public Switched Telephone Network (PSTN) which of course the management also involves these outside parties. This is of course a little annoying if there is a problem with the facilities and infrastructure. The arena of improvement or repairs must also involve outside parties, cannot be handled by internal parties. Internet technology continues to grow until now, the use of these technologies is still limited to entertainment. Actually this technology can be used for applications that are more useful and have
economic value. One of the uses is for VoIP (Voice over Internet Protocol) applications. VoIP offers a different communication alternative to conventional communication which is circuit switching. With the support of network infrastructure, communication between Regional Organizations is not only done through telephone, but also can utilize existing internet network facilities, provided that these parts have devices connected to the Internet network.

Rapid development is also shown by open source technology. This technology allows developers to build a system that is free and easy to develop. There are also many java based programs that are offered free, Asterisk is one of the open source software that can be used as a VoIP server.

The Banten Province Information Communication, Statistics and Coding Agency does not currently have VoIP (Voice over Internet Protocol) Technology in communicating, besides maximizing the existing internet network so that it can reduce the cost of spending on communication needs.

1.1. Purpose and objectives

1.1.1 Purpose

The purpose of this activity is to maximize the existing internet network so that it can reduce the expenses of communication needs.

1.1.2 Objective

The purpose is:

- Designing and Implementing VoIP (Voice over Internet Protocol) technology at the Banten Information Communication, Statistics and Coding Service Office as a free communication channel means that the use of VoIP technology is free and unpaid.

- Implement VoIP (Voice over Internet Protocol) technology as a time-saving, practical and effective communication.

2. Research Methods

The methods used in this activity are:

2.1 Study of literature

Do this literature study with the aim of collecting data by collecting and analyzing documents, both written, drawing and electronic documents. The documents that have been obtained are then analyzed (parsed), compared and combined (synthesis) in applying VoIP technology and forming a systematic, coherent and complete result of the study as a result of analysis of these documents.

2.2 Metode PPDIOO (Plan, Prepare, Design, Implement, Operate, Optimize)

Figure 1. PPDIOO Method Scheme
## 2.2.1 TESTING

System Testing is an investigation conducted to obtain information about the system being built.

## 2.2.2 DESIGN OF TESTING SCENARIO

After everything needed to build a VoIP system is complete, there is a scenario that will be carried out to test the security holes and performance of the VoIP network.

### 2.2.3 Testing Scenario

In this scenario a call is made between the clients using the Zoiper softphone. When communication between clients takes place, the conversation that occurs will be captured (tapping) using a wireshark, and analyzed the data in it and in Play (played again) to find out whether the sound from the conversation between clients. After that the captured data will be analyzed for performance in the network. How much is the Qos of the system. This can be known by analyzing delay, jitter and packet loss. After getting the package, it will be analyzed how the security and performance of the VoIP.

## 3. Measurement And Analisys Performance of VoIP

Measurement and testing of performance includes delay, jitter and packet loss. For the test scenario described earlier. The first experiment will be used in the first scenario.

### 3.1 Measurement and Analysis Delay

Delay is the time needed for data to travel from distance to destination. Delay can be influenced by distance, physical media, congestion or long processing times [5]. According to the TIPHON version (Telecommunication and Internet Protocol Harmonization Over Network), the amount of delay can be classified as follows:

| Category   | Delay |
|------------|-------|
| Excellent  | <150ms|
| Very Good  | <250ms|
| Good       | <350ms|
| Bad        | <450ms|

Delay is an important parameter for determining the quality of a VoIP network. Based on the standards of ITU-T for good VoIP quality, Delay must be <150 ms so there is no overlap in communication. Then the package that passes through will be captured. The process of retrieving data using the Wireshark monitoring tool. The data analyzed is using RTP.

### Table 1. Delay

| No | Client | Source | Objective | Delay (ms) |
|----|--------|--------|-----------|------------|
| 1  | Client A | 192.168.1.12 | 103.36.11.164 | 2000 |
|    |        | 103.36.11.164 | 192.168.1.12 | 3000 |
| 2  | Client B | 192.168.11.41 | 103.36.11.164 | 3000 |
|    |        | 103.36.11.164 | 192.168.11.41 | 1000 |
| **Total Delay** | | | | **5000** |

**Delay** can be counted by equation below:

\[
\text{Voice Payload size} = \text{Long IP Packet} - (\text{Ethernet Header} + \text{IP Header} + \text{UDP Header} + \text{RTP Header})
\]
• **Client A**
  192.168.1.12 🔄 103.36.11.164
  Voice Payload size = 2309 Byte – (14+20+8+12) byte = 2255 byte
  103.36.11.164 🔄 192.168.1.12
  Voice Payload size = 2294 byte - (14+20+8+12) byte = 2240 byte
  Total = 2294 byte + 2240 byte = 4534 byte
  With 2309 byte payload so delay packet is below:
  Delay = 4534 byte / 2309 byte = 1.96 s

• **Client B**
  192.168.11.41 🔄 103.36.11.164
  Voice Payload size = 3296 byte - (14+20+8+12) byte = 3242 byte
  103.36.11.164 🔄 192.168.11.41
  Voice Payload size = 2058 byte - (14+20+8+12) byte = 2004 byte
  Total = 3242 byte + 2004 byte = 5246 byte
  With 2309 byte payload so delay packet is below:
  Delay = 5246 byte / 2309 byte = 2.27 s

From the results of the analysis of data package retrieval above, it can be concluded that the delay of each client can still be accepted based on ITU-T standards for good VoIP quality, delay must be <150.

### 3.2 Measurement and Analysis of Jitter

Jitter is the time variation of periodic signals in electronics and telecommunications commonly called delay variations, which shows the number of delay variations in data transmission on the network. Delay queues on routers and switches can cause jitter. According to the TIPHON version, the amount of jitter can be classified as follows:

| Category   | Jitter |
|------------|--------|
| Excellent  | 0 ms   |
| Very Good  | 75 ms  |
| Good       | 125 ms |
| Bad        | 225 ms |

TU-T recommends good Jitter is <30ms. Jitter that greatly affects sound quality. The bigger the jitter, the sound that is produced will be increasingly unclear (intermittent). The jitter value affects when the RTP packet that comes will be processed into sound. When the jitter value is smaller than the data packet processing time, then before using the packet processing, the next packet has come to wait for processing. So that the sound produced is good too. Following is the analysis of the VoIP system by using Wireshark Monitoring tools.
### Table 4. Jitter

| No | Client | Source | Objective | Jitter (ms) |
|----|--------|--------|-----------|-------------|
| 1  | Client A | 192.168.1.12 | 103.36.11.164 | 0.77 |
|    |         | 103.36.11.164 | 192.168.1.12 | 14.78 |
| **Total Delay** | | | | |
| 2  | Client B | 192.168.11.41 | 103.36.11.164 | 0.07 |
|    |         | 103.36.11.164 | 192.168.11.41 | 8.82 |

**Jitter** can be counted by the equation below:

\[
\text{Jitter average} = \frac{\text{Total Variasi Delay}}{\text{Number of Packages received}}
\]

For counting total variety of delay, it can be counted by the equation below:

\[
\text{Total variety of Delay} = (R_i - S_i)(R_i - S_i)
\]

Notes:

- \( R_i \) = Time when it receive or coming
- \( S_i \) = First time it coming

- Client A
  192.168.1.12 ↔ 103.36.11.164
  e.g Packet received = 99518
  Total Variety of Delay = -0.215026
  \[
  \text{Jitter rata-rata} = \frac{\text{Total Delay Variasi}}{\text{Number of Packages received}}
  \]
  Jitter Average = -0.215026 / 99518 = -216 s

From the results of the analysis of the data package above, it can be concluded that jitter from each client can still be accepted based on ITU-T standards for good VoIP quality, delay must be <30.

#### 3.3 Measure and Analysis Packet loss

Packet loss is defined as the failure of packet transmission to reach its destination. The packet failure reaches its destination, caused by several possibilities, among others: the occurrence of over load in the network, collision or congestion on the network, errors in physical media, occurrence of overflow in the buffer. According to the TIPHON version, the amount of packet loss can be classified as follows:

### Table 5. Packet loss

| Category    | Delay |
|-------------|-------|
| Excellent   | 0 %   |
| Very Good   | 3 %   |
| Good        | 15 %  |
| Bad         | 25 %  |

### Table 6. Example of packet loss in VoIP system

| No | Client | Source | Objective | Packet Loss |
|----|--------|--------|-----------|-------------|
| 1  | Client A | 192.168.1.12 | 103.36.11.164 | 1.3 |
| 2  | Client B | 192.168.11.41 | 103.36.11.164 | 0 |
Packet loss can be count by equation below:

\[
\text{Packet Loss} = \frac{\text{Number Packages Sent} - \text{Amount of Packages received}}{\text{Number Packages Sent}} \times 100\%
\]

- **Client A**: \(\frac{2309 - 2294}{3296 - 2058} \times 100\% = 0\%\)
- **Client B**: \(\frac{2309}{3296} - 2058 \times 100\% = 0.37\%\)

From the analysis of data retrieval above, it can be concluded that the Packet Loss of each Client can still be accepted based on ITU-T standards for good VoIP quality, i.e., Packet Loss must be in the range of 10% to 30%.

3.4 Capture Data VoIP use Wireshark

This is the result of VoIP network monitoring using Wire Shark.

![Figure 2. VoIP Data Capture](image)

4. References

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