Evolution in Electrical and Electronic Engineering Vol. 3 No. 2 (2022) 100-107

EEEE

Homepage: http://publisher.uthm.edu.my/periodicals/index.php/eeee
e-ISSN : 2756-8458

Call Setup Over VoIP in Softswitch System using Web Interface and Application

Mohamad Haikal Jamaludin[1], **Abul Khair Anuar**[1]*

[1]Department of Electronic Engineering, Faculty of Electrical and Electronic Engineering,
Universiti Tun Hussein Onn Malaysia, 86400 Parit Raja, Batu Pahat, Johor, MALAYSIA

*Corresponding Author Designation

DOI: https://doi.org/10.30880/eeee.2022.03.02.012
Received 27 June 2022; Accepted 04 September 2022; Available online 30 October 2022

Abstract: The telephone network has enabled long-distance communication between two people which was previously impossible. As the telephone technology grows, phone calls no longer require a dedicated physical circuit which has now been replaced with VoIP which converts phone calls to internet data. Softswitch system uses a web interface and application is used to set up call over VoIP in many governments and non-government telephone exchange systems. In this work, call setup using Green Packet Global's telecom architecture and Wireshark as a network packet analyser were investigated. A descriptive study and inferential analysis utilising a Wilcoxon signed-rank test were used to establish network traffic dependability and responsiveness with 16 Call ID samples. Local and international network traffic varies slightly in RTP packets, mean jitter, and timestamping. A descriptive examination of RTP packets found 99 percent local and international packet delivery success. Using inferential analysis to analyse mean jitter and timestamping, it was found that 95 percent of call quality was maintained in terms of mean jitter and online record keeping. The measurement result showed that the company's call setup meets its standards. In contrast, the network company's Service Level Agreement (SLA) is similar to other organisations without any server or network difficulties.

Keywords: Voice Over Internet Protocol (Voip), Network Traffic, Wireshark, Wilcoxon Signed-Rank Test

1. Introduction

Cloud technology is one of Malaysia's top three technological investments to enter a new age, covering 35% of investments [1]. The telecommunication sector which is one of the early adopters of cloud technologies consists of service and infrastructure divisions. The service division concentrates on invoicing and marketing, whilst the infrastructure division is where the actual communication takes place. Future technology such as 5G services and infrastructures will have to rely on innovative

Corresponding author: abul@uthm.edu.my

technology to fulfil increasing needs [2]. The Softswitch system is an example of innovative and vital telecommunications technology that may affect the telecommunication industry. Softswitch uses hardware and software. The method operates as an intelligent server for call and terminal control [3].

Every VoIP service, including WhatsApp, Skype, and Telegram, has a Softswitch system that directs calls to mobile users or fixed lines. This includes the Softswitch billing system that handles all VoIP provider purchase rates [4]. In this project, a Softswitch system that utilises a web interface and application is used to set up VoIP calls and later monitor real-time information regarding telecommunication services and infrastructures.

2. Research Methodology

This section describes the project's design methodology and process flow. It includes a full description of the software utilised and its functioning, as well as an explanation of the project's architecture, concept, and application to explain how the system functions for this project.

2.1 Design of Call Setup over VoIP in Softswitch System

Figure 1 provides an overview of the call setup in the Softswitch system that uses Vodia as the PBX and Digitalk as the Softswitch system. Both the IP phone and the softphone application have the number +65 registered in it, which is the Direct Inward Dialing (DID) number for Singapore.

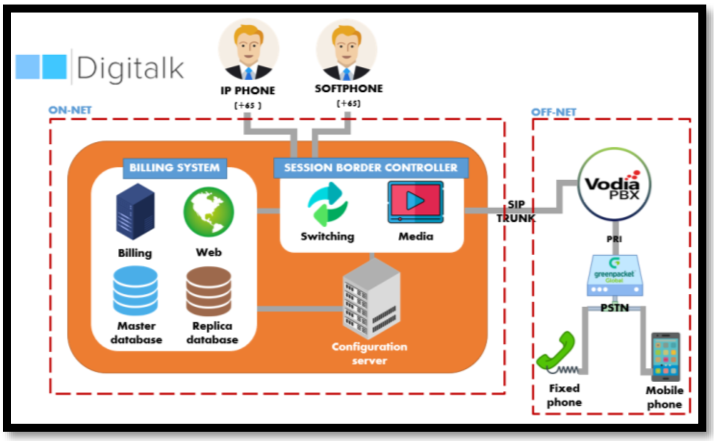


Figure 1: The call setup over VoIP in the Softswitch system

2.2 User interface of the call setup using web interface and application

The user interface of the call setup using web interface and application is applied in an IP Phone and softphone application named Grandstream Wave Lite to test the network of the system. The registration phase is executed as shown in the flowcharts in Figure 2 before the testing phase begins.

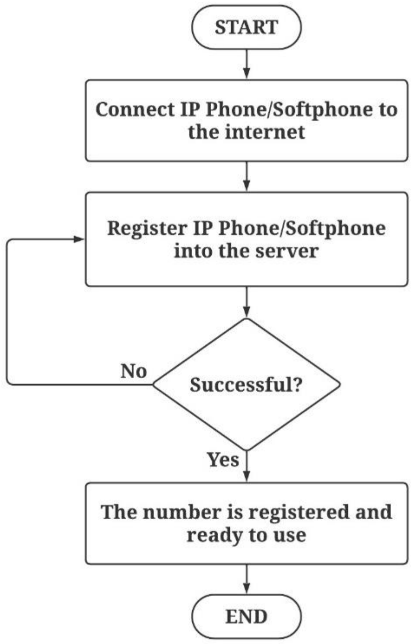


Figure 2: User interface flowchart (IP Phone and Softphone)

2.3 Statistic Equations

This project analyses participant network traffic to assess system reliability. In this investigation, local traffic versus oversea network traffic is an important parameter where time-stamping, mean jitter, and RTP packets were analysed. A data collection of numbers is evaluated using X category (Singapore number calling Singapore number) and Y category (Singapore number calling international number). Eq. 1 [5] calculates the network traffic for this situation.

$$W = \sum_{i=1}^{N_r} [sgn(x_{2,i} - x_{1,i}) \cdot R_i \quad Eq. 1$$

where W is the test statistic, N_r is the sample size of the participants, sgn is a sign function, $x_{1,i}, x_{2,i}$ is corresponding ranked pairs from two distributions, and R_i is Rank i.

3. Results and Discussion

3.1 The Calling ID of the project

As part of the actual research, 16 samples were gathered, which included both local and international phone calls. From Table 1 and Table 2, the samples are indicated as category X and category Y which means local calling and oversea calling. The caller was tested only for outgoing calls using the softphone application (Grandstream Wave Lite). The recorded data is analysed using the Wireshark software to give information regarding the network traffic. A descriptive analysis and inferential analysis is used to identify the reliability of the call system in capturing RTP packets, evaluate whether 36 there is a significant difference in the quality of the call in terms of mean jitter if the jitter 0.5 ms, and evaluate whether there is a significant difference in the quality of keeping records of information online in terms of timestamping. The results are as detailed:

Table 1: Singapore number calling Singapore number (Local)

No.	IP ADDRESS	RTP Packets	Mean Jitter (ms)	Start at (s)
1.	From: 172.XXX To: 103.XXX	Capture: 9190 Expected 9190	0.521933	5.085462 @ 5
2.	From: 172.XXX To: 103.XXX	Capture: 9190 Expected: 9190	0.521933	5.085462 @ 5
3.	From: 172.XXX To: 103.XXX	Capture: 8328 Expected: 8328	0.527623	5.085462 @ 5
4.	From: 172.XXX To: 103.XXX	Capture: 4983 Expected: 4983	0.887144	2.095187 @ 5
5.	From: 172.XXX To: 103.XXX	Capture: 5599 Expected: 5599	0.748909	2.105412 @ 5
6.	From: 172.XXX To: 103.XXX	Capture: 9458 Expected: 9458	0.458690	2.464178 @ 5
7.	From: 172.XXX To: 103.XXX	Capture: 5341 Expected: 5341	0.471675	2.050494 @ 4
8.	From: 172.XXX To: 103.XXX	Capture: 5229 Expected: 5229	0.561679	2.239091 @ 5

Table 2: Singapore number calling international number (Oversea)

No.	IP ADRESS	RTP Packets	Mean Jitter (ms)	Start at (s)
9.	From: 172.YYY To: 103.YYY	Capture:9152 Expected: 9152	0.647123	4.615944 @ 6
10.	From: 172.YYY To: 103.YYY	Capture: 2712 Expected: 2712	0.277572	5.184719 @ 5
11.	From: 172.YYY To: 103.YYY	Capture: 4588 Expected: 4588	0.325661	3.569915 @ 8
12.	From: 172.YYY To: 103.YYY	Capture: 5690 Expected: 5690	0.409408	2.222251 @ 6
13.	From: 172.YYY To: 103.YYY	Capture: 3194 Expected: 3190 (-0.13%)	0.528763	2.919066 @ 5
14.	From: 172.YYY To: 103.YYY	Capture: 4804 Expected: 4804	0.405765	2.188068 @ 5
15.	From: 172.YYY To: 103.YYY	Capture: 4880 Expected: 4880	0.402058	3.952786 @ 8
16.	From: 172.YYY To: 103.YYY	Capture: 5880 Expected: 5880	0.380292	0.995846 @ 6

3.2 Wilcoxon Signed-rank Test Analysis on the Network Traffic

Figure 3 compares the RTP packets that had been captured and are expected to arrive in the Wireshark software analysis. The result shows that the packets arrived with complete no loss. Each with a different number of packets is because of congestion or faults and duration in terms of calls. Due to less congestion or flaws in the system of local calling, there are no errors or losses in packets in the calls. While the RTP packets of the Y category in Figure 4 have one call of packet loss with -0.13%. The congestion or faults in the system may occur during the time of calling.

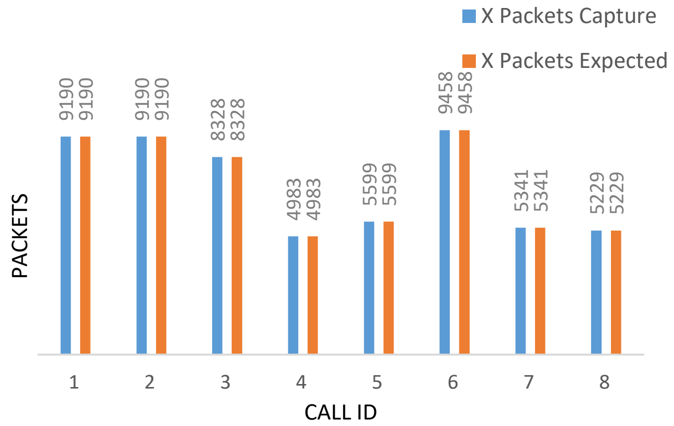


Figure 3: The RTP Packets in the X category between callers

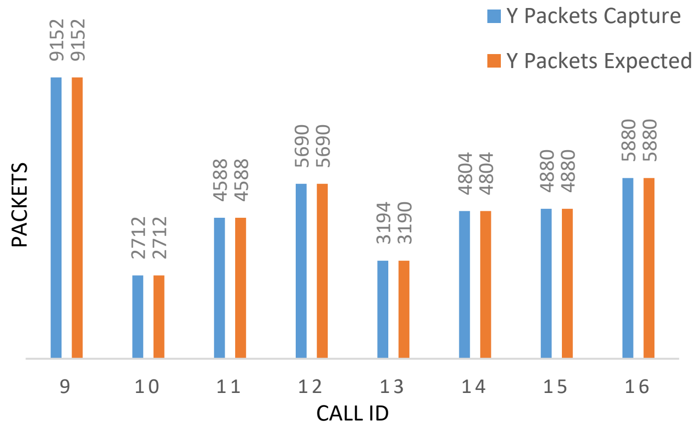


Figure 4: The RTP Packets in the Y category between callers

Figure 5 and Figure 6 show the bar chart of the result in terms of mean jitter in different call IDs of categories X and Y. Based on the observation, it shows the development of data have other numbers of time in jitter. A Wilcoxon signed-rank test was made to analyse the jitter effect on the network connection as the distributional assumptions of the procedures are not satisfied. Using the company's standard Service Level Agreement (SLA), which is 0.5ms maximum jitter, the mean jitter collected in categories X and Y have no significant difference.

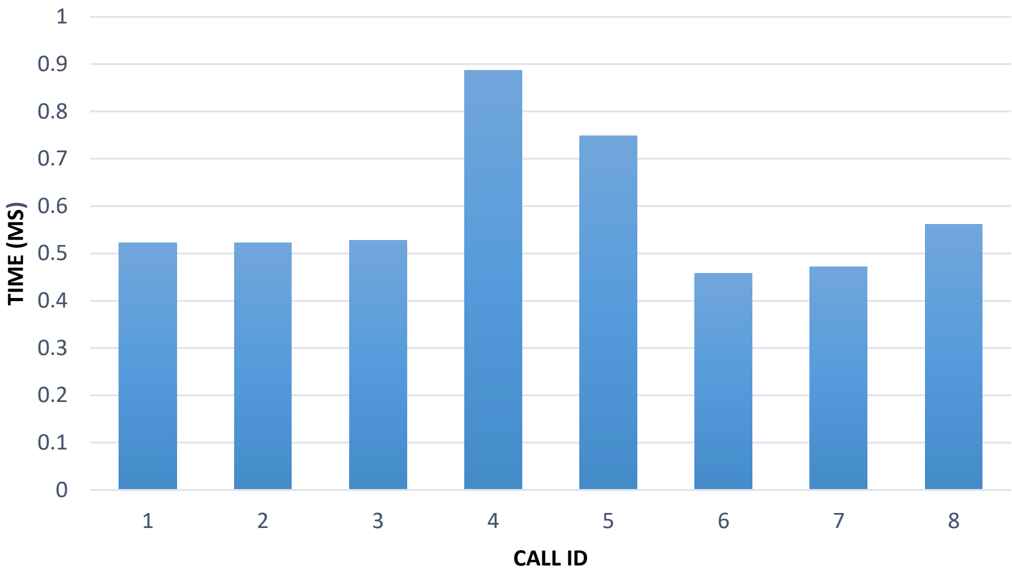


Figure 5: The mean Jitter of X category between callers

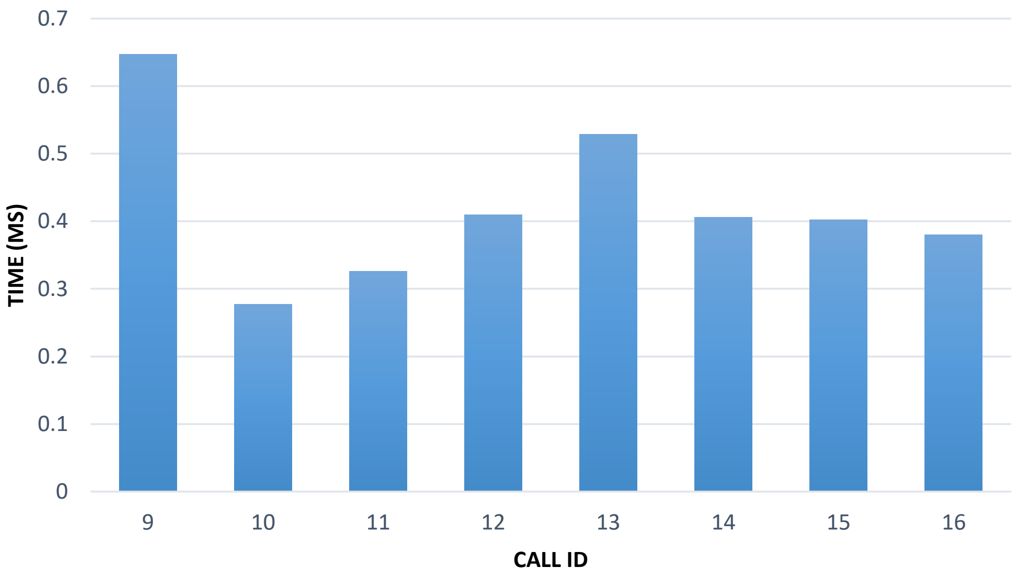


Figure 6: The mean Jitter of Y category between callers

Figure 7 shows the plotted graph of the result of timestamping of the X and Y category between callers. Based on the observation, category Y has delayed timestamping compared to category X. The timestamping efficiency will be measured using Wilcoxon signed-rank test to analyse the timestamping between each of the calls of local calling and oversea calling. The statistics show no significant difference in terms of the time of categories X and Y. As for packets; the result shows a significant difference in terms of packets in categories X and Y. The ability to timestamp high-precision packets requires free error in sequence in the system. Therefore, the analysis of the time difference in timestamping does not affect the data as long as it is received.

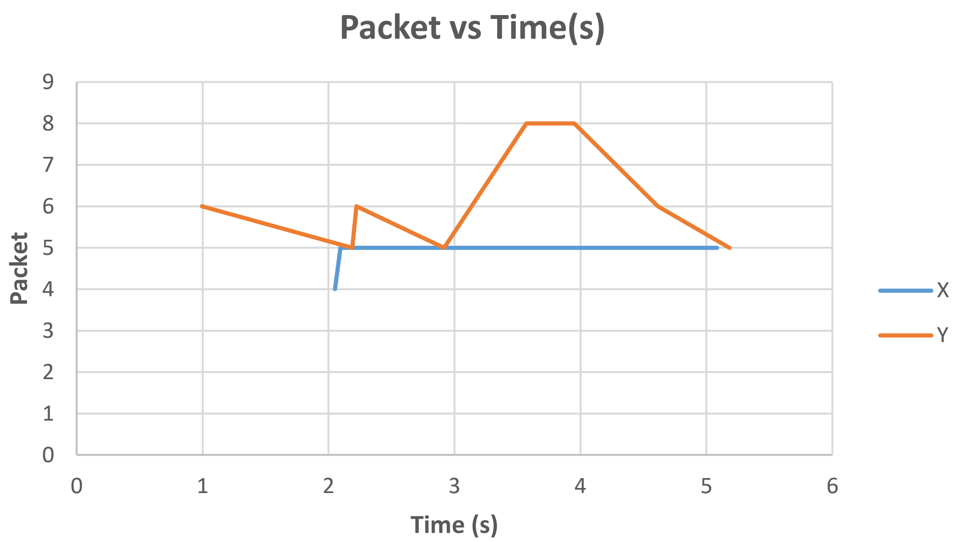
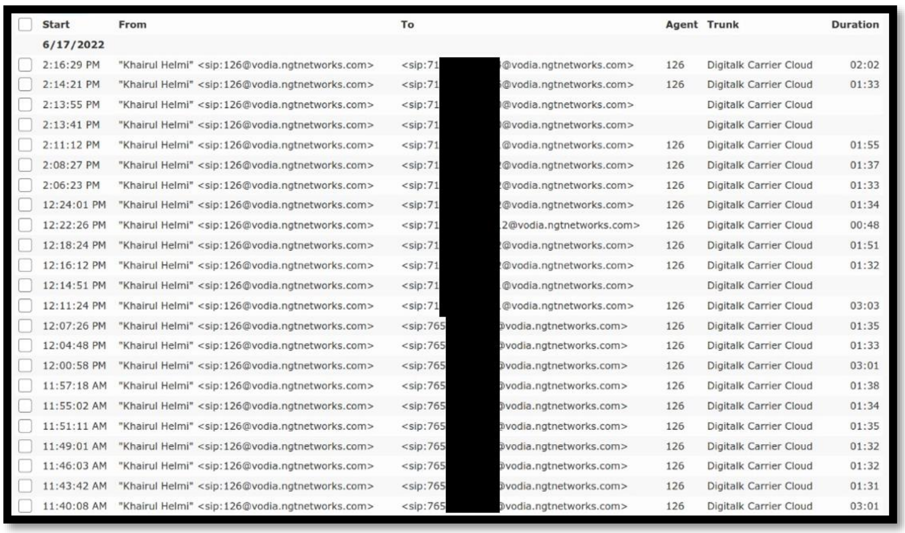


Figure 7: The timestamping of X and Y categories between callers

3.3 Database of Softswitch and PBX system

The database of call IDs is stored in Softswitch and PBX system; the database in Figure 8 shows the outgoing calls of the participants, whether successful or unsuccessful, the system still recorded. The IP address of the participants blurred due to the privacy and confidentiality of the company.



Start	From	To	Agent	Trunk	Duration
6/17/2022					
2:16:29 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	02:02
2:14:21 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:33
2:13:55 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>		Digitalk Carrier Cloud	
2:13:41 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>		Digitalk Carrier Cloud	
2:11:12 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:55
2:08:27 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:37
2:06:23 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:33
12:24:01 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:34
12:22:26 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	00:48
12:18:24 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:51
12:16:12 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:32
12:14:51 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>		Digitalk Carrier Cloud	
12:11:24 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:71[redacted]@vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	03:03
12:07:26 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:35
12:04:48 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:33
12:00:58 PM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	03:01
11:57:18 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:38
11:55:02 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:34
11:51:11 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:35
11:49:01 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:32
11:46:03 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:32
11:43:42 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	01:31
11:40:08 AM	"Khairul Helmi" <sip:126@vodia.ngtnetworks.com>	<sip:765[redacted]vodia.ngtnetworks.com>	126	Digitalk Carrier Cloud	03:01

Figure 8: Database for Outgoing Calls

4. Conclusion

It can be concluded that the call setup over VoIP in the Softswitch system of the company local has a reliable and functional system calling in terms of RTP packets, mean jitter, and timestamping. It is found that the system calls have a minimal difference between local calling and oversea calling, proving that this project's object has been achieved. The accomplishment is achieved by analysing the network traffic between each call. From the results, both local and overseas calling have the effectiveness and minimal error to use as users as the service provider needs to provide the best standard and services.

Acknowledgment

The Faculty of Electrical and Electronic Engineering at Universiti Tun Hussein Onn Malaysia is acknowledged by the authors for its assistance.

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