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VoIP System Implementation Using Issabel as an Integrated IP PBX Server with Telco Vendors

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Abstract. In today's telecommunication world, we cannot be separated from long-distance communication using telephones, either personal needs or mass needs. Therefore, this research aims to introduce a system that may be less familiar today, namely the VoIP System Implementation Using Issabel as an Integrated IP PBX Server with Telco Vendors. In this research, the author conducted experiments using VirtualBox to support thesis writing and using one of the providers that the author borrowed from the office in order to make calls to GSM and PSTN. The author observes that currently there are quite a lot of call centres that need this system to support employee performance and cost efficiency. Through this journal, the author hopes that entrepreneurs or telecommunication companies can gain an understanding of the VoIP System and IP PBX Server to be implemented.

Keywords: VoIP, IP PBX, Issabel, Asterisk

1 Introduction

Computer network technology in the current era of modernization is developing rapidly by bringing very visible changes, especially in the field of communication [1]. The number of VoIP users is currently getting bigger because it has become a basic need. Due to the increasing number of VoIP needs, independent businesses have been created in the field of VoIP service providers.

In the VoIP service provider business, the manager makes every effort to provide optimal service. Therefore, software and hardware are needed that can fulfill these services. VoIP can be used on any type of network, such as LAN, MAN, WAN and WLAN. For VoIP implementation, it can be varied by using a specific OS, such as Elastix, RaspPBX, FusionPBX, Issabel, etc.

Looking at the basis of the problem, the author will discuss the Implementation of a VoIP System Using Issabel as an Integrated IP PBX Server with Telco Vendors. With the implementation of this system, it is expected to make it easier to communicate throughout the region, because this application can be used as a call center, telemarketing, etc. The working principle of VoIP is to convert analog voice into digital data packets, then forwarded through the HUB / router through the internet network and will be received to the destination through the same media. If the user uses telephone media, the principle is almost the same, the telephone media is connected to a phone adapter that is connected to the internet and can be received by the destination telephone [2].

2 Research Method

This research is used to introduce the latest VoIP system using the Issabel server. In its application, Issabel server can create extensions, resource monitoring and call monitoring. The stages of this research are as follows:

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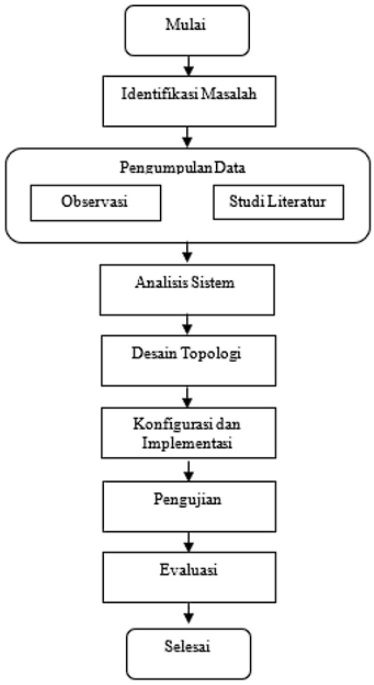

Figure 1. Flow of Research Stages

The initial stage is problem identification. The problem found based on observations made by researchers at several call center companies is the use of telephones that still use local numbers or PSTN. The use of PSTN numbers seen from financing, if making calls to GSM numbers is relatively more expensive. To reduce the cost of outgoing calls to either GSM or PSTN numbers, the implementation of an Issabel Server-based VoIP system integrated with telco vendors will be a solution.

The next stage in this research is determining a solution to solve the problem. In the ease of installation and configuration on the Issabel server, it not only reduces costs but can save time. therefore the solution in this research is the implementation of an Issabel Server-based VoIP system integrated with telco vendors will be a solution.

3 Results

3.1 System and User Interface Preparation

In this research, before installing it is good to create a network topology. The function of creating a network topology is to find out what hardware mapping and specifications will be needed. Here, the network topology that has been created in this study:

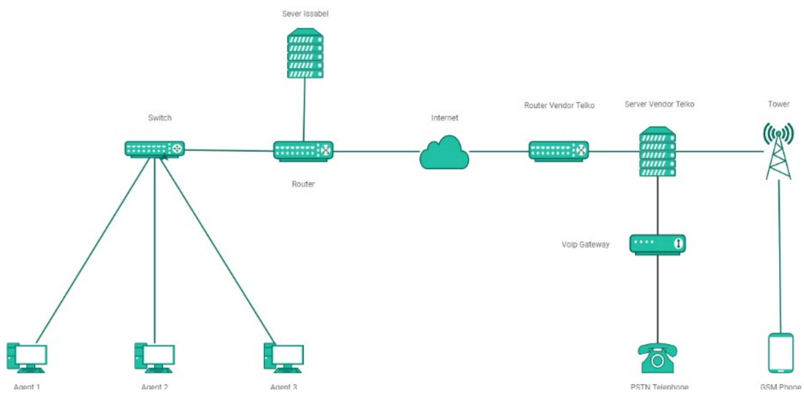

Figure 2. Network Topology

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Next, determine the software and hardware to perform the installation and configuration., software and hardware are needed to fulfill the needs of this research. Here are some of the needs needed during the research:

1. Hardware Specifications

Table 1. Hardware Specification

No	Hardware	Spesifikasi
1	RAM	8 GB DDR4
2	ROM	SSD 512 GB
3	Prosesor	AMD Ryzen 5 5300U
4	VGA	AMD Radeon Graphics
5	Wireless	11AC, 2x2 + BT5.0
6	Flash Drive	Sandisk Cruzer Force 64GB

2. Software Specifications

Table 2. Software specification

No	Needs	Software
1	Issabel and CentOS Installation	VirtualBox
2	Issabel Web Acces	Chrome
3	*Softphone*	Microsip
4	*ISO Image*	Issabel4-USB-DVD-x86_64-20200102.iso
5	*Router Configuration Software*	Winbox ver.3.40

If you have met the software and hardware needs, the next step is to install and configure. To be able to use the VoIP system, the configuration and installation of the server is the main part in order to make calls. In this stage, not only the configuration of storage, memory and network, but in this research requires the integration of Trunk Parameters between the client and the telco vendor.

The configuration and implementation stages are the most important steps in testing the VoIP system using the Issabel server. In the configuration of the Issabel server, trunk parameters are required to integrate with the telco vendor's server. If there are no trunk parameters required in this configuration, then outgoing calls and incoming calls cannot be made from the Issabel server.

In addition to the configuration of the Issabel server, at this stage it is also necessary to configure the Mikrotik router to support the smooth running of the tests in this study. The configured router is used as a medium to distribute the internet network and connect the user agent computer to the Issabel server so that it can be used to make outgoing calls and receive incoming calls.

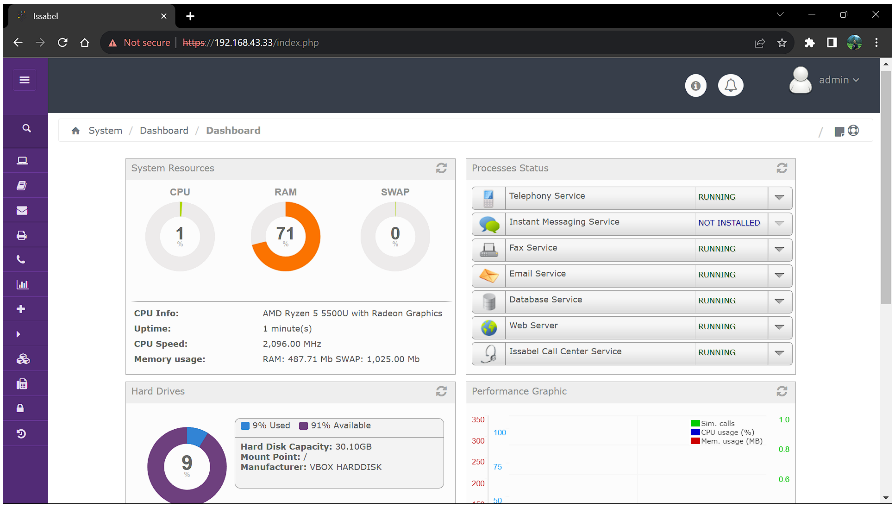

Figure 3. Issabel Dashboard

If the server is already installed, then the next configuration on the issabel server is to be able to make outgoing calls and incoming calls. If the server is already installed, then the next configuration on the issabel

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server is to be able to make outgoing calls and incoming calls. First, configure the trunk with parameters that has been provided by the telco vendor to integrate the Issabel server and the telco vendor server.

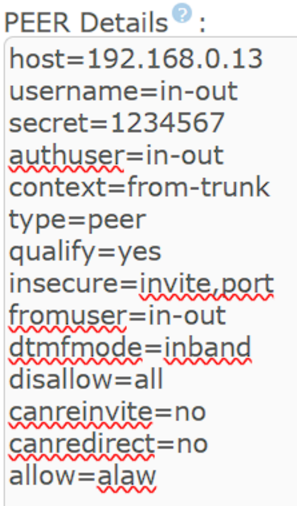

Figure 4. Trunk Configuration

Trunk parameters are registration identities provided from telco vendors that are applied to the Issabel server in order to get VoIP services.

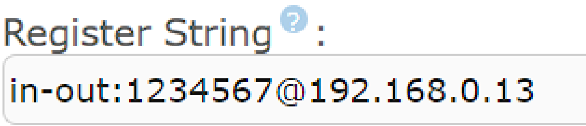

Figure 5. Register String in Issabel

If the trunk parameters have been filled in, then fill in the register string column. The register string is the trunk registration format provided by the telco vendor to integrate the VoIP system from the Issabel server to the telco vendor server. The registration format required for the Issabel server to connect to the telco vendor server where the composition of the filling format is "**username:secret@host**". If the register string field is not filled in, the Issabel server cannot communicate with the telco vendor server and cannot make outgoing calls and receive incoming calls.

Next step is the configuration of outbound routes. This step is a configuration that determines the outgoing call via the trunk to be used. In filling it out, it should be noted because if there are several trunks that have been configured whether to use dial patterns or not. Please note, the prefix is a call prefix or call code that is applied to the VoIP system so that it is easy to separate trunks.

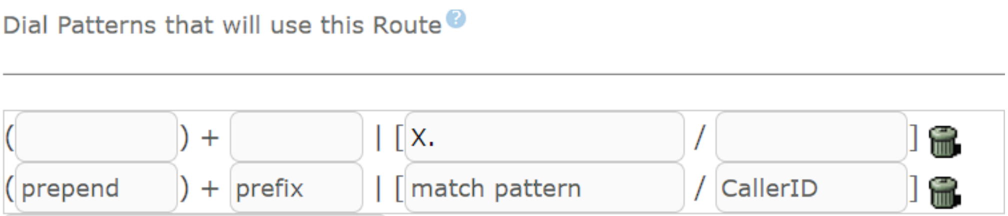

Figure 6. Dial patterns

Figure 6 is the general configuration applied in this study. The "**X**" sign is a rules symbol that includes numbers 0-9 to make calls and the "**.**" is a sign to forward the destination number to be called. If the dial patterns have been filled in, then next determine the trunk that will be used for outgoing calls. The trunk sequence for matched routes column needs to be filled in because if it is left blank, the outgoing call configuration cannot be saved.

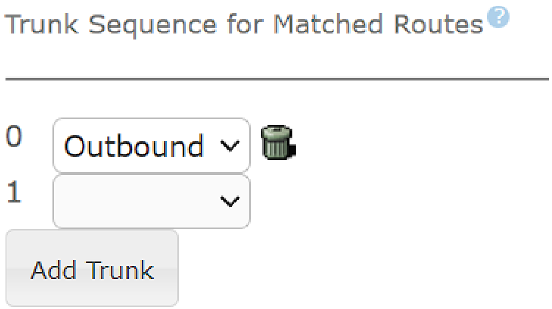

Figure 7. Trunk Sequence for Matched Routes

Figure 7 shows that users can use several pre-configured trunks. The configuration here functions for redundant or backup if at any time the trunk used is constrained (disconnected) then, it will be diverted to a trunk that can be used.

If the trunk and outbound routes have been configured, the next step is to add extensions. Extensions are register formats created on the Issabel server to connect the user agent PC that uses the MicroSIP softphone to the Issabel server so that it can make outgoing calls and receive outgoing calls.

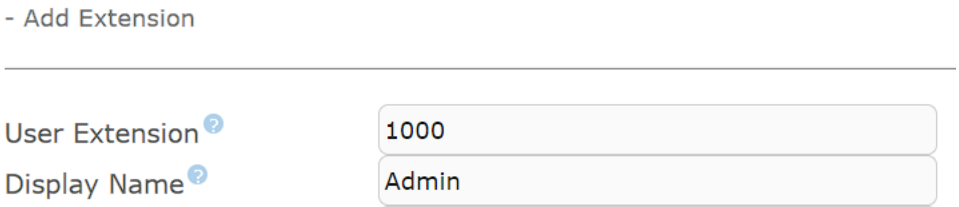

Figure 8. Configuration of User Extension and Display Name

In extensions, users can assign extensions numbers that can be called from other extensions on the same network and can be assigned a name or division to each extension. In Figure 8, the user can provide an extension number and display name to provide an identity that will be distributed to the user agent. Extension function is a numbering in the Issabel server system that can be called from other extension numbers. The display name function is the naming of each extension in order to know the extension number used from the division or user.

The extension configuration also has a password to be able to register on the MicroSIP softphone. Its function is so that each extension user can provide security in making calls and keep it from being confused with other extensions.

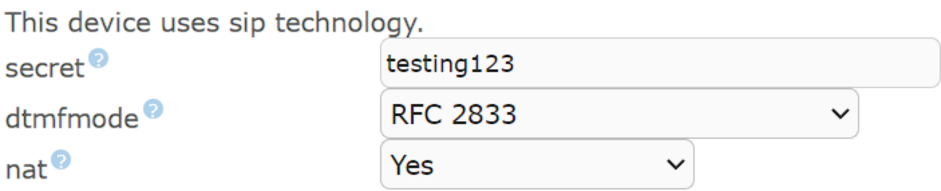

Figure 9. Configurating password of extension

Figure 9 shows the password extension configuration that can be filled with numbers, letters and symbols. It is recommended to use all three elements in filling in the extension password so that security is more guaranteed. In dtmfmode, it can be filled with "RFC2833" as a signaling mode commonly used in VoIP systems. Then at nat, it can be filled with "Yes" so that outgoing calls and incoming calls can be made.

The next configuration is inbound routes. Inbound routes is a configuration to direct incoming calls to extensions in charge of receiving calls. Note that this configuration applies only to trunks that already support inbound call services.

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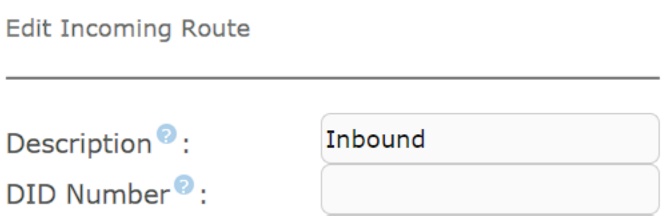

Figure 10. Configuring Inbound Routes

The edit incoming route shown in Figure 10 displays the naming or description given to inbound routes. This is the basic naming in the incoming call configuration. In the DID number, it can be filled or left blank depending on usage. The DID number is the number that can be dialed from the customer or consumer (GSM or PSTN). In this configuration, it can be directed to extensions that are in charge of receiving outside calls.

Figure 11. Configuring Incoming Calls to Extension

In this section, it can be directed to extensions whose job is to receive calls from outside either from GSM or PSTN. This configuration is of course important to direct calls to extensions, if not directed then incoming calls cannot be received from any extensions.

The last configuration is the input of ID extensions that have been created in the Issabel server. To make it easier for user agents to make calls, configure ID extensions on MicroSIP to be registered on the Issabel server.

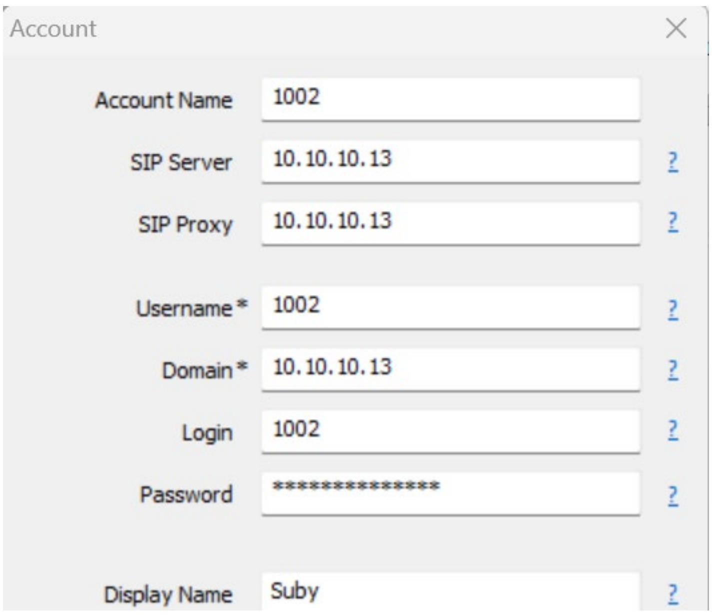

Figure 12. Configuring Extension ID in MicroSIP

The configuration in Figure 12 is fundamental to connecting to the Issabel server. This configuration requires an extension number, server IP, display name and extension password which will be filled in the fields available in the account settings configuration.

3.2 Results of Research

In this research testing will be carried out several two different schemes. First, testing outgoing calls from the user agent to GSM and PSTN destination numbers. Then the second scheme, testing incoming calls from GSM and PSTN numbers to the destination number of the user agent (call center).

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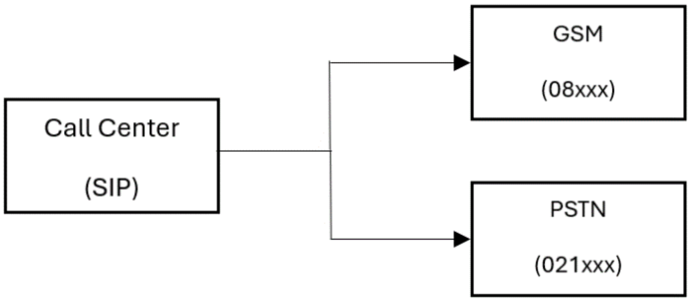

Figure 13. Outgoing call Simulation

Figure 13 shows the outgoing call scheme that will be carried out in this test, this simulation starts from the call center making a call using MicroSIP which has registered to the Issabel server, then entering the GSM (08xxx) or PSTN (021xxx) destination number to be addressed until connected.

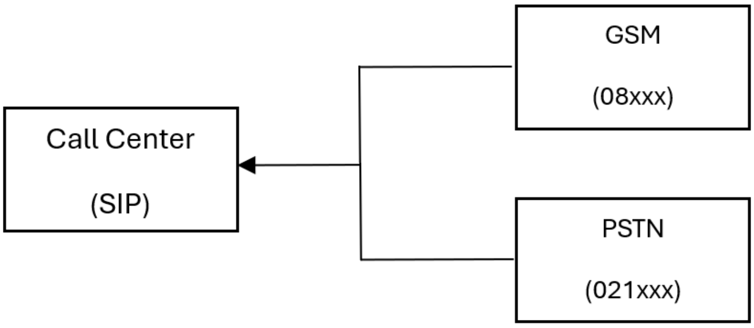

Figure 14. Incoming call simulation

Figure *14* shows the scheme of incoming calls from GSM (08xxx) or PSTN (021xxx) to the call center. This simulation starts from a GSM mobile phone or PSTN telephone set making a call to the intended call center number until it is connected to the user extension on MicroSIP which has been registered on the Issabel server.

From the testing scheme that has been described in the previous section, this section will discuss the test results and tables of the amount of delay, jitter and packetloss in each outgoing call and incoming call. This discussion will be very helpful to see the practice and accurate evidence of each test that has been carried out in this study.

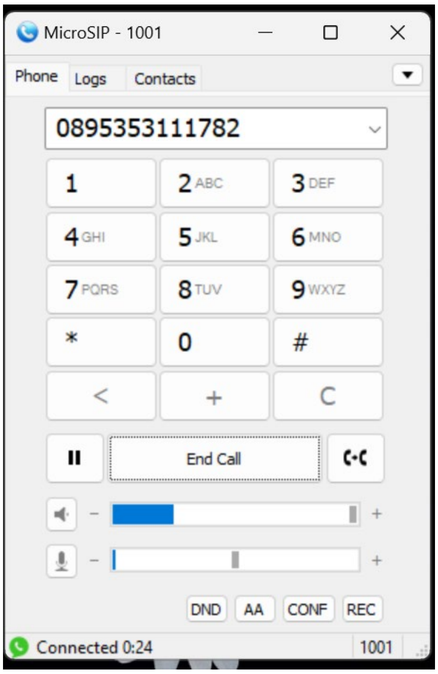

Figure 15. Outgoing Call Test to GSM Number

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In Figure 15 the outgoing call test is made to the GSM sample number (08xxx). This outgoing call is made on the MicroSIP softphone that has registered using the ID that has been created on the Issabel server.

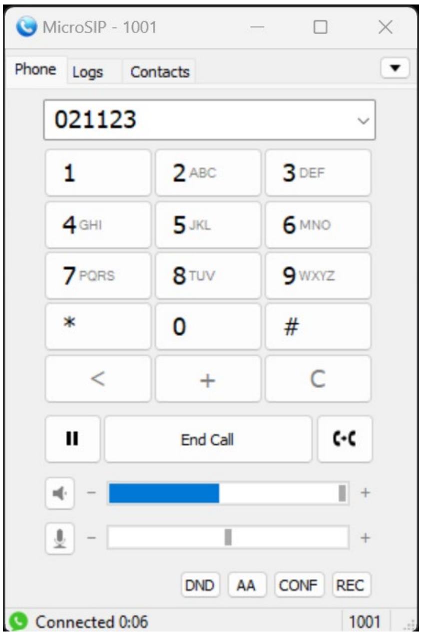

Figure 16. Outgoing Call Test to PSTN Number

Then in Figure 16 the outgoing call test is made to the PSTN call number (021xxx) from the account that has registered with MicroSIP.

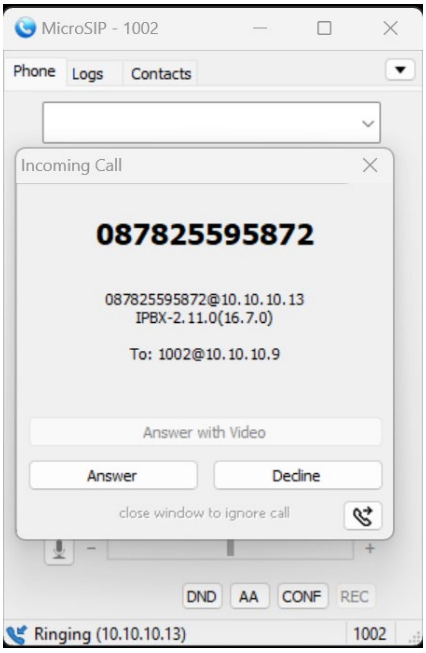

Figure 17. Incoming Call Test from GSM Number

Furthermore, in Figure 17 testing incoming calls made from a GSM number to a call center number (SIP Issabel) using MicroSIP softphones.

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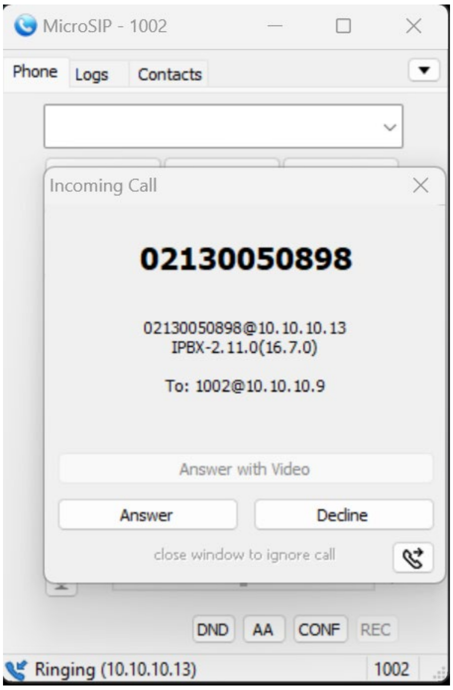

Figure 18. Incoming Call Test from PSTN Number

The last, in Figure 18 an incoming call test was conducted from 021xxx (PSTN) to Issabel's SIP number.

4 Conclusion

The conclusion in this study is that the issabel server can be used as an alternative telephone system that can be used to make PSTN and GSM calls, because it is easy to use and implement. With supporting hardware and software that is qualified to support telephone activities in call center companies, it can help save costs and shorten installation time.

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