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Implementation of a Contact Center with Interactive Voice Response Using Cisco Unified Communications

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ABSTRACT

A popular voice-based communication service used to receive calls automatically is Interactive Voice Response (IVR). Generally, IVR is widely used by institutions to provide services to the public in consulting, offering, or providing information about a particular matter. One of the uses of IVR, among others, is the Contact Center. By using Voice over Internet Protocol (VoIP) technology and using the SIP Trunk service from the service provider, institutions can take advantage of the existing network infrastructure to save on telephone usage. VoIP technology also allows it to be integrated with Private Automatic Branch Exchange (PABX) using Cisco Unified Communications Manager (CUCM). While the IVR system uses Cisco Unified Contact Center Express (UCCX). In this study, the implemented contact center serves to receive calls from external parties. The telephone number uses a call hunting system that will distribute incoming calls to officers on duty at PT. Multipolar Technology Tbk.

1. Introduction

In an era of increasingly developing use of information technology, communication is one of the keys that contributes to maintaining the continuity of business processes and daily life. With the many communication media that exist today such as telephone (PSTN), cell phones, social media, electronic mail (email), instant messaging (Instant Messaging), video calls, etc., of course, communication media that are direct or real-time that the user will choose when in an emergency or requiring a quick response, for example via telephone.

Therefore, telephone service is an important need that must be provided by companies, which in this case are companies that provide services and services to provide excellent and responsive service. For this reason, a contact center is needed that can serve questions, reports, or complaints from customers via telephone lines.

So that this telephone-based contact center service can provide added value and improve customer service and customer experience in the interaction process, IVR (Interactive Voice Response) technology is used in the telephone reception process. In general, the IVR system is used by companies, banks, or government agencies to provide services to the general public in terms of consultation, accommodating complaints, or providing information regarding a particular matter [1].

One IVR system can be installed on an IP network on all Cisco-approved virtual servers [2]. Using a VoIP system provides many benefits for businesses that incorporate a VoIP system into their communications system [3] The VoIP system makes voice call and video call communications cheaper and even free [4]. This telephone technology converts voice into digital code via a data packet network, not analog circuits like ordinary telephones [5].

The working principle of VoIP is to convert analog voice into digital data packets, then the packets are forwarded via the Hub/Router, sent via the internet network and will be received by the destination via the same media [6]. Voice packets are sent to their destinations within countless individual networks on the Internet. Thereby eliminating toll charges, which is why they are cheaper than calls over the PSTN [7]. To build a VoIP-based communication network, you need a set of servers, IP Phone/softphone, router and LAN or internet network connection [8]. IP Phone is a telephone device that has an RJ45 port to connect directly to a network device to carry out or connect a VOIP network [9].

The IVR (Interactive Voice Response) system is a system that can be used to receive and answer calls automatically [10]. IVR works by using algorithms and voice recognition technology to collect user information [11].

In this system, customers can make calls that will be connected to the system by pressing a certain extension number and will be connected to the IVR system which will guide customers in choosing the service they want.

By using virtualization, CUCM and UCCX can be installed on the same server. Virtualization can cut the costs of providing infrastructure and operations independently for each service to be served [12].

Cisco UCCX is used to improve productivity and customer experience. It provides a highly secure, available, virtual, and advanced customer interaction management solution for up to 400 agents [13]. Meanwhile, CUCM acts as a PABX which allows interconnection between conventional telephone lines and VoIP [14].

2. Method

2.1 Data Collection Methods

Data and information collection is carried out through the following stages:

1. Interview

   Interviews are conducted with related parties or stakeholders involving the management structure down to the operator or team on duty in order to obtain complete information on needs.

2. Field Survey or Observation

   Conduct a survey of field conditions to get a clear picture and understand the ongoing processes. The information obtained includes the number of users, hardware and software used, network configuration, and so on.

3. Literature Study

   Look for information from reading user manuals, product documentation, documentation from similar work that has been done before and from other sources that provide knowledge related to implementation.

2.2 Development Method

The development method used uses the NDLC (Network Development Life Cycle) method. NDLC is a method that uses experience or development processes that have been carried out previously such as business strategy planning, application development life cycle, and data distribution analysis.

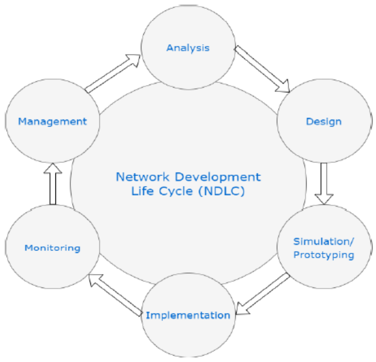

Figure 1. NDLC Development Method

The stages of activities carried out by NDLC include:

1. Analysis Stage (Analysis)

The first stage carried out at the analysis stage is analyzing system requirements, analyzing existing problems, analyzing user needs and desires, and analyzing the current network topology to create a design.

2. Design Stage (Design)

From the data and information that has been collected from the analysis stage, the next stage is carried out, namely making a design of the topology drawing to be built, designing a flowchart, and so on.

3. Simulation Stage (Simulation/Prototyping)

At this stage, simulations and testing are carried out which aim to ensure that the design that has been created runs according to design, and is carried out using user testing and a different path from the one that is running. The results of the simulation will determine whether the design can be implemented or requires revision and improvisation.

4. Implementation Stage (Implementation)

After going through the simulation stage, the design created can be continued to the next stage, namely implementation. In this stage, the design that has been planned and designed previously is implemented.

5. Monitoring Stage (Monitoring)

Monitoring is carried out to ensure that the implemented design runs well within a certain period of time in accordance with initial needs and objectives. After passing a certain period of time, the process can proceed to the next stage.

6. Management Stage (Management)

The management stage is related to institutional policies. At this stage, the design that has been implemented depends on decisions at the institutional management level, whether it will remain in use or be changed according to the institution's future needs or business strategy.

2.3 Research Object

PT. Multipolar Technology Tbk (MLPT) is a leading System Integrator company in Indonesia with a track record of excellence since 1975 providing goods and services in the financial and banking, public, telecommunications and commercial sectors.

MLPT provides the best comprehensive services in the field of Information Technology, which include Hardware and Implementation and maintenance Services, Application Systems and Integration Services, IT Consulting Services, and Business Process Managed Services on a national scale through its subsidiary, PT. Visionet International (VisioNet).

2.4 Running System Analysis

The running system management explains the structure, topology and conditions of the current processes, so that needs analysis and solutions to problems or goals to be achieved can be carried out.

The current flow of the IVR system is intended for the reception team which has the following flowchart:

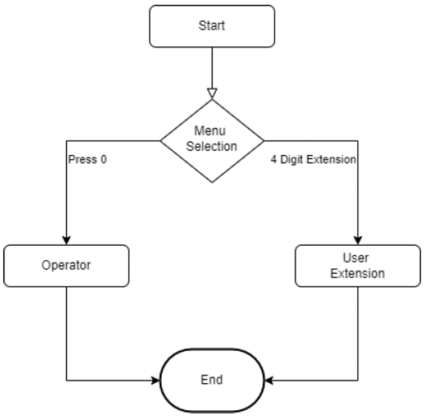

Figure 2. Receptionist IVR flowchart

Then the topology of the currently running VoIP network can be described as follows:

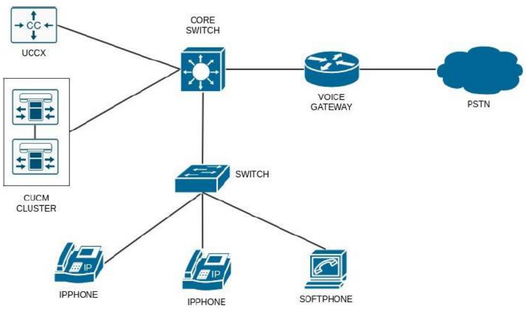

Figure 3. Running Network Topology

3. Result and Discussion

3.1 Proposed Network

From collecting information, we obtain a list of available telephone numbers that can be used as hunting telephone numbers to receive calls

Based on discussions with stakeholders, the telephone number 55777070 installed on voice-port 1/1/1 was chosen as the hunting number for reporting technical problems, with the IVR extension 897070.

As a telephone recipient, you also get a list of extension numbers that will be used by each team member on duty.

For IVR TAC, a new design and implementation was carried out due to the need to provide technical support to customers which was supported by a different team, namely the TAC team.

3.2 Network Topology

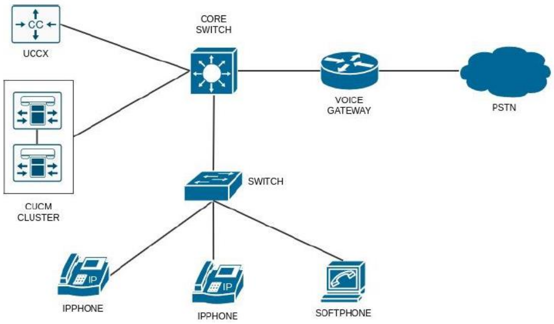

Figure 4. Network Topology

Based on the needs analysis carried out, there is no need for direct topology changes and addition/removal of devices so that the network topology uses the existing one.

3.3 IVR System Scheme

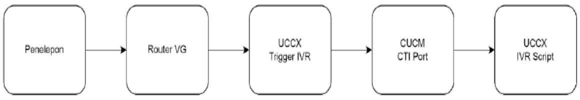

Figure 5. IVR System Scheme

The scheme of the IVR system is:

1. The caller makes a call to the IVR telephone number.
2. The VG router receives and forwards the call to the UCCX.
3. UCCX looks for the appropriate CTI port and sends a redirect request to CUCM according to the CTI port specified for the IVR number.
4. CUCM provides port CTI info and the call is received by UCCX on that port CTI.
5. UCCX starts running the IVR script and handles the call according to the script.

3.4 Flowchart

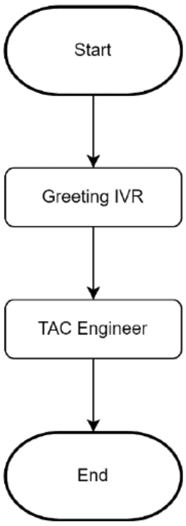

Figure 6. IVR TAC flowchart

The flowchart above illustrates the flow of receiving incoming calls. The caller will receive an IVR greeting automatically before the call will be forwarded to the TAC engineer to fulfill their needs and after that the call can be ended.

3.5 IVR configuration

Add configuration to the Voice Gateway (VG) router as figure 7 below.

```
voice-port 1/1/1
 trunk-group PSTN-Telkom
 input gain 3
 output attenuation -3
 echo-cancel coverage 32
 cptone ID
 timeouts interdigit 0
 timeouts call-disconnect 5
 timeouts wait-release 5
 connection plar 897070
 description 55777070
 station-id name MLPT-CO-111
```

Figure 7. VG Router Configuration

Create a CTI Route Point as an extension number that will trigger IVR on CUCM as figure 8 below.

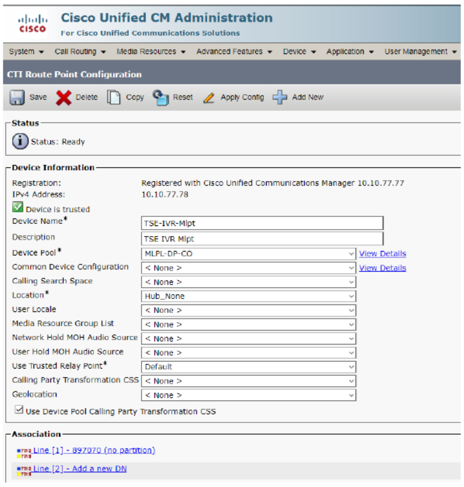

Figure 8. CTI Route Point display

Create and upload recording files for greetings and other prompts required in UCCX as figure 9 below.

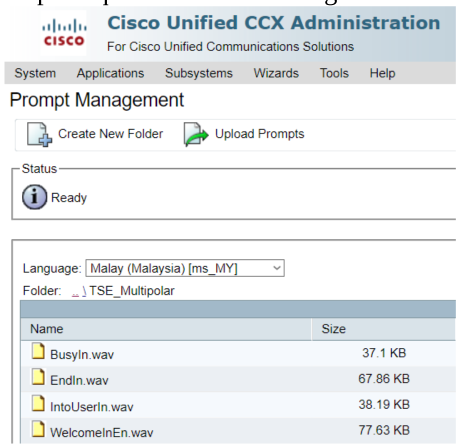

Figure 9. Prompt Management display

Create a Call Control Group on UCCX as figure 10 below.

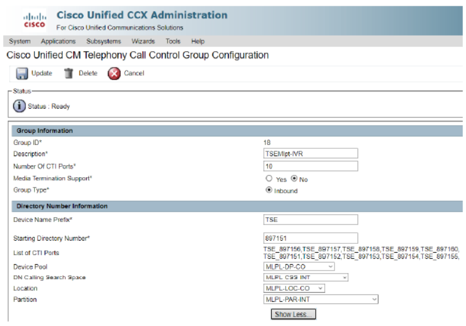

Figure 10. Call Control Group display

Create and upload application scripts from the IVR system on UCCX as figure 11 below.

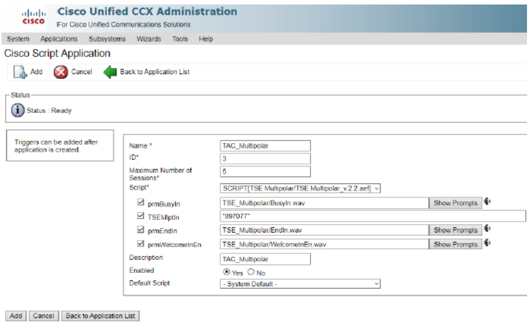

Figure 11. Cisco Script Application display

Script created using the Cisco Unified CCX Editor application as figure 12 below.

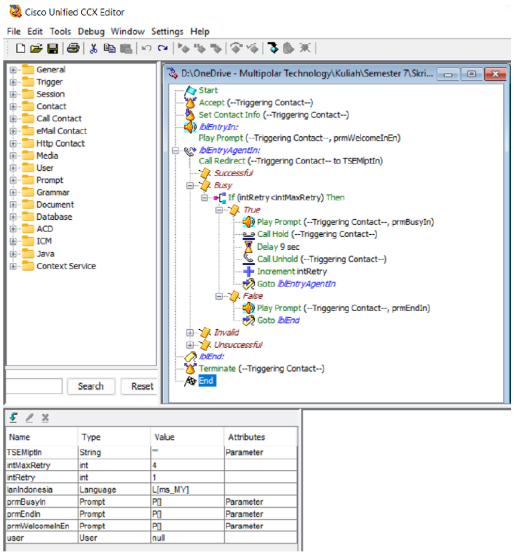

Figure 12. Cisco Unified CCX Editor display

3.6 Testing

The author carried out tests in order to ensure whether the design being built was as desired and could run successfully.

1. Prepare softphones that have extensions that are included in the TAC team members.

Figure 13. Softphone display

2. Check the IP address obtained by the softphone on CUCM

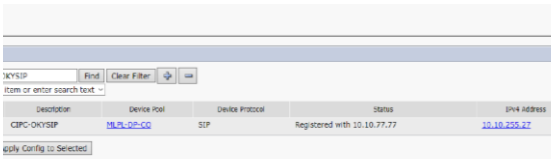

Figure 14. Softphone IP Address

3. Open the Wireshark application then enter the IP address of the softphone in the search field "ip.addr == 10.10.255.27

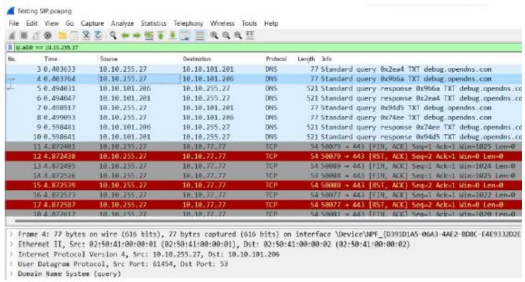

Figure 15. Wireshark display

4. Make a call via mobile (GSM) to the hunting telephone number [15].

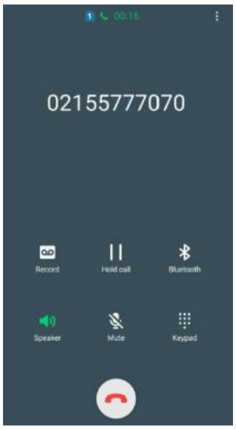

Figure 16. Calls from mobile phones

5. After the ringtone is heard, a greeting is heard and the call is forwarded to the TAC team. At this stage, a telephone speaking test is carried out.

Figure 17. Sofphone Receiving Calls

6. After the call ends, open the VoIP Calls log from Wireshark and click play streams. Here you can also listen to the results of the telephone speaking test.

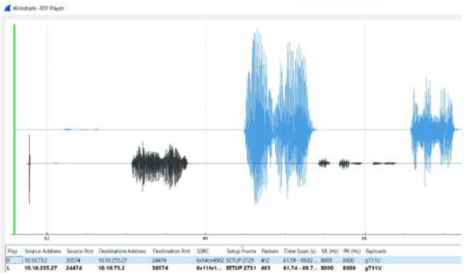

Figure 18. VoIP Calls Wireshark

7. Check the status of the voice port on the voice gateway router. On the voice port used, the status changes to up and off-hook, which indicates that a call is taking place on that voice port.

```
CO-VG-1#sh voice port summary
                                              IN       OUT
PORT            CH  SIG-TYPE    ADMIN OPER STATUS   STATUS   EC
=============== == ============ ===== ==== ======== ======== ==
0/0/0           --  fxo-ls      up    dorm idle     on-hook  y
0/0/1           --  fxo-ls      up    dorm idle     on-hook  y
0/0/2           --  fxo-ls      up    dorm idle     on-hook  y
0/0/3           --  fxo-ls      up    dorm idle     on-hook  y
0/1/0           --  fxo-ls      up    dorm idle     on-hook  y
0/1/1           --  fxo-ls      up    dorm idle     on-hook  y
0/1/2           --  fxo-ls      up    dorm idle     on-hook  y
0/1/3           --  fxo-ls      up    dorm idle     on-hook  y
0/2/0           --  fxo-ls      up    dorm idle     on-hook  y
0/2/1           --  fxo-ls      up    dorm idle     on-hook  y
0/2/2           --  fxo-ls      up    dorm idle     on-hook  y
0/2/3           --  fxo-ls      up    dorm idle     on-hook  y
0/3/0           --  fxo-ls      up    dorm idle     on-hook  y
0/3/1           --  fxo-ls      up    dorm idle     on-hook  y
0/3/2           --  fxo-ls      up    dorm idle     on-hook  y
0/3/3           --  fxo-ls      up    dorm idle     on-hook  y
1/0/0           --  fxo-ls      up    dorm idle     on-hook  y
1/0/1           --  fxo-ls      up    dorm idle     on-hook  y
1/0/2           --  fxo-ls      up    dorm idle     on-hook  y
1/0/3           --  fxo-ls      up    dorm idle     on-hook  y
1/1/0           --  fxo-ls      up    dorm idle     on-hook  y
1/1/1           --  fxo-ls      up    up   idle     off-hook y
1/1/2           --  fxo-ls      up    dorm idle     on-hook  y
1/1/3           --  fxo-ls      up    dorm idle     on-hook  y
2/0/0           --  fxo-ls      up    dorm idle     on-hook  y
2/0/1           --  fxo-ls      up    dorm idle     on-hook  y
2/0/2           --  fxo-ls      up    dorm idle     on-hook  y
2/0/3           --  fxo-ls      up    dorm idle     on-hook  y
2/1/0           --  fxo-ls      up    dorm idle     on-hook  y
2/1/1           --  fxo-ls      up    dorm idle     on-hook  y
2/1/2           --  fxo-ls      up    dorm idle     on-hook  y
2/1/3           --  fxo-ls      up    dorm idle     on-hook  y
```

Figure 19. Voice Port Status

8. Testing was also carried out on the distribution algorithm used, namely broadcast, which makes all extensions ring if there is an incoming call via IVR. Testing was carried out using 2 IP phones containing 3 extensions of TAC team members. The result is that all IP phones ring when a call is made to the IVR telephone number.

Figure 20. IP Phone Receiving a Call

4. Conclusions And Recommendations

4.1 Conclusions

Based on the results of the analysis, design and discussion presented previously, the following conclusions can be drawn:

1. To configure IVR for PT's needs. Multipolar Technology Tbk, namely by making changes and/or additional configurations to the Voice Gateway Router, Virtual Machine Cisco Unified Communication Manager (CUCM) and Cisco Unified Contact Center Express (UCCX).
2. Implementation is carried out by adding or changing the configuration as required on the Voice Gateway, CUCM and UCCX Routers.
3. The flow design is made using symbols that can show the steps and decisions regarding the process being carried out, adjusted to the needs and elicitations that have been carried out.

4.2 Suggestions

From the results of this analysis and research, several suggestions can be added for developing the IVR design, namely:

1. The IVR configuration can be made in more detail regarding the IVR system or flow, such as greetings adjusted to the time, for example good morning, good afternoon, and so on.
2. The language used in the service can also be added, for example using English.
3. The analog telephone used cannot receive telephone calls for more than one session so it is recommended to migrate from an analog telephone to a VoIP (SIP Trunk) based telephone.
4. The IVR system can be supplemented with a voice recording system which records conversations between customers and the team on duty to provide evidence if something unusual happens and requires checking the conversation recording.
5. In the old IVR system, you can add an automatic telephone answering configuration to answer incoming telephone calls outside operational hours that the company is not operating.

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