

Efficient Response Solution for Integrated Command and Control Center Using Automatic Interactive Voice Response System

Daniel GÎRBEA and Petrică CIOTÎRNAE

Abstract—Current communication networks are constantly expanding, the services offered are diversifying and the number of special users is increasing. These facts lead to an increasing number of special requests from military user applications and implicitly to a long waiting time in the calling queue. The purpose of this paper is to design an Automatic Call-Center integrated with an Interactive Voice Response (IVR), which improves call routing and increases special users satisfaction through various specific modern capabilities, but also have call monitoring and error reporting options from a management center. The call-center configuration solution has been implemented in accordance with the needs of command and control centers that is constantly developing and takes into account special user requests.

Index Terms—Call-Center, IVR, Features, Special Military User Requests, Command and control center.

I. INTRODUCTION

Nowadays, when technology is developing and many opportunities are available for special users to ask various special requests, they will no longer tolerate a tiring process of waiting on the phone and it is no longer enough for operators to depend on various written reports when serving special users.

A command and control center must have a business strategy that offer opportunities for expansion and increase user satisfaction by interacting with them using various communication channels (e-mail, messages, call-centers) [1-4].

A call-center is an essential part of any organization that deals with requests/calls from users. An integrated call-center reduces call waiting time and improves call routing. The end result means satisfied users who solve their problems. Because the costs of such a system are not very high, most organizations that want to improve relationships with their users certainly need to have a call-center implemented [5-7].

A call-center is a specific case of using an inventory management method, namely FIFO (First In, First Out), which assumes that the first to enter the system is also the first to be served. The basic technologies of a call-center are: a telephone exchange (PBX – Private Branch Exchange) defined by an ACD (Automatic Call Distributor), a CTI

system (Computer Telephony Integration), IVR Server (Interactive Voice Response), Mail Server, Database Server, Human Agents with a desktop and an IP phone [8-10].

II. PROPOSED METHODOLOGY

The paper proposes the design of a call-center, also presenting the chosen technical configuration solution that ensure maximum operating performance through various specific current features and call monitoring capabilities.

The design of the technical solution for a call-center was chosen so that the implemented system is divided into three separate subsystems: the call-center itself, the management center and the special user's network. Fig. 1 shows the interconnection scheme of the three centers, which will receive IPs from the subnet 192.168.40.0/24.

The call-center consists of an open-source soft telephone exchange (Freeswitch 1.10) installed on a Debian 10 Virtual Machine (Buster) with the IP address: 192.168.40.135/24, which interconnects several soft telephones (softphones) installed on a Windows 10 Virtual Machine with the IP address 192.168.40.137/24. This Virtual Machine hosts 10 Blink phones for operators 1002-1011 and X-Lite phone for 1012.

The management center is a Virtual Machine with Debian (192.168.40.136/24) through which the network manager will manage the entire system, having access to all the equipment.

He will also have a tSIP softphone with number 1001 on it, through which he will be able to contact any operator within the command and control center or even all of them at once through a conference. This softphone also offers monitoring calls in the call-center. For advanced call control, the network manager can easily access the Freeswitch control panel via SSH.

The management center also contains the real-time monitoring application The Dude, which is open-source and offers the ability to make a network map.

Also here runs a Postfix mail server managed by the ISPConfig 3 system. It has been used to create various accounts, that can be accessed from the Round cube mail client, so that system managers the received by email messages registered by special users in call-center voicemail.

The network manager also has access to a call-center monitoring web page, initialized by running a script in Python, which uses the Flask architecture to develop a web application and the “mod_xml_rpc” module in Freeswitch to interconnect the WebAPI with the console FS.

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D. Gîrbea is with “Ferdinand I” Military Technical Academy, Communications Department, Bucharest, Romania.

P. Ciotirnae is with “Ferdinand I” Military Technical Academy, Communications Department, Bucharest, Romania (e-mail: petrica.ciotirnae@mta.ro).

The special users network includes two analog telephones and an ECV (Voice Switching Equipment), that is a hard telephone exchange from Topex, used as a gateway that stands for the interface between the local special users network and the public network, by converting the TDM (Time Division Multiplexing) in IP (Internet Protocol).

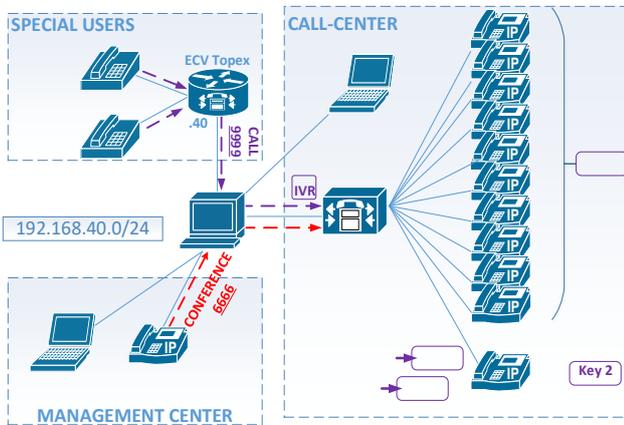


Figure 1. Interconnection scheme

The steps for setting up the call-center are as follows: Debian 10 virtual machine installation, Freeswitch installation, dialplan configuration, modules configuration, users configuration, Windows 10 Virtual Machine installation and softphones registration.

The core of the call-center consists of a default configuration of a stable, flexible and scalable *Freeswitch* system, with all the necessary basic configurations. Freeswitch 1.10 was chosen, because it is considered the most stable version at the moment.

Both the dialplan and the rest of the Freeswitch files are written in XML (Extensible Markup Language) and are stored in the `/etc/freeswitch` directory. The dialplan is the “heart” of the Freeswitch configuration. Depending on its characteristics, an incoming call will match a specific “*context*” in the dialplan. In order to make calls, a careful configuration of the dialplan and then the registration of subscribers are necessary.

The dialplan of this system consists of the following separate contexts:

- *default.xml* - basic context used for calls between subscribers that are registered in Freeswitch (1001 - 1012);
- *in.xml*, which transfers the call to the *ivr.xml* file - contexts used for incoming calls to the call-center, from special users (2001, 2002);
- *out.xml* - used for outgoing calls from the call-center from any subscriber registered in Freeswitch (1001 - 1012) to special users: 2001 and 2002;
- *conference.xml* - used for conferences between subscribers registered in Freeswitch (1001 - 1012).

In the following lines, some sequences of the extensions used for incoming calls are presented.

The *in.xml* context contains the extension “*cc_call_entry*” and deals with external calls from users (2001 and 2002), which dial “9999” and reach the call center. The users listen to a greeting message asking them to press a number on the telephone keypad. The value of the key will be stored in the variable *val_ivr*, and the call will be transferred to *ivr.xml* file.

```
<condition field="destination_number" expression="^9999$">//the first
condition for the call to go through the extension is that the dialed number
(destination_number) to be 9999
<condition wday="2-6" time-of-day="08:00-16:00" continue="true">//the
second condition is that the call to be made from Monday to Friday
between 08: 00 - 16:00
<action application="read" data="1 1 misc/in.wav val_ivr 10000 #"/>//
play the audio file in.wav, then in the variable val_ivr the digit will be
stored (no digits: min = 1, max = 1), that is pressed on the keypad within 10
seconds
<action application="record_session"
data="\${recordings_dir}/${strftime(%Y-%m-%d-%H-%M-
%S)}_\${destination_number}_\${caller_id_number}.wav"/>// record the
call in the /var/lib/freeswitch/recordings directory
<action application="transfer" data="\${destination_number} XML
ivr"/>//transfer the call to ivr.xml file
<anti-action application="playback" data="misc/program.wav"/>//if the
call is made outside of business hours, the program.wav audio file in which
the call-center program is specified is played: "The call center program is
from 8 to 16. Thank you!"
```

The *ivr.xml* file contains 5 extensions, depending on the user’s selection from the phone keypad. Key 1 will send him to the *ivr 1* extension, key 2 to *ivr 2*, key 3 to *ivr 3*, key 4 to *ivr 4* and key 0 to *ivr 0*.

```
<extension name="ivr 1">//extension name
<condition field="\${val_ivr}" expression="^1$">//variable value
<action application="callcenter" data="cc_atm"/>//send call to queue
cc_atm, where agents 1002 - 1011 are defined in mod_callcentermodule
```

The first extension sends the call to the *cc_atm* queue in the call center module shown below, in which agents 1002 to 1011 are defined.

```
<configuration name="callcenter.conf" description="CallCenter">
<queues>
<queue name="cc_atm">//call center queue called cc_atm
<param name="strategy" value="longest-idle-agent"/>//the rule by which
calls are distributed to agents is the one that sends the call to the agent with
the fewest calls received
...
<agents>
<agent name="1002" type="callback"
contact="[leg_timeout=10]user/1002@192.168.40.135"
status="Available" max-no-answer="3"/>//agent type specifies that he is
always online; if the agent does not answer within 10s, the call goes to
another agent; after 3 missed calls, the agent status changes from
"Available" to "On Break" etc.
... //identical settings for the rest of the agents
```

III. RESULTS

The IVR (Interactive Voice Response) system is the mechanism of interaction with special users.

The implemented IVR system benefits from integrated management and offers various capabilities such as: MOH (Music On Hold), recording calls, CDR (Call Detail Records) messages with call logs etc. Fig. 2 shows the welcome IVR, specific to the call-center implemented for a Command and Control Center.

Each branch of the voice menu reacts to the DTMF signals transmitted by the caller and executes a dialplan application (callcenter, transfer, playback, etc.).

1. *Key 1* transfers the call to one of the ten operators in the General Information Office (1002 - 1011);
2. *Key 2* transfers the call to the operator of the Technical Problems Office (1012);
3. *Key 3* plays an audio message of general interest about the Command and Control Center and its attributions;
4. *Key 4* allows the recording of an audio message with

the details of a Special Request from the Command and Control Center, the file being then sent by e-mail;

5. Key 0 allows listening to the greeting message again.

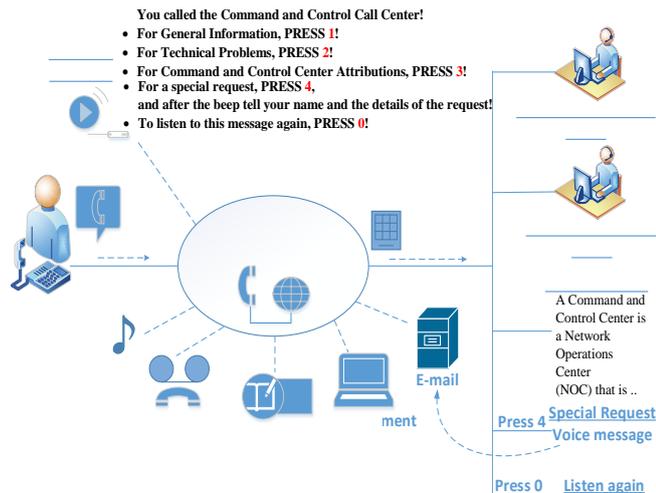


Figure 2. IVR scenario

The first step in the implementation process of the management center is the installation of various servers and applications: Apache web server, Postfix mail server, Roundcube mail client, BIND DNS server and the main server ISPConfig 3, accessible from the WEB Browser, which was used first to create “@atm.ro” Mail Domain and then two mailboxes, accessible from the Roundcube client.

Fig. 3 shows an e-mail with a wav extension recording from the user “2001”, in case of pressing Key 4. Mails are sent from freeswitch@atm.ro to daniel@atm.ro.

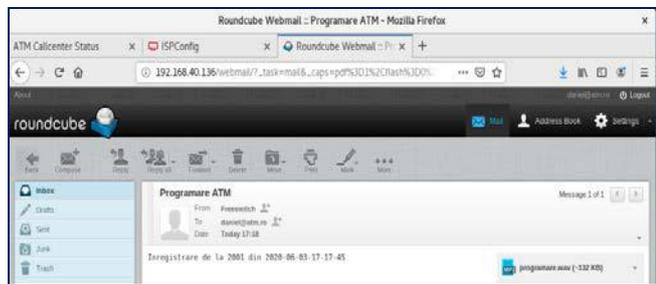


Figure 3. E-mail with a request

To develop a network map and to monitor in real time all the entities, The Dude application will be used.

To monitor calls, the network manager will use several tools. One of these is the fs_cli (Freeswitch command line interface), which provides real-time information about subscribers and calls. Fig. 4 shows the result of the “show registrations” and “show calls” commands, with an internal call between 1012 and 1002.



Figure 4. Fs_cli monitoring

Another call monitoring application is tSIP, used to record the phone number 1001 and to monitor the status of other subscribers by two indicators: BLF (Busy Lamp Field) and Presence. The differences between them are illustrated in Fig. 5 as follows: The BLF shows with whom the monitored number is in conversation and when taking into account the orange color, the BLF shows it when the number is in the “ringing” state and Presence indicates that the phone is unregistered.

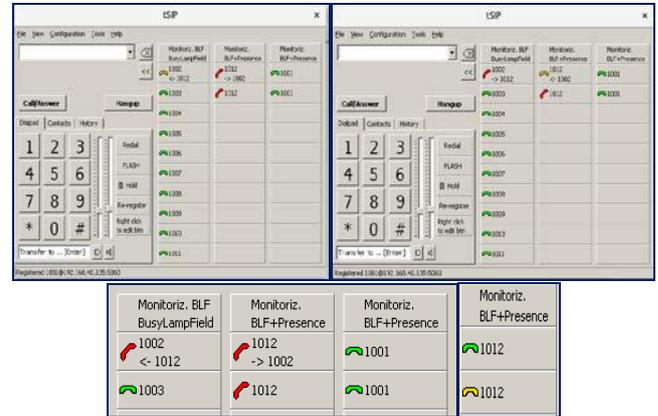


Figure 5. tSIP monitoring

The special users network contains the ECV and two analog telephones. From the Web, the IP was set on the device, then from SSH, a series of configurations were made so that the voice packets reach Freeswitch and back to ECV, without the call being interrupted. The special users (2001 and 2002) were configured from OAM (Operation, Administration and Maintenance), then the routing table where the rule so that they can only call 9999 is created and then the output direction with “Freeswitch” rule. The input direction is 0.0.0.0/0, so the ECV subscribers can receive calls from anyone.

In addition to programming, the OAM application provides real-time traffic tracking. Fig. 6 illustrates an ongoing call described by the “CONNECTED” state, initiated by 2001, but also a call initiated by 1001 to 2002, which has not yet answered, thus being in the “ALERTING” / “RINGING” state.

We can also view the codes according to the Q.850 standard that specifies the causes of error after the end of calls (16 - Normal Call Clearing, 17 – User Busy etc.).



Figure 6. OAM monitoring

In addition to the monitoring methods presented above, the most powerful tool is the web page “Callcenter Status”, which mainly monitors the status of agents. The page is created by running a script implemented in Python, which uses the Flask architecture for developing a web application and the “mod_xml_rpc” module in Freeswitch for interconnecting the WebAPI with the Freeswitch console.

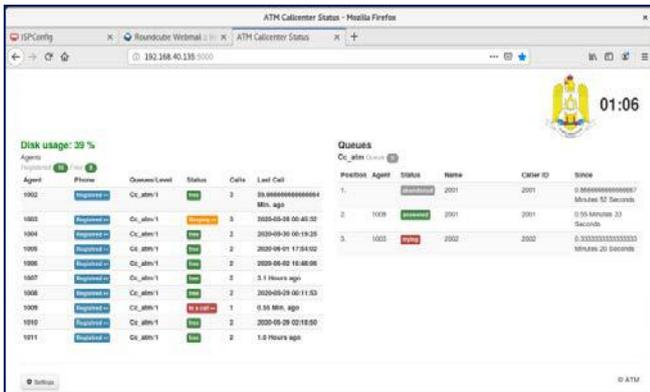


Figure 7. Call-center monitoring

The *conference.xml* context includes two extensions with two types of conferences that can be made by FSW subscribers. *Conference1* can only be performed by the network manager (1001) by calling 6666. This will create a conference between him and Freeswitch and then automatically initiate instant calls to all other operators to invite them to the conference. *Conference2* is done by calling 6665 and is a type of conference that will require a 5-digit PIN. It can be created by any subscriber and can be joined by anyone who knows the PIN.

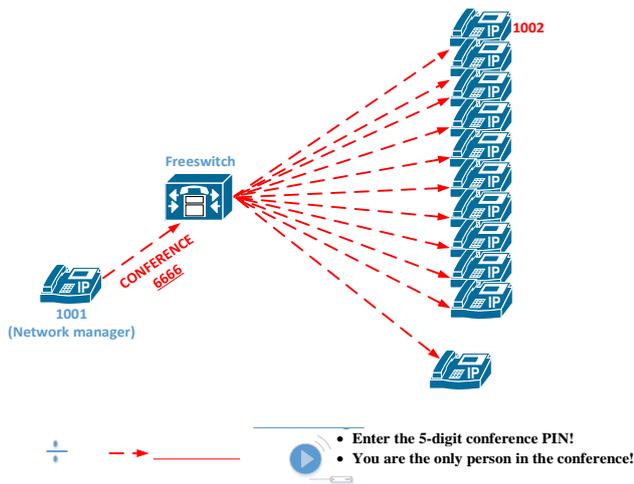


Figure 8. Conferences

IV. CONCLUSION

It can be concluded that the proposed objectives have been successfully met, and the Freeswitch offers advanced capabilities for configuring this type of call-center.

The callcenter will significantly reduce the waiting time for special users in the queue and will increase the quality of calls.

In order for the system to be truly efficient and the waiting time to be almost zero through a quick response to the user's call, call-centers must be integrated with IVR technology, which will group calls on various areas.

The ability to record calls is a necessary feature in a call center for possible misunderstandings or complaints.

When the call center is closed, the IVR provides an audio message presenting the work schedule.

Implementing a server and a mail client is imperative within the command and control center's network, as it helps a lot in quickly reporting call statistics.

The importance of monitoring was also demonstrated, because the network manager has call control in real time.

And finally, considering these times when time presses us a lot and the ability to hear us on the phone is a great bonus, it has been shown that for critical problems, one of the best ways to communicate with center staff in real time is by using an audio conference. Thus, the network manager will provide and receive useful information efficiently from all the departments of the command and control center.

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