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Interactive Response System (IRS) with Text & Call Support in Contact Center Application

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***Abstract*— Now-a-days, a major part of service industry utilizes contact center solution for customer support and troubleshooting purposes. Hence, customers interact with these applications on a day-to –day basis making it a priority for the organizations to provide contact center services in a cost-effective manner. This paper introduces a new system design for interactive response in Contact Center similar to the existing IVR system. The new system was designed to overcome the flaws in the current IVR system. This new system replaces the traditional voice response to text-based response thereby reducing the time to generate the response after comprehending all the options. Unlike in IVR systems where the responses from the customer are collected on call, we propose a web-based request-response system which results into a call (VoIP/PSTN) as per requirement.**

Keywords*— Contact Centre, Interactive Response System (IRS)

I. Introduction

Telephone based call centers are now called as Contact Center, which enable the delivery of services via telephone lines.[1] Call Center technologies allow computers to handle first level of customer support, text mining and natural language processing. In telecommunications, IVR allows customers to interact with a company's host system via a telephone keypad or by speech recognition, after which they can service their own inquiries by following the IVR dialogue. IVR applications can respond with prerecorded or dynamically generated audio to further direct users on how to proceed. IVR applications can be used to control almost any function where the interface can be broken down into a series of simple interactions. IVR applications deployed in the network are sized to handle large call volumes. Historically, IVR solutions have used pre-recorded voice prompts and menus to present information and options to callers, and touch-tone telephone keypad entry, to gather responses. Modern IVR solutions also enable input and responses to be gathered via spoken words with voice recognition.[1]

There are two types of IVR system,

I) Touch Tone IVR

II) Speech Enabled IVR

In current IVR system customer need to hear all the options and then need to give the response (press key if Touch Tone IVR or Speaking an option if Speech enabled IVR). IVR tones are long and customer need long time to hear all the options. The customer has to opt for repeated announcements of the same options, if the call drops or due to hearing problem caused by network coverage issues.

To avoid flaws in current IVR system and to save valuable time of the customer, we need to change contact center system (IVR System) which will provide service fast and comprehensive. If we provide Text based options to customer with voice based tones then time required to read the option is less as compared to hear all the options.

II. Existing System

In existing system, the user initiates the call to the service specific phone number which terminates at the contact center of that service provider (e.g. Bank contact center, sim card service provider number, etc). As shown in figure, the user can connect to the service provider by using either a PSTN network or a VoIP network. The call is established between the user and the service provider instantly via the secure border element (session border controller). At this point, there is no direct call connection between the agent and user.

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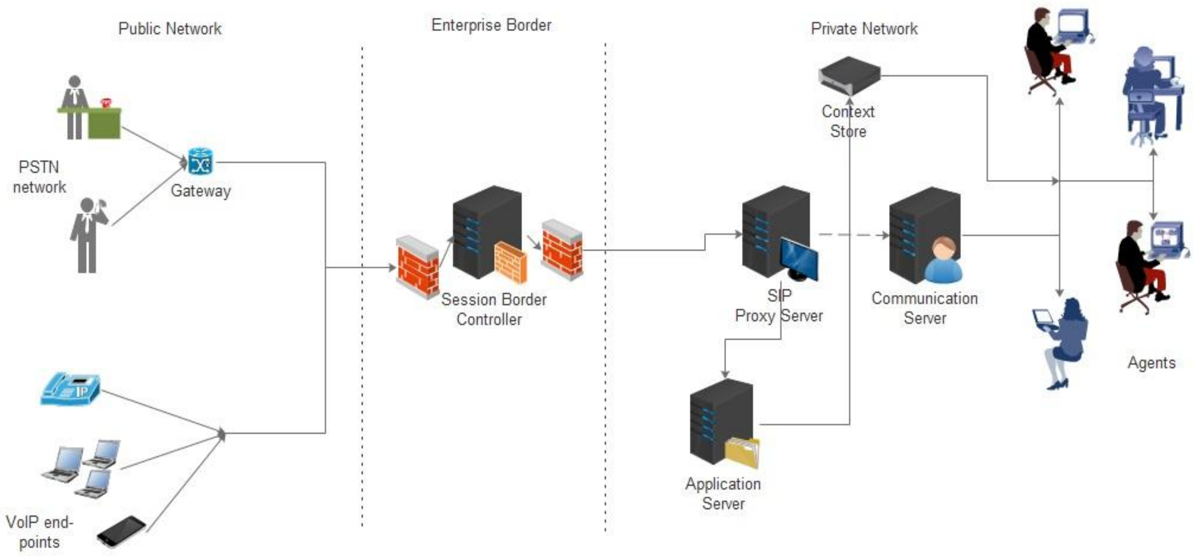

Fig. 1. Architecture of Exisiting System

After the call is initiated by the user, the call request is forwarded by the SBC to the SIP Proxy Server. The proxy server forwards the request to the application server which invokes the application script for executing the written task flow containing IVR execution flow. A complex IVR execution flow follows a hierarchical model which includes multiple simple IVRs connected to each other that are, executed one after another till the call is terminated. The application server passes an IVR to the TTS server, which convert these options into voice and redirects the voice output to the user. On the basis of options present in the current IVR, the user makes a selection via the phone keypad (generating the DTMF tones) which is forwarded to the application server. This request-response flow continues until his/her queries are resolved (resulting in call termination) or the call is routed to the agent (automatic call connection with a contact center agent). If the call is to be routed to the agent, then before connecting the call to the agent the request-response values and user-related information is stored in the context store. All the agents (active) and their details are stored in the communication server. The agent is selected based on the response values (given by user). The communication server forwards the selected agent's detail to the proxy server for successfully routing the user call to the agent. The agent can have access to the user details & responses (for current call) stored in the context store.[2]

In short, if the user chooses; call is routed to agent after a lengthy process of selecting appropriate options for the IVRs. This process of patiently listening to all the options, deciding on an option and then making a selection is quite lengthy and time-consuming. Also, it is fairly important to remember the options with corresponding digits to make the final choice.

At a time, a network can support only a specific number of calls due to bandwidth limitations. Being a time-consuming process, a user has to stay on the call till the end. Hence, the same call utilizes the resources; which could have been dedicated to some other call. Also, for some companies these calls are billed and hence it becomes non-economical to continue this call with a bad network coverage (voice breakage and other issues may render the user to listen to the options again & again to be clear thereby increasing the call time).

III. DEATILED ARCHITECTURE OF PROPOSED SYSTEM

There are various components present in the proposed system design. This segment gives in brief explanation of their role in the network.

a. WebRTC plugin

WebRTC stands for web real-time communications. WebRTC leverages multiple standards and protocols, like STUN/TURN servers, signaling, JSEP, ICE, SIP, SDP, NAT, UDP/TCP, network sockets, and more. WebRTC is primarily used for real-time peer-to-peer audio and video (i.e., multimedia) communications. It enables end users to make secure voice calls from their Web browser. The call destination can be any endpoint to which Session Manager is able to route calls. For example, a customer can initiate a call directly into the contact center from a Web page on a company's public Web site.

b. Session Border Controller

Session border controllers are usually implemented as SIP Back-to-Back User Agents (B2BUA) that is placed between a SIP user agent (end point) and a SIP proxy. A session border controller (SBC) is a dedicated hardware device or software application that governs the manner in which phone calls are initiated, conducted and terminated on a Voice over

Internet Protocol (VoIP) network. Phone calls are referred to as sessions.

c. Application Server

The application server contains a number of scripts running on it as a daemon which can perform the allocated tasks in a continuous manner when the scripts are specifically invoked.

d. Context Store

It provides a centralized, scalable and low latency in-memory data cache for applications to store, retrieve and share contextual information about customer interactions throughout the customer journey and eventually feed that information into database for persistence and analytics. It provides RESTful interfaces and Java SDK for any application to store and retrieve real-time contextual data.

e. EDP Server

Engagement Development Platform provides a virtualized and secure application platform where Java programmers can develop and dynamically deploy advanced collaboration capabilities. Engagement Development Platform acts as the platform for many applications such as WebRTC, Real-Time Speech Snap-in, Context Store and Work Assignment.

f. Work Flow Assignment

Work Assignment is a highly available work distribution system that assigns work to resources across the enterprise. All resources across the enterprise are maintained in a single pool by the work assignment and work is assigned using a single universal matching engine and attributes-driven routing.[5]

g. SIP proxy Server/Session manager

SIP proxy server/Session manager provides the facility to monitor and manage a VoIP network end-to-end for maximum quality of service and availability. It is a SIP routing and integration tool. It enables interconnection across a variety of disparate VoIP networks and legacy systems while providing centralized voice and data management. Session Manager integrates all SIP devices across the entire enterprise network within a company and leverages the existing PBX infrastructure.

h. Communication Manager

A call server/communication manager is a particular form of application server that manages the setup or connection of telephone calls. The call server will receive call setup request messages, determine the status of destination devices, check the authorization of users to originate and/or receive calls, and create and send the necessary messages to process the call requests.

IV. Working of Proposed System

We propose a system, in which the voice response is replaced by web (text-based) pages where the user has to select the options from the menu. This request-response mechanism will be supported by web pages. To connect to an active agent, a call shall be generated by the user end-point (mobile phones, desk-phones, web-call, etc).

A. Contact Center Application

The contact center application specific to the service provider should be pre-installed on the user phone. Using this app, the user can connect to the contact center of the service provider. In the figure, the phone has the contact center application (for exchanging requests & responses), webRTC plugin (for web/IP calls) and pre-loaded mobile calling application (for non-IP calls). [4]

The user places a request using the CC application. This request reaches the secure border element and forwarded to the application server. Web server can be implemented on the application server, or separate application server and web server can be maintained. The application server invokes the task flow & sends the user request to the web server, which sends the user menu for selecting the options. The user makes his choice and sends the corresponding request to the application server. When the option is selected from the last menu in the flow and it is required to generate a call to agent, then all the user-related information & the user-selected options (current transaction) will be stored in the context store.

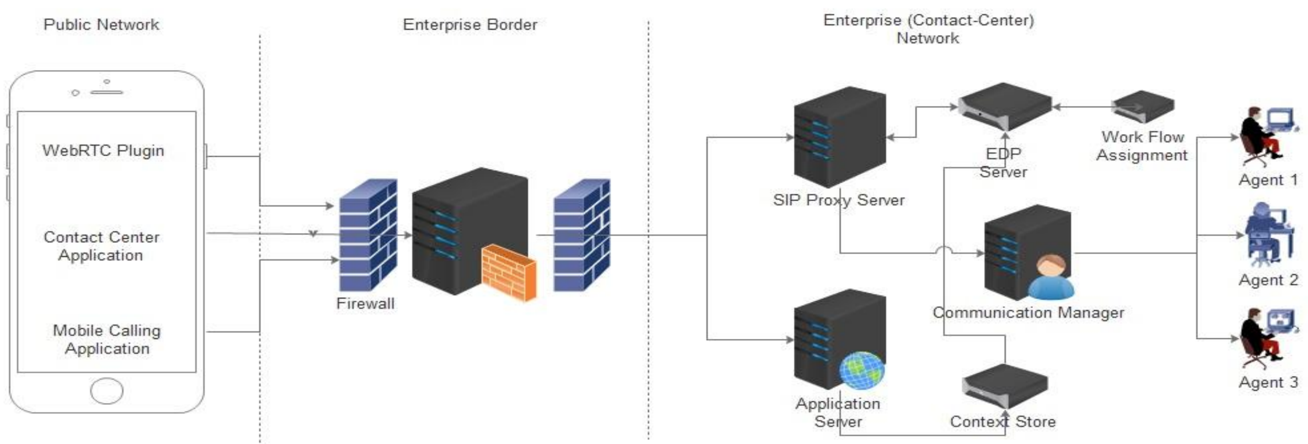

Fig. 2. Detailed architecture of Proposed System

B. WebRTC/PSTN Call

When the automated call generation will take place, depending on the 2G/3G/4G networks (coverage) of the user end-point, the call may take place via the end-point calling application (2G) or IP call requiring WebRTC plugin (3G/4G).

The data related to the current transaction (with the user) are sent to the EDP server from the context store. The EDP server processes this data and forwards it to the work flow assignment device. The work flow assignment device contains the best-agent selection algorithm. This will select the agent based on the information received from the EDP server. The details for contacting the agent will be forwarded by the EDP server to the SIP proxy server which in turn connects the call to the respective agent via the communication manager.

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V. Case Study

Let us consider an example of banking application. When a bank customer wants to contact the customer care for some query, he/she dials the customer care number. Instead of this approach, the customer can connect to the customer care center of the bank by simply pressing 'Connect to Customer Care' button on the bank app. The customer is greeted with a 'Welcome' message by the bank app and the language selection menu is presented. The customer selects the language for interaction and this is reverted to the application server present at the customer care premises.

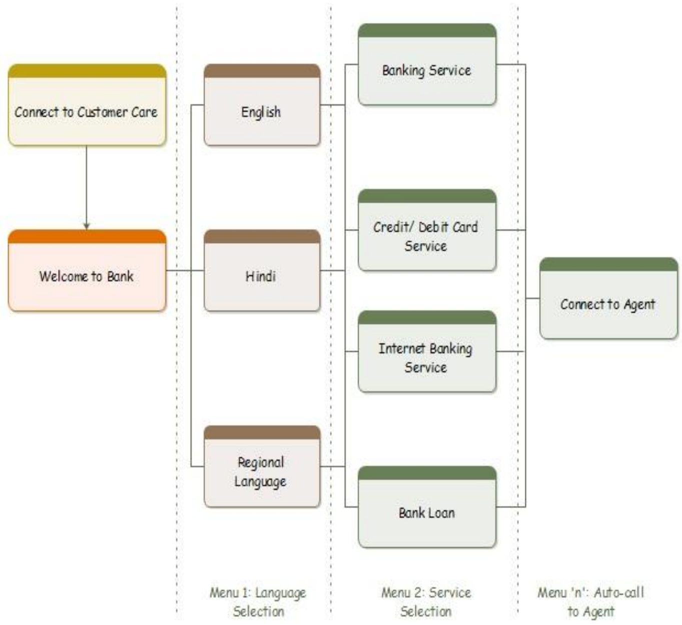

Fig. 3. Interactive Response Service Menu

On the basis of the language selected (for display), the next menu is presented to the customer, enlisting the type of services (e.g banking services, debit card services, internet banking services, etc). The customer selects the relevant option forwarded to the customer care. This process continues till the customer finds solution to his query or else for solving the query, a call is generated which is connected to the call center agent.

Based on the available network, the call may take place via the end-point calling application (2G) or IP call requiring webRTC plugin (3G/4G).

VI. Conclusion

This paper proposes the new system similar to IVR .This system design concept can be good in the present scenario where the network coverage issues may be present, etc. The use of web page exchanges for selecting the appropriate service is a better approach than voice call.

Acknowledgment

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References

[1] Mudili Soujanya and Sarun Kumar, "Personalized IVR system in Contact Center" 2010 International Conference on Electronics and Information Engineering (ICEIE 2010)

[2] Shanchuan Xu, Wanlin Gao, Zhen Li, Shuliang Zhang, Jianing Zhao, "Design of Hierarchical and Configurable IVR System" 201O Second International Conference on Computational Intelligence and Natural Computing (CINC).

[3] Chitralekha Bhat, Mithun BS, Vikram Saxena, Vrushali Kulkarni, Sunil Kopparapu , "Deploying Usable Speech Enabled IVR Systems for Mass Use".

[4] Ramazan Karademir 1 and Emre Heves, "Dynamic Interactive Voice Response (IVR) Platform", EuroCon 2013 - 1-4 July 2013 ,Zagreb, Croatia.

[5] M.H. Bokaei, H. Sameti, H. Eghbal-zadeh, B. BabaAli, KH. Hosseinzadeh, M. Bahrani, H. Veisi, A. Sanian, "Niusha, the first Persian speech-enabled IVR platform", 2010 5th International Symposium on Telecommunications

[6] Xia Xiaofeng, Xiong Ao, Liu Ya, Meng Xiangping, "The Design of IVR Dial-test System Based on Os-Workflow Engine" , 2009 International Forum on Computer Science-Technology and Applications.

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