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Enhanced VoIP Based Virtual PC Troubleshooting

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Abstract— According to the businesses and marketing strategies, customer relationship management and call center outsourcing are becoming major concerns since customer satisfaction is considered as the ultimate goal of any kind of business organization. This paper discusses how the VoIP technology can be used in an automated remote PC troubleshooting environment along with Interactive Voice Response and remote desktop assistance, for the purpose of creating ultimate software based PC troubleshooting solution capable of operating with reduced human involvement in order to provide optimum availability, reliability and cost-effectiveness for PC related business organizations. In this solution, a call processing engine pre-processes incoming calls and redirects them into the Interactive voice response engine which is enforced with a built-in PC troubleshooting knowledge base capable of guiding the caller to a solution with voice commands. If the knowledge base fails to provide a reasonable solution, the remote assistance enables real-time interaction between the caller and a human trouble-shooter. This approach minimizes the current drawbacks of traditional PC troubleshooting and increases the productivity and the efficiency.

Index Terms— PC troubleshooting, Session Initiation Protocol (SIP), Interactive Voice Response (IVR), Remote Desktop Assistance, Decision tree.

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1 Introduction

With the emergence of the personal computer, human life style became more and more fascinating, simple and automotive. In 1957 the first commercial PC was introduced by IBM[1] and thereafter many PC companies were born such as HP[2], DELL[3], Apple[4] etc. While the commercial PC became a daily partner of human life, its main goal was to create the computer era and simplify the human lifestyle as much as possible. The number of personal computers in use worldwide hit one billion in June 2008, and is expected to reach 2 billion by 2014[5]. But the main issue is; is there any systematic mechanism to increase the after sales maintenance and customer care services for those PC's vendors in order to provide good customer satisfaction and an acceptable level of recovering PC related issues.

The currently accepted solutions for the above addressed problem domain is traditional PC troubleshooting mechanisms where, if a particular customer needs assistance in resolving an issue of a PC, the customer can inquire from the customer care division by phone or using online support which normally provides FAQs and forums [6],[7] dedicated for PC troubleshooting services and after that it is necessary to bring the PC to a service center of the relevant vendor organization whether the issue is in hardware or software. According to the existing business marketing strategies, the technical and maintenance services are still in their primary stages because those operations are still handled manually.

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PC vendor companies are still expending a considerable amount of their annual budget for customer relationship management and service maintenance. A traditional call center solution is fairly unacceptable in modern state of art technologies and those are becoming bottlenecks in modern day business development strategies by wasting customer time, company's efficiency and productivity as well as reputation and degrading customer satisfaction. Here are the main problems faced by vendors related to the existing procedure;

i. A company needs lots of staffing facilities and ground spaces for call centers or outsourcing the service to a separate company.
ii. If the company has globally expanded their business, they need to maintain a regional or country vice service and technical centers which cost huge annual expenditure.
iii. Initial cost of a call center is highly expensive.
iv. It's not cost effective to expand the call center capacity.
v. An organization is has not any standard provable way to measure the call center agents productivity.
vi. Technical agent's skills are always not effectively used to their maximum potential.

The enhanced VoIP based virtual PC troubleshooting solution 'BackSpace', which has the ability of breaking the traditional barriers of PC troubleshooting services and used the combination of virtual call center architecture alone with interactive voice response aided automated remote PC troubleshooting procedure, so the PC vendor organizations will be able to use the state of art technologies such as VXML[8](We used VXML 2.0), VoIP[9], SIP[10] etc to improve the productivity, efficiency and customer satisfaction and achieve their business goals more confidently.

2 METHODOLOGY

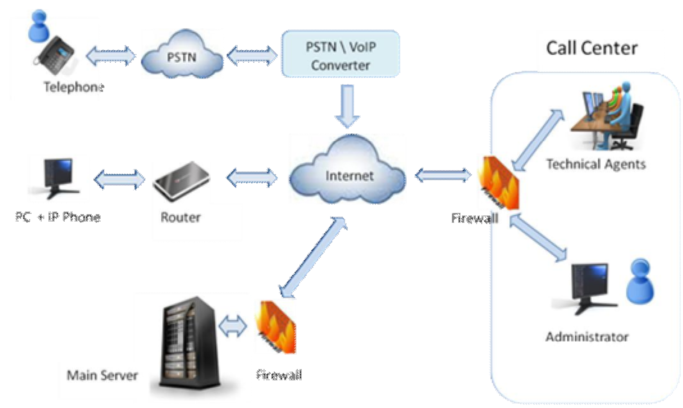

Fig. 1. System diagram

The design is mainly based on 3-tier architecture which separates the entire solution into three separate service layers. It accompanies VoIP technology to transfer voice through IP [11] and use internet and PSTN [12] as communication networks, SIP as application layer protocol and RTP [13] as transport layer protocol. It also provides virtual call center architecture so the employees can connect to the system as shown in Fig. 1. irrespective of the location and the system will be available anytime anywhere.

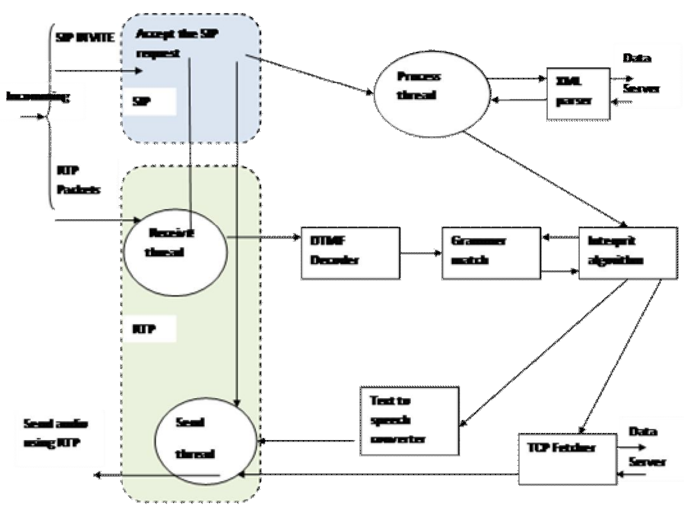

Fig. 2. Main server Process flow diagram

Fig. 2. shows the diagram of the process flow of the Main server. When the server receives an incoming request, it accepts the call and invokes the process thread and RTP thread. RTP thread receives the incoming RTP packets and invokes the DTMF [14] decoder. If any DTMF signal is detected, then it immediately sends to the grammar matching module and checks for any valid DTMF digit. The process thread is responsible of handling initial VXML page requests using the XML parser which fetches the page to a tree structure. Then the process thread calls the interpreter algorithm and it accepts the selected page as the input. It has the ability to invoke several modules based on the VXML page contents such as invokeText to Speech synthesizer to synthesize any <prompt> or invoke another TCP request. This VXML page also can specify the grammar rules too. Send thread is used to pre-process and send the data to the caller which is coming from speech synthesizer or TCP fetcher.

The BackSpace solution is consisting of three major components;

A. Call processing Engine and DTMF Decoding subsystem
B. Interactive Voice Response (IVR) sub system
C. Remote Desktop Assistance Sub System

A. *Call Processing Engine and DTMF Decoding*

Call processing Engine is responsible for handling all the call processing tasks and internal system monitoring activities. User friendly advanced Graphical User Interfaces (GUIs) are provided for necessary controlling and monitoring of processes.

The call processing unit provides functionalities and services to balance the overall system load as follows;

- Mapping PSTN to IP
- Call Queuing
- Call Transferring
- System monitoring and log file creation

BackSpace is designed to use some advanced features of call center architecture. It continuously accepts incoming calls from both IP phones as well as the POTS (Plain Old telephony Service) via PSTN networks. Since we are using VoIP, we converted PSTN analogue signals into SIP supported data using a FXO (Foreign Exchange Office) [15] analog voice card.

Advanced call routing and call queuing is used to handle the multiple user connections of the system simultaneously. Our implementation has the capability of handling hundred (100) simultaneous calls. When the number of calls exceeds more than hundred, the next incoming request will be queued without disconnecting. Each call is handled by creating a separate thread in order to provide unique service for each request.

The primary stage of providing a solution for a particular issue is achieved by providing automated step by step advices through a problem solving hierarchy. If the automated system is unable to solve the problem as expected by the user, due to lack of information in the PC trouble shooting knowledge base it is needed to transfer the call for human attention, as a secondary stage. To achieve this functionality we have designed and implemented a call transferring algorithm to find out an available best suitable person in order to transfer the call.

System monitoring and log file creation is added to this solution for the administration purpose. Administrators can log into the system and view the call history, agent productivity and the system records.

1) DTMF Decoding: DTMF decoder is primarily used for decoding the incoming analog signals of the caller and detect whether the incoming signal has any DTMF digit, if so decode it and send it to the IVR unit in order to interact and browse through the PC troubleshooting hierarchy.

When considering commercial products, all the systems use hardware based DTMF decoding solutions but in our solution we used a totally software based DTMF decoder (developed using C# 4.0) and it acts as a bridge between the caller and the IVR unit.

We have used 'Goertzel algorithm [16]' when designing the DTMF decoder because Goertzel algorithm does not require much memory and is optimal when used to compute to deal with frequencies.

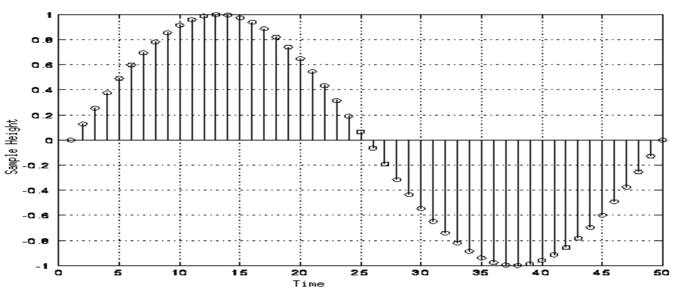

Fig. 3. Sampling of the wave signal

As Goertzel algorithm implemented is based on sampling method, we had to sample the wave signal (Fig. 3.) in to a number of discrete values (640 samples) and, using coefficient values it will detect the client pressed key.

DTMF component is responsible for performing following tasks.

- Listen for client DTMF tones
- Generate the wave signal for integer values
- Sampling the wave signal to frames
- Identifying the tone pressed by the client
- Send identified DTMF tone to IVR unit

B. *Interactive Voice Response (IVR)*

The IVR component directly communicates with the server and the DTMF decoder. There is a separate DTMF decoder specially designed for this purpose and it continually listens for DTMF data from the incoming RTP packets received by the caller's phone.

Interactive Voice Response unit is the main module which is responsible for interacting with the human actors who are connecting to the BackSpace system through PSTN telephones, IP phones or soft phones. Once the system is up, the IVR module is responsible for continuously listening on incoming client requests which come to the IVR unit as an initial caller request or as a DTMF digit. The IVR module consists of three sub components (Fig. 4.).

- SIP application server (deploys the application logic)
- Media server (IVR commands are interpreted, executed, DTMF detected and recording audio signals)
- Data Server (database for prompts and repository that stores VXML pages and other data bases)

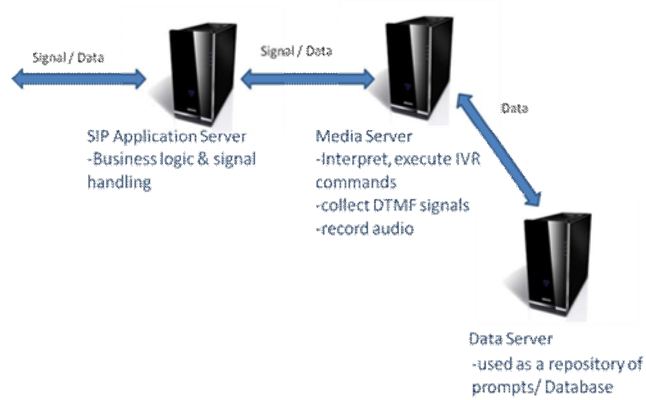

Fig. 4. Sub components of the IVR

1) *SIP Application Server:* The functionality is similar to a normal HTTP web server. But the difference in here is only the VoIP applications requests can be handled. It has the ability of taking care of multiple thread handlings, running several VoIP applications simultaneously with better memory handlings. SIP application server is responsible for accepting SIP requests from the outside. It uses port 5060 which is used for UDP [17].

2) *Media server:* Responsible for handling audio/video stream flows via RTP sessions. So the media server is the entity that interacts directly with the end-user devices in order to play prompts, record audio streams and detect DTMF digits. Here TRP flows between the media server but SIP signal does not. SIP only does the session initiation between two end points and then resides silently until the RTP session terminates.

3) *File Server:* File server is the repository of the BackSpace solution and acts as the data tier of the system. File server communicates only with the media server using TCP connection. This is a repository which is used for storing VXML pag-

es, pre-recorded audio files and SQL database. The primary objective of placing these files in a separate server is to achieve more scalability, security and reliability.

The IVR unit is uses the VXML 2.0 technology to interact with the caller. The solution is capable of analysing DTMF (Dual Tone Multi Frequency) dial pad input of the caller's telephone and response with the relevant audio output. For this purpose VXML browser/parser is used. It gets the caller input, divides a particular VXML page's contents which are surrounded by <prompt> tag into its corresponding audio stream and played back to the caller. Following (Fig. 5.) is the example VXML content which fetches the user selection.

```
<grammar version="1.0" root="top" mode="dtmf">
<rule id="top">
<one-of>
<item>1</item>
<item>2</item>
<item>3</item>
<item>4</item>
</one-of>
</rule>
</grammar>
```

Fig. 5. VXML code fragment

When considering the IVR component, it addresses most drawbacks of the current call center or any other IVR systems in the market. But this is a customized version when compared to a typical IVR application; some unnecessaryfeatures which are not related to the objectives of the IVR component used in BackSpace are removed in order to achieve high quality and the simplicity of the solution.

IVR administration panel shows in Fig. 6. provides a highly user-friendly environment to customize and edit the IVR knowledge base. According to the current usage, it has the ability to customize the IVR unit into another problem domain. (As an example this solution can be transformed in to an automated call center which can be used in a travel agent company). Through this panel, a client can do all the necessary modifications.

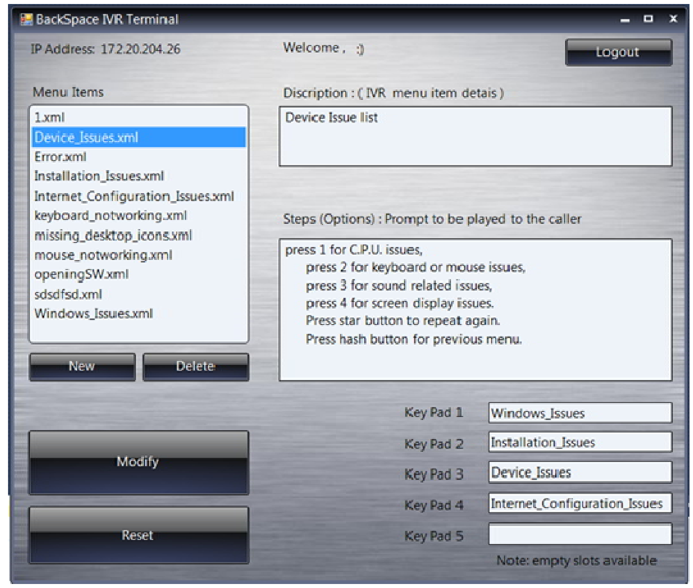

Fig. 6. IVR administration panel

C. *Remote Desktop Assistance Sub System*

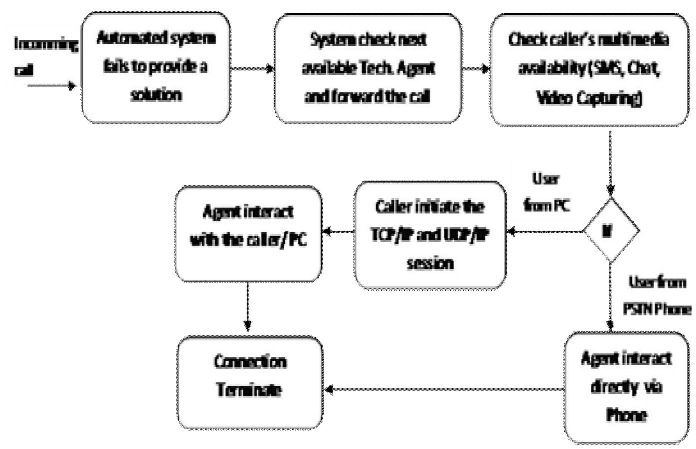

Fig. 7. Remote desktop assistance process flow

The remote desktop assistance is used to achieve the real-time communication between a caller and a technician as the secondary stage of problem solving phase of the solution.
Fig. 7. shows the process flow of the remote desktop assistance. The connection initialization is done by the caller. That will be a TCP / IP connection and UDP / IP and TCP / IP connection. This design is totally implemented using .Net socket programming [18]. There are three communication methods used to interact with the caller.

- Text Communication
- Voice Communication
- Desktop Capturing

1. *Text Communication:* For text communication implementation we have used library called conference library [18].

Text message typed in the textbox will be sent to destination IP address using UDP/IP connection and socket programming.

2. *Voice Communication:* For chat communication implementation we have used compressed methods called Alaw [19], MuLaw [19] to compress voice data before sending over the internet. Voice data will be sent to destination IP address using UDP/IP connection and socket programming.

3. *Desktop Capturing:* Remote desktop capturing designed and developed using TCP/IP connection and socket. When sending initial Bitmap images to the other side we convert and compress the data into JPEG [20] model before sending through the network since it is very efficient and very easy to handle and process. Screen will be captured every second and if the screen is same like the previous one that will not be sent and save as bit map image for fast processing.

3 Results and Discussions

The main goal of this research was to develop an automated software based solution for medium to large-scale PC vendor organizations to automate their traditional PC maintenance and customer care services. This solution has the ability of breaking the traditional barriers of PC related technical service providence. Furthermore, the highly customizable user-friendly environment enables the system to change its decision tree according to any business domain so it can be use anywhere as a replacement for the traditional call center. It uses the combination of virtual call center architecture along with automated remote PC troubleshooting to overcome above addressed issues, so the PC vendors will be able to use this solution to improve the productivity, efficiency and customer satisfaction and achieve their business goals.

We have successfully designed, developed and implemented BackSpace PC troubleshooting solution to handle hundred simultaneous calls and the troubleshooting decision tree contains solutions up to considerable level. When developing the solution we used SipekSdk [21] which is a C# based free and open source SDK, together with pjsip[22] (Open source SIP stack) and pjmedia[23] (Open source media stack) for VoIP based call processing and media handling. The decision tree contains only Windows XP [24] related issues and it provides automated audio instructions only in the English language. We successfully demonstrated our achievements in the final presentation of our research project.

4 Conclusion and Future Work

The current solution is capable of improving its functionality and features since the whole solution is developed using a component based approach with well-defined interfaces. The following are the key features that we are planning to implement in the near future.

1. Change the system to deploy and function in Linux environment.
2. Design and implement a VPN to connect administrators and technicians to the main server.
3. Improve the functionality and efficiency of the VXML parser.
4. Currently the PC troubleshooting knowledge base is includes solutions only for Microsoft Windows XP operating system. We are planning to extend it to have troubleshooting solutions for a Linux based operating system.
5. Provide language selection facility for the callers before entering to the troubleshooting process.
6. Improve the overall security of the system.

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