6-2009

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Introduction

Voice over Internet Protocol (VoIP) has progressed from a limited, untested technology to a feature-rich service that continues to find a place in all types of businesses. Companies such as Vonage and Skype have built successful business models based on VoIP services, and major telecommunications companies such as AT&T and Time Warner have followed suit with bundled offerings of phone, cable, and Internet services. Research has suggested that the Small-to-Medium Business (SMB) marketplace is quickly moving towards VoIP for productivity and efficiency benefits, convenience, and cost savings (Frost & Sullivan Company, 2006). The size of the VoIP market is increasing, with forecasts in 2006 predicting a compound annual growth rate of 31.4%, reaching nearly $3.3 billion in 2010 (Kretkowski, 2009). Although a large growth rate does not directly correlate to cost savings, it indicates that there are advantages in upgrading to either a hybrid or pure VoIP-based system. While total savings vary for each organization, the ability to route multiple calls over a single Internet connection reduces the per-call cost when compared to one-call-per-line PSTN service. Currently, many VoIP service plans offer unlimited inbound calling and outbound at a rate of $0.02 per minute.

Problem Analysis

Recently, a local financial services company had outgrown its proprietary PBX system and was in dire need of a replacement. Business conducted in the office involves real estate, insurance, tax preparation services, and management of the company's employees. The office deals directly with customers on a daily basis, with most of the trade being handled over the telephone. A normal call volume is approximately 15 calls per hour. During the tax season (January through April), the call volume spikes to an exceptionally high level, often reaching 200 calls per hour. As a result, customer calls could not be handled in a timely manner due to the limited number of staff to route incoming calls and the enormous amount of time needed for searching for information and working with multiple databases.

To help call routing, a Private Branch Exchange (PBX) system was installed in 2000. Unfortunately, this PBX system had reached the end of its useful service life and was no longer supported by the vendor. Past experience with proprietary systems uncovered problems in finding support and service for the equipment after reaching end-of-life status. Core upgrades and new features were often not available for older equipment, and home or consumer-grade devices were not capable of meeting the demands of this business environment. Additional upgrades to the old system were not possible, since manufacturer support had ended, and aftermarket accessories were difficult to find. Services such as voicemail, music-on-hold, and Interactive Voice Response (IVR) were not available. Because new instruments were no longer available, the only option for replacing broken or malfunctioning station instruments was to buy used items from surplus dealers. In tax season (January through April) when call volume spiked to an exceptionally high level (200 calls an hour), with only four POTS lines for inbound and outbound calling, call failure on outbound calls was common during peak times, and outbound calling was limited because there were no more available lines to provide outbound service. Employees had to monitor their stations and wait for one of the busy lines to clear, thereby disrupting their normal workflow.

Inbound calls were answered by a receptionist at the front desk. As new calls came in, the receptionist had to place current calls on hold, answer the incoming call, and place that caller on hold. At times, all four lines were in use, with three callers on hold. Transferring calls to other employees was difficult, for most of them were servicing in-office clients. Call analysis showed a majority of the calls fell into three areas of inquiry: hours of operation, fees for preparation, and times for appointments. Associated with each of these questions was a standard answer, which the front desk employee usually read from a script. Calls from these three areas were easy to handle, but they consumed over 90% of the front desk time.

In addition, after-hours messaging relied on a one-line, standard answering machine. This type of service only provided the ability for a caller to leave a general message for a particular employee. The messages had to be transcribed to customer contact forms and delivered to each employee afterwards. This process was time-consuming and often error prone, leading to lost messages and multiple callbacks to the customer in order to clarify the requested information. Another limitation to the old messaging system was that it could only take one call at a time. If another call came in, it would fail to connect to the messaging system and continue to ring until the caller disconnected. These shortcomings lead to frustration of the customers and lost productivity for the employees. A replacement of the PBX system was, therefore, more than necessary to support the daily business operations.

The Solution

A detailed evaluation of three open source VoIP software packages, Asterisk, Trixbox, and PBX in a Flash (PBXiF), was conducted over a three-week period to determine a possible replacement of the PBX system. The packages were selected based on three criteria: cost of acquisition and support, availability of forums or technical support groups, and compatibility with industry standard hardware platforms.

After three weeks of testing, scores based on the three criteria were presented to the management team with the recommendation to install Trixbox as the VoIP replacement for the legacy PBX. When Trixbox was installed on the base computer configuration, all BT-200 phones were configured to use the Trixbox system as their VoIP gateway. Each phone was given an extension from the PBX management software within Trixbox and configured with voicemail capability.

After implementation, the results of the new VoIP system were encouraging. The IVR in the new system dropped calls answered by front desk personnel by 75%. Such reduction can be directly attributed to the modification of the IVR to prompt for the most frequently asked questions. An added benefit to the callers is the ability to dial directly to a particular employee. This option is explained at the beginning of the IVR announcement and can be performed

anytime while the caller is in the IVR. To avoid disturbing the employee while he or she is occupied, employees can select the "do not disturb" button on their phone stations to send all calls immediately to their voicemail box. Telephone stations with a waiting voicemail are able to show a flashing light to alert the employee of a saved message. Table 1 shows the average number of calls answered by the front desk before and after the VoIP installation.

Call Type	Average Number of Calls (per hour)	
	Before	After
Office Hours	45	1
Service Fees	50	5
Appointments	15	13
Other	10	10
TOTAL	**120**	**29**

Table 1. Call Volume Analysis

What the Future Holds

With VoIP being a software-driven technology, there are many additional features that can be implemented in a VoIP system at a very low cost as compared to previous technologies, where such features may not be possible.

First, a move from hardware phones to softphones (a computer-based software application for phone functions). Softphones will reduce dependence on additional hardware components and provide additional flexibility to users. Features such as videoconferencing and instant messaging can be incorporated in a softphone with just a code upgrade. Compared to hardware phones that are limited in upgradeability, many softphone programs currently can be updated on a regular basis.

Second, use VoIP between offices to reduce telecommunications costs. Routing VoIP calls between offices can be accomplished with few modifications to existing systems. Trixbox is already capable of establishing a VoIP session with another VoIP system across the Internet. By obtaining a static IP address for each office, an intra-office call could be routed over the Internet using Trixbox. Many large corporations currently use this type of service, bypassing traditional POTS lines and their associated costs.

Third, route voicemails to e-mail. When a customer leaves a voicemail message, an employee could set the option to have the message sent to an e-mail address. This enhancement could be useful for employees who work from home.

Conclusion

The goal of the project was to implement a VoIP system in order to improve the efficiency and effectiveness of the company's operations. The use of call analysis provided critical data for establishing an effective IVR menu. In addition, detailed evaluation of the three proposed systems helped provide objective feedback on several performance factors. The minimal hardware requirements, coupled with freely available open source software, offered a low-cost VoIP solution that required little maintenance, provided for future

upgrades, and showed acceptable reliability and stability. The successful implementation of the VoIP system also fosters an updated IT plan that will help this organization chart its business strategy for future years.

References

Frost & Sullivan Company (2006) North American Enterprise IP Telephony End Point Markets, Report #F696-62

Kretkowski, P. (2009) http://www.voip-news.com/feature/9/reports.html

Note: This case is developed based on a Master's of Information Systems capstone project completed in 2008. A detailed description of this case will be available in the 2009 Proceedings of the 15th Americas Conference on Information Systems (AMCIS).