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RETOOLING FOR SUCCESS: A CASE STUDY OF VOIP IMPLEMENTATION TO IMPROVE CUSTOMER SERVICE AT A MIDWESTERN FINANCIAL SERVICES OFFICE

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Abstract

This article presents a case study of the acquisition of a Voice over Internet Protocol (VoIP) system to replace an outdated telephone system at a Midwestern financial services company. In the Midwest retail office of ABC Financial Services, the old Private Branch Exchange (PBX) phone system was incapable of handling customer inquiries during the busy tax return season. The inefficient systems exposed the organization to missed and delayed calls, which lead to a considerable number of customer complaints and lost revenue. Guided by the systematic approach of technology retooling, computer engineers followed the steps of problem diagnosis, analysis of competing solutions, implementation, and assessment of the VoIP system as the replacement telecommunications platform. System performance and evaluation data were collected during and after system implementation. Assessment of the new VoIP system demonstrated improved availability, speed, and reliability of the information provided to customers. New functionalities, such as customer inquiry of the database, provided through the VoIP system pushed the self-service adoption to a record high level. The system implementation also fosters an updated IT plan that will help this organization chart its business strategy for future years.

Keywords: VoIP, retool, customer service, financial services

1. Introduction

Customer service and customer retention are critical to an organization's success in the service sector. ABC Financial Services[1], founded in 1991, is a financial services company that operates four offices in a two-county area. In 2000, the company expanded its portfolio of services to include tax preparation. This study focuses on the office branch located downtown in a Midwestern city that serves as both a retail location and the company's headquarters. 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. In such cases, customer calls could not be handled in a timely manner due to the limited number of front desk staff who needed 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. The lack of an adequate PBX system was also causing problems throughout the entire organization. During tax season, the old PBX system was unable to handle the increased workload and staff members were overwhelmed, resulting in decreased customer satisfaction, increased complaints, and a potential loss of market share. An effective system was therefore required to support the daily business operations and to provide quick access to users while minimizing overall maintenance and support costs.

Motivated by the problems with the outdated PBX system and the risk of losing their market share, the company decided to replace the inefficient phone system. Due to the competitive nature of the industry, systems supporting the operations at the headquarter office must be quickly adapted to changing needs of management and customers. Often, system upgrades needed to be implemented within hours in order to alleviate customer complaints. In addition, the company needed to provide a streamlined process to handle routine questions for customers and minimize any additional workload on the staff. The new system must also provide automated routing of calls to the correct staff member and should be capable of routing standard calls to designated customer service representatives. Considering all the above, the purpose of the project was to implement a Voice over Internet Protocol (VoIP) system at the Midwestern headquarter office of ABC Financial Services to address the major challenges to the efficiency and effectiveness of the company's operations.

2. Retooling and Problem-Solving

According to the dictionary, retool means "to make changes to (something) in order to improve it" (www.m-w.com). In practice, the concept of retooling is process-oriented and includes the actions of updating, reorganizing, and improving (Pitman, 1994). In summary, the process of retooling involves introduction, adoption, and implementation of new and improved technologies. To be successful, the retooling process will also need to involve the participation of a wide range of stakeholders at all levels and the consideration of a number of key components, technical and non-technical. The central concept of retooling, like many other technology updates and implementations, often involves problem-solving to address a specific technical issue of ineffectiveness, inefficiency, or dysfunction.

The extensive literature on solving problems demonstrates its importance as well as its broad impacts in almost all fields of scientific and practical operations. In the field of information technologies, problem solving is one of the most frequent activities carried out to address technical concerns such as trouble-shooting. The most widely accepted theories and frameworks for problem solving originated from the work of George P. Huber's approach to problem solving, especially when solving problems related to information systems and decision sciences (Huber, 1980, 1984, 1991; Huber & McDaniel, 1986). Huber introduced five steps or stages for problem solving: 1) Understand Problem, 2) Plan Solution, 3) Evaluate Alternatives, 4) Implement, and 5) Monitor/Review (Huber, 1980). This approach was widely adopted and used for technical problem solving (Huber, 1984) as well as for problem-based education and training (Kamis & Kahn, 2009).

In this study, we adopted the five steps of problem solving as a set of guiding principles for the phone system retooling project at the ABC Financial Services. The section below details each step and explains its process and outcome.

3. Retooling with VOIP

Step 1: Understand the Technical Problem

ABC Financial Services used a proprietary, fifteen year old Nortel PBX system that could connect four external Plain Old Telephone Service (POTS) lines for inbound and outbound calling with a maximum of 12 stations and functions of

[1] The identity of the firm is disguised. The first author was the IT Director of the firm at the time the project took place.

intercom, speed dial, and caller ID display. The manufacturer had ended the technical support; therefore, no more upgrades were possible and aftermarket accessories were difficult to find. Services such as voicemail, music-on-hold, and Interactive Voice Response (IVR) were not supported with the old phone system. The only option for replacing broken or malfunctioning station instruments was to buy used parts from surplus dealers because new instruments were no longer manufactured.

Inbound calls coming to the office were answered by a receptionist at the front desk. As additional calls came in, the receptionist had to place current calls on hold, answer the incoming call, and place that caller on hold. Very often all four lines were in use and three callers were on hold. Transferring calls to other employees was difficult because most of them were occupied with in-office clients. The normal inbound call volume at the business office was approximately 15 calls per hour. During the tax season (January through April), call volume spiked to an exceptionally high level, often reaching 200 calls an hour. With only four POTS lines for inbound and outbound calling, call failure on outbound calls happened frequently during peak times, and outbound calling was limited because there were no available lines to provide outbound service. Employees at the office had to monitor their station and wait for one of the busy lines to clear before making an outbound call, thereby disrupting their normal workflow and significantly decreasing their productivity.

In addition, after-hours messaging relied on a one-line, standard answering machine. This 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. 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 previous caller disconnected. These shortcomings lead to frustration of the customers and extremely low productivity of the employees.

Despite the limitations of the current telephone system, the network infrastructure was in good condition. New wiring was installed five years previously and met CAT-6 standards. Switching was provided by a Netgear GS724T, a 24-port switch, with routing provided by a D-Link DFL-300 router. Typical traffic loads were at less than 2% of switch capacity, with 17 out of 24 ports being used. The current business model projected an increase in customers for the next several years. To support such growth, additional stations and call capacity, both inbound and outbound, had become a requirement. The company was determined to improve the front desk call handling, messaging, and routing processes by implementing a much-needed new phone system.

Step 2: Plan Solution

Integrated phone systems used to be the sole domain of major telecommunication companies. Large Fortune 500 companies, such as AT&T, Nortel, and Siemens, provided Private Branch Exchange (PBX) systems and services either as standalone systems (e.g. Merlin) or bundled with other products (e.g. Audix). The landmark 1968 Carterfone decision by the Federal Communications Commission (FCC, 1968) allowed third-party and non-phone company devices to be connected to the Public Switched Telephone Network (PSTN), thereby opening the door to increased competition in the then monopolized PBX market (Ismail, 2011). Most PBX systems operate on a hub-spoke configuration, with each station (a telephone instrument or plain telephone) connected to a central unit via twisted pair cabling (Figure 1). When a station places a call, the central unit recognizes the calling station, provides a dial tone, and waits for the dial. After the digits have been entered, the system determines if the call will be routed internally or externally. In this case, the user does not interact directly with the PSTN because the PBX system handles the interfacing of the external subscriber lines to the internal stations. The majority of the PBX systems in use today rely on proprietary signaling methods between the stations and the central unit (Sulkin, 2004). This particular method results in incompatibility between competing systems and often even between different models of product from the same company, which became a noticeable drawback for choosing a PBX system for any organization.

VoIP stands for Voice over Internet Protocol. The technology has been in various stages of design and improvements since the 1970s and has recently emerged as a viable platform for meeting small to medium-sized organizations' telecommunication needs (Varshney, Snow, McGivern, & Howard, 2005). The concept of moving voice traffic across IP connections was first formalized in 1973 with the experimental Network Voice Protocol, funded in part by the Advanced Research Projects Agency (ARPA). The goals of that project were "... to develop and demonstrate the feasibility of secure, high-quality, low-bandwidth, real-time, full-duplex (two-way) digital voice communications over packet-switched computer communications networks" (Cohen, 1977, p. ii). In 1970s, the IP infrastructure was not designed to handle the amount of bandwidth required, thereby resulting in less than acceptable call quality, despite many experiments to test and validate the protocols and concept. As technology improved, the idea of VoIP calls surfaced once again in the early 1990s with Vocaltech's commercial offering of a functional VoIP product. Although calls could be completed with acceptable quality, the software had several limitations. First, a successful call required Internet access and sufficient bandwidth at each end to provide acceptable throughput. Secondly, each computer was required to run the software package in order to use the VoIP function, necessitating the purchase of multiple copies of software. Finally, all calls had to be made via the Internet because the software had no way to connect to the PSTN.

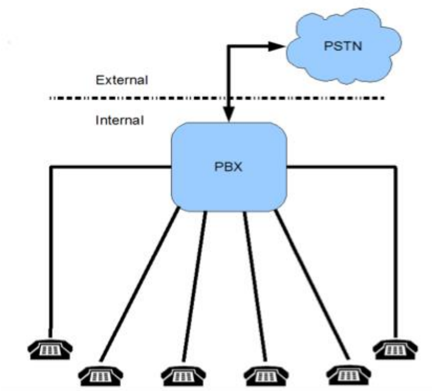

Figure 1: Typical PBX Configuration

In the late 1990s, a new market began to emerge. The increasing availability and decreasing cost of high-speed Internet, coupled with a demand for rich features in business telephony systems, sparked the first low-cost, open source, VoIP PBX system: Asterisk. The system was initially aimed at home or small business users who wanted the flexibility of a proprietary PBX system, but without the cost or possible obsolescence that might result. Other companies offered VoIP services such as Vonage, Skype, and Clearwire, but required the use of proprietary equipment and provided fewer services than dedicated VoIP PBXs. After the turn of the century, hardware costs further decreased and reliable open source applications became more powerful and flexible. The early 2000s saw the barriers lowered for integrating VoIP in small to medium-sized businesses. Recognizing the cost savings and competitive advantage that VoIP could provide, many organizations began the transformation from proprietary PBX systems to VoIP solutions that could combine PSTN and pure VoIP technologies. These hybrid systems allowed organizations to continue to use traditional PSTN lines for local calls while leveraging VoIP connections to reduce local-toll and long distance costs. As the reliability and quality of VoIP connections improved, many organizations dropped their PSTN services and converted to 100% VoIP.

Research shows that the Small-to-Medium Business (SMB) marketplace is quickly moving towards VoIP for productivity and efficiency, convenience, and cost savings (Goode, 2002; Frost & Sullivan Company, 2006; Sulkin, 2001, 2004). The size of the VoIP market is also increasing. The U.S. VoIP market has experienced 16.7% annual growth rate over the last five years and generates $15 billion in annual revenue in 2013 (VoIP Market Research Report, 2014). Although a large growth rate does not directly correlate to cost savings, it indicates great benefits and advantages in upgrading to VoIP. For instance, 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 technology and service plans offer unlimited inbound calling and outbound rates at $0.02 per minute or less.

Step 3: Evaluate Alternative Solutions

Three VoIP software packages, Asterisk, Trixbox, and PBX in a Flash (PBXiF), were selected for testing as a replacement for the old PBX system. These selections were made based on three criteria: 1) cost of acquisition and support, 2) availability of forums or technical support groups, and 3) compatibility with industry standard hardware platforms. To meet the business requirements at ABC Financial Services, the new VoIP system needed to support the following ten capabilities:

- A minimum of 15 stations utilizing a VoIP telephone device with ability to support up to a total of 30 stations
- IVR capability
- Voicemail for individual users and for groups, available during and after-hours
- Time conditions for open and closed settings
- Five incoming (FXO) phone lines
- Three analog (FXS) station lines
- Station to Station calling
- Transfer of calls
- Intercom function
- Historical call records

To ensure comprehensive and comparable evaluation of the technical capacities and performance of all possible solutions, each package was tested using the hardware configuration of 2.2 Ghz AMD processor, 1 GB RAM, 80 GB SATA Hard Drive, VGA Video (built-in), 10/100 Mbs LAN connection, with CD/DVD drive.

Table 1 provides a side-by-side evaluation of the three systems on three main criteria:

Platform	Cost/Support	Forum/Technical	Industry Standards
Asterisk	Software is free, company support requires payment	User and developer forums available on the Internet with heavy volume of posts	Compatible with Digium, Sangoma, and Rhino hardware cards
Trixbox	Community Edition is free. Support packages are available in different tiers based on software package selected	User and developer forums available on web page and from other sites. Heavy volume of posts	Compatible with Digium, Sangoma, and Rhino hardware cards. Pre-built appliance available
PBXiF	Free. Support is via forum or through third-party providers, costs vary	User and developer forums available from web page. Moderate postings, limited third-party offerings	Tested with Digium cards, but no explicit support for hardware cards. Base package is designed for a particular hardware configuration

Table 1: VoIP Software Installation Criteria

To interface the incoming telephone lines from the phone company, a Digium TDM-400 analog modular gateway card was installed. This particular card was selected because it was supported by all three software packages without any additional software or drivers. Digium was the first company to provide an analog gateway card that was open source and not system specific. The telephone instruments used for the test were four Grandstream GXP-2000s and eleven BT-200s. These instruments resembled standard phones; however, they could be used only with an Ethernet connection and a VoIP system. Each software package was installed and allowed to run for one week at the headquarters office. A computer system was built using the hardware configuration listed above as a platform for each software package, thereby allowing each package to be evaluated using the same hardware. Raw call statistics were collected using the call logs integrated in each software package.

Cole and Rosenbluth (2001) described a detailed method for monitoring and evaluating VoIP applications using the ITU-T's E-Model (ITU, 1998) and the Mean Opinion Score (MOS) contained within the model. Although this type of analysis was designed for a large installation of VoIP systems and applications, there were two elements that were applicable to the project's particular deployment: 1) codec (coder-decoder or compressor-decompressor) type and 2) passive monitoring techniques. The chosen codec for each system was G.711 because it provided a reasonable trade-off between processing requirements of the VoIP system and voice quality. Passive monitoring was chosen over active monitoring for two reasons. First, passive monitoring is non-intrusive to the network under test and adds no additional load. The passive monitoring ensures that measurements are based on real network traffic and not artificially generated numbers. Furthermore, passive monitoring allowed a more extensive measurement capability (specifically the type of measurement), thereby providing improved granularity of data and subsequent information.

In addition to the system performance metrics mentioned above, to minimize the subjectivity and quantify factors such as call quality, end users (employee of the financial services firm) were instructed to rate the quality of each call (0-10, low to high) on 1) call connection time, 2) dropped words or breaks in conversation, and 3) echo or feedback of audio. Specifically, call connection time was the time elapsed from entry of the last digit of the calling number to the beginning of the second ring tone. The second ring tone was chosen because each VoIP system provides one ring tone internally before connecting the caller to the PSTN. The second ring tone was chosen to mask the call setup procedure that occurred for every outbound call; otherwise the user would be presented with silence and might think the call could not go through. This measurement was directly related to processing time on the system and how efficient the software was managing the call. Dropped words or breaks were missed words or syllables, stuttering, and dropping of audio. This measurement was related to jitter or delay in packet reception due to network congestion. The congestion could be attributed to saturation of the network, an overloaded switch or router, incorrectly set jitter buffer in the phone, or the VoIP software's inability to handle several simultaneous calls. Out of these four causes, the failure of the software to handle calls was the most common problem. Echo or feedback occurred when the callers heard in the headset their voices mixed with a slight delay or high-pitched whine or chirp. It could be experienced by both the caller and called party; however, feedback was limited to the caller because of software filtering. The echo was usually the result of an imbalance in the POTS lines and the modular gateway card and could be cancelled by setting transmit and receive audio levels in the gateway card. Once the levels were adjusted correctly, the problem rarely returned.

Furthermore, customers who called into the VoIP system were asked to rate the quality of their experience (0-10, low to high) from two perspectives: 1) ease of use with the IVR and 2) overall quality of the call. More specifically, employees rated the quality of calls on a scale of 0 to 10 and were asked to take notes if any of the above resulted in a termination of the call, which was rated a score of zero. In a period of 10 days, eight employee rated ten calls per day, with a total of 800 ratings. Analysis of previous call statistics showed that the peak calling time was between 11:00am and 1:00pm; therefore, employee were instructed to rate two calls per hour within that time period and one call per hour at other times.

In addition to ratings on call quality, special attention was placed on the maintenance and upgrade of the system under evaluation. It was likely that management would appoint an individual who would not be familiar with PBX administration to perform maintenance on the system in the future, thereby increasing the importance of straightforward, user-friendly, and uncomplicated upgrade and maintenance procedures. The IT person who was in charge of all system upgrades and maintenance evaluated the ease of configuration and the ease of maintenance on three criteria: 1) amount of interaction required to install and update the software, 2) effort required to add an extension, IVR prompts, and voicemail, and 3) number of unscheduled reboots required during the test period. Table 2 shows the average scores of call quality, ease of configuration, and maintenance for Asterisk, Trixbox, and PBXiF.

Criterion of Evaluation (out of 10)	Asterisk	Trixbox (recommended)	PBXiF
Call Quality	7	9	6
Ease of Configuration	10	10	8
Ease of Maintenance	7	9	8
TOTAL	8.0	9.3	7.3

Table 2: Scores for Three Systems under Evaluation

Call quality favored Trixbox, with Asterisk and PBXiF falling short in this area. Initially, it was expected that an outdated driver for the TDM-400 card may have affected timing issues for the system. After obtaining an updated driver from the support board at Digium, tests were carried out for both Asterisk and PBXiF, but the result showed no changes in the quality. Testing was also carried out using both phone types provided by the organization, but no correlation could be found between call quality and a particular phone type. Although the base code of each system was Asterisk, each package had modified certain parts of the code base, resulting in slight differences between each system, which could account for some of the score differences.

In reference to ease of configuration, both Asterisk and Trixbox garnered perfect scores on the ease of configuration but PBXiF fell short in support for FXS and FXO card support. The PBXiF software recognized the cards, but it would only do so part of the time. Several attempts were made to work with the underlying configuration files to remedy the issues; but ultimately, PBXiF failed to meet the requirements for the category. Ease of maintenance showed Trixbox as the highest scoring, with PBXiF and Asterisk in second and third places respectively. PBXiF scored well in installation and maintenance, with only occasional reboots of the system during and after initial setup. Asterisk required several reboots and additional files to complete the setup process along with a great deal of expertise in Linux-based command-line skills. Because there was no Graphical User Interface (GUI) within Asterisk, the process relied totally on the ability of the user to enter a series of long text strings to initiate any updating of the system. In comparison, both PBXiF and Trixbox offered a GUI that allowed upgrades and maintenance to be accomplished with a few clicks of a mouse.

The scores of testing were presented to the management team with the recommendation to install Trixbox as the VoIP replacement for the legacy PBX.

Step 4: Implement

Upon approval, Trixbox was installed on the base computer configuration, seventeen 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. One of the primary problems at the office was the large call volume during peak times. An analysis of calls indicated that the 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 canned answer that the front desk employee usually read from a script. Calls from these three areas were easy to handle and they consumed over 90% of the front desk time. Therefore, an interactive voice response (IVR) was created to route inbound calls. The IVR was configured to prompt the callers to determine if their questions fell into one of the three areas. If it did, they were instructed to press a corresponding number, routing the call to an automated message. The message relayed the appropriate information and offered two choices to the caller: returning back to the main menu or being connected to an operator.

Step 5: Evaluate

The Trixbox installation was configured with two IVR menus: one was active during regular office hours, and the other was active when the office was closed. The switch between IVRs was automatic and controlled by a time scheduler within the program. The prompts were recorded using one of the BT-200 phones, and were easily activated by the menu-driven administration page. Individual users could create their own voicemail recordings using the same BT-200 phones, and users were prompted by the system for configuration; no interfacing with the administration page was required. The time scheduler was easily modified from the administration page, and was utilized several times during the implementation period. This allowed any authorized user to adapt the system to an immediate need, and provided flexibility with minimal training. Table 3 shows the results of the user and customer evaluations.

Quality Evaluation Criteria (out of 10)	
Call Quality	9
Dropped Words, etc.	10
Echo/Feedback	9
IVR Interaction	8
Overall Quality	8
Average Score	8.8

Table 3: System Scores

The analysis on call volume was also conducted after implementation and the results of the new VoIP system, Trixbox, were encouraging. Table 4 shows the average number of calls answered by the front desk before and after the VoIP installation.

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

Table 4: Call Volume Analysis

The IVR provided by the new Trixbox system reduced calls to be answered by front desk personnel by 75%, and reduced the office hours to handle incoming calls at the front desk from 45 hours to 1 hour. Such reduction can be directly attributed to modifying the IVR to prompt for the most frequently asked questions. An added benefit to the callers was 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 now select the “do not disturb” button on their phone station 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.

The Trixbox system is fully functional. It keeps detailed call records and allows for precise tracking of long-distance and other toll related calls. All records are stored in a MySQL database and are accessible using the internal, web-based interface located on the company’s Intranet.

4. Discussion

The initial challenge ABC Financial Services faced was an old phone system that could not adequately support call volumes. Guided by the problem-solving strategy suggested by Huber (1980), the project analyzed the technical issue, evaluated three feasible technological solutions, and implemented the Trixbox VoIP system with satisfactory results. With VoIP quickly becoming a standard for telecommunications, there are many additional features that can be implemented. After the successful implementation of the Trixbox VoIP systems, the following recommendations were made to the organization’s future IT plan. First, the company should move from hardware phones to softphones (a software application for phone functions). The softphones program 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. Hardware phones have limited upgradeability. Many softphone programs currently have these capabilities available. Second, the company should 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. Finally, the system should 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.

5. Conclusion

The goal of the project at the headquarters of a Midwestern financial services firm was to implement a VoIP system to improve the efficiency and effectiveness of the company's operations. A systematic retooling process for technical problem solving was adopted to guide the project to identify and implement a VoIP solution for the company. The use of call analysis to determine the nature of incoming calls and the subsequent discovery of three commonly asked questions provided critical data for establishing an effective interactive voice response (IVR) menu. In addition, detailed evaluation of the three test systems by staff members revealed potential problems with each system and 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. Finally, post-implementation analysis indicated a reduction in calls to front desk personnel by 75%, freeing valuable resources for other organizational needs. The implementation of the VoIP system was successful, and on completion of the project, the organization had replaced their outdated PBX system with a fully functional, upgradeable VoIP system.

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