A Case of VoIP Implementation at A Financial Services Office

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ABSTRACT

Customer service and customer retention are critical to an organization’s success in the service sector. In the Midwest retail office of Anthem Consultants, the old PBX 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. This paper presents a case for implementing a VoIP PBX system at Anthem Consultants. Several VoIP systems were tested to determine the suitability of VoIP as the replacement telecommunications platform. The new system increases the availability, speed, and reliability of the information provided to customers and at the same time reduces the staff load. The system implementation also fosters an updated IT plan that will help this organization chart its business strategy for future years.

Keywords

VoIP, customer service

STATEMENT OF THE PROBLEM

Anthem Consultants, 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. The office in this study is located downtown in a city in the Midwest and 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. 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 proprietory 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. A replacement of the PBX system was, therefore, required to support the daily business operations.

The lack of an adequate PBX system was causing problems throughout the entire organization. Because of the spike in customer calls during tax season, the old system was unable to handle the increased workload, resulting in decreased customer satisfaction, increased complaints, and a potential loss of market share. Because staff members could be overwhelmed quickly during tax season, a method was needed to provide quick access for users and customers while minimizing overall maintenance and support costs. A replacement of the PBX system was therefore required to support the daily business operations.

Motivated by the current problems with the outdated PBX system, the organization had decided to replace it. Because of the competitive nature of the industry, systems supporting the operations at the headquarters office must be quickly adaptable to changing needs of management and customers. Changes often needed to be implemented within hours in order to alleviate customer complaints. In addition, the organization needed to provide a streamlined process to handle routine questions for customers, minimizing any additional workload on the staff. Although replacing PBX system was one step of the solution, providing automated routing of calls to the correct staff member was also critical; therefore, a system that was capable of standard call routing functions and rapid changes with minimal effort was needed. The purpose of the project at the
headquarters office of Anthem was to implement a Voice over Internet Protocol (VoIP) system to address the major challenges to the efficiency and effectiveness of the company's operations.

**BACKGROUND OF VOIP SYSTEM**

Until recently, integrated phone systems were 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 bundled with other products (i.e., Audix) or as standalone systems (i.e., Merlin). The landmark 1968 Carterphone decision by the Federal Communications Commission (FCC, 2009) 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.

Most PBX systems operate on a hub-spoke configuration, with each station (a telephone instrument or just plain telephone) connected to a central unit via twisted pair cabling (Figure 1). When a station wants to place a call, the central unit recognizes the calling station, provides a dial tone, and waits for digit input. After the digits have been entered, the system determines if the call will be routed internally or externally.

![Figure 1. Typical PBX Configuration](image)

The user does not interact directly with the PSTN, for 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 between different models from the same company. Organizations that choose to install a PBX system need to weigh the costs versus benefits carefully before choosing.

VoIP, which has been in various stages of design since the 1970s (Varshney, 2005), has recently emerged as a viable platform for meeting small to medium-sized organizations’ telecommunication needs. 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 put forth in RFC 741: “...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.” (RFC, 1977, p. ii) While testing and validating the protocols and concept, the IP infrastructure of that era was not designed to handle the amount of bandwidth required, thereby resulting in less than acceptable call quality.

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. These
requirements necessitated the purchase of multiple copies of software to support even small offices. Finally, the only calls that could be made were via the Internet because the software had no way to connect to the PSTN.

In the late 1990s, a new market began to emerge. The increased 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 such as Vonage, Skype, and Clearwire offered VoIP services, but they required the use of proprietary equipment and provided fewer services than dedicated VoIP PBXs.

With hardware costs decreasing and reliable open source applications becoming more powerful and flexible, the early 2000s saw the barriers for integrating VoIP in small to medium-sized businesses lowered. 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 total VoIP use.

Telecommunications services are a need for any business, regardless of size. 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). Figure 2 illustrates these factors and how they ranked among businesses.

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 can indicate that there are advantages in upgrading to VoIP.

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 plans offer unlimited inbound calling and outbound rates a $0.02 per minute.

**PROBLEM ANALYSIS OF THE PBX SYSTEM**

The organization used a proprietary Nortel PBX system (Table 1). The system had a maximum capacity of twelve stations and was approximately fifteen years old. Four external Plain Old Telephone Service (POTS) lines could be connected to the system for inbound and outbound calling.

- 4 POTS lines
- Maximum of 12 stations
- Intercom

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**Figure 2. VoIP Adoption among Businesses**

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No more upgrades were possible to the old system, for 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. The only option for replacing broken or malfunctioning station instruments was to buy them used from surplus dealers because new instruments were no longer available.

Normal inbound call volume is approximately 15 calls per hour. During the tax season (January through April) call volume spikes 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 was common during peak times, and outbound calling was limited because there were no available lines to provide outbound service. Users must monitor their station 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. 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. 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 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 for the customers and lost productivity for the employees.

Despite the limitations of the current telephone system, the network infrastructure was in good condition. New wiring was installed five years ago and met CAT-6 standards. Switching was provided by a Netgear GS724T, 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, and additional stations and call capacity, both inbound and outbound, had become a requirement. Improvements to the front desk call handling, messaging, and routing processes had been instituted, but they had reached the limits of the telephony technology provided and were quickly overloaded.

**ANALYSIS AND TESTING**

To meet the requirements of the organization, 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

Three VoIP software packages, Asterisk, Trixbox, and PBX in a Flash (PBXiF), were chosen for testing as a replacement for the PBX system currently in use. The VoIP software packages were chosen based on three criteria: cost of acquisition and...
support, availability of forums or technical support groups, and compatibility with industry standard hardware platforms. Table 2 compares the three products based on these criteria.

To ensure a level playing field, each package was tested using the following hardware configuration:

- 2.2 Ghz AMD processor
- 1 GB RAM
- 80 GB SATA Hard Drive
- VGA Video (built-in)
- 10/100 Mbs LAN connection
- CD/DVD drive

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

**Table 2. VoIP Software Installation Criteria**

To interface the incoming telephone lines from the phone company, a Digium TDM-400 analog modular gateway card was installed. The decision to use this particular card was based on the fact that it is supported by all three software packages without the need for 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. Raw call statistics were collected each week using the call logs integrated in each software package. 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.

Cole and Rosenbluth (2001) described a detailed method for monitoring and evaluating VoIP applications using the ITU-T’s E-Model (ITU-T, 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: codec (compressor-decompressor) type and 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. Secondly, passive monitoring allowed a more extensive measurement capability (specifically the type of measurement), thereby providing improved granularity of data and subsequent information.

To minimize the subjectivity and attempt to quantify a factor like call quality, end users were instructed to rate the quality of each call placed in the following manner:

- Call connection time (high, medium, low) (Q1)
• Dropped words or breaks in conversation (high, medium, low) (Q2)
• Echo or feedback of audio (high, medium, low) (Q3)

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.

End users 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, 8 users rated 10 calls per day. Analysis of previous call statistics showed that the peak calling time was between 11:00am and 1:00pm; therefore, end users rated two calls per hour within that period and one call per hour at other times.

Besides ratings on call quality, special attention was placed on the maintenance and upgrade process and 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. One IT person who was in charge of the system upgrade and maintenance evaluated the ease of configuration and the ease of maintenance on the following criteria:
• Amount of interaction required to install and update the software (Q4)
• Effort required to add an extension, IVR prompts, and voicemail (Q5)
• Number of unscheduled reboots required during the test period (Q6)

Table 3 shows the averaged scores of call quality, ease of configuration, and maintenance for Asterisk, Trixbox, and PBXiF.

<table>
<thead>
<tr>
<th>Factor/System</th>
<th>Asterisk</th>
<th>Trixbox</th>
<th>PBXiF</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Quality (out of 9)</td>
<td>7</td>
<td>9</td>
<td>6</td>
</tr>
<tr>
<td>Configuration (out of 10)</td>
<td>10</td>
<td>10</td>
<td>8</td>
</tr>
<tr>
<td>Ease of Maintenance (out of 9)</td>
<td>7</td>
<td>9</td>
<td>8</td>
</tr>
<tr>
<td>TOTAL</td>
<td>24</td>
<td>28</td>
<td>22</td>
</tr>
</tbody>
</table>

Table 3. Total System Scores

IMPLEMENTATION AND RESULTS

Although both Asterisk and Trixbox garnered perfect scores on the required 10 capabilities (described under Analysis and Testing), PBXiF fell short in support for FXS and FXO card support. The 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.

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.
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. When 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. A prior call analysis indicated that the three most commonly asked questions concerned the hours of operation, fees for tax service, and appointment scheduling. An 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.

After implementation, the results of the new VoIP system were encouraging. Table 4 shows the average number of calls answered by the front desk before and after the VoIP installation.

<table>
<thead>
<tr>
<th>Call Type</th>
<th>Average Number of Calls (per hour)</th>
<th>Before</th>
<th>After</th>
</tr>
</thead>
<tbody>
<tr>
<td>Office Hours</td>
<td>45</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>Service Fees</td>
<td>50</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Appointments</td>
<td>15</td>
<td>13</td>
<td></td>
</tr>
<tr>
<td>Other</td>
<td>10</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>TOTAL</td>
<td>120</td>
<td>29</td>
<td></td>
</tr>
</tbody>
</table>

Table 4. Call Volume Analysis

The IVR in the new system dropped calls to be answered by front desk personnel by 75%. Such reduction can be directly attributed to modifying 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 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 now keeping detailed call records, allowing 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.

CONCLUSION

The goal of the project at the headquarters of Anthem Consultants was to implement a VoIP system in order to improve the efficiency and effectiveness of the company’s operations. 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 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.
RECOMMENDATION FOR FUTURE PLANS

With VoIP quickly becoming a standard for telecommunications, there are many additional features that can be implemented. Based on the strategic goals of the organization, the following recommendations are made to the organization’s future IT plan.

First, 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 are limited in upgradeability. Many softphone programs currently have these capabilities available.

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.

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.

REFERENCES