AAMGA Automation Committee

Paths to VOIP

Prepared for the AAMGA Automation Conference, March 9-11, 2008 Dallas, TX

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3/9/2008

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VOIP Overview

Voice over IP, a group of technologies developed in the mid-1990s by hobbyists and then formally by Cisco and Nortel, was heralded, initially, as a panacea for the issues companies were experiencing with the public switched telephone network. Over time, many realized that the technology, while very powerful, needed careful implementation and maintenance to realize the dream of leaving the Bells. While some corporations were able to make early versions work, often showing fantastic presentations about the integration of data system and phone network, many companies experienced major outage issues and call audio quality issues. As time has passed, internet data speed has increased, prices have fallen, and expertise has become more widely available, a new round of VOIP rollouts seems to be underway.

VOIP uses data networks, such as the internet, to transfer audio using packets of information, similar to how normal program data flows over the internet. The promise of VOIP is comprised of cost savings in line charges, long distance costs, and government line taxes; all while adding many new connectivity features and weaving company phone call data with internal data applications.

The information on VOIP is voluminous on the internet. There have been countless books written on the topic and this document is not intended to supersede the sterling work of others. Rather, this document is meant to demonstrate the stories of two companies on their own roads to VOIP systems. We have, additionally, included an appendix with further reading materials on the topic as well as an FAQ of commonly reported issues that many experience with VOIP and the general framework for a solution.

Case Study: Small to Medium Sized Business

The following case study describes the migration from an old phone system to a hosted VOIP solution. The company employs between 50-100 people and occupies 5 offices in different areas of California. The company's name has been changed for the purpose of this study.

Prior Configuration

SampCo's original configuration was a fairly standard office configuration. An aging PBX switch provided service to users at the home office through approximately 10 external lines with multiple inbound phone numbers in a hunt group. A toll free number also provided service to customers; the toll free number forwarded to the main line. Internet access was provided by two DSL lines on an internet NAT router.

A capital cost of approximately twenty thousand dollars was paid for the PBX at the inception of this phone system configuration with additional cost for each of the phones. Adding phones into the system required configuration by SampCo, requiring someone who was trained in the case of additions and subtractions of users and phones.

Monthly recurring costs for this configuration were: approximately $2500 monthly. ~ $1200 funded the lines and call charges themselves, ~ $400 for line taxes and government surcharges, and ~ $900 for long distance charges.

While four branch offices exist, one site, in Los Angeles, CA will be used for this case study. The Los Angeles office's phone charges summed to ~ $800 providing phone lines for all office employees, one DSL line for internet access, and a long distance number.

Moving

When the company decided to move headquarters to a new office space a phone system move was required. This move left SampCo with multiple options on the phone system. They could move the existing phone system, purchase a new phone system – similar to the current one with updated phones, change to a Voice Over IP self hosting system, or change to a fully hosted Voice Over IP configuration.

The Old

Moving the old phone system, at first glance, would have seemed to be the simplest way of migration to the new office. The same phone company could be used while the PBX could be moved and connected to provide phone service to users. The largest consideration was in the implementation of the actual process. Would the PBX be able to be moved in one weekend for a minimal lapse of service to users? Would the PBX be able to be hooked up correctly once moved? Once moved, would the PBX function correctly, or would the system break in transit, requiring the purchase of a new PBX or repaired parts at the last second and a premium paid on a quick replacement, or would the PBX plug and play correctly? If the old PBX was moved, would the system play nice with newer T1 phone line blocks and, if so, how much would the T1 upgrades cost for the older PBX?

While some of these questions were easy to answer, others simply could not be. One of the requirements for the new phone system was a savings in the budget. Moving the phone system while keeping the phone lines and PBX would not save SampCo in monthly recurring costs, so the T1 option was looked at.

In the current market, partial T1 lines are generally offered at near or higher prices than full T1s due to the configuration required splitting the line. The only potential option was to use part of the T1 for voice and the other for data. This use of the T1 would give SampCo 768 KBPs data up and down and provide the phone lines needed. The new required configuration, which included increased speed of internet connectivity, would need an additional T1 for increasing data plans. This would require aggregation of the T1 with the partial T1 left from the voice lines.

T1 lines, at the time, went for approximately $425 per month. Purchasing two T1 lines would provide voice servicing as well as plenty of data. This configuration, however, would not remove government line charges, long distance charges, or 800 number charges. Additionally, it was found that upgrades to the PBX would cost around $1500 dollars while the PBX model currently owned was listed on E-Bay for less than $1000 dollars. Given these factors and the danger of moving the PBX and upgrading the old box, SampCo decided to not use this option.

The New

A second option was to replace the old PBX with a new PBX. This option would avoid the issues of moving the old PBX (downtime, lapse of service due to breakage, and upgrades) but would include the capital cost of a new PBX and purchase of new phones. While PBXs were not found to cost as much as PBXs when the original box was purchased, a significant capital investment was still required to purchase the new PBX (~$5000). The new PBX would be implemented and prepared before the phone system would cut over the lines and the entire phone system switchover could be as close to seamless as possible. While this option would cost more, the peace of mind it would provide left this as a good option to consider.

The VOIP

The Final option considered was VOIP. While the term VOIP truly covers an entire slough of options SampCo was interested in three major varieties: First, internal hosting with analog lines coming into the office; second, internal hosting using SIP trunking; third, external hosting. Reading about VOIP had provided details on some of the great features, described how it was the future of communication, and would allow SampCo previously unheard of connectivity options, but unfortunately also revealed some of the problems that accompany many VOIP solutions, such as dropped calls, sound quality issues, and call echoing. All in all, if the features could deliver, it appeared that solutions were available for the problems.

Internal Hosting With Analog Lines

Internal hosting requires a software package running on a server that allows SIP devices (phones) to make calls. The server routes calls using the analog lines it has plugged into it. The server can be expensive, as in the case of a Cisco SIP PBX, or inexpensive, in the case of a UNIX based Asterisk box. Given the sky high price of the Cisco box and the incredibly low cost of a UNIX box running free Asterisk software, we didn't truly consider the Cisco box so please do not consider this section to be a discussion of the reasons to use the Cisco option versus an Asterisk server.

The Asterisk server configured with analog lines would be very similar to the configuration of a newer PBX when looking at the system from the phone company. To the phone company, analog lines, or a T1 line is still offered out and charges are still calculated the same way. Long distance charges are the same, government line charges are the same, and line charges are the same. The difference is on the internal side of the equation. Once the phone lines reach the VOIP server SampCo would have full control over the system, able to have telecommuting users connecting to the Asterisk server from remote locations as if they were sitting at the home office. The new SIP phones would also allow a greater amount of flexibility compared to the old PBX style phones. Finally, configuration of the hard layout of the new office would be easier, instead of running phone lines and network lines to each desk, multiple Cat6 network lines could be run to each desk allowing the connection of data or phone irrespective of which port the device was plugged.

Internal Hosting with SIP trunking

Sip trunking is a fairly new option that is not being offered universally yet. The basic idea is that, instead of phone lines coming into an Asterisk server, a network connection allows the server a connection to the phone company and avoids having actual phone lines coming in. This method has many improvements over having lines coming into your server with one gigantic problem.

The improvements of Sip Trunking are in cost and configuration. Since the way the company connects to the phone company is through its data lines, government surcharges are removed which, on SampCo's bill totaled around $400 dollars per month. Second, long distance charges, depending on the phone company you use, can be drastically reduced as the old phone company networks are no longer involved and do not charge by the minute for long distance calling. In SampCo's case, the company was offered a 5000 minute/month block over the entire United States and Canada for $100 dollars a month, in essence removing the idea of long distance from North America. Configuration is simple and straightforward as the VOIP server is simply pointed at a VOIP server on the phone company's network to route calls. Finally, data speed can be improved because instead of a T1 having channels split off for voice all the time, when not being used this configuration allows the bandwidth to flow back into the normal data stream.

The big issue with SIP trunking is in how the local VOIP server connects to the phone company's VOIP server. If the company is not using the phone company's backbone to connect their VOIP servers together, the VOIP phone calls would be routed over the internet. This configuration is possible, and may be increasingly reliable, but for SampCo's situation this configuration was not to be trusted in order to avoid calling troubles. For some companies the SIP trunking option will not work as the VOIP provider should be the same as the data line provider, but if the company has this option, the SIP trunking method holds great value.

Hosted Solution

The hosted solution takes the demarcation of the VOIP server (the Asterisk UNIX server as mentioned before) and moves the line back to the phone company provider. This configuration gives the phone company the power to create new users, phone lines, setup voice mail, troubleshoot voice mail passwords, etc. The phone company also then becomes the single point of failure when something goes wrong with the system. One of the issues with an Asterisk server is keeping a trained staff member around to configure the thing. Of course the hosted solution, for all its merits also tends to make some of the more tricky configurations of VOIP unavailable due to the nature of hosted systems. A per seat charge is also added to the phone bill for each user that is provided phone service on the phone companies servers. This charge is dependent upon the services that the users' phone will be given such as basic dial tone or voice mail.

One could make the argument that hosted solutions are more reliable for external users because an extra point of failure is introduced with SIP trunking. The phone companies lines are point of failure one, while the VOIP server at the client side is point of failure two.

When boiled down, hosting is the option that small to midsized companies should use, while large companies that can afford staff to maintain an Asterisk VOIP server should consider that option.

Decision

With all these options on the table, the tendency was to go with VOIP but the fear was all the horror stories one hears about issues with VOIP. Since SampCo was moving offices it was decided that these issues with proper planning – famous last words!

Not wanting to deal with the old style phone systems and feeling SampCo would save a great amount of money going with a VOIP solution, SampCo set upon making a decision. After everything was considered at the headquarters office, SampCo could offer our users a VOIP system and 3 T1 lines at 4.5 MBPs aggregated with no long distance charges for the same monthly recurring price as the old system with a couple DSLs.

The major savings from this system though, would not be simply in one office, but in external users, future offices, and staffing.

External users would now be supported by the VOIP system, using a DSL line or cable line to provide phone support. Those users would now be rolled into the headquarters office long distance minute block, get headquarters phone lines, appearing as though they were calling from the office, get transfer and conference ability, and they could do all this without the cost of the extra phone line provided at their house.

Further, it was decided that branch offices would be moved into this system if it functioned as intended. The first office, Los Angeles, was given a T1, internal extensions, and external lines, local to their area. The internal receptionist could take over the answering of phones for that office, alleviating a staffing need. All of the calls would then be rolled into the system saving further.

Finally, SampCo could save on staffing as shown in the branch offices no longer requiring receptionists, and in the home office, being able to transfer receptionist functions to any phone in the system.

If the system could work as intended, it would be a no-brainer, there are truly too many great features and abilities with VOIP to mention in this case study. For more information about what your office can take advantage of, please take a look at the references attached to this document.

While there was certainly a risk going with new technology, the savings were evident; not only would SampCo save monthly on the phone charges, but the entire package would now include 3 T1 lines increasing data presence on the internet and the hosted package, removing the need for maintenance staff to perform upkeep on a VOIP server. Finally, since SIP devices (phones) are not proprietary, market forces have lowered prices on phones to the point where SampCo was able to purchase new phones for all users for under $3000 dollars.

Implementation

The process of switching over was planned over a 3 month period in which SampCo's external numbers were setup to be taken over by the new phone company on the transfer date and ported to the hosted VOIP system. It was hoped that the numbers would transfer over within an hour or so when the final go order was given. SampCo also was able to configure direct in dialing for all of their users going from a situation where external callers would have to dial in to a single phone line and be transferred to a receptionist to a situation in which external callers could call into the main line and be transferred but could also dial into a direct dial line for each user.

Old system:

User would call 555-1000 and then transfer to extension 500.

User would call 555-1000 and then transfer to extension 532.

New System:

User could call either 555-1000 and be transferred to extension 500 or simply call 555-1500.

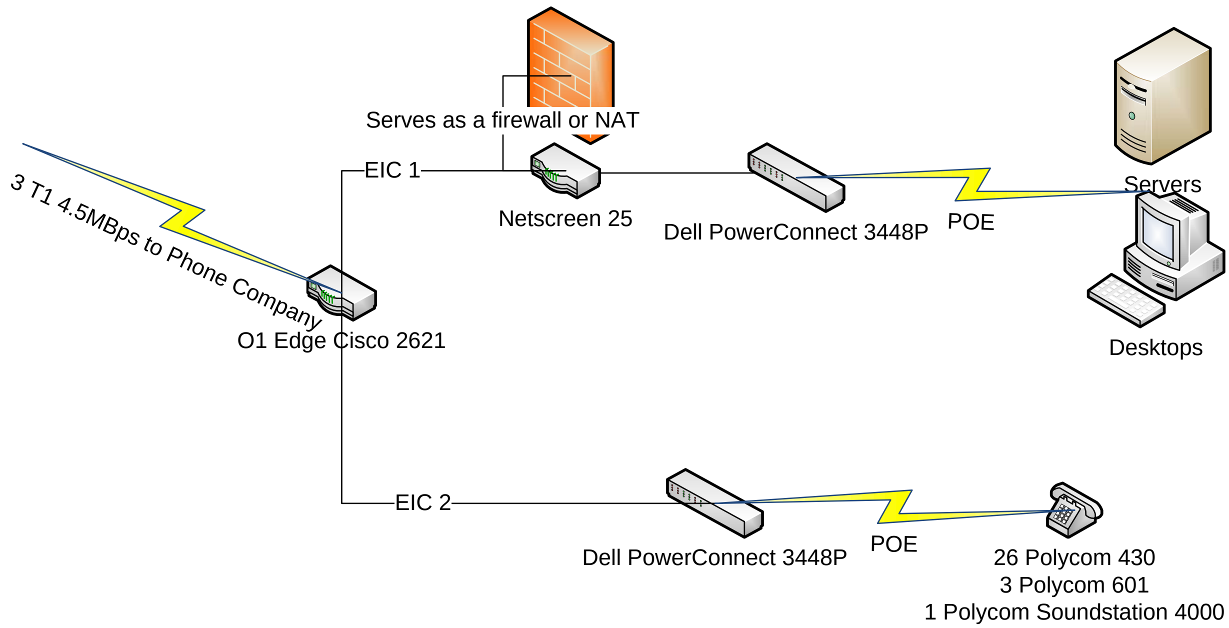
User could call either 555-1000 and be transferred to extension 532 or simply call 555-1532.

The network configuration that was selected was a hybrid system that would allow for aggregation of data bandwidth when calls were not being made as well as protection of the phones network segment so that company data transfer would not intermingle with SIP traffic. This system would place Quality of Service functions in the hands of the phone company's on site router.

There would be two interfaces on the phone company's router, one for the phone system and one for the data side. SampCo would then put its own router between the phone company's router and its desktops and servers to provide firewall servicing and network address translation.

Both network segments would be serviced by Dell PowerConnect switches which would power the phones but could be interconnected with VLAN technology if extra ports were needed for one side of the network or the other.

Consider the following figure:



The above figure illustrates the configuration for the new SampCo Network. The IP segments would be setup in concert with each other for routing between being performed by the Cisco 2621 at the edge of the network. With this configuration, QOS is provided by the Cisco 2621 router from the phone company, completely avoiding the data segment of the network. This way, the plausible points of fault with SampCo would be only the Dell PowerConnect 3448 switch.

While other possible configurations were possible, this configuration gave SampCo a clear demarcation from the phone network to allow the phone company to reliably troubleshoot any problems without SampCo's data network convoluting the process of troubleshooting.

Cutover day nears

As the day to cutover grew closer, the new office was prepared with multiple network ports at each workstation tied back to a patch panel in a server room. Patches were not immediately made between the patch panels and the switches so that only connections that would be used would be turned live.

The phone company sent representatives to verify the layout of the new office and to verify that the configuration would function correctly. The verification team was a subcontracted group that appeared to be knowledgeable and was in and out within a couple hours. This process would almost assuredly be longer and more arduous if the VOIP cutover is in an existing versus a new office.

About a week before the final cutover the T1s were installed on schedule and the Cisco router was turned on. SampCo then put its data router in place to being testing. The data side came up immediately and there were no troubles. The phones had to be configured to talk to the VOIP server at the phone company. This configuration consisted of something similar to configuring an IP address into a printer. The hardest single thing with this configuration is working through the menus on a device that does not have a full keyboard and mouse. Once the configuration was in the phone came right up and started working on the Power over Ethernet – this means that the phone worked without an AC adapter plugged into the wall and worked simply with the Cat6 cable coming out of the wall.

At this early stage call quality was perfect as there was no data being transferred over the lines and only one phone attached. If three T1s could not support one VOIP phone successfully we would have major problems. Luckily, everything worked just fine. All that was left was to wait on the movers and connect up the other phones.

Cutover

The day after moving day the process was to call the phone company and have them turn on the lines and cutover from the old site. At this point all internal functions were supposed to work as expected.

At 10:00am the phone company was called and the process began. Everyone waited patiently as clients calling in received busy signals. At around 10:30am the phone company called and the line tests began. Everything appeared to be working fine at the beginning. Direct dials were checked and functioned correctly, our users could call out and were getting nice call clarity without drops, voice mail was up and functioning, the toll free number was not up yet but was being worked on. Finally, the time was right to check the main phone numbers to see if they all worked as expected.

At SampCo there are 3 main lines that clients call and one line that is the primary line that is printed on all the business cards. The first two lines checked out without any problems. The third, primary line was called and on the other end the old phone system in the old office picked up – not good. This administrator's heart leapt up into his throat and vision began to blur. The call was repeated and again the same response. A call was put in to the phone company to see what was going on. The phone company said they would check into the problem and reply later. They also mentioned the toll free number was now up (this checked out fine).

After an excruciating wait, the phone company called back to give up the bad news. Somehow, through an administrative error, the main phone number had failed to be added to the LNP transfer list from the phone company. This means that in the process of telling the old phone company which numbers to transfer to their system, they forgot one – the most important one. One could spend the next page describing how many times the phone company was contacted to double and triple check the transfer lists but this administrator will assume that the reader understands that the numbers were checked multiple times.

The fix would take another three days while the LNP could be put through again on the primary line. In the meantime the old PBX would be left in operation with a message saying to call the new number. The number could not be forwarded normally due to the configuration of the lines at the old office.

When performing an autopsy on the situation the phone company was completely apologetic and took full blame of the situation. One of the things discovered was that this phone company was much smaller than the large providers. While they seemed to have super technical people working for them who could handle all the technical details, it seemed that their administrative functions lacked as compared to the large institutionalized systems of the large bells.

On the bright side, the VOIP system was functioning perfectly. We had some early trouble due to users not talking directly into their microphones but overall the experience was very good. Users were provided training, however the new SIP phones tend to function like most cell phones so most users are already pre-trained. There are no confusing number combinations to type in to access the phones features. Features are presented in a context specific way on the soft menu that comes up on the phone's LCD screen.

Once the main phone number finally made its appearance on the new phone system some time was available to reflect on the process. Although there were some administrative problems, overall the process was unbelievably smooth. The phone system came up as expected, the data speed was more than we had ever expected and the call quality was crystal clear. The direct dial lines alleviated the use of a full time receptionist so some worker time was saved. While the loss of the phone number for three days was painful, the benefits were clearly worth the switch.

Continued Experience

Over the next six months SampCo has had a few outages here and there as the provider has had different issues. Outages coming in an hour here and there have been put in context with the savings on the new phone system. The support staff at the provider is very responsive and very technical. When called, the caller gets straight through to the network operations center without any major holds as compared to the old phone company that callers would spend hours on the phone waiting to speak to someone who would attempt to boot the call off to another department where the CSR wouldn't have any idea how to help.

A new office has been integrated into the fold, increasing the savings for the overall company and the connectivity. The new site, as mentioned before, used to spend ~$800 per month on their phone system. The new system rolls them in with a T1 allowing for a very reliable VPN between the offices and an interconnected phone system. The price to roll the new site into the phone system comes out to ~$600 a month. The great part is the more sites that are added, the more savings received.

The new VOIP system has performed quite beautifully and this administrator can truly give a recommendation without reservation. With a headquarters in Northern California and a suite in

Southern California this administrator's office is connected through a rented office space's shared internet connection. The day a VOIP phone was plugged in was the same day that clients could call in a local number to the headquarters office and get this administrator in Southern California.

Case Study: Medium to Large Sized Business

McClelland and Hine(MHI) is a Texas based E&S broker writing approximately $50 Million in premium with a main office in San Antonio and branch offices in Houston and Dallas. The agency targets smaller commercial and personal lines business and is approximately 75% to 80% binding authority, 20% to 25% brokerage, and 90% commercial and 10% personal lines. MHI processes over 25,000 policies annually.

MHI has an internally developed agency management and policy issuing system and utilizes DocuCorp policy issuing and ImageRight scanning systems. The IT department consists of the CTO and two systems analysts. The Houston and Dallas offices are connected via T1 lines to San Antonio.

MHI's existing phone system provided the ability to utilize a central telephone reception service in San Antonio for all offices.

MHI was using an Intertel Axxes 64 PBX System under a 36 month lease at $3,508 per month with a June 30, 2007 expiration. The renewal cost for the identical system was $1,763 per month totaling $63,472.68 for a three year term. Upgrading to an IP PBX system would have cost $2,101.65 per month. In addition, certain desired modules and training and internal maintenance would have increased the cost substantially.

VOIP Decision

MHI was faced with an imminent telephone system decision as the current system lease was expiring within 8 months.

The key considerations in selecting a new system were:

a) Ability to interface with the management system.

b) Ability to control service and productivity enhancements.

c) Long term savings

d) Comprehensive reporting capabilities.

MHI began an extensive review of its phone system in November of 2006 and investigated the following options:

a) Renew existing system

b) Upgrade existing system

c) Implement VOIP Hosted Alternative

d) Develop Internal VOIP Solution

While the existing system was functional and provided basic needs service, it did not provide the needed flexibility to integrate with MHI's management system. In addition, vendor service had deteriorated substantially during the review period.

Given the service issues and the increased cost for upgrades and additional modules and anticipated training time, upgrading the existing system quickly exceeded the costs of a VOIP alternative.

Two hosted VOIP alternatives were considered: Fonality and Digium. Neither option provided a flexible and controllable system that could be easily integrated with MHI's management system.

MHI decided to develop and install its' own VOIP telephone system using Asterisks Open Source Software. The initial analysis indicated a cost of approximately $32,585 for software and hardware. The final cost was:

Actual Cost				
Quantity		Per Item	Total	Location
1	Software	$0.00	$0.00	San Antonio
3	PE 1950	$2,076.00	$6,228.00	
3	Sangoma A101d	$999.00	$2,997.00	
1	Misc	$350.00	$350.00	
57	Phones	$163.04	$9,293.28	
0	POE 20 Port	$1,400.00	$0.00	
1	ATA Adapter	$674.00	$674.00	
13	Phones	$163.04	$2,119.52	Dallas
0	POE 20 Port	$1,400.00	$0.00	
1	ATA Adapter	$465.00	$465.00	
22	Phones	$163.04	$3,586.88	Houston
0	POE 20 Port	$1,400.00	$0.00	
1	ATA Adapter	$465.00	$465.00	
Total			*$26,178.00*	

In the pipeline are 5 POE 20Ports at a cost of $667.00 for an additional cost of approximately $3,335.00.

During implementation, MHI had to purchase additional T1 Cards with built in Echo Cancellers when it was discovered that regular T1 cards with Echo Canceller software would not handle more than 10 lines.

In addition, implementation required the full time services of a system analyst for approximately 90 days.

Implementation

Implementation began with the assignment of development to a system programmer with six years of Linux experience and no Asterisks or telephone system experience.

Full implementation was completed in five months as follows:

- February 15th, 2007- Initiated implementation with download of Trix Box (Fonality Mfg.) open source software, which included Asterisks Open Source Software and a web interface product.
- February 21st, 2007- Determined Trix Box and Web Interface would not function properly for a multi-office, centralized phone system configuration.
- March 6th, 2007- Decided to continue with full internal development of Asterisks VOIP system. Utilized published material and on line chat with Asterisks software developers and other resources for development guidance. Intensive development was started immediately. We utilized “Free PBX” web interface software to replace Trix Box web interface software.
- March 30th, 2007- A functioning test system was initiated with five test stations (phones) in San Antonio.
- April 18th, 2007- Testing was completed and T1 Cards were purchased.
- May 20th, 2007- San Antonio went live for all external incoming and out going calls. A parallel Intertel system continued functioning for intra office calls from and to Houston and Dallas.
- May 21st, 2007- Initial T1 cards and Echo Canceller software proved ineffective for high volume system.
- May 22nd, 2007- Replacement T1 Cards with hard Echo Canceller hardware were installed.
- June 16th, 2007- Dallas bought on line
- July 15th, 2007- Houston bought on line.

Throughout the process, each step, each new software package, and each piece of hardware was tested extensively.

Implementation and Usage Issues

The following were some of the issues and challenges MHI faced:

a) Faxing-We had issues with three different models of digital to analog converters used for internal fax machines. Use of Hylafax with IAXModem software as a fax server has receive issues with “Brother” fax machines. This is still a challenge.

b) New phones were different- i.e. user training

c) Initial and periodic, but very infrequent line disruption- echo, jitter.

d) New Voice Mail protocols- i.e. user training

Training of office personnel was not sufficient. An informal learning system developed, which delayed full system use.

Productivity Enhancements

The following are areas where the new telephone has or will be utilized:

1) Established full conference call capabilities including pin authorization and web interface with 30 minutes of programming.
2) Maintain a user data base of telephone and computer equipment, which automatically updates the user database.
3) Maintain a MYSQL call database.
4) Developed on computer screen caller ID as well as pushing database information to the phone display.
5) Developed “click to call” which allows user to click computer to automatically initiate call.
6) Developed text cell phone notification of office voice mail.
7) Rerouted 1411 calls to Goog411 to eliminate information service charges.
8) Develop automatic DashBoard to generate caller information on ring in.

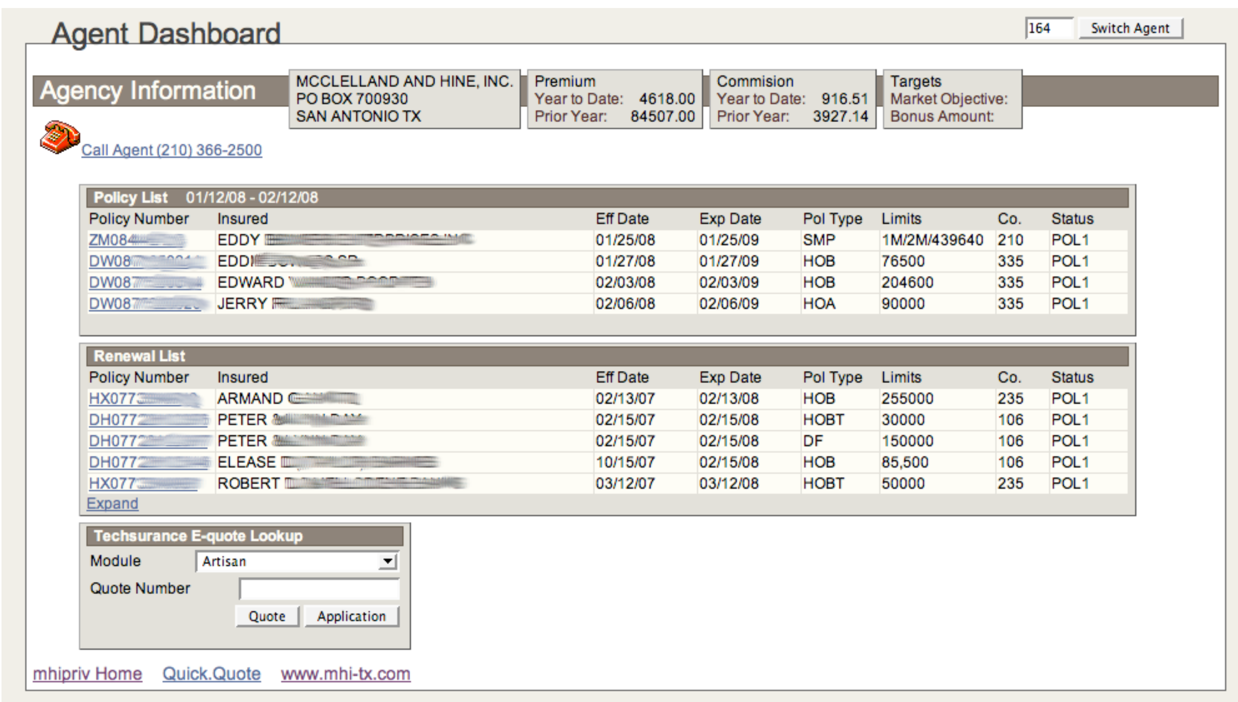


Figure 1. Agent Dashboard that automatically opens when a telephone call is identified. This project is currently in testing.

Appendix

Common VOIP Issues

Voice quality issues:

Voice quality is determined by many things with VOIP. Think of your network as a road system with freeways, two lane highways, and back country roads along the path to a call's destination. Think of the phone call then as a steady stream of cars from one destination to another. If one of these sections gets backed up or in a traffic jam, so to speak, the packets (cars) travelling from one destination to the other start coming in at delayed intervals or unexpected intervals. These packets are bits of data that, when delayed, cause all sorts of problems in a VOIP infrastructure. Here are some tips on avoiding these types of issues

- Your internal network is the one you have the most control over. If your calls do not even make it out of your network without issue, you have a severe problem. The first step is always QOS. QOS (Quality of Service) can be implemented, and is generally available, on most routers and allows one to tell the router that voice traffic should be dealt with before anything else – sort of a jump to the front of the line ticket.
    - Unfortunately, QOS is not always a panacea to all issues that occur on your internal network. If there is heavy traffic that does not leave your network, such as a disk cloning operation or video transfer it is conceivable that one would have issues with Voice Quality. To put it another way, voice quality suffers when quality is downgraded before the router gets a chance to perform QOS.
    - A common way to deal with voice quality issues on your internal network is to split out your network into two VLANs with two interfaces on your outbound router. One segment is primarily used for phones while the other is primarily used for data. By not allowing your data network to intermingle with your voice data, your packet switched network can do a better job of getting data right to the router so that QOS can take place. Many VOIP phones also have network jacks on the back to allow pass through while performing their own QOS. This does not; however preclude other phones from allowing pass through of a BitTorrent stream while the receiver remains on the hook.
        - Separating your phones from your data network also carries value when troubleshooting. By isolating your phone system you can effectively rule out issues on your data network causing phone problems.
- Your outbound connection to the internet is the second bit you have almost (Budgets always seem to kill that extra T-1 you wanted right?) complete control over.
    - A typical VOIP call generally costs your network about 32 kb/s depending on the Codec used to compress the audio.

- DSL and cable modems, while having huge download, usually have little bitty straws for uploading data. An average DSL connection usually gets 784 kb/s down and 128 kb/s up. This means that your typical DSL can handle about 4 calls simultaneously in a perfect world. Obviously, worlds are not perfect, and one would be lucky to get 3 calls working nicely at the same time on a 128 kb/s upload.
    - This means that your DSL can handle the audio from the caller quite well but the audio coming from your phone would have to compete for almost one quarter of the entire upload speed of your DSL.

One Sided Calls

In the case of many small businesses, VOIP systems are configured behind NAT routers which employ a method of taking one IP address and sharing it with multiple hosts. VOIP takes issue with this situation and, in many cases, will simply not work. In other cases, one might experience one sided calls. To better explain one sided calls, one must describe, briefly, how a VOIP call takes place.

- The SIP protocol is a signaling protocol that attempts to make a connection to another SIP device; this process is done in TCP and, therefore generally makes its trip through a NAT router successfully. The problem occurs when the SIP signal starts a UDP signal in the audio broadcast of RTP. Unfortunately, many NAT devices cannot strip away the local IP address of RTP packets in place of the public facing IP and, therefore the call does not know how to reach one of the parties. This situation can lead to a phone call looking like it is working with no audio present.

- The unfortunate part of this situation is that many higher end routers support NAT transversal for VOIP but situations in which companies use VOIP tend to be an attempt to get bang for the buck, leading them to the low cost routers such as Linksys and Netgear. Most of these low cost routers deal with VOIP NAT transversal very poorly.

Dropped Calls

Dropped calls can happen for a number of reasons but in the case of the only symptom your phone system is dropped calls, one can usually tie this problem down to server registration. The phones in your network must register with their VOIP server every so often so the VOIP server knows that the phone is in operation. In the case where a phone does not register correctly, the server will simply drop the call. By changing registration settings, usually in the phone’s config file, some of these problems can be eliminated.

Suggested Reading

http://www.asterisk.org/ - Asterisk is the foremost open source VOIP server software available. Used in one of the case studies in this document, Asterisk is very reliable and allows for custom integration.

http://www.voipuser.org/forum_index.html - VOIPUser is one of the better forums on the internet for VOIP related topics. Troubleshooting, product support, all the way to rollout tips is contained in these forums.

http://en.wikipedia.org/wiki/Voice_over_IP - Wikipedia has a wealth of information about VOIP and the background of the protocol suite.

http://www.voip-info.org/wiki-NAT+and+VOIP – This page and site is a great article on the issues surrounding NAT and VOIP interoperability.