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Introduction

The purpose of this document is to summarize the knowledge acquired by Legal Aid of the Bluegrass (LAB) as part of the Legal Service Corporation Technology Initiative Grant. Contained within is the knowledge gathered over a 3 year period, and it is the hope of LAB that this information will be used to seed success for Open Source VoIP telephone systems in other LSC programs.

What is VoIP

Definition

***Voice over IP** (**VoIP**) is a general term for a family of transmission technologies for delivery of voice communications over IP networks such as the Internet or other packet-switched networks. Other terms frequently encountered and synonymous with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, and broadband phone.* (*http://en.wikipedia.org/wiki/VoIP*)

VoIP means simply that your voice on a telephone (or other device) is divided up into short sound clips sent over a network to another device and reassembled to deliver the audio to the device on the other end and vice versa for two-way communications.

VoIP can represent different ideas to many people. It has become a widely used buzz word in the industry due to the fact that VoIP has completely changed the landscape of the telecommunications market. Adding a level of flexibility, integration and choice to the market. Prior to VoIP, telecommunications needs were meet by local telephone companies and often was the only choice. The local telephone companies were protected from outside competition by existing as a utility or regulated monopoly.

Assumptions

Many assumptions are now made about what VoIP provides and typically come from marketing efforts. VoIP could mean some (or more) of the following.

- Free Long distance
- High Tech telephones
- Feature rich PBX
- Very poor call quality

Yes, VoIP could mean any of these things, but generally VoIP is just the method at which these products are delivered. When exploring the market do not just accept VoIP as an all-expenses paid trip to massive return on investment or a preconceived collection of benefits or features.

What is Asterisk

Open Source Software

Open Source Software (OSS) is software developed by a community with a common need and is created under a "Public Domain" sort of license, which means the software is free to use and may not be resold. Also, the source code of the software is open to the public as well. This means that the software can be openly evaluated and modified to fit any particular needs.

Many successful business models exist under which OSS is developed. Often OSS is developed just to create a support market for the developer. Companies that manufacture hardware products will start OSS projects to create the software needed for their products. This happens to be how Asterisk was created, by Digium, Inc. A manufacture of telephony hardware for use with Asterisk.

The OSS community produces thousands and thousands of software products; however, not all are created equal, are supported, or work at all. When choosing OSS software a watchful eye must be placed on the state of the community developing the software, the existence of recent activity, and the support options available.

Asterisk

Asterisk is an Open Source Software (OSS) that turns a computer into a voice communications server. Asterisk is the core piece of software that powers various PBX systems, VoIP Gateways, Conference servers and more. It is running in small and large organizations and everywhere in between. More information can be found on http://www.asterisk.org/asterisk.

Other Companion Open Source Projects

Asterisk is the core software enabling voice communications on a server. However, Asterisk by itself does not provide a graphical interface for configuration. It solely relies on configuration files to provide details for every aspect of how to perform voice communications. Asterisk is a very complicated, feature-rich application and has prompted others to create graphical interface projects like FreePBX which is a very popular OSS configuration tool for Asterisk. However, FreePBX also brings many Asterisk dial plan scripts that provide additional powerful PBX features. There are also many other tools and packages that can be used with Asterisk some of which may be OSS and some commercial packages.

Asterisk is also delivered as a packaged distribution. These packaged distributions are basically complete system installation software, available from both Open Source and Commercial sources. Simply download the CD-Rom image, create the installation CD, start it up in a server and install the software. After installation is complete you have a fully ready-to-go communications server waiting to be configured. Legal Aid of the Bluegrass (LAB) uses such a packaged distribution called the Trixbox Community Edition and has had great success. Trixbox installs CentOS (Linux), Asterisk, FreePBX and other various OSS packages, all in one installation package.

Other Open Source VoIP options

Most of the major commercial PBX providers, such as Nortel, Avaya, InterTel, Cisco have created successful VoIP products as well. LAB's has had some experience shopping for these types of systems but have generally been well out of financial reach and is what lead LAB to pursue the use of Open Source solutions. The commercial solutions often require the purchase of both hardware and licensing and in some case support contracts may be required as well. LAB's only practical experience was with a commercial solution was the Nortel BCM system purchased in 2006 and found it to be an expensive and time consuming to modify. License requirements often added excessive lead times from the reseller and vendor to add new features or expand the system.

Another option is VoIP Hosted PBX in which a communications server (typically Asterisk) is hosted off site at a data center. Connections are typically provided over the Internet or a dedicated point-to-point connection. These Hosted solutions are typically lengthy contract based solutions that come in a wide range of terms. LAB has not had direct experience with any Hosted PBX solutions. It has, however, spoken with many organizations that have largely been dissatisfied with reliability, management or call quality issues. Extensive negotiations, contract review and reference checks are highly recommended measures to use when evaluating this type of solution.

Benefits and Costs

Features

The following is a list of features that come standard with most Asterisk based systems.

- Caller ID
- Blacklisting (Block callers by Caller ID)
- Call Waiting
- Call Transfer
- Call Conferencing
- Conference Center/Bridging
- Do-Not-Disturb
- Call Forwarding
- Call Parking
- Call Detail Reporting
- Dial by Name Directory
- Interactive Voice Response (Auto Attendant)
- Time Condition Call Routing
- Call Queuing
- Call Back
- Hold Music/Recordings (MP3)
- Application Integration
- Extended Call Reporting
- DISA (Direct Inward System Access)
- Dictation
- Follow Me
- Paging/Intercom
- Ring Groups
- Graphical Call Manager
- Day/Night Controls

Integration

LAB has completed several development efforts to better integrate the new VoIP telephone system with its business systems. The in-house development efforts are Click-to-Dial, Inventory Endpoint(IP Phone) Management , Queue Manager and XHTML Directory. All of these efforts have had great success and LAB will continue to innovate.

The Click-to-Dial system allows the user of the case management system (CMS) and Intranet website to simply click a button or intranet link to place a telephone call. The system also determines if the call is long distance, which requires dialing 1 + area code. This saves time in Intake call back operations because the staff no longer has to re-enter a phone number from the CMS to the physical phone on the desk or determine if the number requires long distance toll dialing.

The Inventory Endpoint Management system uses the existing inventory data to manage the location and configuration of endpoints. The endpoints are assigned to users within the inventory data and a process generates the endpoint configurations and uploads them to the VoIP servers. This allows LAB to easily add or move endpoints within the organization. Also, the data being integrated provides excellent data for accurate directory and location information, a foundation for the XHTML Directory that will be discussed later.

The Intranet Queue Management interface was built to help users sign in and out of the telephone queues. The standard process for this was completed via the endpoint though the use of feature codes and extensions. This process did not provide adequate validation and agents often made dialing errors that would not correctly logon the on or logoff their respective queues, often resulting in a poor experience for callers. The new system gives the agents a graphical user interface that provides the logon/logoff functionality as well as information regarding their current queue status, something not provided by the endpoint. Additional work is currently being planned to provide queue status and logon/logoff functions on the endpoint's large LCD screens to indicate queue membership.

The XHTML Directory allows the Polycom Endpoint's XHTML browser to browse Intranet directory data. This allows users to view and dial from a master directory for the entire program from the endpoint. This is only the beginning of the use of this technology. Many more features are to be developed over time such as Speed Dial, Local Resource directory and much more.

Costs

The Asterisk software is free to use. As previously mentioned, the Asterisk software is Open Source, leaving the only expense to be the purchasing of hardware and support resources. A typically single office system will require a server and endpoint (Telephone) hardware. If the system needs to be connected to a local telephone service, additional interface hardware may be required. Generally this cost is around $200 - $500 per seat, which largely depends on the type of end points and interface hardware chosen. PSTN, VoIP and Internet connection cost and quality vary greatly depending on the area of the country. All of these services can range from $100 per month for a couple of analog telephone lines to thousands of dollars per month for PRI, T1 and fiber optics services. Real world cost data of our LABs remote office deployments are provided in Appendix A.

Design

Setting Requirements

One of the hardest things to do is to set the requirements for a new phone system. Be sure to understand the available features and try to apply them to your business needs. Does the organization need simple telephone service or maybe something complicated like a call center or something in between?. It is important to do as much up front planning as possible. It is a good idea to educate and survey staff about the existing systems in an attempt to determine the current requirements and what may be missing. *See Appendix F for an actual survey with response data*

Core Selection/Distribution Selection

LAB started using the Trixbox packaged distribution. This is a packaged distribution that includes an installation CD that will completely install and configure a server with Linux, Asterisk, FreePBX and other supporting software. This is a great place to start evaluating, testing or implementing. LAB currently has 4 Trixbox systems running in production. There are other packaged distributions, such as AsteriskNOW and PBX in a Flash.

One issue that arises from the use of a packaged distribution is that often times the OSS used is forked from its project. This means that the project that created the packaged distribution has modified the source code from the host project. This often will cause a situation where bugs fixed in the original project will not be available to the packaged version. Packaged distribution can also contain a lot of extra software that exceeds requirements and adds unnecessary complexity. LAB has recently begun to install Linux, Asterisk and FreePBX as separate packages which are a bit more complicated to install but offer greater version control. This combination has been found to meet all of LAB's requirements.

VoIP, PSTN, Internet

Choosing a VoIP or PSTN (Public Switched Telephone Network) for external or outbound call routing is a complicated choice typically driven by the local market. Asterisk is a VoIP based system, so it makes sense that it can natively communicate in this manner without any additional requirements. However, if you would like to connect to the PSTN via analog lines or digital circuits with the telephone company, it will required the purchase interface hardware adding additional capital expense.

Generally speaking, VoIP is always going to be in use to connect your End Points to the server. So when you pick up a VoIP telephone it will connect via VoIP to the server. The Asterisk server at that point will connect the call to a VoIP provider or PSTN. Note that interface hardware can be purchased to connect analog telephones and fax machines to the Asterisk system as endpoints.

VoIP calls to the outside world will typically be placed through an external provider. Each provider will have their own contract terms, prices and offerings. This is a good way to test your system, as many providers offer month to month low cost services that you can used to test. It will also give you a good idea of how to connect to different providers.

PSTN connections will require the purchase of interface hardware. LAB has generally used Digium products. However, there are other manufactures such as Sangoma and Rhino which appear to be other popular choices.

First look at your current voice services which typically is the local telephone company. Gather 6 months of billing and call usage data. Get a firm understanding of your costs. If possible, determine how many calls are placed each month, how many are long distance, if there are a lot of calls being placed to another office or partner?

Second, take a look at your internet service. Know how much bandwidth you are purchasing and how much it costs. If VoIP services are selected this will be a very important connection, as it will directly dictate the call quality. The quality of the Internet connections will dictate the audio quality of the VoIP communications to a VoIP provider. When selecting a VoIP provider always check the latency of the connection.

See Appendix B for a visual diagram of how Asterisk integrates with various forms of communication.

End Point Selection

LAB reviewed and tested many types of VoIP endpoints or telephones. It found quite a difference in physical, audio and software quality. LAB reviewed softphones, Grandstream, Polycom and Cisco.

Software Based Phones – LAB tested several software based phones with the Asterisk based phone systems. The best one was an OSS package called X-Lite by Counter Path. LAB's impression of this software was that it works well and is an invaluable testing tool. However, it quickly determined that the soft phone was going to be limited to special case scenarios. A success story using this type of endpoint is documented in the success stories section of this document.

Grandstream – LAB purchased two of the Grandstream GXP2000 endpoints. These units were priced very low at around $80 per phone. The units worked well but were missing some important features. The units had to be plugged in to the wall for power as they did not support being powered by the Ethernet connection. They also presented quite a bit more echo during use than the soft phones we had tested before. Also, the units also did not feel as if they would last very long. The unit was light weight and had a cheap feeling construction.

Cisco – LAB never tested a Cisco phone due to the research done on community forums. While the phone was very popular it was also very unsupported by Cisco without a costly relationship with an authorized reseller. Very special requirements and support contracts were necessary to obtain a new version of the firmware for the devices. Given that LAB was working in such a fast changing environment LAB really did not want to have to maintain expensive support contracts just in case we needed the latest firmware release was needed.

Polycom – LAB tested Polycom's IP Soundpoint series of phones. This included several models. IP330, IP501, IP550 and IP650. All of these models supported Power over Ethernet (PoE) which is a method for powering the phones via the Ethernet network. This offers the advantage of keeping telephones up and running in the event of a power outage if the PoE enable Ethernet switching equipment was supported by a UPS (Uninterruptable Power Supply or battery backup). All of these phones have speaker phone and head set capabilities. The speaker phone feature on these devices is exceptional. All the firmware except the most recent release is readily available for download directly from Polycom without the need for support contracts or resellers. The level of configuration is extensive, easy to use and are highly documented.

- The IP330 ($130) is a smaller 2 line phone with a 2 line non-back lit LCD display. These phones were a great phone for common areas such as Copy rooms, break rooms or temporary work spaces. Headsets are also supported.
- The IP501 ($160) was a cheaper predecessor to the IP550. The phone only supports 3 lines and requires some special cabling to support the PoE feature. While the LCD was large it was not back lit. This phone did not appear to fit our environment. Turns out this line of phones is due to be discontinued.
- The IP550 ($260) and IP650 ($300) quickly became the models of choice. Loaded with features, the IP550 and IP650 both came with easy to read back lit LCD displays which offer feature rich application support such as a caller log, directories and XHTML browser. The IP650 has all the same great features as the IP550, but it possesses 2 more line buttons for a total of 6. It also supports the addition of up to 4x 14 button side cars for management and receptionist use. One nice feature of standardizing with these models is that the users will be comfortable using either of the phones.

Accessories – LAB tested a verity of Plantronics head set models. Plantronics head sets were first to be tested as they are the de-facto standard for call centers. Ranging in price from about $70 - $330 we first started with the lower end wired headset. They worked well but we also tested the more moderately priced wired noise-cancelling headsets; which proved to provide much improved voice quality. However, after our Intake staff moved to a new cubical style office environment in our new building a request was made for Binural or Dual Ear headsets and LAB now provides that option. The headsets turned out to be a great success. However, LAB wanted to investigate the use of Wireless headsets. While considerably more expensive, these proved to be very valuable to the receptionist staff and have been consistently providing the same high level of voice quality.

Support Options

Asterisk and its associated OSS companions are complicated requiring technical knowledge to install, configure and support. Choosing a support model greatly depends on the complexity of implementation and the existence and skill set of in-house technical staff.

In-house support requires that organization has technical staff that is familiar with Linux and Networking. The staff also must have an online research skill set for sifting through internet search, wiki, and online forms for support information. The nature of OSS is that all inclusive documentation rarely exist.

Asterisk does have many commercial support options. The bigger names that come to mind are Digium, Trixbox and FreePBX. All have a pay for support services. LAB has not had to use these services and thus does not have experience with them.

Local Support in some areas is becoming increasingly available. The popularity and success of Asterisk has created a market for these services. Many local telephone system providers are now offering Asterisk based products. Keep in mind that Asterisk is very supportable from a remote stand point and that if no resource exists in the immediate area, service can be found a little further out.

Operating System Selection

Asterisk runs on Linux; however, a recent project has been launched which allows Asterisk to operate in a windows environment as well. However, support may be very limited and bug fixes may be slow. Asterisk will run on just about any Linux distribution. Selecting a Linux distribution will greatly depend on the supporting staff or service. The most popular choice is CentOS version 5 better and is, known for being a few steps behind in their repositories of the newer "bleeding edge" updates offering improved stability. I might add that most of the package distributions use CentOS.

Facilities

Premise Wiring

Asterisk endpoints typically use Ethernet for IP connectivity. Thus the location in which Asterisk is to be deployed must have sufficient Category 5 wiring to support the use of Ethernet. This typically exist in an office with existing workstations. A complete inventory of network drops should be complete to help determine if additional wiring needs to be completed. LAB found that when possible having 2 network drops at each location is ideal, 1 network connection for the computer and 1 for the endpoint. Some endpoints have a built-in 2 port Ethernet switch that allows an endpoint and computer to share one connection. LAB has found that this adds minor complexity to the cabling for the desk, increases the points of failure and the phone becomes difficult to relocate around the work area.

Existing Service Demarcations

Document the location of any demarcations within the building. This will help for both troubleshooting connectivity problems and the installation of new services. Often phone equipment and computer equipment are not located in the same part of the building. Older phone switching equipment were often installed in basements or very small spaces. Asterisk being a software product, running on a server may influence the operating location. LAB typically would extend wiring from the demarcation points to the server closets in the remote offices.

Legal Aid of the Bluegrass

Back Story

As a result of the Kentucky Legal Services Planning Initiative in January of 2002, Legal Aid of the Bluegrass undertook its second merger, merging with Central Kentucky Legal Services, Inc. The program service area expanded to 33 counties with a poverty population of over 137,000. The combined staff of LABG grew to 55, spread between the four offices in Lexington, Covington, Morehead, and Ashland. The challenge to this newly merged organization was to create a technology framework to increase access and integrate services provided to clients, enhance the effectiveness and efficiency of staff, and expedite communication between staff and the community-based organizations with which it partners. In the five years since that merger LABG has accomplished a great deal of those goals with computer networking through the use of ADSL connections, cable, Internet T1s and point-to-point T1s. The remaining challenge is to upgrade old, limited telephone systems in its offices.

While attending a disaster & recovery session at 2007 TIG conference, Legal Aid of the Bluegrass (LAB) learned of Asterisk from William Guyton of Legal Services Alabama who was presenting the session. Following the conference LAB applied for LSC Technology Initiative Funding to help underwrite the cost of a Kentucky prototype voice over IP system using the Open Source Asterisk software to achieve "the most economical and effective delivery of legal assistance" in LABG's service area, and possibly for the state as a whole. The project connects LABG's main office to 3 outlying offices, integrate two bar association sponsored Pro Bono programs that already share case management capabilities into the telecommunications system, help streamline intake through enabling a single 800 intake number, enhance client service through a custom call tree/IVR system that is easy to use and connect advocates in all offices to each other and clients. All of these functions provides the foundation to facilitate other legal services programs and partners to share in the project's successes.

Old outdated, feature-limited PBX systems in 3 offices cause significant issues with client service. Specifically, when LSC recently conducted Legal Services Corporation when conducted a quality review, they pointed out the problem for LAB's intake in their report dated January 31, 2007

"LABG's intake system is a work in progress. The program's intake is generally unified in terms of policies and procedures. …there appears to be some resistance to further consolidation of the intake system, based on the perception that it would deflect scarce resources from extended service cases only to produce more advice and brief services. However, a well-functioning distributative phone system should allow the Covington office to take advantage of intake staff in other offices without creating the need to move resources to Covington….Many of the obstacles to improvement of LABG's centralized intake can be overcome by replacing the phone system, which is weak throughout…"

Outdated telephone equipment causes problems for clients, staff and partners—the Northern Kentucky Volunteer Lawyers, Inc. (NKVL) and the Fayette County Bar Association Pro Bono Program, Inc. For example, intake is centralized in the Covington office where it occurs on a "callback" basis. LABG intake staff also takes the NKVL's intake calls by callback. For calls coming to the Lexington office, clients call a separate 800 number and non-emergency calls are sent to a voice mail system where Covington intake staffs retrieve the messages by calling the Lexington office. Sometimes frustrated callers end up talking to local Lexington staff who often will perform an intake, thus effectively moving people to the head of the line and bypassing others who have waited. Sometimes, intake staff in Covington then calls back the applicant after Lexington staff has performed an intake, creating greater amounts of inefficiency. In Ashland, the applicant for service calls the local office to hear a long message before getting the information to call yet another 800 number if he or she is seeking help. In Morehead, intake occurs locally, but old equipment limits the capabilities of intake staff and efforts are disjointed. The Fayette County Bar Association Pro Bono Program, Inc. performs its own intake and shares LABG's client management software. When an applicant calls the local Lexington office but the case is appropriate for Pro Bono referral, LABG staff must give the applicant yet another number to call.

The old systems in several offices do not allow for caller ID, call forwarding, and efficient, effective interactive voice response systems; therefore, advocate efficiency is affected and client services are not as streamlined as possible. For non-English speaking clients and the attorneys and paralegals representing them, the tired, old systems cause problems with voice mail translations.

LABG anticipates that the client groups to be served include any applicant for service in the program's service area who calls a single 800 number. That includes the 138,000 poor people in the service area who encounter legal problems, vulnerable immigrant victims of domestic violence regardless of immigration status, the elderly in the communities, and frail and vulnerable nursing home residents.

Increasing advocate efficiency translates into an increase in the amount of brief service and advice that LABG can provide. Of particular importance is the ease with which those who speak a language other than English as their primary language can leave voice mail messages for translation services. Clients or applicants experiencing emergencies or not adept with voice mail and call back systems will benefit when they can immediately reach a live person.

The project will ease client frustration, increase advocate efficiency and enhance client service by eliminating the inefficiencies currently built into the system and easing the amount of awkward steps clients must take to access services. The examples of how this will occur are many.

The project is unique and practical, too. For one thing, from this project, LABG could offer to other LSC-funded programs, at the very least, lessons learned in its successes and failures and, at most, replicable systems or the potential for practical, feasible integrations. A unique aspect is bringing the Pro Bono programs right into the integrated system.

Research/Trial

In 2007 Legal Aid of the Bluegrass (LAB) started researching Asterisk. This began with the Trixbox packaged distribution. The Trixbox community had several "getting started" articles that LAB followed. Reusing an old outdated workstation to host the software LAB began testing Asterisk using a soft phone to connect to the system and investigate various features.

LAB continued testing and setup an account with several different VoIP service providers in order to begin testing calls to the outside world. Quality was sometimes poor and we largely suspected our internet connection speeds were to blame. Analog PSTN interface hardware was purchased and tested with great success and achieved a proof of concept.

LAB continued to research and test various aspects of the system and purchased interface hardware, VoIP End Points and improved server hardware. This process continued though implementation and underscored the advantage of Asterisk's low cost and commitment. LABG was able to test software and hardware at very little cost or commitment to a provider. During this testing LABG also realized the added benefit of being able to modify and support its telecommunication system in-house at a moment's notice. Previously these changes would have at least a day or more lead time resulting in additional costs. It might also be noted that this in-house expertise greatly improves DB&R ability.

Implementation

LAB began implementation with a pilot in the Intake department. The office had a PRI digital circuit with 23 channels of voice services that the current provider was converting into 8 Analog lines. This left 15 unused channels and LAB had the additional used channels configured as separate PRI for testing. A Digium TE110P PRI interface was added to the Asterisk server and configured which gave the system access to 15 channels of voice services. Endpoints where purchased and installed for all the Intake staff. The pilot began with Intake staff placing outbound callbacks using the new telephone system. Staff reported improved call quality over the existing system and they liked the new Endpoints, which were much more modern than the old telephones.

Following the successful pilot, LAB made the decision to go live with the system. Additional endpoints were purchased and installed, and a training program developed and executed. The system went live and was a huge success. LAB continued to deploy the new systems to the remote offices over the following two years with similar success.

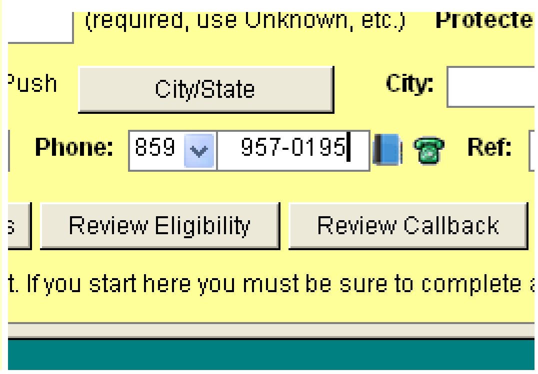
Following the Probono, Ashland and Morehead offices there were frequent reports of poor audio quality during calls. The quality issues were often sporadic or a direct result of minor network issues which made troubleshooting very difficult. Before the deployment to the last office LAB surveyed staff to determine how wide spread of a problem existed. *See Appendix C for the survey and results*. After surveying the staff LAB determined that the problem was somewhat minimal and most staff were accepting of the minor issues given the significant cost savings the new system represented. This view was expressed at the training in the Lexington office. LAB's technical staff continued to researched the audio quality issues extensively and found that two issues existed. First echo cancelling hardware firmware was out of date and updates were not being applied due to a software driver issue. The second issue may have been digital to analog conversion times and most of those issues were resolved with PRI digital circuits being installed, which had recently become more affordable. To date little or no troubles have been reported. *See Appendix C for complete deployment details for each of LABs offices.*

Success Stories

Application Dialing - Long Distance Lookup

Asterisk provides an API that allows functions to be called from external applications. LAB has leveraged this technology to provide one click dialing of phone numbers and extensions on the LAB Intranet and directly from Kemps.

While at their desk, users can now simply click the extension of the person they would like to call directly from the LAB Intranet, as shown in the image below. Once the user clicks the extension, the user's phone begins to ring. Once the user answers the phone the call is then placed to the person they wished to call. Below you will also find a screen shot from the Kemps CMS with a new green phone icon which allows the user to dial number directly from the Kemps CMS



Ashland Intake Routing

During the installation of the Ashland office it was discovered that the previous voice system instructed the callers to hang up and dial 1-800-888-8189 if they did not have an open case with LAB. While configuring the new automated attendant on the Trixbox, LAB configured the system to allow the caller to dial 1 if they did not have a current case with LAB, at which point the system would connect the caller to the Covington office without having to hang up and dial any other numbers. This also provides the caller with another toll free option to reach LAB; specifically, the caller calling the Ashland office in the Ashland area would be dialing the local Ashland office number. Selecting 1 would send the call to the Covington office toll free to the Client and LAB.

Internal Office Connectivity Routing

LAB has deployed the Trixbox system in four offices: Covington, Ashland, Morehead and the ProBono Partner program. All of these systems are now connected together over the Wide Area Network (WAN). This allows LAB to route calls from one system to another. All offices with the new system can now call other offices simply by dialing the extension of the user they would like to contact. This places the call over the existing WAN avoiding long distance charges and time spent navigating automated attendant systems.

In addition to the extension dialing, the systems in each office have addition programming that redirects calls over the WAN from one office to another. For example, if a user in our Covington offices dials the main number of the Morehead office that call would be placed over the WAN rather than it being placed on regular phone service, thereby avoiding long distance charges between offices.

VoIP at remote home based offices

This was not a brave new world. LAB had previously tested this concept between the Network Administrator’s home office and the central office yielding results that were less than satisfactory. Calls placed back and forth were usable but the quality of the conversation was severely limited. The configuration had been left in place for some time and was occasionally used but most often calls ended with “… hold on, I’ll call you from my cell”. It wasn’t until the phones at the new building were installed that we realized that the phone that had been used at the network administrator’s home was of lesser quality.

While the new central office was nearing completion, preparations and testing had began for moving LAB’s technology to the new location. A big part of this task was to install internet and limited phone service at the new building. After installing the internet the Network Administrator was able to bring over spare equipment and configure Virtual Point to Point Connection (VPN) between the two buildings over the internet. Next, a Polycom VoIP phone was installed at the new building and connected to the TrixBox server over the VPN to the old office building. The VoIP phone worked extremely well and brought phone service to our new building for the first time.

After moving into the new building and experiencing a good success with the remotely located phone one was installed at the Accountant's home office. The Accountant's home office had been previously connected to the network with the same methods mentioned above using a broadband internet connection and a VPN. The Polycom 550 was deployed in the home office and was a great success and is still used today for daily business. The Accountant is now more connected and involved in the daily operations and is much more accessible than before with the single home phone line. The feature that comes with a quality phone and TrixBox, like unified messaging and caller ID, make it a very powerful tool.

Remote Intake Staffing

An opportunity presented itself when a contract attorney performed for part time intake work from home in the evening hours. This has been done before and has proved to be a very successful way to reach clients who are unavailable during normal business hours. Typically this is done by having the contract attorney logon to the remote access system to access client data and begin returning calls from his/her home phone and making sure to disable caller id before placing calls. The employee would have to reports long distance usage detail for reimbursement. This typically generates a lot of unwanted administration time. Using the Direct Inward System Access (DISA) feature, the contract attorney was able to call the TrixBox from their home phone, enter a Pin and then dial the client's number. This accomplished two things it first would appear to a client that they were in fact getting a call from Legal Aid of the Bluegrass, not another bill collector, and second, it was allowed the contract attorney to use long distance resources without the need to separately account for them.

SHIP Hotline

LAB was charged with creating the The State Health Insurance assistance Program (SHIP) Hotline to provide assistance with various Medicare, Medicaid and other health insurance issues. The SHIP staff that will be assists callers we located in three of LAB’s offices.

The CALS-Net project was up to the task of overcoming barriers with the help of the new VoIP communications systems. The hotline was designed and created to take live callers and place them in a call queuing system that would distribute the load across three offices. The system calls each SHIP agent in sequence until the call is answered. The system also provides the option for callers to leave a message. Callers can also choose one of three mailboxes to leave their message, in depending on the type of issues they have. The voice mail messages are then converted to an audio file and emailed to a shared e-mail box which allows all the SHIP agents access to these messages. To date, the system has come to serve many clients efficiently.

U-Interim Relief Clients Hotline

During the U-Interim Relief project the need arose to have a special intake hotline setup. A large mailing was due to go out and it was targeted for primarily Spanish-speaking clients. The attorneys requested that a new 800 number be setup to specifically deal with these clients, a number that could be published on the outgoing letter. However, many of the attorneys working on this project were located in several different offices, which made checking messages somewhat of a training issue and a time-consuming task. An 800 number was ordered, a new voicemail box was configured on the TrixBox and unified messaging was enabled using a common email address. This would allowed all the attorneys access to the voicemail's left by U-Interim Relief clients via their own email inboxes. Using previous systems, attorneys would have had to manage and navigate a common mailbox which in the past leads to one person relaying the messages to everyone else, or multiple users accidentally deleting others' voicemail messages.

This was added proof that the system was highly flexible and brought great value. We were able to perform all of these tasks in-house with minimal effort, validating Proving once again that TrixBox is a very powerful tool.

Unified Messaging

Unified messaging has been rolled out along with every user on the TrixBox system. Unified messaging is simply the ability to receive voicemails via an email system. So far the system has work beautify. The email contains details about the voice mail message, such as the caller ID information, the length, and date the message was received. The emails users receive also contain an audio file of the message that is playable on just about anything smart enough to receive email. This allows users to play the messages on their computer, email it to someone else or save the voicemail message with other related data on a hard drive or network location. The messages can also be played on a cell phone capable of receiving emails and playing attachments, which is a very powerful feature for those "on the go". Traveling users also now have the advantage of simply being notified when a new voice mail has arrived at their desk without having to call in to check. Many users now opt to not use the traditional voicemail box over this new method and request that the messages not be retained on the phone once emailed. This feature is by far one of the most valued assets of the system.

Works In progress

Interactive Voice Response (IVR)

The IVR is the system that most everyone is familiar with today. *"You have reached ACME, Press 1 for sales, Press 2 customer Service,..."* LAB in the past had a different independent phone system for each office. Each with system configured with a different IVR. This worked in the past but now that the systems are inter-connected LAB has updated the IVRs to work together by routing calls from one office to another. *See Appendix D* for a sample of the existing IVR after modification recent modifications.

Our Ashland IVR used to instruct callers that did not have active cases to hang-up and dial our toll free number for the Covington office which handles intake. The IVR now instructs those callers to choose option 1 which will then routes the call over LAB internal VoIP trunks to the Covington office. This not only provide a better experience for the client but also saves LAB and the caller long distance charges.

Our Lexington office telephone system used to have a voicemail box that new clients were instructed to leave messages. Staff in Covington would then call and check that voicemail box and log call backs. The IVR now routes the call over LAB internal VoIP trunks to the Covington office, where it will be handles directly by Covington staff.

The IVR has proven to be a challenging aspect to the new system. Also note that this is less of a technology problem and more of a business model issue that applies to all telephone system open source or commercial.

References

Technical Resources

http://cdn.oreilly.com/books/9780596510480.pdf ("Asterisk: The future of Telephony" PDF book, a must read.)

http://www.voip-info.org/ (Great over all Wiki for anything VoIP)

http://www.asterisk.org/ (The core Open Source VoIP Package)

http://www.trixbox.org/ (Free Community Edition Trixbox "CE")

http://www.trixbox.com/ (Pay For Trixbox)

http://www.digium.com/en/ (Interface Manufacture and sponsor of the Asterisk development effort)

http://www.freepbx.org/ (GUI Based controll module for Asterisk. Components used in Trixbox)

http://www.polycom.com (Manufacture of Voice endpoints (Telephones) and more)

http://www.voiplink.com/ (Hardware Vendor) http://www.vitelity.com/ (VoIP Trunking services "The way to place calls over the internet")

CALS-Net Overview Presentation (Presented on December 12, 2008 Regular Tech Committee Conference Meeting)

http://www.freepbx.org (FreePBX is a HTML adminstration interface for Asterisk)

Additional Software of interest

PuTTY - PuTTY is an OSS SSH client found at www.putty.org. This software can be used to remotely access a Linux terminal session. Used for remotely supporting Asterisk servers.

FileZilla - FileZilla is an OSS FTP client/server offering found at http://www.filezilla-project.org used to remotely connect to Linux SSH FTP connections. Very useful for managing Linux files from a windows computer.

Notepad++ - Notepad++ is an OSS SSH text file editor found at www.notepad-plus-plus.org

Glossary

Endpoint - Endpoint refers to and telephony device can place or receive calls. This could be a VoIP or Analog telephone, a software based VoIP phone (allows the computer to be used as a telephone) or other announcement speakers that can be placed in ceilings.

IVR - **Interactive Voice Response (IVR)** is a technology that allows a computer to detect voice and dual-tone multi-frequency signaling (DTMF) keypad inputs. IVR technology is used extensively in telecommunication, but is also being introduced into automobile systems for hands-free operation. Current deployment in automobiles revolves around satellite navigation, audio and mobile phone systems. In telecommunications, IVR allows customers to access a company's database via a telephone keypad or by speech recognition, after which they can service their own inquiries by following the instructions. IVR systems can respond with pre-recorded or dynamically generated audio to further direct users on how to proceed. IVR systems can be used to control almost any function where the interface can be broken down into a series of simple menu choices. In telecommunications applications, such as customer support lines, IVR systems generally scale well to handle large call volumes.

OSS - Open Source Software is software developed by a community of people with a common need. The this software is developed under a "Public Domain" sort of license that means the software is free to use and may not be resold. Also the source or code of the software is open as well. This means that the software can be openly evaluated and modified to fit any particular needs.

PBX - A **private branch exchange** (PBX) is a telephone exchange that serves a particular business or office, as opposed to one that a common carrier or telephone company operates for many businesses or for the general public. PBX is often also used to describe telephone system that included set of common telephony features such as voicemail, Automated Attendants, call routing and more.

PoE - Power over Ethernet or **PoE** technology describes a system to safely pass electrical power, along with data, on Ethernet cabling. PoE requires category 5 cable or higher for high power levels, but can operate with category 3 cable for low power levels. Power can come from a power supply within a PoE-enabled networking device such as an Ethernet switch or from a device built for "injecting" power onto the Ethernet cabling, dubbed **midspan**.

PSTN - The **public switched telephone network (PSTN)** also referred to as the Plain Old Telephone Service (POTS) is the network of the world's public circuit-switched telephone networks. It is a worldwide net of telephone lines, fiberoptic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables connected by switching centers, which allows any telephone in the world to communicate with any other. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital in its core and includes mobile as well as fixed telephones.

SSH - Secure Shell is a network protocol that allows data to be exchanged using a secure channel between two networked devices.[1] The two major versions of the protocol are referred to as **SSH1** or **SSH-1** and **SSH2** or **SSH-2**. Used primarily on Linux and Unix based systems to access shell accounts, SSH was designed as a replacement for Telnet and other insecure remote shells, which send information, notably passwords, in plaintext, rendering them susceptible to packet analysis.[2] The encryption used by SSH is intended to provide confidentiality and integrity of data over an insecure network, such as the Internet.

Telephony - In telecommunication, **telephony** (pronounced /təˈlɛfəni/ *or* teh-LEH-fuh-nee) encompasses the general use of equipment to provide voice communication over distances, specifically by connecting telephones to each other.

Unified Messaging - Unified Messaging (or UM) is the integration of different electronic messaging and communications media (e-mail, SMS, Fax, voicemail, video messaging, etc.) technologies into a single interface, accessible from a variety of different devices.[1] While traditional communications systems delivered messages into several different types of stores such as voicemail systems, e-mail servers, and stand-alone fax machines, with Unified Messaging all types of messages are stored in one system. Voicemail messages, for example, can be delivered directly into the user's inbox and played either through a headset or the computer's speaker. This simplifies the user's experience (only one place to check for messages) and can offer new options for workflow such as appending notes or documents to forwarded voicemails.

Appendix A. - Equipment and Service Costs

Probono – Scheduled for deployment on 4/2008 (2-6 Staff)

Equipment Costs

Description	**Cost**	**Qty**	**Totals**	**Notes**
LAB Trixbox Server	$1,000.00	1	$1,000.00	
(4) FXO Analog DGM-TDM04B	$378.00	1	$378.00	4 Local Dial Lines
Polycom IP 550	$235.00	5	$1,175.00	Standard Desk Phone
Polycom IP 650	$324.00	1	$324.00	Front Desk and Administration Phones
Polycom IP 330	$139.00	2	$278.00	General Use / Lawyer phones
Cisco 2611XM Router	$500.00	1	$500.00	Conference Room Phone
pfSense Firewall	$0.00	1	$0.00	Open Source using repurposed retired hardware
Grand Total			**$3,655.00**	

Monthly Service Costs

Description	**Cost**	**Qty**	**Totals**	**Notes**
Private T1 to Covington	$400.00	1	$400.00	
Analog Lines	$50.00	4	$200.00	4 Local inbound/outbound
DSL (1.5Mbps/384Kbps	$69.00	1	$69.00	
Grand Total			**$669.00**	

Morehead – Scheduled for deployment on 11/2009 (12-18 Staff)

Equipment Costs

Description	Cost	Qty	Totals	Notes
LAB Trixbox Server	$1,000.00	1	$1,000.00	
(8) FXO Analog TDM-808E	$378.00	1	$378.00	8 Local Dial Lines Ports
Plantronics Noise-Canceling Headset	$99.00	5	$495.00	
Polycom IP 550	$235.00	15	$3,525.00	Standard Desk Phone
Polycom SoundPoint Expansion Module	$229.00	2	$458.00	Front Desk Side Cars
Polycom IP 330	$139.00	2	$278.00	General Use / Lawyer phones
Polycom IP 650	$299.00	2	$598.00	Front Desk and Administration Phones
Cisco 2511 Router	$500.00	1	$500.00	
pfSense Firewall	$0.00	1	$0.00	Open Source using repurposed retired hardware
Equipment Total			**$7,232.00**	

Monthly Service Costs

Description	Cost	Qty	Totals	Notes
Private T1 to Covington	$400.00	1	$400.00	
Analog Lines	$40.00	7	$280.00	4 Local inbound/outbound
DSL (1.5Mbps/384Kbps)	$69.00	1	$69.00	
VoIP (Vitelity.com)	$80.00	1	$120.00	Direct Dial Lines
Service Total			**$869.00**	
Grand Total			**$8,101.00**	

Ashland – Scheduled for deployment on 11/2008 (3 - 6 Staff)

Equipment Costs

Description	Cost	Qty	Totals	Notes
LAB Trixbox Server	$1,000.00	1	$1,000.00	
(4) FXO Analog DGM-TDM04B	$378.00	1	$378.00	4 Local Dial Lines
Cisco 2111 Router	$500.00	1	$500.00	
Polycom IP 550	$235.00	4	$940.00	Standard Desk Phones
Polycom IP 330	$139.00	2	$278.00	General Use / Lawyer phones
Polycom IP 650	$299.00	2	$598.00	Front Desk and Administration Phones
Polycom SoundPoint Expansion Module	$229.00	2	$458.00	
pfSense Firewall	$0.00	1	$0.00	Open Source using repurposed retired hardware
Equipement Total			**$4,152.00**	

Monthly Service Costs

Description	Cost	Qty	Totals	Notes
Private T1 to Covington	$400.00	1	$400.00	
Analog Lines	$40.00	4	$160.00	4 Local inbound/outbound
Cable Broadband (1.5Mbps/768Kbps)	$169.00	1	$169.00	
VoIP (vitelity.com)	$60.00	1	$60.00	
Service Total			**$789.00**	Direct Dial Lines
Grand Total			**$4941.00**	

Lexington – Scheduled for deployment on 6/2010 (12 - 18 Staff)

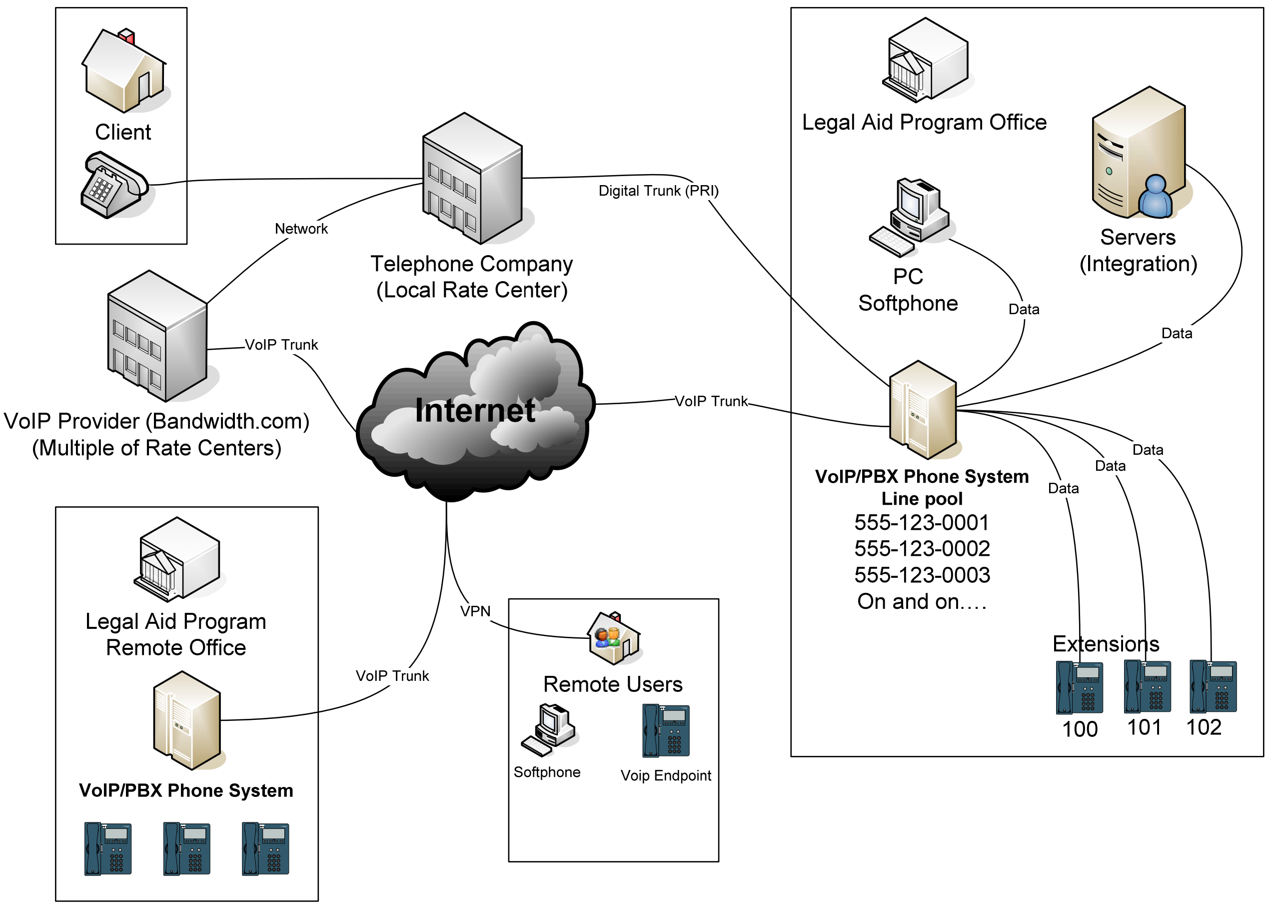
Equipment Costs

Description	**Cost**	**Qty**	**Totals**	**Notes**
LAB Trixbox Server	$1,500.00	1	$1,500.00	
PRI T1 Interface DGM-TE110P	$499.00	1	$499.00	
Polycom IP 550	$239.00	12	$2,868.00	Standard Desk Phone
Polycom IP 650	$289.00	2	$578.00	Front Desk and Administration Phones
Polycom SoundPoint Expansion Module	$182.00	4	$728.00	Front Desk Side Cars
Polycom IP 450	$199.00	2	$398.00	General Use / Lawyer phones
Cisco 2611XM Router	$500.00	1	$500.00	
pfSense Firewall	$0.00	1	$0.00	Open Source using repurposed retired hardware
Grand Total			**$7,309.00**	

Monthly Service Costs

Description	**Cost**	**Qty**	**Totals**	**Notes**
Voice and Internet (16 Channels Voice + 3Mbps [2xT1] Internet)	$1,580.00	1	$1,580.00	LD over 1000 minutes extra
Private T1 to Covington	$400.00	1	$400.00	
Grand Total			**$1,980.00**	

Appendix B. - Implementation Options Diagram



Appendix C. - Office Deployments

Ashland Office Trixbox Deployment.

Planning

Facilities

Review of facilities indicates that additional wiring is needed to fully support the use of the new system in the location. The Ashland office currently has twenty category 5e wiring runs to computers and telephones in eight locations. The additions of two category 5e wiring runs are needed to provide connections for the placement of Voice Endpoints in the common area and conference room. The existing runs to telephones also required modification to convert them from phone lines to Ethernet lines.

Communications

The Ashland Office currently has six telephone lines, four of which are used for inbound/outbound voice and the fifth is a dedicated attorney referral line and the sixth is dedicated to the fax DSL services.

Current data communications consisted of a DSL Broadband Internet service providing Internet access and supporting a VPN connection over the Internet to the Legal Aid of the Bluegrass programs main office providing access to email, file storage and case management systems. The current DSL broadband service provides adequate quality and will also provide voice connectivity to the other offices in the program.

Equipment

Current equipment consisted of one small office Internet VPN router which consumed the DSL Broadband Internet connection and provided firewall and VPN services.

No existing equipment was available to house the Trixbox VoIP system. New server equipment will be ordered and is to include the purchase of a UPS or battery backup device.

The current switch in used is one that is integrated into the Internet router providing eight connections and with the addition of six VoIP endpoints is now under capacity and is incapable of providing power to its connections a new power over Ethernet switch will need to be purchased too provide adequate service and power to VoIP End points.

Interfaces

The Trixbox VoIP system will require the purchase of interface hardware to connect the system to the analog telephone services. The interface will need to support eight total connections. The Digium brand hardware will be used as it has proven successful in the deployment at the Covington and Morehead offices of Legal Aid of the Bluegrass.

Voice Endpoints

Reviewed the requirements of the Ashland office needs and discovered the need for six Polycom IP 550 endpoints as the standard workstation phone and two Polycom IP 330 endpoints to be placed in common areas. One Polycom IP 650 with expansion modules was also required for the receptionist.

Design

User extension will be assigned from 2301-2350

All other extension numbers will be assigned from 2351-99, this will include common areas and special programming.

All workstations will have high quality Polycom IP 550 endpoints providing four line keys, large LCD display and a rich feature set.

Common areas and temporary workstation will have Polycom IP 330 a simple two line key phone.

Inbound calls will terminate to an Automated Attendant where callers can self select there destination or dial the extension of there party. Unanswered calls are to be routed to a common voice mailbox.

Unified messaging will be configured for each user of the system. Allow voice mail to be emailed to the user.

VoIP trunk needs to be configured to provide connectivity to the Covington and Morehead offices of Legal Aid of the Bluegrass. This allows for direct toll free extension dialing to Legal Aid of the Bluegrass offices. In the case of Ashland the automated attendant option for intake routes calls to the Covington number for intake services.

Installation

Services Installation

Wiring must be completed first before other systems and services can be installed..

Analog telephone line changes must take place during the installation as to not interfere with the current voice system.

Hardware Installation

The equipment closet will be the location for equipment and voice/data patch panel. This area needs to have a plywood mounting panel installed.

New category 5e wiring runs will be need to be installed and terminated to a patch panel located on the new plywood mounting panel.

The existing Netgear FVS318 router will be replaced with a pfSense open source BSD based firewall to provide the necessary performance gain and bring the office to LAB standards.

The Netgear FS726TP switch will be mounted to the plywood panel configured and tested.

The server and interface equipment needs to be assembled, installed and configured. This includes the installation of the Trixbox software.

The Polycom endpoints need to be installed at there appropriate locations and configured.

Cut Over

All installation work must be completed and a cut over date must be established. The cut over date will be driven primarily by the availability of the local service provider. The new analog services must be ordered.

Training must be completed.

Once the new lines have been activated and configured. The wiring needs to be removed from the old voice equipment and connected to the new Trixbox system.

Testing

The following must be tested.

- Internetworking must perform with out error. This includes logging on to workstations and accessing various remote applications which include local network printers and copiers, Internet, Legal Aid of the Bluegrass resources.
- Place outbound calls ensure calls are connected and clear.
- Place inbound calls ensure calls are connected and clear.
- Ensure caller ID is working correctly inbound and outbound.
- Place internet offices calls to Legal Aid of the Bluegrass
- Place inbound calls and leave voicemail messages. Ensure messages are delivered clearly

Testing Result

All test passed

Training

Training must be done and everyone must understand how to effectively receive and place calls, check and receive voicemail and understand the features of the system. *See Appendix E for Training Manual*

Training was completed in one hands on session

- Angela Rigsby
- Annie Francis
- Brian Dufresne

- Mary Rudd

Lessons Learned

This office was small and did not face many challenges except for handling the Attorney Referral line that was hosted by the office. This line needed to be separate process from the rest of the standard system. The lines in this office were not a digital circuit which did not allow routing. However, a module of the new phone system which had not previously been tested allow routing for the analog telephone lines. This enabled LAB to remove the old answering machine system that was previously used and add the line to the main system which benefited the office by being able to use the additional line for general use.

Covington Office Trixbox Deployment.

Planning

Facilities

Review of facilities indicates that the current wiring will be sufficient to implement VoIP endpoints throughout the building. However, the VoIP endpoints must be equipped with a computer Ethernet port as each office only has one Ethernet connection. Thus for the computers and phones to share the same network the Ethernet connection must pass through the phone to the computer.

Communications

The facility also had a pre existing T1 with 23 channels of voice that were each converted to an analog or POTS line.

Minor testing was completed with Trixbox using a Digium analog interface card. It was determined that the use of a T1 PRI which existed was a much more advanced connection method and provided digital switching. This enhances the use of the line over the analog method due to the fact that far more telephone numbers or DIDs could be assigned to the available channels. LAB proceeded to contact NuVox communications who provided the PRI service and had a reconfiguring completed that provided the existing equipment eight analog lines and the fifteen left over lines were provided via a PRI connection. Additional Digium interface hardware was purchased to connect to the PRI service.

Current data communications consisted of a two bonded T1s providing Internet access and supporting a VPN connection over the Internet to the Legal Aid of the Bluegrass programs remote offices providing access to email, file storage and case management systems.

Equipment

Current equipment consisted of one small office Internet VPN router which consumed the Cable Broadband Internet connection and provided firewall and VPN services.

No existing equipment was available to house the Trixbox VoIP system. New server equipment will be ordered and is to include the purchase of a UPS or battery backup device.

The current switches were 24 port non Power over Ethernet. New Power over Ethernet switches were purchased too provide connectivity and power to the VoIP End Points

Interfaces

The Trixbox VoIP system will require the purchase of interface hardware to connect the system to the T1 PRI telephone services.

Voice Endpoints

Reviewed the requirements of LAB and discovered the need for twenty five Polycom IP 550 endpoints as the standard workstation phone and two Polycom IP 330 endpoints to be placed in common areas. A single Polycom IP 650 with two expansion modules was purchase for the receptionist.

Design

User extension will be assigned from 1200-1299

All other extension numbers will be assigned from 1700-1799, this will include common areas and special programming.

All workstations will have high quality Polycom IP 550 endpoints providing four line keys, large LCD display and a rich feature set.

Common areas and temporary workstation will have Polycom IP 330 a simple two line key phone.

Inbound calls will terminate to an Automated Attendant where callers can self select there destination or dial the extension of there party. Unanswered calls are to be routed to a common voice mailbox.

Unified messaging will be configured for each user of the system. Allow voice mail to be emailed to the user.

Installation

Services Installation

The PRI/Analog configuration needed to be changed over to a pure T1 PRI service to provide all 23 channels to the Trixbox system for use.

Hardware Installation

The Netgear FS726TP switches were mounted to the current system rack and replaced the older Linksys switches.

The server and interface equipment needs to be assembled, installed and configured. This includes the installation of the Trixbox software.

The Polycom endpoints need to be installed at there appropriate locations and configured.

Cut Over

All installation work must be completed and a cut over date must be established. The cut over date will be driven primarily by the availability of the local service provider.

Training must be completed.

Once the new lines have been activated and configured. The wiring needs to be removed from the old voice equipment and connected to the new Trixbox system.

Testing

The following must be tested.

- Internetworking must perform with out error. This includes logging on to workstations and accessing various remote applications which include local network printers and copiers, Internet, Legal Aid of the Bluegrass resources.
- Place outbound calls ensure calls are connected and clear.
- Place inbound calls ensure calls are connected and clear.
- Ensure caller ID is working correctly inbound and outbound.
- Place internet offices calls to Legal Aid of the Bluegrass
- Place inbound calls and leave voicemail messages. Ensure messages are delivered clearly

Testing Results

All tests were completed successfully

Training

Training must be done and everyone must understand how to effectively receive and place calls, check and receive voicemail and understand the features of the system. *See Appendix E for Training Manual*

Training completed in 3 separate sessions.

- Acena Smith
- Alma Yaros
- Barbara O'Connor
- Brad Christy
- Carl Melcher
- Cindy Millay
- Connie Berkemeier
- Donna Hackett
- Eileen Zell
- Glenda Harrison
- Jim McHugh
- Jocklyn Jouett
- Lisa Goetz
- Monica Burke
- Richard Cullison
- Susan Watts

Lessons Learned

The Covington office originally had a "Key Based" telephone system. This meant that all phones had buttons for each line the system made available. This meant that users would select lines 1 - 6 and place a call. The new Asterisk phone system was of a "PBX" model and the phones did not have buttons bound to individual telephone lines. The new system instead had 4 line buttons that would select a free line from a pool of available lines. This is because the new system had 23 channels of voice and having 23 buttons would not be practical. Many users found this to be very confusing and became a significant training challenge but was over come during extended hands on training sessions.

Lexington Office FreePBX Deployment.

Planning

Facilities

Review of facilities indicates that additional smaller workgroup Power Over Ethernet (PoE) network switches are going to be necessary due to limited premise wiring. Installation of additional wiring in the building would have greatly exceed the cost of purchasing and supporting smaller distributed PoE switches.

Communications

The office had and existing 23 channel PRI connected to a Nortel BCM50 telephone switch. This was an acceptable resources for the new FreePBX system.

The office had recently been part of the program wide switch to Windstream Ethernet Internet Access (EIA) data circuit. This provides the office with 3Mb of up/down bandwidth. This service provides the necessary bandwidth to support interoffice connections.

An analog line for the FAX machine is required. The Nortel BCM50 provided the analog fax line via the PRI. The FreePBX system will support this configuration but has not be extensively tested by LAB.

Equipment.

No existing equipment was available to house the FreePBX VoIP system. New server equipment will be ordered.

Interfaces

The FreePBX VoIP system will require the purchase of interface hardware to connect the system to the PRI telephone services. The Digium brand hardware will be used as it has proven successful in the deployment at the Covington office of Legal Aid of the Bluegrass.

Voice Endpoints

Reviewed the requirements of the office needs and discovered the need for fifteen Polycom IP 550 endpoints as the standard workstation phone and four Polycom IP 450 endpoints to be placed in common areas. Two Polycom IP 650 phones with expansion modules were ordered for the receptionist staff.

Design

User extension will be assigned from 2200 -2250

All other extension numbers will be assigned from 2251-99, this will include common areas and special programming.

All workstations will have high quality Polycom IP 550 endpoints providing four line keys, large LCD display and a rich feature set.

Common areas and temporary workstation will have Polycom IP 450 a simple two line key phone.

Inbound calls will terminate to an Automated Attendant where callers can self select there destination or dial the extension of there party. Unanswered calls are to be routed to a common voice mailbox.

Unified messaging will be configured for each user of the system. Allow voice mail to be emailed to the user.

VoIP trunk needs to be configured to provide connectivity to the Covington office of Legal Aid of the Bluegrass. This allows for direct toll free extension dialing to Legal Aid of the Bluegrass offices.

Installation

Services Installation

Analog telephone line changes must take place during the installation as to not interfere with the current voice system.

Hardware Installation

The server closet will be the location for equipment and voice/data patch panel.

The four Netgear FS108p switches will need to be placed where necessary to supply connections to common areas and additional equipment such as printers and copiers.

The server and interface equipment needs to be assembled, installed and configured. This includes the installation of the FreePBX software.

The Polycom endpoints need to be installed at there appropriate locations and configured.

Cut Over

All installation work must be completed and a cut over date must be established.

Training must be completed after equipment has been installed but has not yet been put into production. This gives staff time to familiarize themselves with the new equipment.

The wiring needs to be removed from the old voice equipment and connected to the new FreePBX system.

Testing

The following must be tested.

- Internetworking must perform with out error. This includes logging on to workstations and accessing various remote applications which include local network printers and copiers, Internet, Legal Aid of the Bluegrass resources.
- Place outbound calls ensure calls are connected and clear.
- Place inbound calls ensure calls are connected and clear.
- Ensure caller ID is working correctly inbound and outbound.
- Place internet offices calls to Legal Aid of the Bluegrass
- Place inbound calls and leave voicemail messages. Ensure messages are delivered clearly
- Ensure the voicemail notices are being sent to, received and delivered by the mail server.

Testing Results

- The EIA was tested. Connectivity was confirmed and latency was around 10ms. This is will provide quality interoffice voice communications.
- Outbound calls were placed. local, long distance and VoIP calls. All were connected successfully.
- Inbound calls were placed both from long distance and local sources. All calls were received by the system and routed to the automated attendant.
- Inbound calls were not displaying the correct caller ID information on the phones correctly. This issue was resolved.
- Inter office calls were placed between extensions in the office and the Covington, Morehead, Ashland and Probono offices. The calls were completed successfully and clear.
- Additional calls were placed to test the voicemail system. Minor adjustments to the recording gain levels for the system were made to improve the sound level of the voice mail messages. After which all was in good working order.

Training

Training must be done and everyone must understand how to effectively receive and place calls, check and receive voicemail and understand the features of the system. *See Appendix E for Training Manual*

- Amanda Hitt
- Amy Dougherty
- Amy Hayden
- Angela Zeek
- Catherine DeFlorio
- Jennifer Casey
- Josh Fain
- Karen Jones
- Kristen Gordon
- Lisa Johnson
- Mary Lou Campbell
- Melissa Miller
- Rhonda Stanger

- Sarah Tolliver

Lessons Learned

The Lexington office had recently just before this project began received a new Nortel BCM telephone system. This system was connected with a PRI digital circuit. Previous installations at other offices that had dedicated fax lines that did not interact with the telephone systems. The Lexington fax line was provided by the PRI and the analog circuit to the fax machine was provided by the Nortel BCM telephone system. After the cut over to the new Asterisk system the fax machine was disconnected. This was not accounted for prior to the system install. The fax line was redirect by the provider to our Covington office until at which point a dedicated line could be install in the Lexington office.

Morehead Office Trixbox Deployment.

Planning

Facilities
Review of facilities indicates that additional wiring is needed to fully support the use of the new system in the location. The Morehead office currently has category 5e wiring runs to all workstations. The additions of one category 5e wiring run to the Kitchen area are needed to provide connections for the placement of Voice Endpoints.

Communications
The Morehead office currently has a total of seven analog lines telephone lines, four of which are used for voice configured in a hunt group. One line dedicated to fax communications, one spare line and one dedicated to the DSL internet service.

Current data communications consisted of a DSL Broadband Internet service providing Internet access and supporting a VPN connection over the Internet to the Legal Aid of the Bluegrass programs main office providing access to email, file storage and case management systems. In order to perform VoIP communications to the other Legal Aid of the Bluegrass a higher quality connection was needed. The installation of a point to point T1 is required which will replace the Internet VPN. The Cable Broadband Internet service will remain in place as a backup connection option and to provide local Internet access.

Equipment
Current equipment consisted of one small office Internet VPN router which consumes the DSL Broadband Internet connection and provides firewall and VPN services. The addition of a T1 point to point connection will require the addition of a Cisco 2611XM router to provide interfacing to the T1 circuit. The other end of the T1 will terminate in to the Covington office of Legal Aid of the Bluegrass and will required an additional interface card to be installed in the Cisco 3640 router in that location.

No existing equipment was available to house the Trixbox VoIP system. New server equipment will be ordered.

The current switch in used is one that is integrated into the Internet router providing eight connections and a single 24 port 10/100 switch which is incapable of providing power to its connections a new power over Ethernet switch will need to be purchased to provide adequate service.

Interfaces
The Trixbox VoIP system will require the purchase of interface hardware to connect the system to the analog telephone services. The interface will need to support seven total connections. The Digium brand hardware will be used as it has proven successful in the deployment at the Covington office of Legal Aid of the Bluegrass.

Voice Endpoints

Reviewed the requirements of the Morehead Office needs and discovered the need for fifteen Polycom IP 550 endpoints as the standard workstation phone and two Polycom IP 330 endpoints to be placed in common areas. Two Polycom IP 650 phones with expansion modules were ordered for the receptionist and managing attorney.

Design

User extension will be assigned from 2100 -2150

All other extension numbers will be assigned from 2151-99, this will include common areas and special programming.

All workstations will have high quality Polycom IP 550 endpoints providing four line keys, large LCD display and a rich feature set.

Common areas and temporary workstation will have Polycom IP 330 a simple two line key phone.

Inbound calls will terminate to an Automated Attendant where callers can self select there destination or dial the extension of there party. Unanswered calls are to be routed to a common voice mailbox.

Unified messaging will be configured for each user of the system. Allow voice mail to be emailed to the user.

VoIP trunk needs to be configured to provide connectivity to the Covington office of Legal Aid of the Bluegrass. This allows for direct toll free extension dialing to Legal Aid of the Bluegrass offices.

Installation

Services Installation

Wiring must be completed first before other systems and services can be installed.

Prior to system installation the T1 service must be ordered, installed and configured. This process can take up to 1 month.

Analog telephone line changes must take place during the installation as to not interfere with the current voice system.

Hardware Installation

The server closet will be the location for equipment and voice/data patch panel.

New category 5e wiring runs will be need to be installed and terminated to a patch panel located on the new plywood mounting panel.

The Cisco 2611XM router will be mounted to the existing equipment rack

The Netgear FS726TP switch will be mounted to the existing equipment rack.

The server and interface equipment needs to be assembled, installed and configured. This includes the installation of the Trixbox software.

The Polycom endpoints need to be installed at there appropriate locations and configured.

Cut Over

All installation work must be completed and a cut over date must be established. The cut over date will be driven primarily by the availability of the local service provider. The new analog services must be ordered.

Training must be completed.

Once the new lines have been activated and configured. The wiring needs to be removed from the old voice equipment and connected to the new Trixbox system.

Testing

The following must be tested.

- Internetworking must perform with out error. This includes logging on to workstations and accessing various remote applications which include local network printers and copiers, Internet, Legal Aid of the Bluegrass resources.
- Place outbound calls ensure calls are connected and clear.
- Place inbound calls ensure calls are connected and clear.
- Ensure caller ID is working correctly inbound and outbound.
- Place internet offices calls to Legal Aid of the Bluegrass
- Place inbound calls and leave voicemail messages. Ensure messages are delivered clearly

Testing Results

- The T1 was tested. Connectivity was confirmed and latency was around 20ms. This is required for quality voice communications.
- Outbound calls were placed. local, long distance and VoIP calls. All were connected successfully.
- Inbound calls were placed both from long distance and local sources. All calls were received by the system and routed to the automated attendant.
- Inbound calls were also displaying the correct caller ID information on the phones.
- Inter office calls were placed between extensions in the Morehead office and the Covington office. The calls were completed successfully and clear.
- Additional calls were placed to test the voicemail system. Minor adjustments to the recording gain levels for the system were made to improve the sound level of the voice mail messages. After which all was in good working order.

Training

Training must be done and everyone must understand how to effectively receive and place calls, check and receive voicemail and understand the features of the system. *See Appendix E for Training Manual*

- Angela Dailey
- Beverly Kirch
- Brenda Combs
- Cathy Pettit
- Cheryl Barber
- Daniel Mason
- Debbie Ratliff
- Dianna Reynolds
- Katie Ackerman
- Melinda Jennies
- Teresa Mason

Lessons Learned

Following the Probono, Ashland and Morehead offices there were frequent reports of poor audio quality during calls. The quality issues were often sporadic or a direct result of minor network issues which made troubleshooting very difficult. Before the deployment to the last office LAB surveyed staff to determine how wide spread of a problem existed. After surveying the staff LAB determined that the problem was somewhat minimal and most staff were accepting of the minor issues given the significant cost savings the new system represented. LAB's technical staff continued to researched the audio quality issues extensively and found that two issues existed. First echo cancelling hardware firmware was out of date and updates were not being applied due to a software driver issue. The second issue may have been digital to analog conversion times and most of those issues were resolved with PRI digital circuits being installed, which had recently become more affordable.

Fayette County Pro Bono Partner Trixbox Deployment.

Planning

Facilities

Review of facilities indicates that additional wiring is needed to fully support the use of the new system in the location. The Pro Bono office currently has networks six category 5e wiring runs to computers in the office The additions of six category 5e wiring runs are needed to provide connections for the placement of Voice Endpoints next to existing workstations and 2 common areas. Additionally 3 more category 5e runs are needed to the common equipment room where voice and data communications enter the building.

Communications

The Pro Bono partner program currently has three telephone lines, two of which are used for voice and the third is a dedicated fax line. After working with Tammie Haddix we have determined that four voice lines would better serve the office but was not supported by the current voice equipment. Two additional lines are needed and will be placed in to a hunt group with the current voice lines. This will allow the office to have one main telephone number that will access each of the four lines based on the line's availability. The fax line will remain independent from the system.

Current data communications consisted of a Cable Broadband Internet service providing Internet access and supporting a VPN connection over the Internet to the Legal Aid of the Bluegrass programs main office providing access to email, file storage and case management systems. In order to perform VoIP communications to the other Legal Aid of the Bluegrass a higher quality connection was needed. The installation of a point to point T1 is required which will replace the Internet VPN. The Cable Broadband Internet service will remain in place as a backup connection option and to provide local Internet access.

Equipment

Current equipment consisted of one small office Internet VPN router which consumed the Cable Broadband Internet connection and provided firewall and VPN services. The addition of a T1 point to point connection will require the addition of a Cisco 2611XM router to provide interfacing to the T1 circuit. The other end of the T1 will terminate in to the Covington office of Legal Aid of the Bluegrass and will required an additional interface card to be installed in the Cisco 3640 router in that location.

No existing equipment was available to house the Trixbox VoIP system. New server equipment will be ordered and is to include the purchase of a UPS or battery backup device.

The current switch in used is one that is integrated into the Internet router providing eight connections and with the addition of six VoIP endpoints is now under capacity and is incapable of providing power to its connections. A new power over Ethernet switch will need to be purchased too provide adequate service.

Interfaces

The Trixbox VoIP system will require the purchase of interface hardware to connect the system to the analog telephone services. The interface will need to support four total connections. The Digium brand hardware will be used as it has proven successful in the deployment at the Covington office of Legal Aid of the Bluegrass.

Voice Endpoints

Reviewed the requirements of the Pro Bono Partner programs needs and discovered the need for four Polycom IP 550 endpoints as the standard workstation phone and two Polycom IP 330 endpoints to be placed in common areas.

Design

User extension will be assigned from 2401-2450

All other extension numbers will be assigned from 2451-99, this will include common areas and special programming.

All workstations will have high quality Polycom IP 550 endpoints providing four line keys, large LCD display and a rich feature set.

Common areas and temporary workstation will have Polycom IP 330 a simple two line key phone.

Inbound calls will terminate to an Automated Attendant where callers can self select there destination or dial the extension of there party. Unanswered calls are to be routed to a common voice mailbox.

Unified messaging will be configured for each user of the system. Allow voice mail to be emailed to the user.

VoIP trunk needs to be configured to provide connectivity to the Covington office of Legal Aid of the Bluegrass. This allows for direct toll free extension dialing to Legal Aid of the Bluegrass offices.

Installation

Services Installation

Wiring must be completed first before other systems and services can be installed.

Prior to system installation the T1 service must be ordered, installed and configured. This process can take up to 1 month.

Analog telephone line changes must take place during the installation as to not interfere with the current voice system.

Hardware Installation

The back wall of the break room will be the location for equipment and voice/data patch panel. This area needs to have a plywood mounting panel installed.

New category 5e wiring runs will be need to be installed and terminated to a patch panel located on the new plywood mounting panel.

The existing Netgear FVS318 router will be mounted to the plywood panel.

The Cisco 2611XM router will be mounted to the plywood panel, configured and tested.

The Netgear FS726TP switch will be mounted to the plywood panel configured and tested.

The server and interface equipment needs to be assembled, installed and configured. This includes the installation of the Trixbox software.

The Polycom endpoints need to be installed at there appropriate locations and configured.

Cut Over

All installation work must be completed and a cut over date must be established. The cut over date will be driven primarily by the availability of the local service provider. The new analog services must be ordered.

Training must be completed.

Once the new lines have been activated and configured. The wiring needs to be removed from the old voice equipment and connected to the new Trixbox system.

Testing

The following must be tested.

- Internetworking must perform with out error. This includes logging on to workstations and accessing various remote applications which include local network printers and copiers, Internet, Legal Aid of the Bluegrass resources.
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- Ensure caller ID is working correctly inbound and outbound.
- Place internet offices calls to Legal Aid of the Bluegrass
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Test Results

- The T1 was tested. Connectivity was confirmed and latency was around 20ms. This is required for quality voice communications.
- Outbound calls were placed. local, long distance and VoIP calls. All were connected successfully.
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- Inbound calls were also displaying the correct caller ID information on the phones.

- Inter office calls were placed between extensions in the Probono office and the Covington office. The calls were completed successfully and clear.
- Additional calls were placed to test the voicemail system. Minor adjustments to the recording gain levels for the system were made to improve the sound level of the voice mail messages. After which all was in good working order.

Training

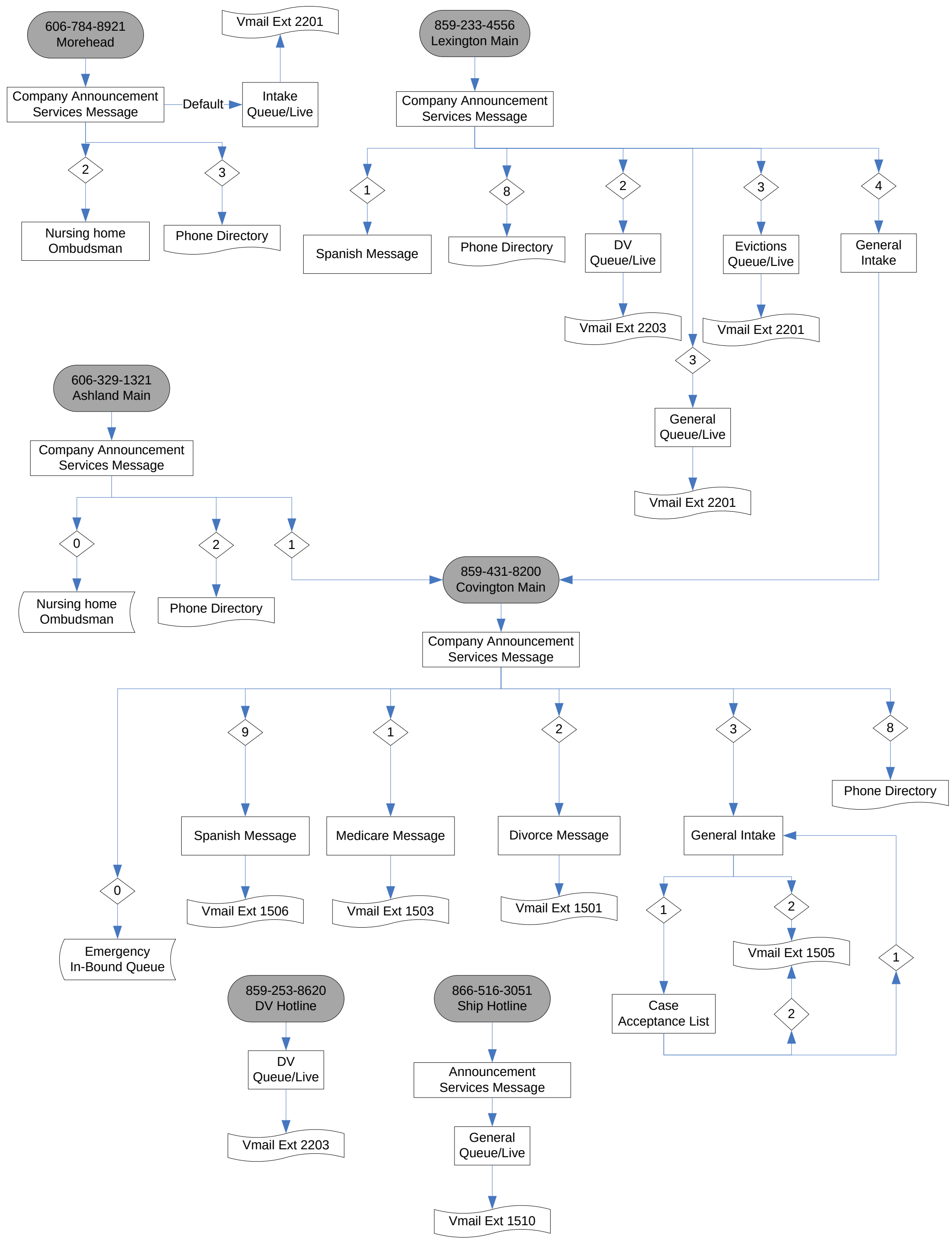
Training was completed and everyone understands how to effectively receive and place calls, check and receive voicemail and understand the features of the system. *See Appendix E for Training Manual*

- Malika Sanders
- Tammie Haddix
- Trinia Clemons

Lessons Learned

This office was very small and previously had a two line phone system The new system was 4 lines and was the first to not have a dedicated receptionist. This posed a unique configure that had not yet been used by LAB. The staff was accustom to having all the phones in the building ring when calls came in to the office. This was different from typically setting where a receptionist would handle the call routing. Due to considerable intake the staff would often let the phones ring to voicemail. The office also lacked a direct line for not intake business calls. LAB configured the phones to have two line appearances one for intake and one for the direct business line. This improved operations for the staff.

Appendix D. - IVR Diagrams



Appendix E. - Training Manual

Asterisk Phone System Training Guide

Legal Aid of the Bluegrass

Written by: Brian Miller

Edited by: Eileen Zell

Table of Contents

Asterisk Phone System

The Asterisk phone system is a digital phone system much different from older “Key” based systems. The Key system generally has a designated extension and all available lines are selected by static buttons on the façade of the phone itself. This type of system often minimizes the number of lines the system can handle because it is impractical to place 12 or more ‘line buttons’ on all phones. Typically with Key systems, inbound calls enter into the system by the next available line and the callers are given the option to dial an extension number or transfer to the operator. If the call is transferred to the operator, a live operator would then determine course of action.

The Asterisk phone system will work very similar to the Key system with the exception of the way lines are handled. Lines are now pooled together so it is not possible to see that a phone call exists on “Line 4” or “Line 2”. When someone initiates a call or an inbound call originates, the Asterisk system chooses an open line and then simply handles the call. If the call is direct to the extension, the call appears as a new phone call with caller ID information if it is available. To ‘move’ calls around in the Asterisk system, the user must “Transfer”, “Conference,” or “Park” the call in the system. The following sections will discuss general call handling procedures and describe the Asterisk phone features and buttons.

The phones connected to the Asterisk system are Polycom IP 650, 550, and 330 models. The IP 650 and IP 550 models are generally identical except the 650 has more line buttons and can carry an expansion module. The IP 650 is only used where these features are necessary. The standard desk phone is the IP 550, which has 4 line buttons. The IP 330 model phones are very basic two line phones which are for general area use, (i.e., copy rooms, waiting rooms, kitchen, etc.). The Polycom phones are fairly intuitive but if more detailed information about this is needed please refer to the Polycom IP 550 User guide.

Asterisk Call Handling

Common call handling procedures:

Placing a Call

Calls can be placed in a number of ways. Common to all of these methods is the “Dial Map”. When a user enters a series of numbers, the Dial Map determines whether the series entered is a valid number and then determines whether the user is done entering numbers.

For example, the phone system is programmed to recognize that the office extensions are 4 digits. Therefore, once the phone receives 4 digits of input from the user, the system automatically dials the number. If only 6 digits were entered however, the phone would continue to wait for the 7^{th} digit or until the user instructs the phone to dial by pressing the dial or send button.

Call Placing Options:

- Dial an extension, number, or feature code. Pick up the handset, select a line button or press the “Dial” soft key.
- Pickup the handset and dial an extension, number or feature code.
- Select a line button and dial an extension, number or feature code
- Selecting the Speaker Phone or Headset via the buttons on the phone and dial an extension, number or feature code
- Pressing a buddy or speed dial button.

Note: Because our extensions start with 1 and are 4 digits long, please be aware that dialing a long distance number can be tricky. If you don’t begin the 7 digit section of the phone number soon enough the phone will identify 4 digits as being and extension and dialing them. Example, if you are dialing 16067848921 and you pause to review your number after 1606, the phone will dial this as an extension.

Receiving a Call

Receiving calls is a fairly simple process. If not on a call, the phone will begin to ring and will display the caller id information (if available). There will be 3 soft key options available, "Answer", "Reject", and "Forward".

By pressing "Answer", the system will answer the call and place it on speaker phone. Alternatively, choosing "Reject" will immediately send the call to voice mail, and choosing "Forward" will allow the user to forward the call to another extension. Additionally, the user can also answer the phone by simply picking up the handset, pressing the speaker, or by pressing the headset button.

If the user is on an already existing call and a new call comes in, the new call will revert to the next available line and the light next to the line button will begin to blink green and caller ID information will be place along the bottom of the LCD screen

Transferring a Call

If the user has determined the need to transfer the current call to another extension or outside number, the user may do so by pressing the "Transfer" button, at which point the current call will be placed on hold and a new line opened. Then the user should dial the extension or phone number of the party intended to receive the transfer and "Transfer" when ready.

Blind Transfer:

A blind transfer is the ability to transfer the call immediately without being in contact or having any communication with the caller. Select the **"Blind"** soft key before dialing the destination party number. Once a valid number has been entered the system will immediately transfer the call. This is very useful when transferring a caller to voice mail by prefixing the extension with **"*"**. Below is a listing of steps.

On a Call -> Press Transfer -> Press Blind Soft Key -> * + 1234

This would immediately transfer the call to extension 1234's voicemail.

Call Conferencing

If the user has determined the need to conference another extension or outside number, the user may do so by pressing the “Conference” button, at which point the current call will be placed on hold and a new line opened. Then the user should dial the extension or phone number of the party to conference and press “Conference” when ready to connect the two lines.

Splitting the Call:

While on a conference call, the user will notice that there is a “Split” soft key. This soft key will split the conference call among the lines on the phone placing both calls on hold. This feature is useful to hold the entire call, or to engage in communications with only one party to the call.

Call Parking

Call parking allows a user to put a call on hold and make the call available to other users on the system. The system has a parking lot with 8 parking spots, numbered from 71 through 78. A call should be transferred to Extension 70, then the system will park the call in the next available slot and inform the user transferring the call of the parking slot number (ext. 71 – 78).

For example, when the user transfers a call to ext. 70, the system attendant will say, “71”, then the user completes the transfer. Now the call is on hold and is available to any user that dials the 71 extension/parking spot number. However, if the call is not picked up in 3 minutes the call will be automatically transferred back to the user that originally placed the call in the parking spot. This feature is useful for taking a call to a conference room or if the user plans on picking the call up elsewhere in the building. Additionally, if the user needs to physically locate someone in the building, this feature will allow the call to be picked up anywhere.

Asterisk Feature Guide

Feature codes are used to activate certain Line/Extension and system features, such as "Call Forwarding", "Do Not Disturb", "Paging", etc.

Listed below are brief descriptions and instructions:

Call Forwarding

Call forwarding generally means to forward calls to another extension or outside line. This is useful if the user is not going to be at his or her desk for an extended period of time or would like someone else to handle calls of a particular Line/Extension.

72 *Call Forward All Activate

Activates call forwarding of all inbound calls to the Line/Extension. Simply dial ***72** the system will prompt the user for the extension to forward, then the system will prompt for the number or extension to forward calls too.

73 *Call Forward All Deactivate

As the name implies, this deactivates the feature set using ***72**. Dial ***73** the system will confirm the feature was deactivated

90 *Call Forward Busy Activate

Activates call forwarding to another outside number or Line/Extension if the Line/Extension is busy. Simply dial ***90** the system will prompt the user for the extension to forward, then the system will prompt for the number or extension to forward calls too.

91 *Call Forward All Deactivate

As the name implies, this deactivates the feature set using ***90**. Dial ***91** the system will confirm the feature was deactivated

52 *Call Forward No Answer/Unavailable Activate

Activates call forwarding to another outside number or Line/Extension if the Line/Extension is unanswered. Simply dial ***52** the system will prompt the user for the extension to forward, then the system will prompt for the number or extension to forward calls too.

***53 *Call Forward No Answer/Unavailable* Deactivate**

As the name implies, this deactivates the feature set using ***52**. Dial ***53** the system will confirm the feature was deactivated.

Call Waiting

Call waiting generally means to allow new incoming calls to roll to another extension assignment, if deactivated the system will consider the phone to be busy if the line is used. By default Call Waiting is activated and new incoming calls will light on the second Line/Extension.

70 *Call Waiting Activate

Activates Call waiting allowing inbound calls to use the next available Line/Extension. Simply dial ***70** the system will confirm the feature was activated

71 *Call Waiting Deactivate

Deactivates Call waiting, if the phone is in use new inbound calls will not be allowed to ring on the next available Line/Extension. Simply dial ***71** the system will confirm the feature was activated

User Intercom

User Intercom is a method of placing a call to another phone that will automatically be received and be placed on speaker phone (a ring tone will first alert the receiver before the intercom is automatically engaged). This feature is commonly used to engage in intercom communications between offices.

80 *Intercom Prefix

This feature code is used as a prefix. Simply dial *80 + Extension and the system will intercom the Extension. In the example (*801234) and press send. This will dial the Extension 1234, the phone configured with Extension 1234 will ring once and then be placed on speaker phone automatically allowing for two way communication.

54 *Intercom Allow

This feature code tells the system to allow intercom calls. Dial ***54** and the system should confirm the feature's activation.

55 *Intercom Disallow

This feature code tells the system to disallow intercom calls. Dial ***55** and the system should confirm the feature's deactivation. Other system users attempting to intercom an intercom deactivated station using the ***80** feature will be notified the Extension is not accepting Intercom calling.

Feature Code Quick Reference

Call Forwarding

72** ***Call Forward All Activate

73** ***Call Forward All Deactivate

90** ***Call Forward Busy Activate

91** ***Call Forward Busy Deactivate

52** ***Call Forward No Answer/Unavailable Activate

53** ***Call Forward No Answer/Unavailable **Deactivate**

Call Waiting

70** ***Call Waiting Activate

71** ***Call Waiting Deactivate

User Intercom

80 + [Extension]** ***Intercom Prefix

54** ***Intercom Allow

55** ***Intercom Disallow

Voice Mail

98** ***Dial Voicemail

97** ***Dial My Voicemail

*** + Extension** ***Call Voice Mail Directly***

Asterisk Voice Mail

Accessing

There are a number of ways to access voice mail on the new system. Listed below are the different methods of accessing:

1. Press the "Message" button on the phone. This will display the messages screen and will provide brief information about the number of messages for that line. Selecting "Connect" will dial the voice mail system and it will prompt for the password for the Line/Extension of that phone. Enter the password and press **#**. (If the phone has more than one Line/Extension assigned, after pressing the messages button, the system will first prompt for which line to the user is requesting information.)
2. Dialing ***97** will dial the voice mail system and it will prompt for the password for the Line/Extension of that phone. Enter the password and press **#**.
3. Dialing ***98** will dial the voice mail system. Using ***98** the system will not automatically identify the mail box for the Line/Extension currently being used. Instead the system will prompt the user for a mailbox number. Enter the mailbox number and press **#**. If you enter a valid mailbox number the system will prompt for the mailbox password. . Enter the password and press **#**. This method of access is useful for accessing the voice mailbox of a Line/Extension from a foreign phone, such as common area or other user's phones. This is also required if you need to access voice mailboxes that are not assigned to a phone.
4. Dial the main office number (859) 431-8200 and press the ***** key. This will effectively dial ***98**, please see item 3 and duplicate the steps.

Message Notification

The Asterisk voice system has two methods of notifying users of new messages. The first method is the “read message” indicator light on the top of the phone. The light will begin to flash until new message are checked. The second method is email notification. The voicemail system will send an email to the user of the mailbox when new messages arrive. This message will contain information about the message and the user that left the message, similar to what is shown below. The message will also contain an attachment which is a WAV format sound file of the message, allowing the user to play the message on a computer. Accessing remote email now allows the user to check for voicemail messages.

```
Brian Miller,

There is a new voicemail in mailbox 1239:

        From:   "Intake Agent 1" <8100>
        Length: 0:03 seconds
        Date:   Tuesday, September 25, 2007 at 11:43:37 AM

Dial *98 to access your voicemail by phone.
```

*The “**From:**” will only be available from internal extensions or incoming calls that provide caller ID.*

Deleting email voicemail notifications will not delete the message from the voicemail box. Message must be deleted by accessing the voicemail box.

Using Voicemail

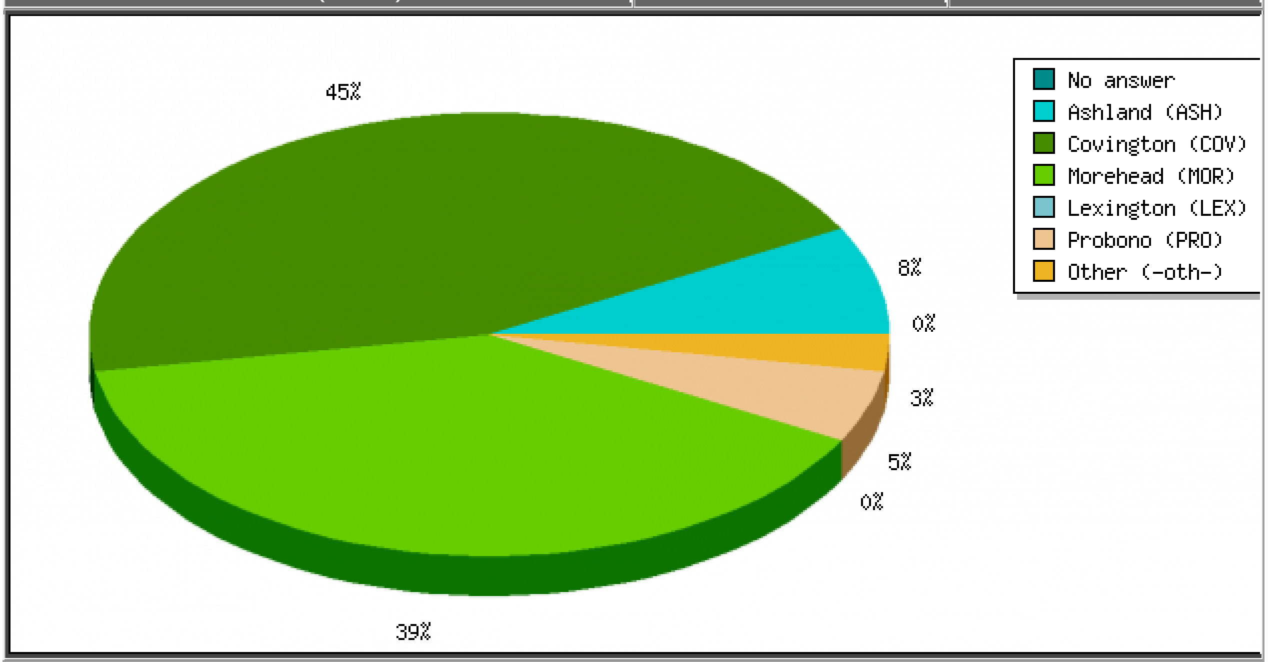
The Asterisk voicemail is a generally standard voicemail system that is used by most phones today. Generally most users will be able to navigate the voicemail system with ease just by accessing the system and listening to the instructions. Below is the voicemail menu map and a link to the graphical quick guide, a useful tool for viewing the menus and menu items as they relate to the phone key pad.

- **1** Read voicemail messages
    - **3** Advanced options
        - **1** Reply
        - **2** Call back
        - **3** Envelope
        - **4** Outgoing call
        - **5** Send Message
    - **4** Play previous message
    - **5** Repeat current message
    - **6** Play next message
    - **7** Delete current message
    - **8** Forward message to another mailbox
    - **9** Save message in a folder
    - ***** Help; during message playback: Rewind
    - # Exit; during message playback: Skip forward
- **2** Change folders
    - **0** Switch to New Messages
    - **1** Switch to Old Messages
    - **2** Switch to Work Messages
    - **3** Switch to Family Messages
    - **4** Switch to Friend Messages
- **3** Advanced Options
    - **5** Send Message
- **0** Mailbox options
    - **1** Record your unavailable message
    - **2** Record your busy message
    - **3** Record your name
    - **4** Record your temporary message
    - **5** Change your password
    - ***** Return to the main menu
- ***** Help
- # Exit

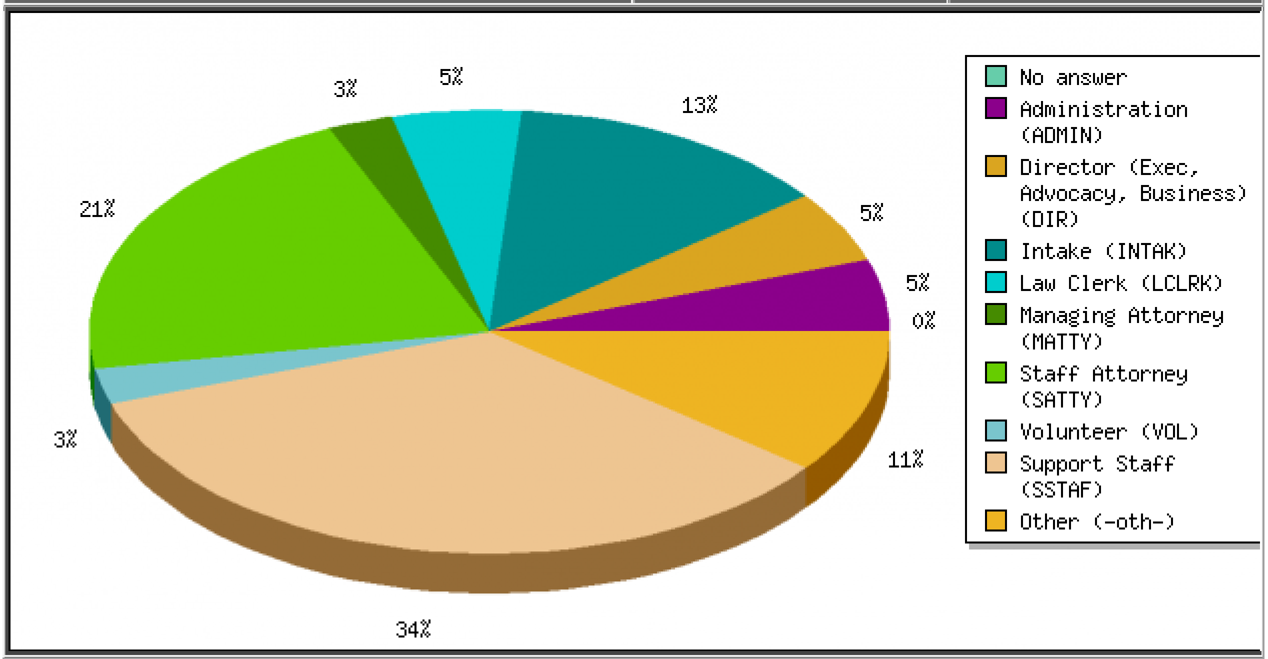
Version: 10.8.07

Appendix F. - Survey and Results

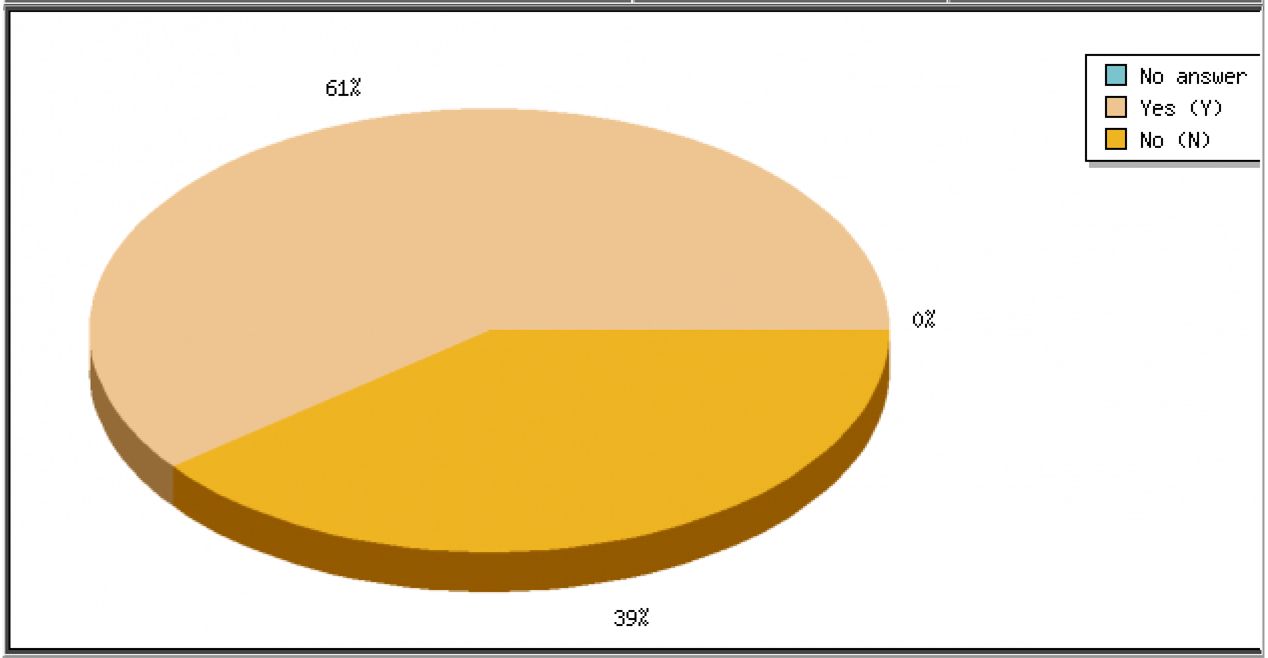
Field Summary for G1Q1:		
What office are you located in?		
Answer	Count	Percentage
No answer	0	0.00%
Ashland (ASH)	3	7.89%
Covington (COV)	17	44.74%
Morehead (MOR)	15	39.47%
Lexington (LEX)	0	0.00%
Probono (PRO)	2	5.26%
Other (-oth-)	1	2.63%



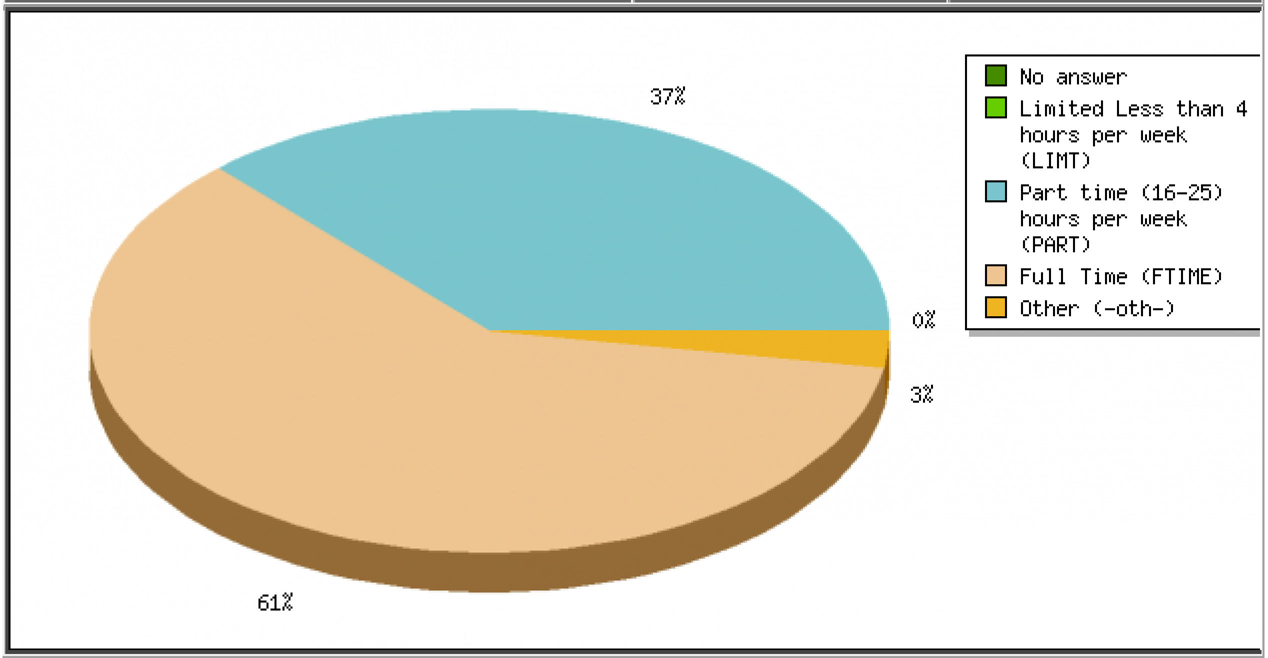
Field Summary for G1Q2:		
What is your function at Legal Aid of the Bluegrass?		
Answer	Count	Percentage
No answer	0	0.00%
Administration (ADMIN)	2	5.26%
Director (Exec, Advocacy, Business) (DIR)	2	5.26%
Intake (INTAK)	5	13.16%
Law Clerk (LCLRK)	2	5.26%
Managing Attorney (MATTY)	1	2.63%
Staff Attorney (SATTY)	8	21.05%
Volunteer (VOL)	1	2.63%
Support Staff (SSTAF)	13	34.21%
Other (-oth-)	4	10.53%



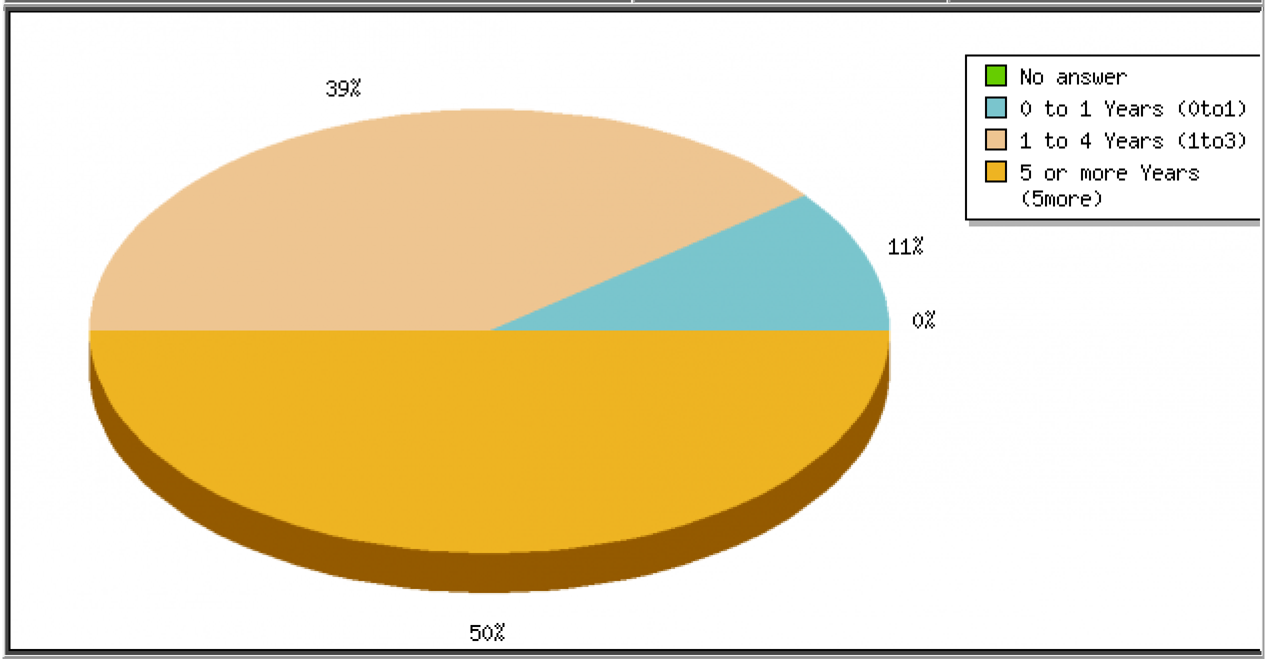
Field Summary for G1Q3:		
Is answering, forwarding or routing calls a major part of your job?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (Y)	23	60.53%
No (N)	15	39.47%



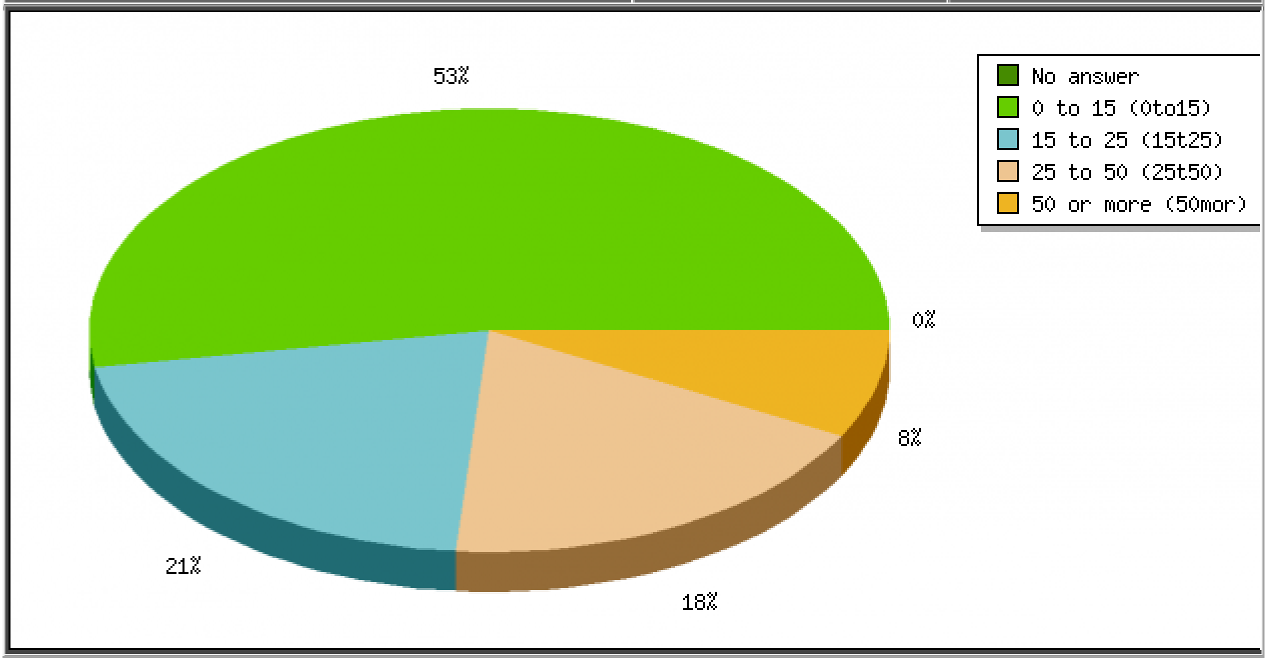
Field Summary for G1Q4:		
What is your standard work schedule		
Answer	Count	Percentage
No answer	0	0.00%
Limited Less than 4 hours per week (LIMT)	0	0.00%
Part time (16-25) hours per week (PART)	14	36.84%
Full Time (FTIME)	23	60.53%
Other (-oth-)	1	2.63%



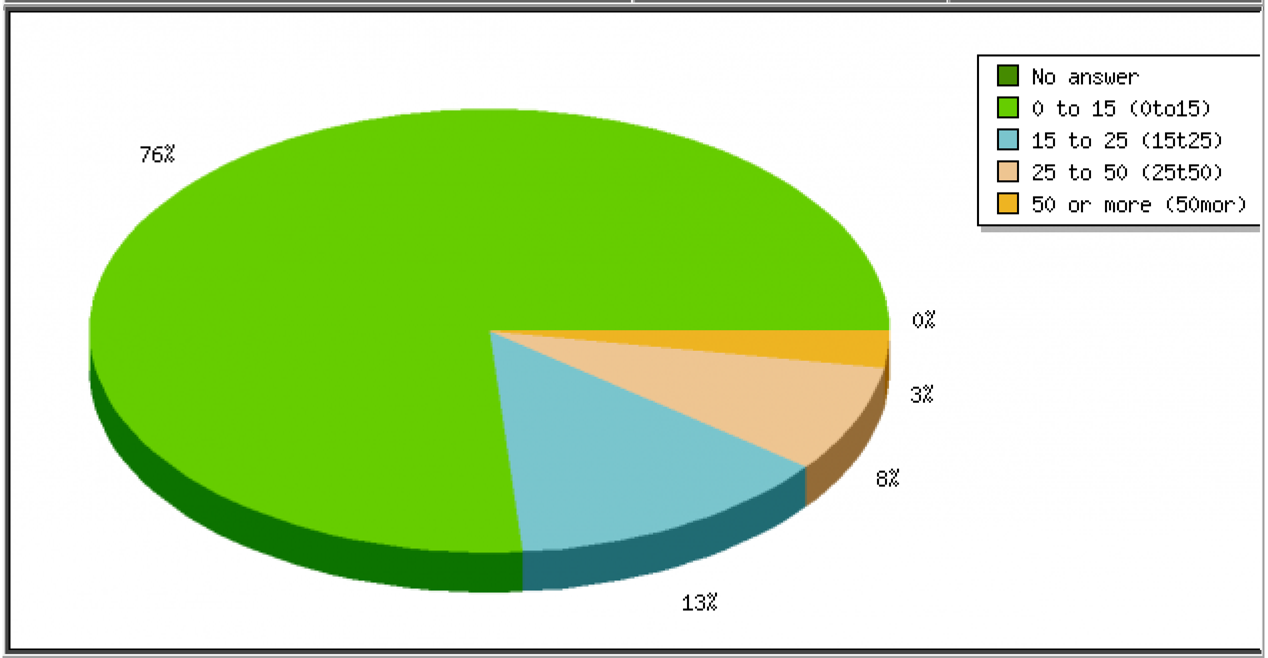
Field Summary for G1Q5:		
How long have you been with Legal Aid of the Bluegrass?		
Answer	Count	Percentage
No answer	0	0.00%
0 to 1 Years (0to1)	4	10.53%
1 to 4 Years (1to3)	15	39.47%
5 or more Years (5more)	19	50.00%



Field Summary for G1Q6:		
How many calls do you place or receive in a day		
Answer	Count	Percentage
No answer	0	0.00%
0 to 15 (0to15)	20	52.63%
15 to 25 (15t25)	8	21.05%
25 to 50 (25t50)	7	18.42%
50 or more (50mor)	3	7.89%



Field Summary for G1Q7:		
How many voicemails do you receive in a day?		
Answer	Count	Percentage
No answer	0	0.00%
0 to 15 (0to15)	29	76.32%
15 to 25 (15t25)	5	13.16%
25 to 50 (25t50)	3	7.89%
50 or more (50mor)	1	2.63%

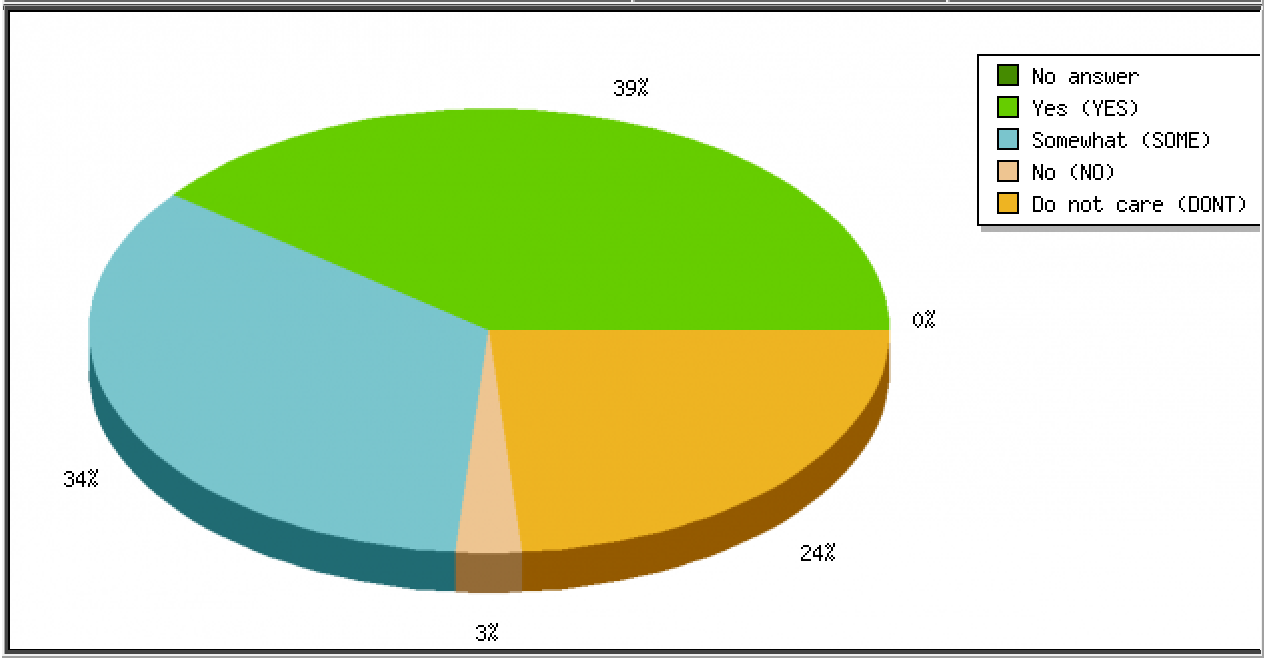


Field Summary for G1Q8:		
Is there something else about your job you feel is important in relation to the way you use your phone?		
Answer	Count	Percentage
Answer	14	36.84%
No answer	24	63.16%

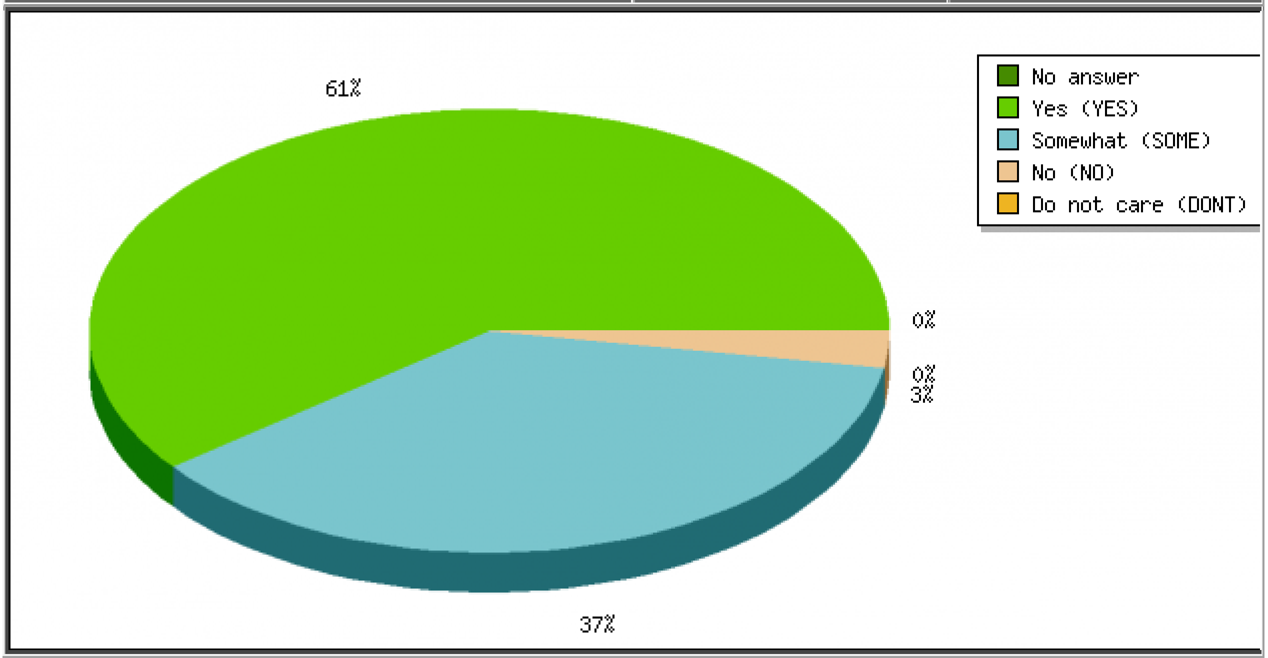
I make lots of calls between the Morehead and Ashland offices .
The people we talk with are most often elderly.
I often have conference calls and I often use a conference call to have a telephone hearing
Recently I have been answering the SHIP hotline
I use phone messages to screen calls as a means of time management.
Dependent on other offices to relay messages or take messages
no
No
NA
The clarity of the caller\'s voice
Point of clarification - I answered with averages; there are certainly many days when I place or receive well more than 50 calls and when I receive well more than 25 voicemails in a day, but on average, more like 25-50 and 15-25. I use my phone to keep track of who I recently spoke to, who I should get getting a call back from, how many calls I have made in an interval, and I use it as a directly in house.
No.
The wireless headset is awesome
no

Wireless headset is helpful @ the LABG office. Would like to have one here @ PB.

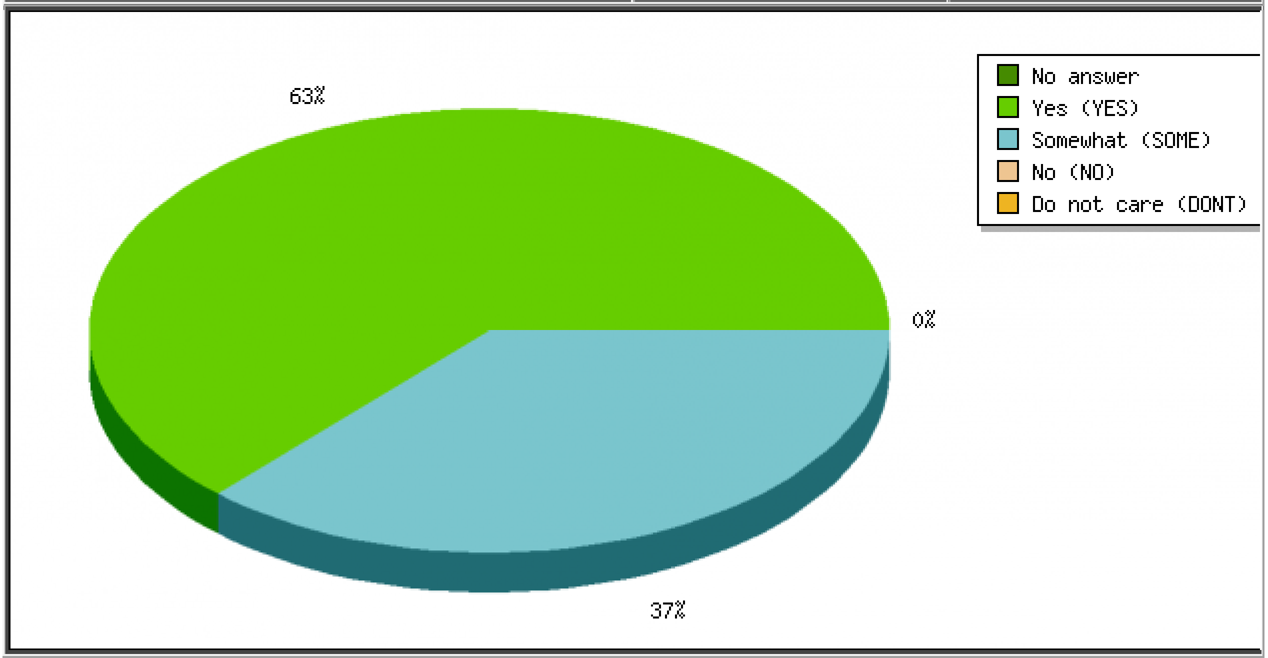
Field Summary for G2Q1:		
Is the phone appealing?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	15	39.47%
Somewhat (SOME)	13	34.21%
No (NO)	1	2.63%
Do not care (DONT)	9	23.68%



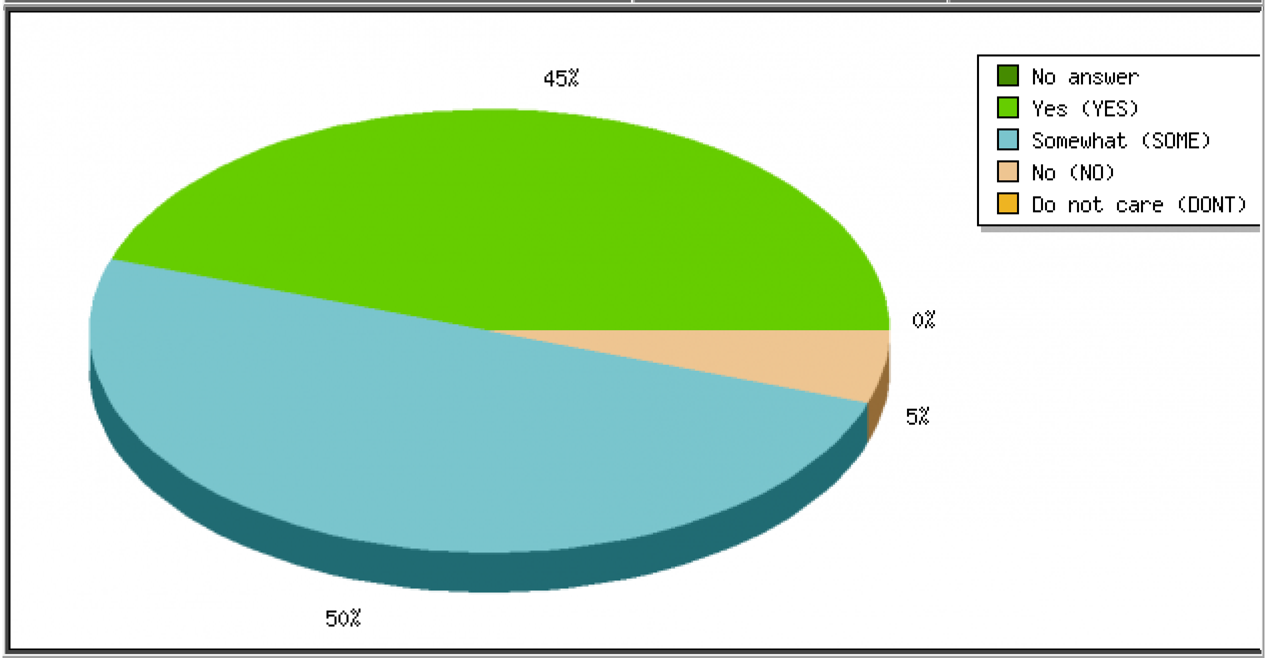
Field Summary for G2Q2:		
Is the phone comfortable to use?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	23	60.53%
Somewhat (SOME)	14	36.84%
No (NO)	1	2.63%
Do not care (DONT)	0	0.00%



Field Summary for G2Q3:		
Is the phone display easy to read and understand?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	24	63.16%
Somewhat (SOME)	14	36.84%
No (NO)	0	0.00%
Do not care (DONT)	0	0.00%



Field Summary for G2Q4:		
Is the button layout easy to navigate?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	17	44.74%
Somewhat (SOME)	19	50.00%
No (NO)	2	5.26%
Do not care (DONT)	0	0.00%



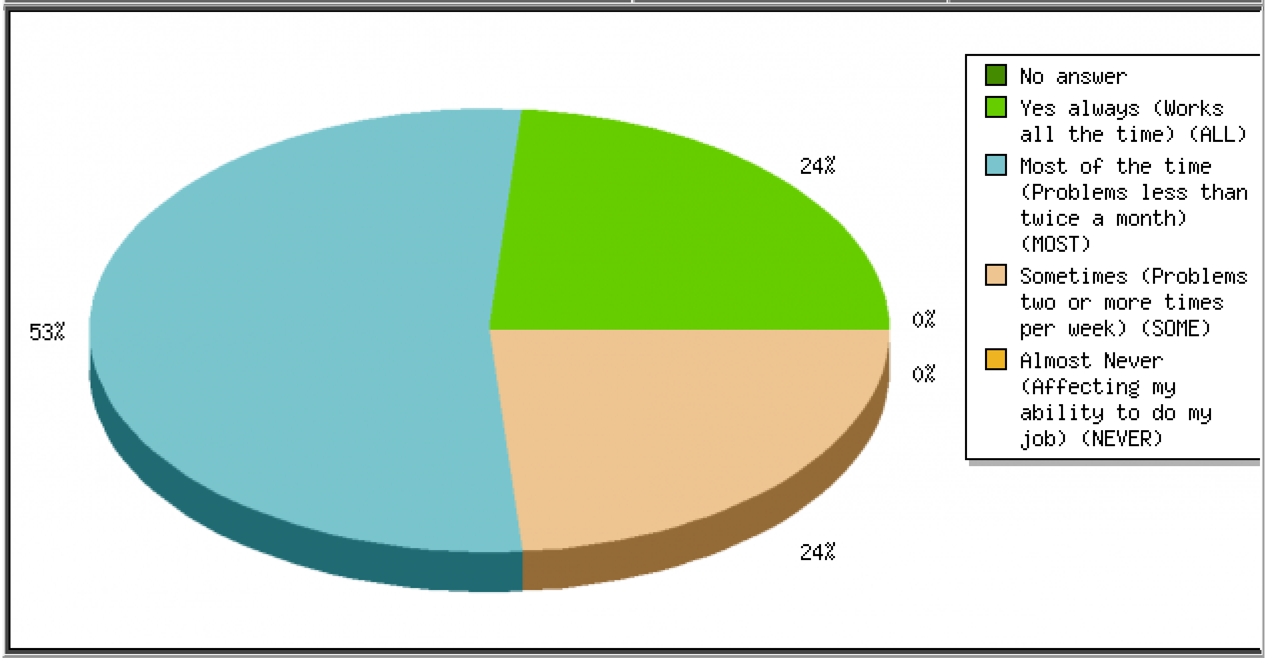
Field Summary for G2Q5:		
Are there any other features missing from the phone that you would like to have? Please explain?		
Answer	**Count**	**Percentage**
Answer	13	34.21%
No answer	0	0.00%

voice mail
I would like the ability to have some personal speed dial numbers on the phone
No
put people on hold then transfer
no
show time on display; easier navigation to check for missed calls
No
NA
No
General caller ID similar to a home phone where the \'whitepages?\' name and number show up, not just the number.
The time and date on voicemail messages.
No
no
Would like to see a company directory. Need date and time of call on messages. Need the private number separated from the rollover. Thee is a loud beep that you hear when on a call and another comes in that we\'d like to eliminate if possible.

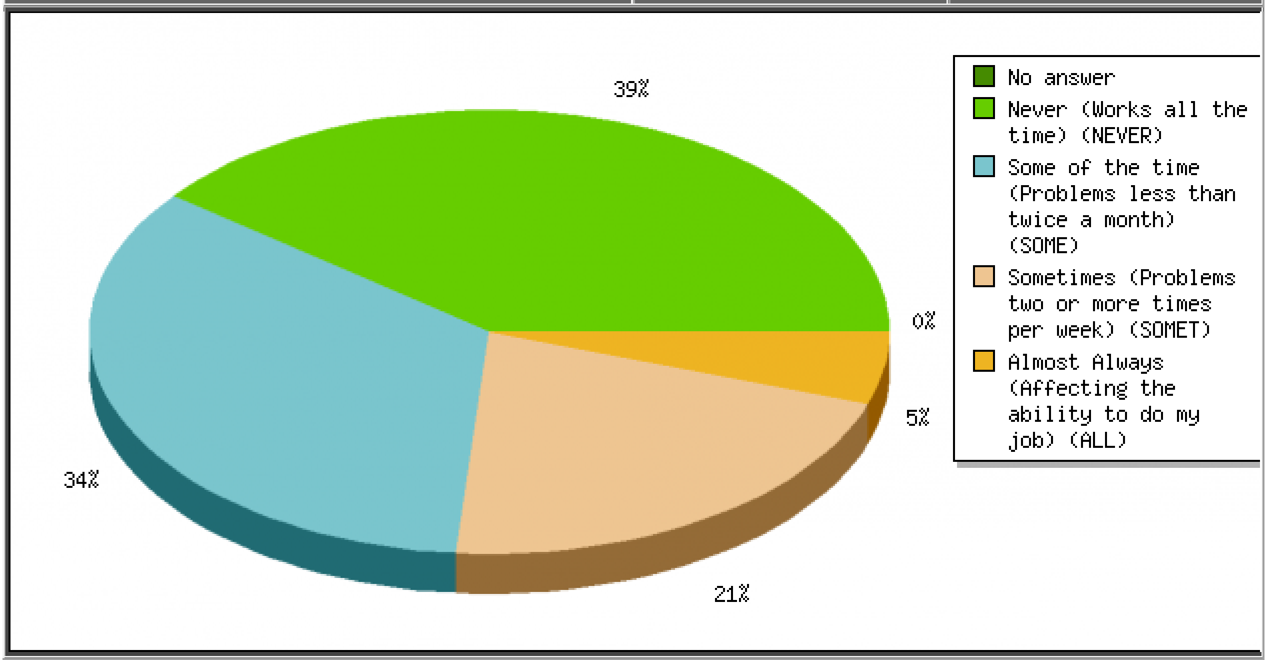
Field Summary for G2Q6:		
Do you have any other hardware related comments?		
Answer	Count	Percentage
Answer	15	39.47%
No answer	0	0.00%

Very hard to place the phone between my ear and shoulder when I need to grab a file really quick while talking with a client. I tried this twice and both times it slipped to the floor - I\'m not sure if that is because the phone is so narrow and curved or not.
We have ongoing problems connecting between the Morehead and Ashland offices with the calls breaking up when you use the 4 digit calling . I have pretty much given up on trying that , but it would be nice if we could get that fixed .
At some point I would like a longer phone cord so I can move the phone to the other side of my desk
Directories button and applications button is confusing.
I need a new headset since it cuts in and out at times.
I find the lists, such as the ones in applications, hard to read. They are too dark or something. I don\'t know, but I have to raise the phone up to read the names w/the dark backgrounds. But, for example, the extensions w/out the dark backgrounds are very easy to read.
I wish we had a \"90%\" mute button so people didn\'t think we hung up on them because it didn\'t go creepy quiet but we could still attend to other matters (e.g. people asking questions in the office)

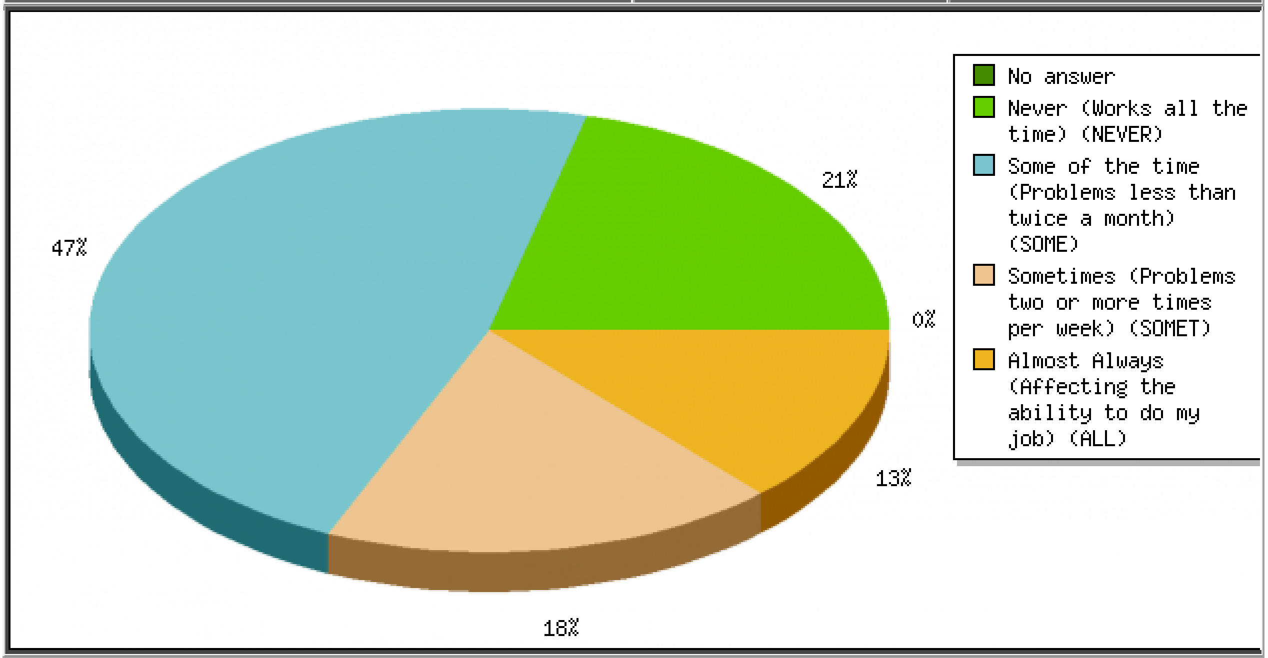
Field Summary for G3Q1:		
Are your phone calls clear?		
Answer	**Count**	**Percentage**
No answer	0	0.00%
Yes always (Works all the time) (ALL)	9	23.68%
Most of the time (Problems less than twice a month) (MOST)	20	52.63%
Sometimes (Problems two or more times per week) (SOME)	9	23.68%
Almost Never (Affecting my ability to do my job) (NEVER)	0	0.00%



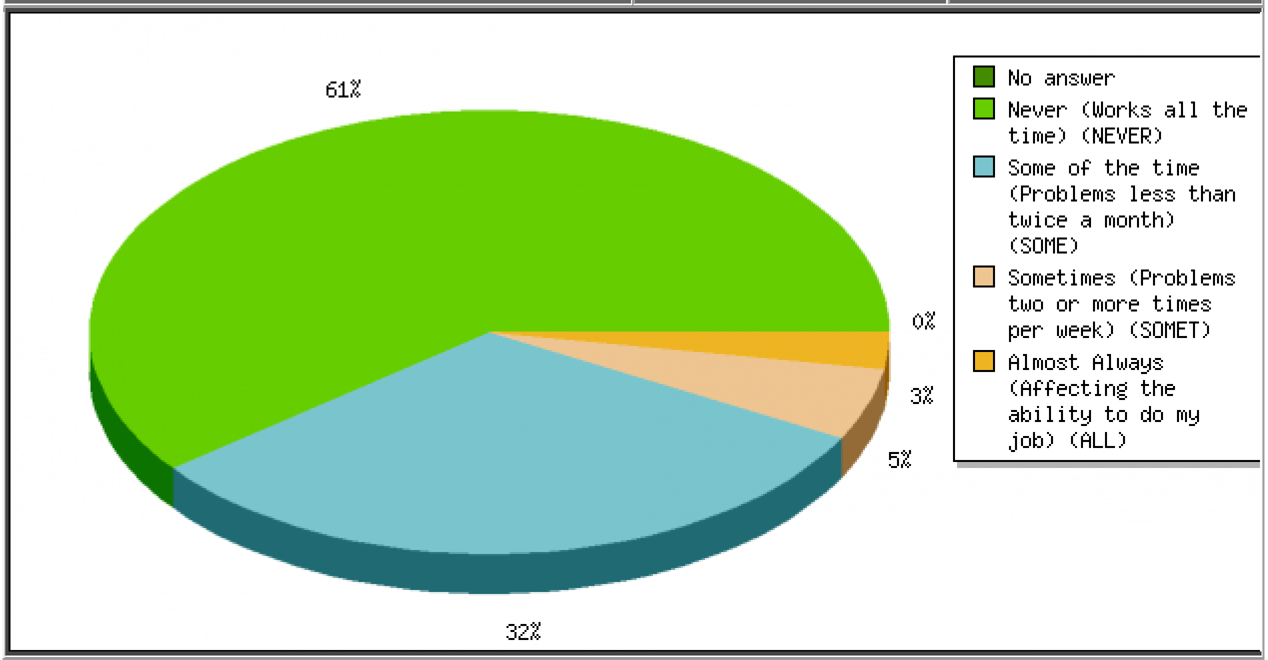
Field Summary for G3Q2:		
Do you experience static while using the phone?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	15	39.47%
Some of the time (Problems less than twice a month) (SOME)	13	34.21%
Sometimes (Problems two or more times per week) (SOMET)	8	21.05%
Almost Always (Affecting the ability to do my job) (ALL)	2	5.26%



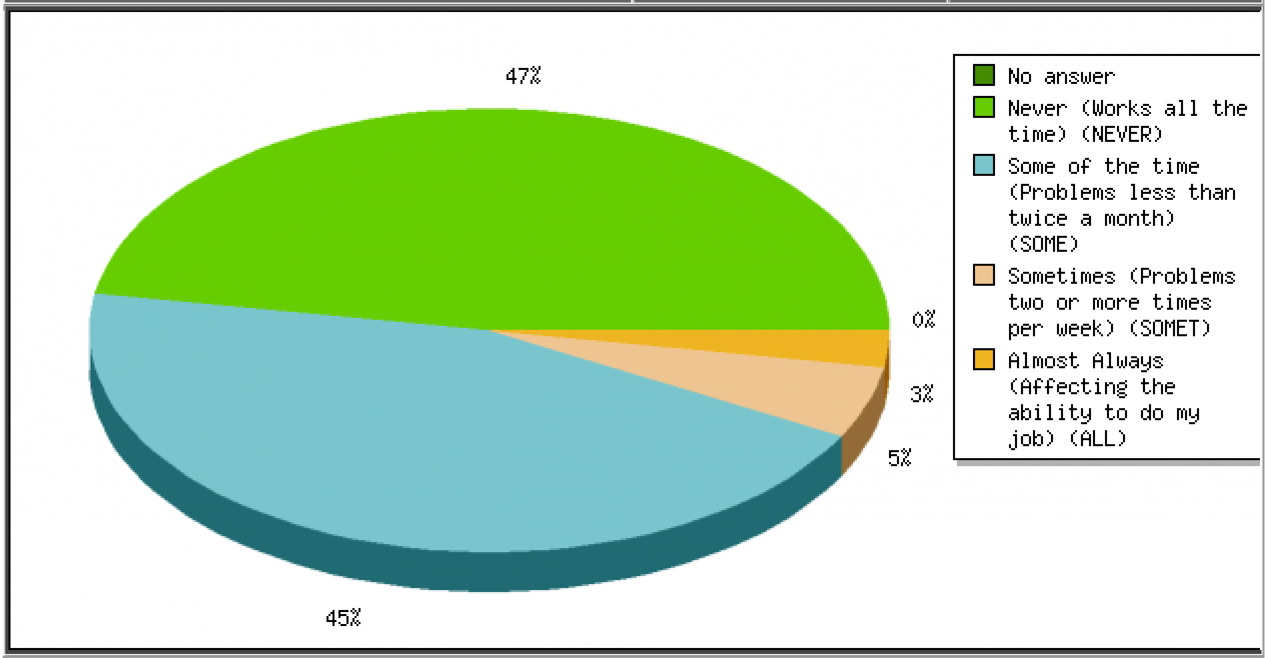
Field Summary for G3Q3:		
Do you experience echo while using the phone?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	8	21.05%
Some of the time (Problems less than twice a month) (SOME)	18	47.37%
Sometimes (Problems two or more times per week) (SOMET)	7	18.42%
Almost Always (Affecting the ability to do my job) (ALL)	5	13.16%



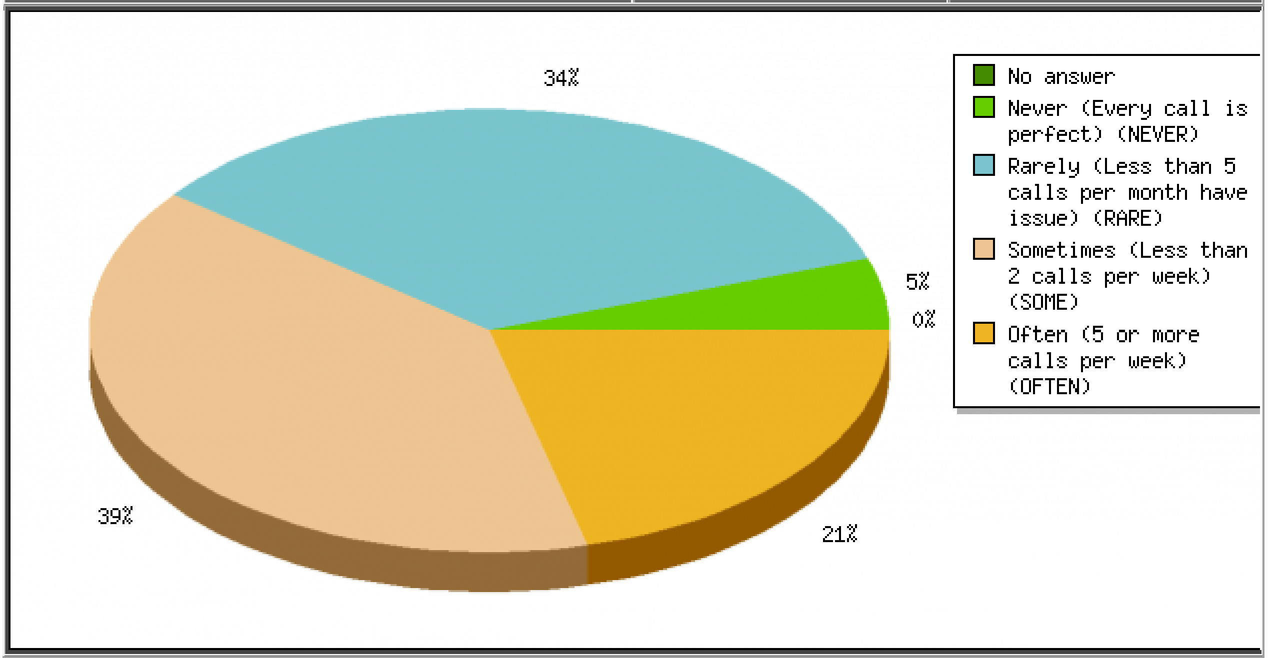
Field Summary for G3Q4:		
Do you experience one way audio while using the phone for example you can hear the person you are calling but they can't hear you and vice versa?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	23	60.53%
Some of the time (Problems less than twice a month) (SOME)	12	31.58%
Sometimes (Problems two or more times per week) (SOMET)	2	5.26%
Almost Always (Affecting the ability to do my job) (ALL)	1	2.63%



Field Summary for G3Q5:		
Do you experience dropped or disconnected calls while using the phone?		
Answer	**Count**	**Percentage**
No answer	0	0.00%
Never (Works all the time) (NEVER)	18	47.37%
Some of the time (Problems less than twice a month) (SOME)	17	44.74%
Sometimes (Problems two or more times per week) (SOMET)	2	5.26%
Almost Always (Affecting the ability to do my job) (ALL)	1	2.63%



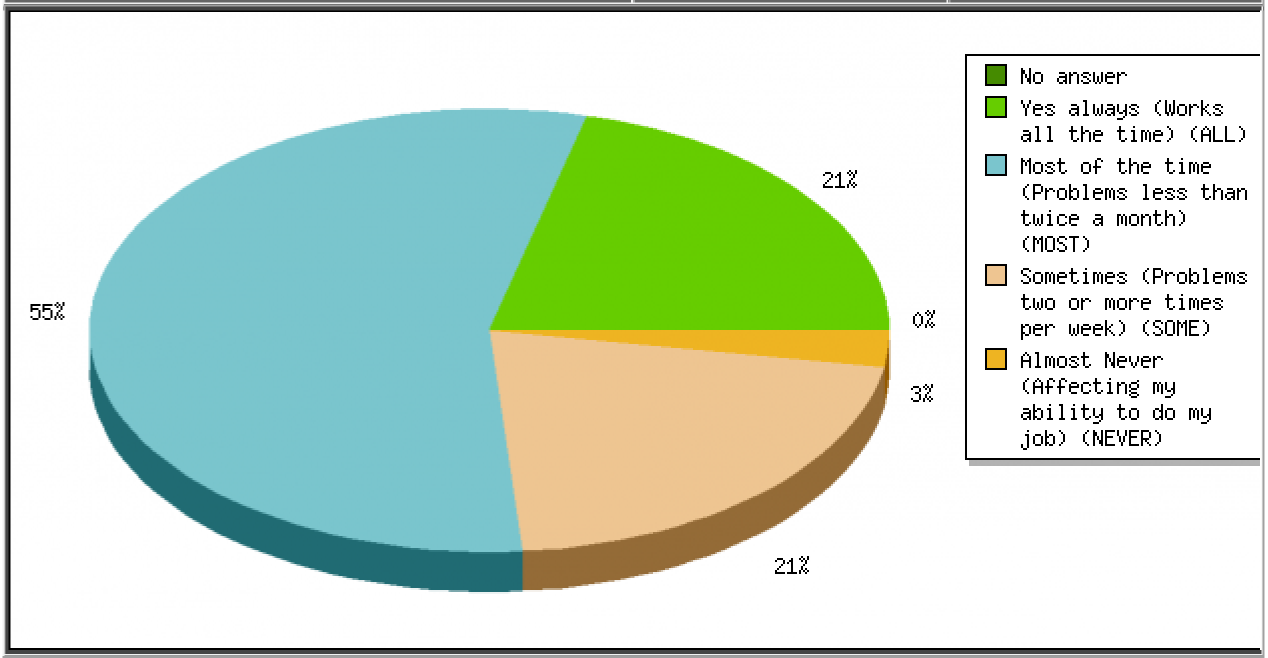
Field Summary for G3Q6:		
How willing are you to tolerate "cell phone" type quality issues if it saves significant long distance costs.		
Answer	**Count**	**Percentage**
No answer	0	0.00%
Never (Every call is perfect) (NEVER)	2	5.26%
Rarely (Less than 5 calls per month have issue) (RARE)	13	34.21%
Sometimes (Less than 2 calls per week) (SOME)	15	39.47%
Often (5 or more calls per week) (OFTEN)	8	21.05%



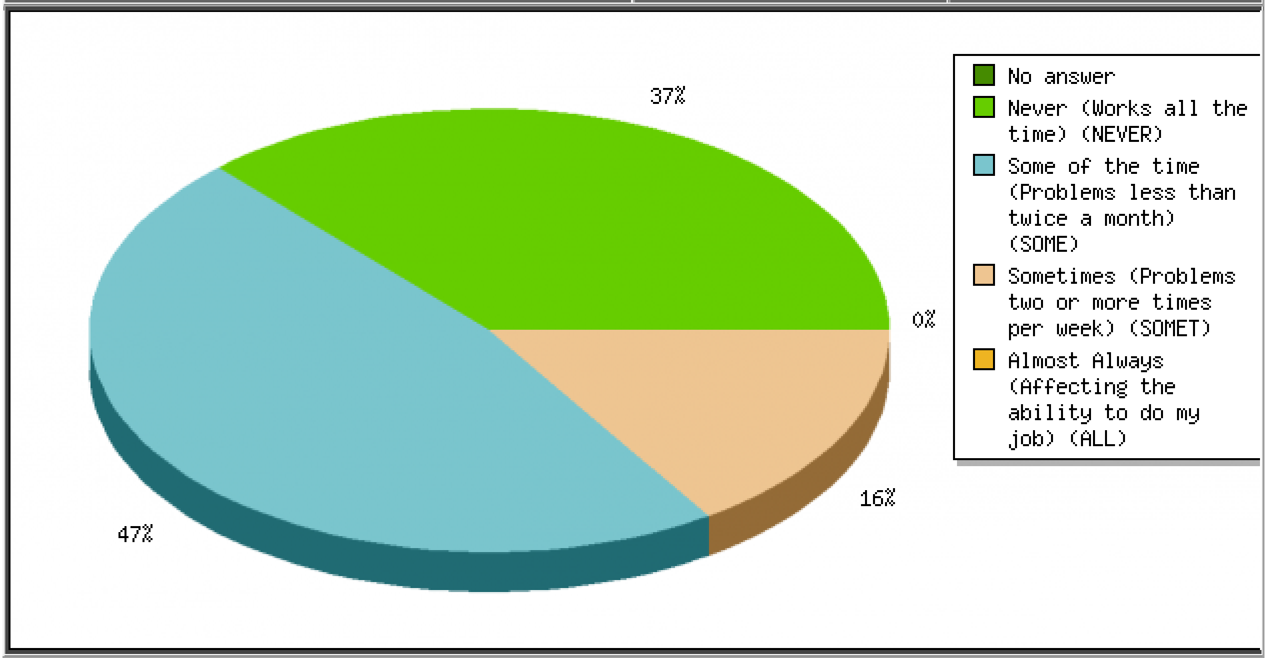
Field Summary for G3Q7:		
Do you have any other comments on topics not covered in this section?		
Answer	Count	Percentage
Answer	9	23.68%
No answer	1	2.63%

I have never used the four digit numbers to call anybody in the other offices so I don't know how good the quality of these interoffice calls are. I really didn\'t have an answer for any of the questions above. I was still under the impression that Morehead could not contact the Ashland office through the 4 digit numbers (which is the office I contact most). I remember when I did try to use this method in the beginning I was never able to get through to the other party I was calling or if I did the call was dropped every single time.
My answers above about static and phone calls being clear only relate to when I use the 4 digit calling between Morehead and Ashland .When I dial the entire number there are no problems . Also no problems calling from Morehead to Covington .
Sometimes when a person is speaking the voice completely cuts out. If I am talking with them, I can ask them to repeat. However, if they are leaving a message, that part of the message is missing. Fortunately, it is more likely to happen during a direct conversation than in a left voice message.

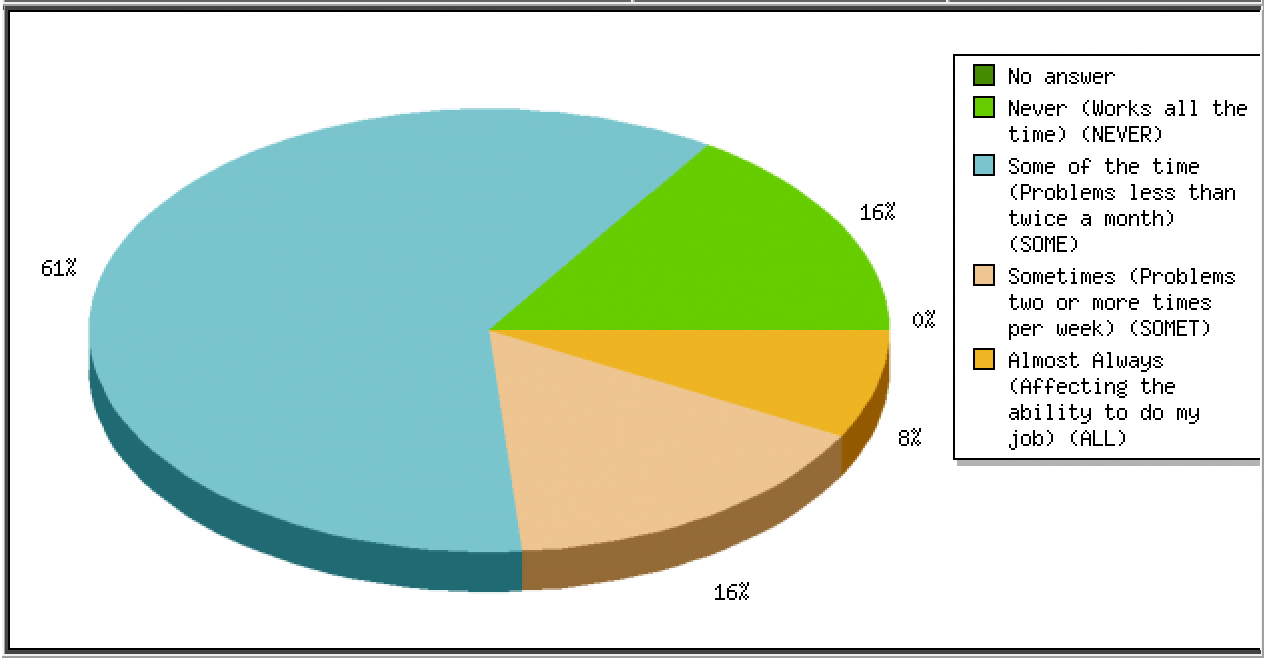
Field Summary for G4Q1:		
Are your phone calls clear?		
Answer	Count	Percentage
No answer	0	0.00%
Yes always (Works all the time) (ALL)	8	21.05%
Most of the time (Problems less than twice a month) (MOST)	21	55.26%
Sometimes (Problems two or more times per week) (SOME)	8	21.05%
Almost Never (Affecting my ability to do my job) (NEVER)	1	2.63%



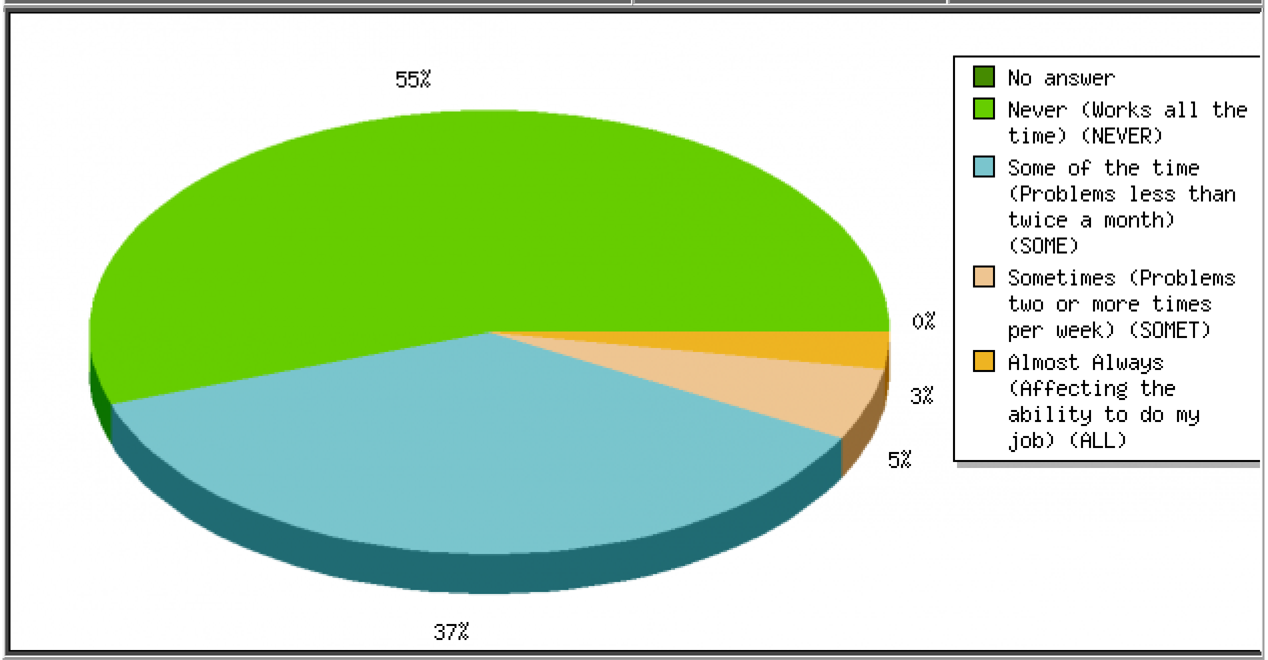
Field Summary for G4Q2:		
Do you experience static while using the phone?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	14	36.84%
Some of the time (Problems less than twice a month) (SOME)	18	47.37%
Sometimes (Problems two or more times per week) (SOMET)	6	15.79%
Almost Always (Affecting the ability to do my job) (ALL)	0	0.00%



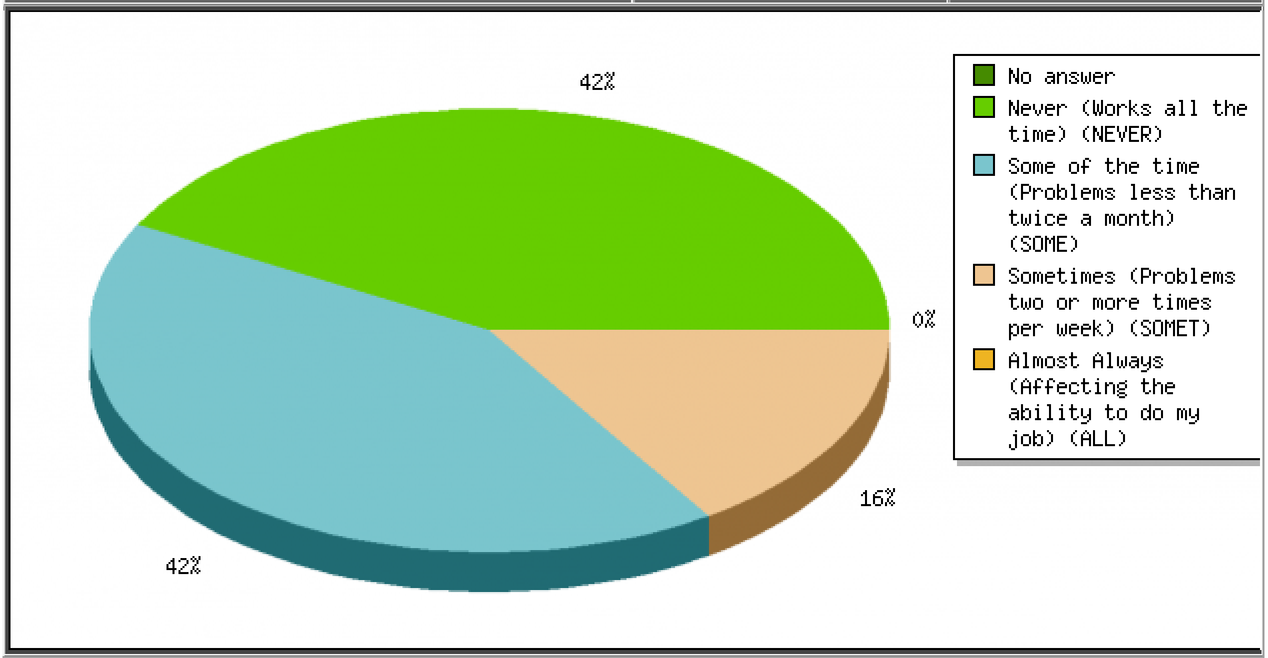
Field Summary for G4Q3:		
Do you experience echo while using the phone?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	6	15.79%
Some of the time (Problems less than twice a month) (SOME)	23	60.53%
Sometimes (Problems two or more times per week) (SOMET)	6	15.79%
Almost Always (Affecting the ability to do my job) (ALL)	3	7.89%



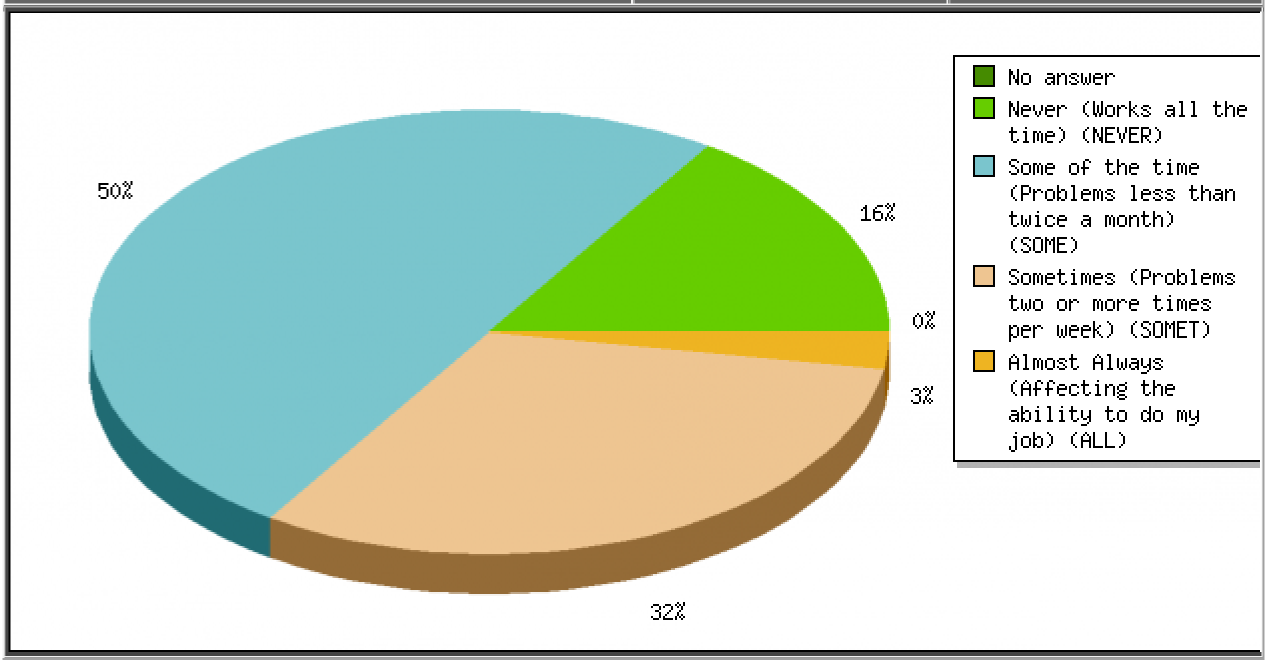
Field Summary for G4Q4:		
Do you experience one way audio while using the phone, for example, you can hear the person you are calling but they can't hear you and vice versa?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	21	55.26%
Some of the time (Problems less than twice a month) (SOME)	14	36.84%
Sometimes (Problems two or more times per week) (SOMET)	2	5.26%
Almost Always (Affecting the ability to do my job) (ALL)	1	2.63%



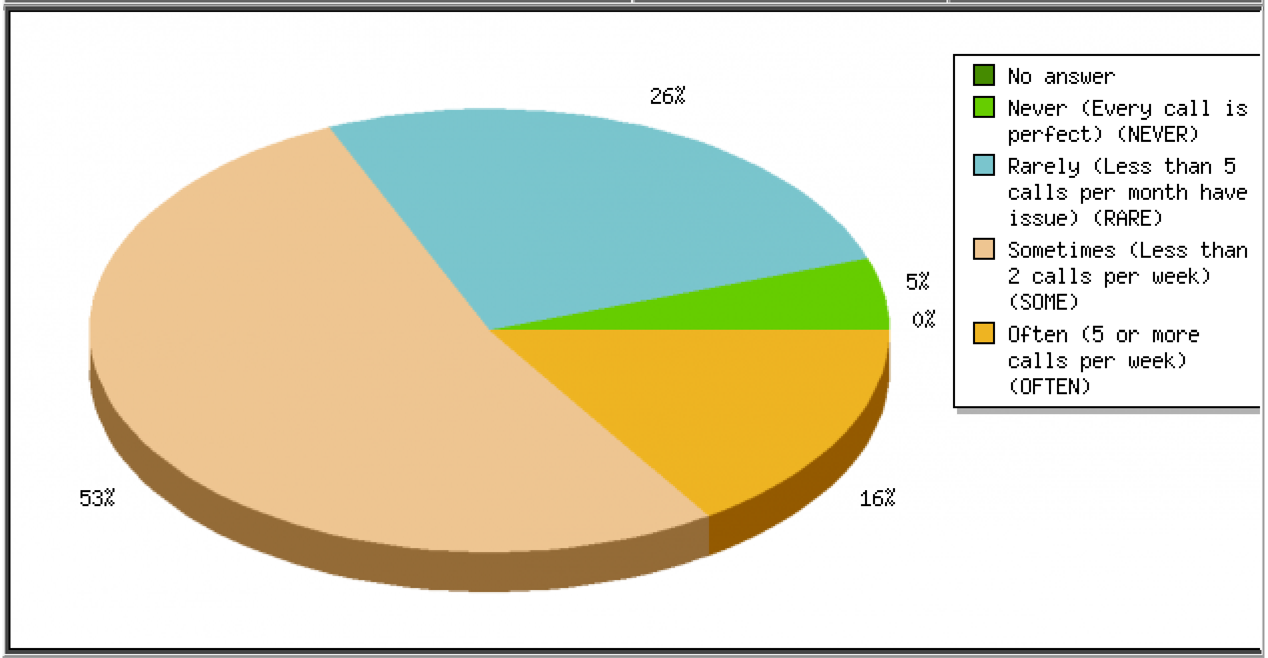
Field Summary for G4Q5:		
Do you experience dropped or disconnected calls while using the phone?		
Answer	**Count**	**Percentage**
No answer	0	0.00%
Never (Works all the time) (NEVER)	16	42.11%
Some of the time (Problems less than twice a month) (SOME)	16	42.11%
Sometimes (Problems two or more times per week) (SOMET)	6	15.79%
Almost Always (Affecting the ability to do my job) (ALL)	0	0.00%



Field Summary for G4Q6:		
Do you experience more problems with calls to and from cell phones?		
Answer	Count	Percentage
No answer	0	0.00%
Never (Works all the time) (NEVER)	6	15.79%
Some of the time (Problems less than twice a month) (SOME)	19	50.00%
Sometimes (Problems two or more times per week) (SOMET)	12	31.58%
Almost Always (Affecting the ability to do my job) (ALL)	1	2.63%



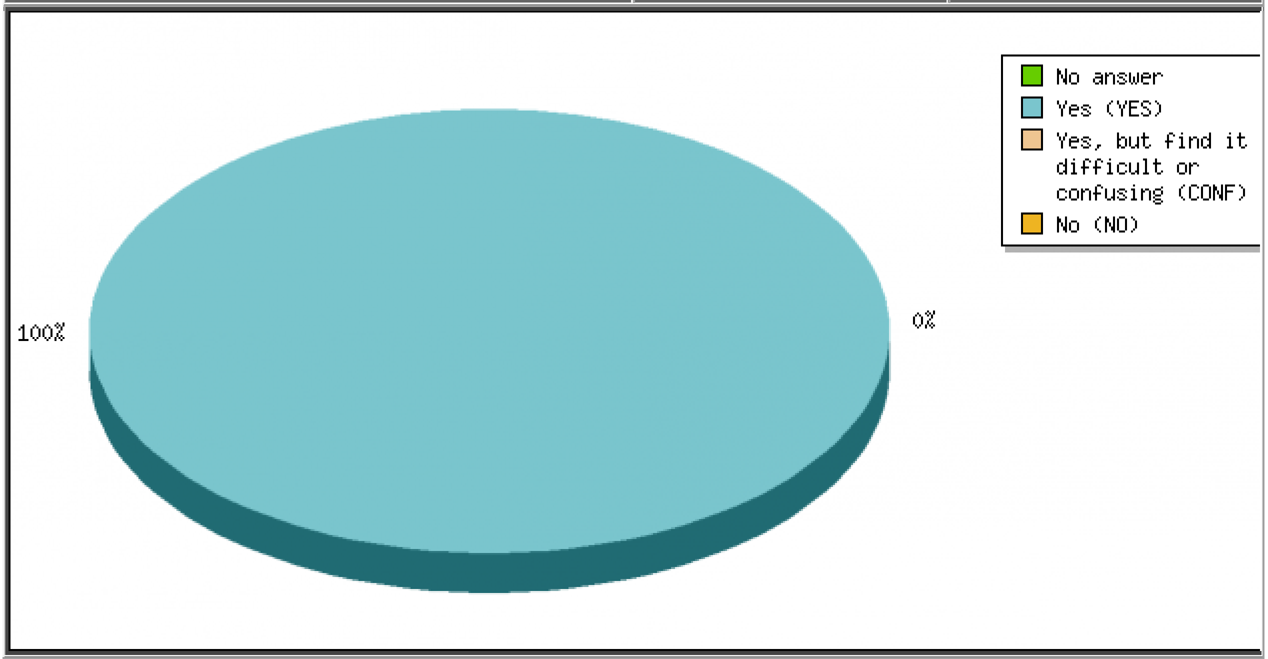
Field Summary for G4Q7:		
How willing are you to tolerate "cell phone" type quality issues if it saves significant costs.		
Answer	Count	Percentage
No answer	0	0.00%
Never (Every call is perfect) (NEVER)	2	5.26%
Rarely (Less than 5 calls per month have issue) (RARE)	10	26.32%
Sometimes (Less than 2 calls per week) (SOME)	20	52.63%
Often (5 or more calls per week) (OFTEN)	6	15.79%



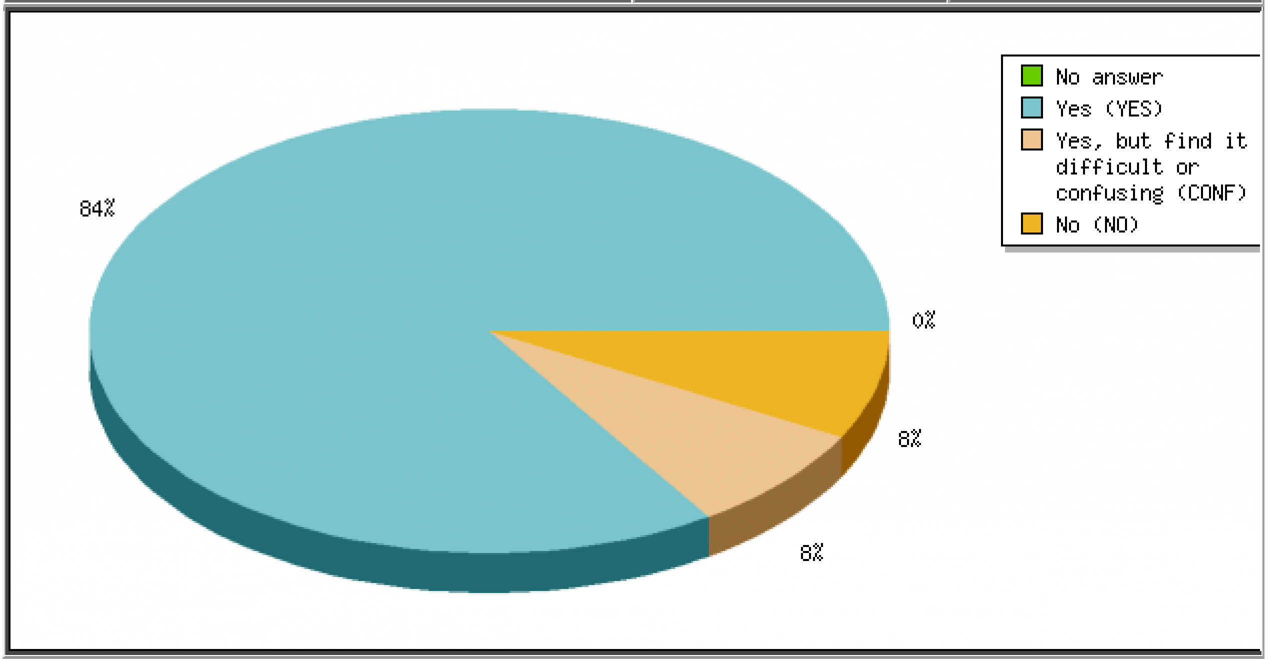
Field Summary for G4Q8:		
Do you have any other comments on topics not covered in this section?		
Answer	Count	Percentage
Answer	13	34.21%
No answer	0	0.00%

I have had dropped calls several times and it is very embarrassing to have to explain to my clients who all have prepaid cell phones that my phone system is the problem and not theirs. I end up wasting more of their minutes having to call them two or three times in a row. Although this is embarrassing I am even more concerned about this occurring while I am speaking with a Judge or another attorney. I have also had the problem where I have left important messages for clients and been told they never received them. This has happened with more than one client.
The one way audio can be annoying , especially when doing hearings over the telephone .
I am not as observant as I should be regarding documenting when it is happening. I usually just deal with it and go on unless it is repeated multiple times within a short time period.
Cell phone answer isn\'t with our phone lines, the problem is with the cell phones clients use, they cut out often Use to have a lot of problems with placing a call out, but that seems to be fixed now.
Most problems that I experience with call back messages are with the caller's use of their cell phone on the road, or in an area where this a weak signal.
Everytime I lose a call or there is poor call quality, I\'m fairly certain it is due to the cell phone on the other end.
I've noticed that the cell phone will cut out while the person leaves a message. I feel that poor external call quality is due to the cell phone or a dying cordless phone.

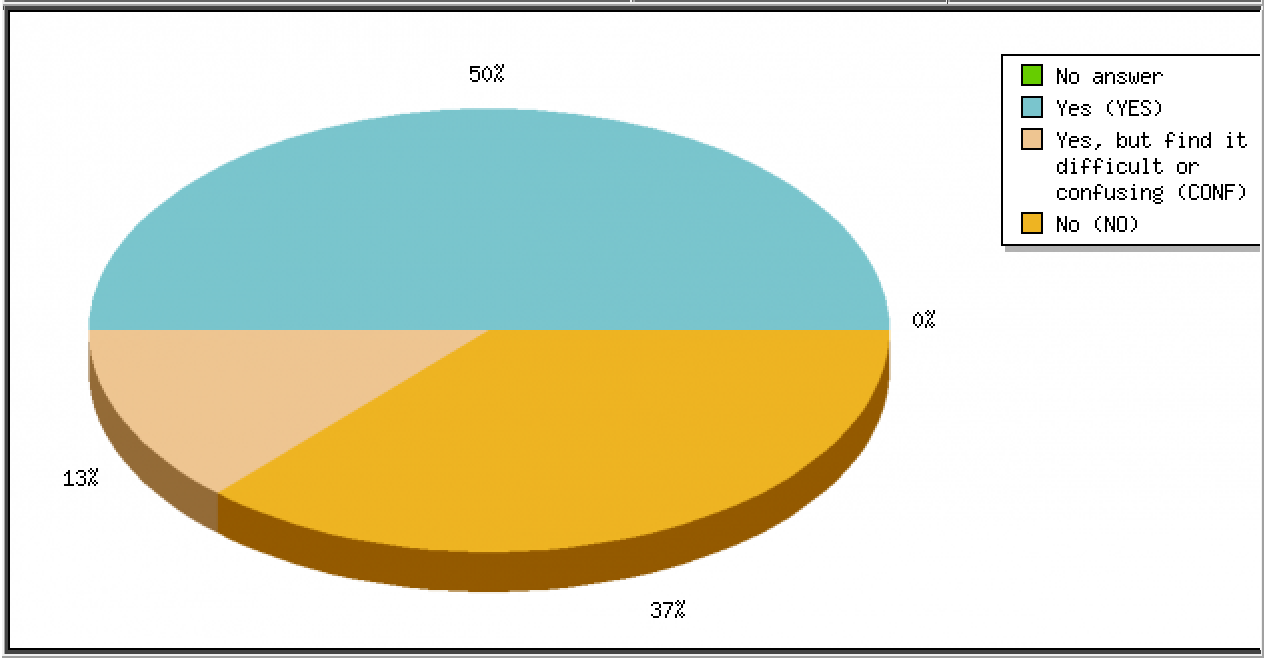
Field Summary for G5Q1:		
Do you feel comfortable placing and receiving calls?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	38	100.00%
Yes, but find it difficult or confusing (CONF)	0	0.00%
No (NO)	0	0.00%



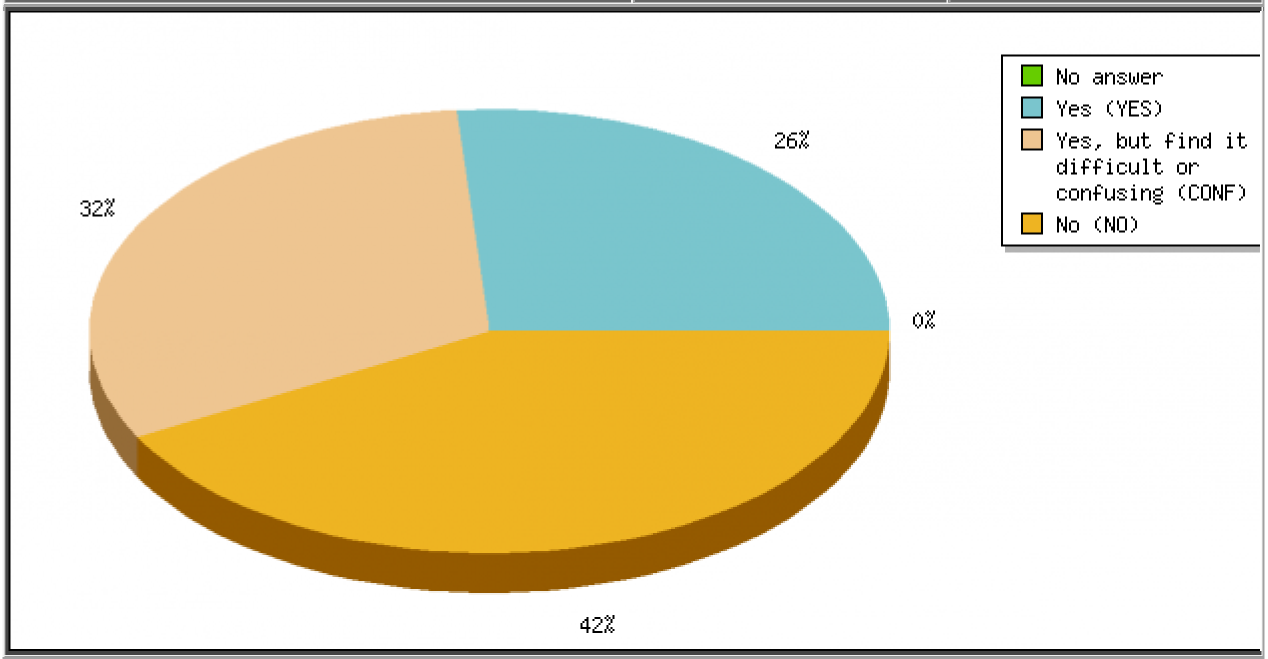
Field Summary for G5Q2:		
Do you feel comfortable placing calls on hold?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	32	84.21%
Yes, but find it difficult or confusing (CONF)	3	7.89%
No (NO)	3	7.89%



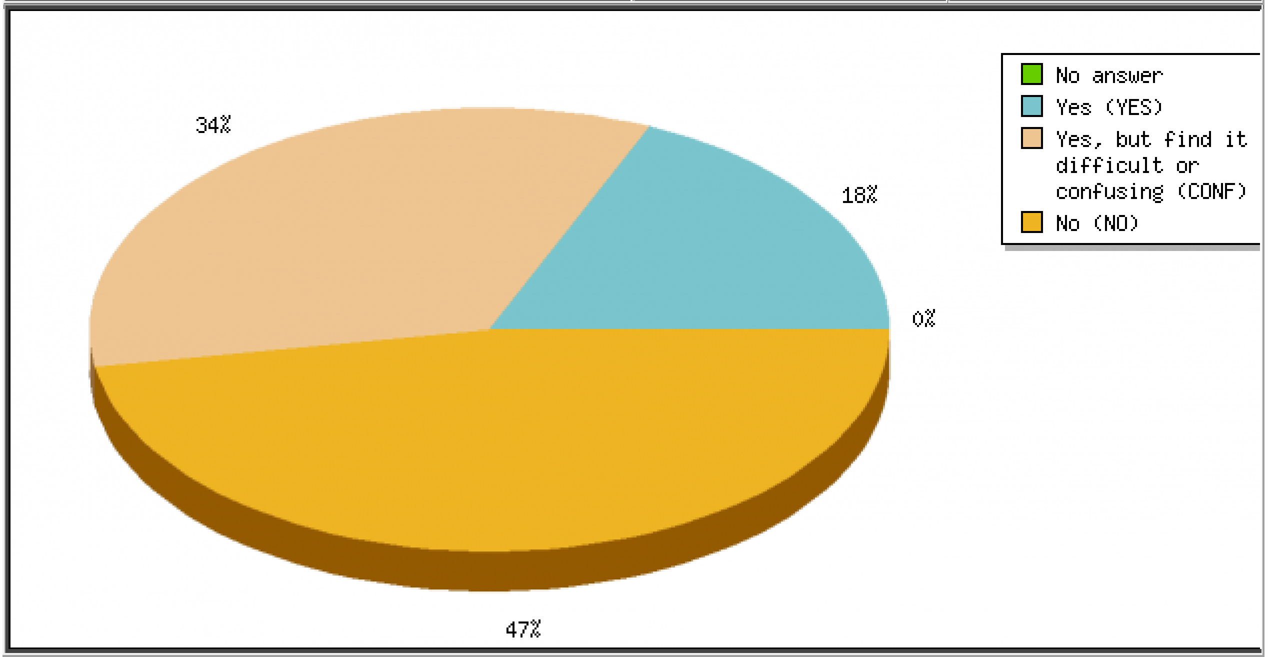
Field Summary for G5Q3:		
Do you feel comfortable answering additional calls while on a call? Example: Placing your current call on hold and picking up a second line.		
Answer	**Count**	**Percentage**
No answer	0	0.00%
Yes (YES)	19	50.00%
Yes, but find it difficult or confusing (CONF)	5	13.16%
No (NO)	14	36.84%



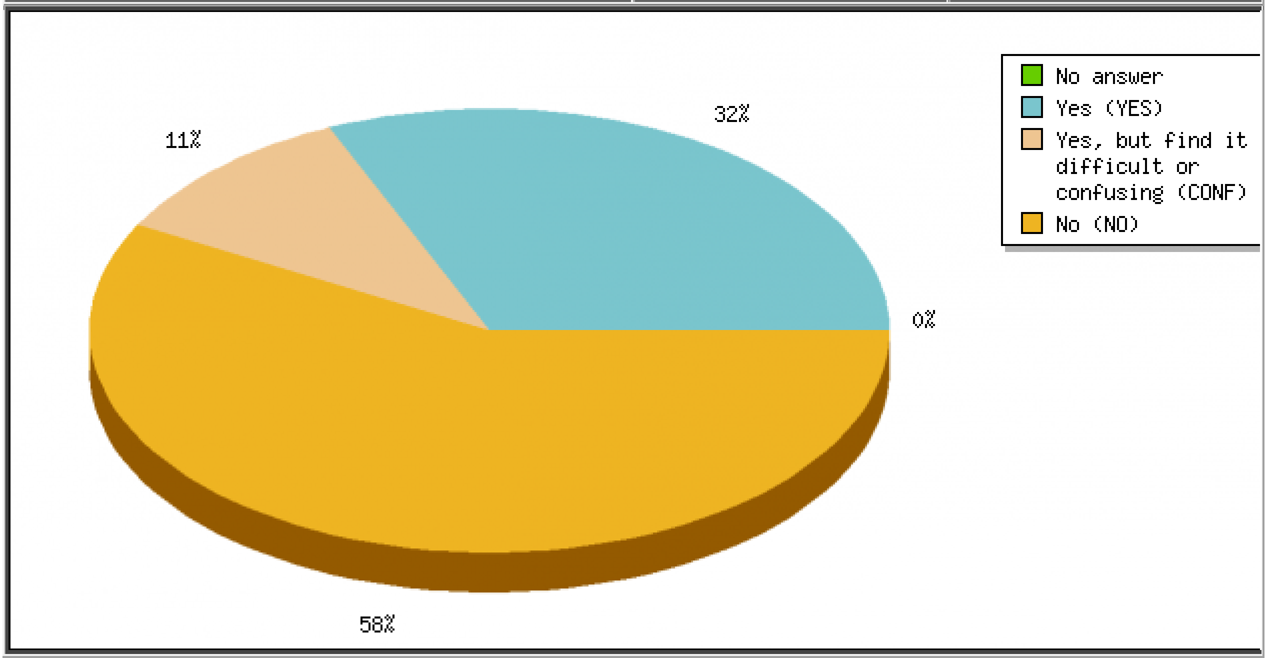
Field Summary for G5Q4:		
Do you feel comfortable transferring calls to others?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	10	26.32%
Yes, but find it difficult or confusing (CONF)	12	31.58%
No (NO)	16	42.11%



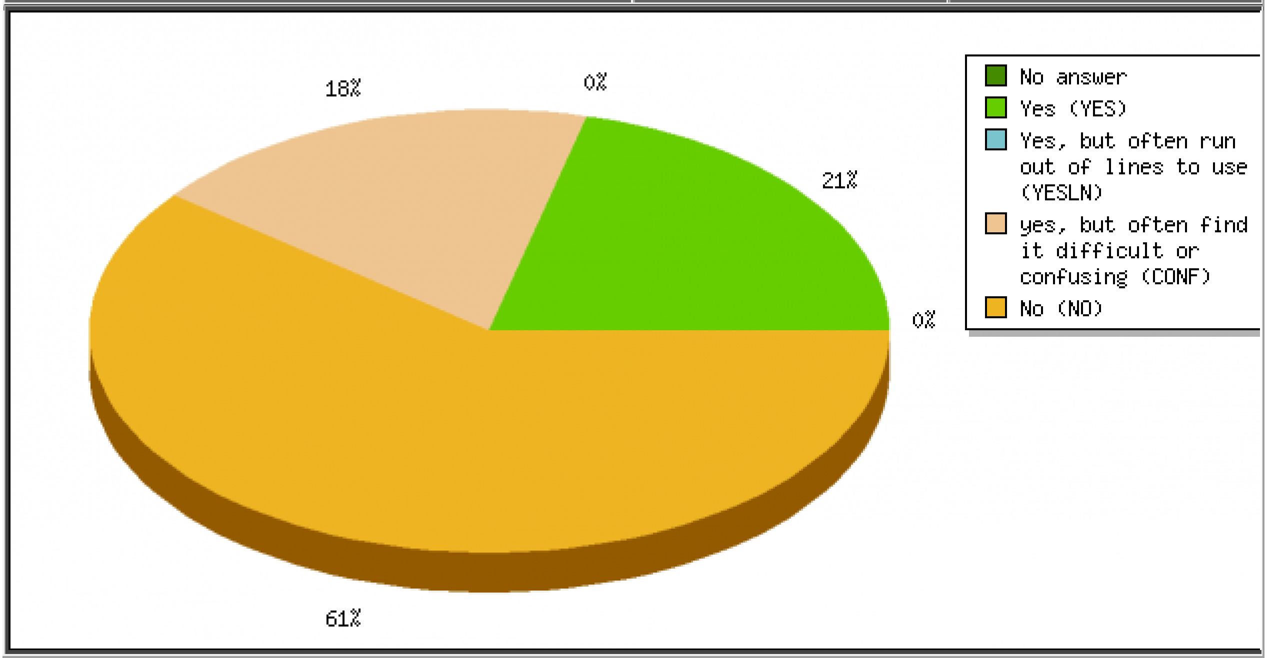
Field Summary for G5Q5:		
Do you transfer calls to others in one step? Example: You press transfer, contact another person introduce the caller and then press transfer again?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	7	18.42%
Yes, but find it difficult or confusing (CONF)	13	34.21%
No (NO)	18	47.37%



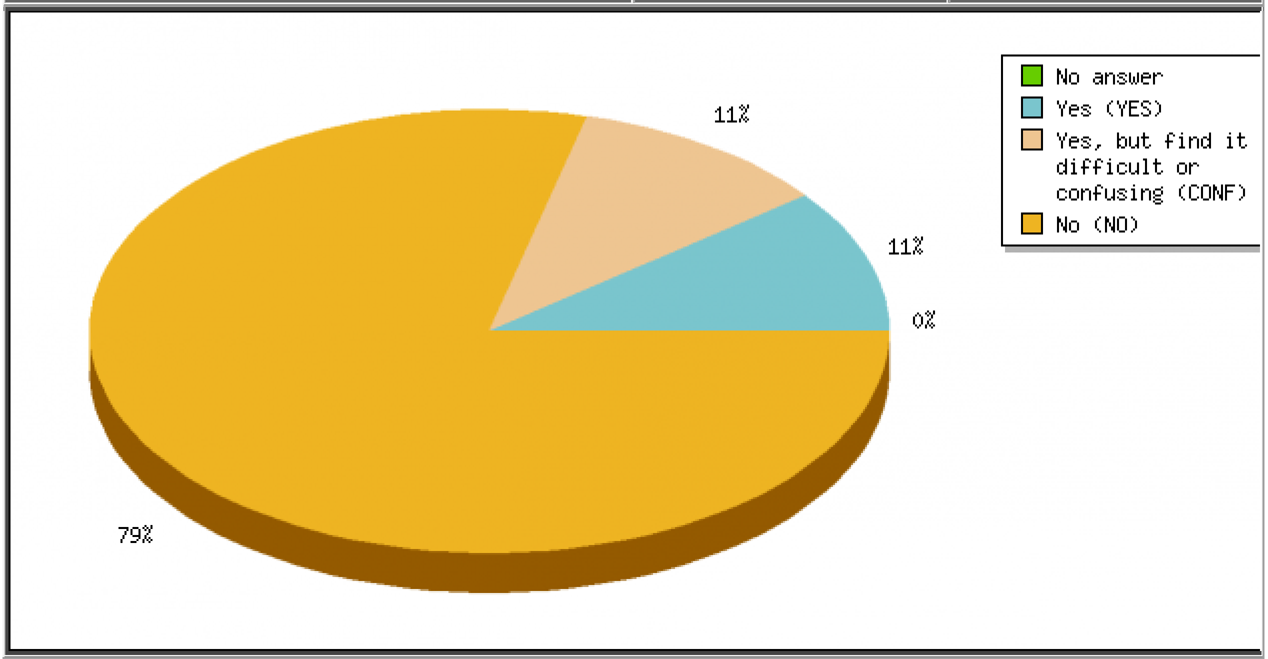
Field Summary for G5Q6:		
Do you know how to blind transfer calls? Example: You transfer call to someone without calling first?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	12	31.58%
Yes, but find it difficult or confusing (CONF)	4	10.53%
No (NO)	22	57.89%



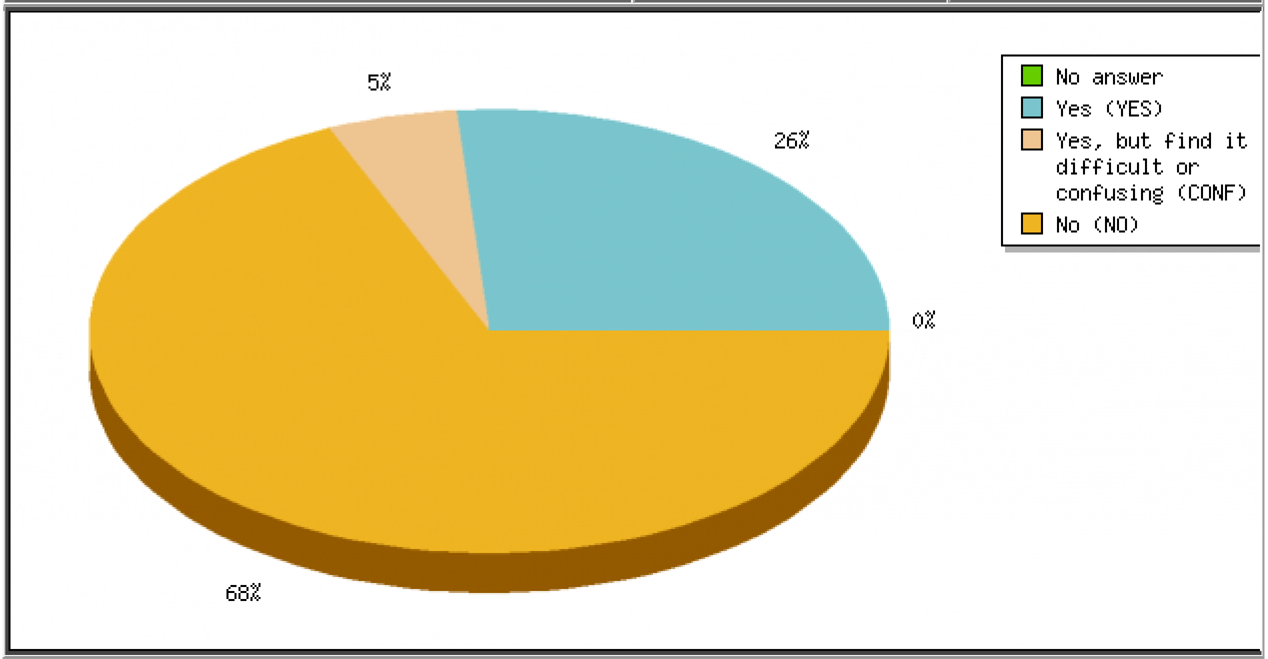
Field Summary for G5Q7:		
Do you know how to conference more than one call?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	8	21.05%
Yes, but often run out of lines to use (YESLN)	0	0.00%
yes, but often find it difficult or confusing (CONF)	7	18.42%
No (NO)	23	60.53%



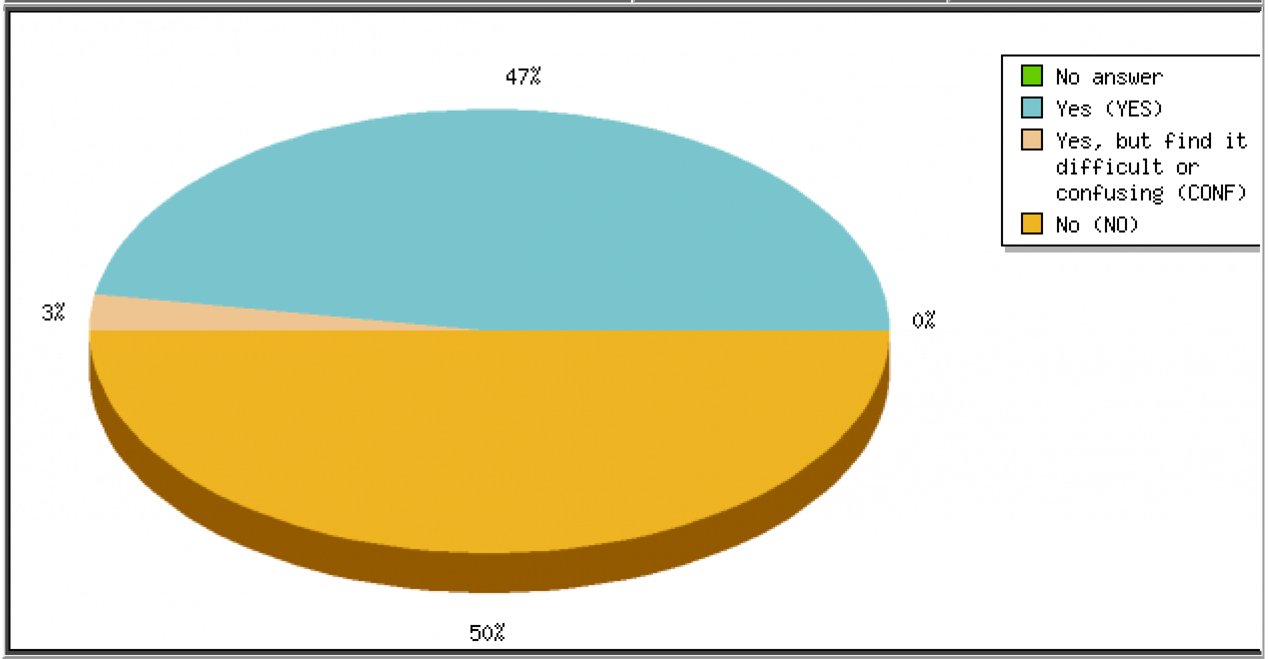
Field Summary for G5Q8:		
Do you know how to Park a call?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	4	10.53%
Yes, but find it difficult or confusing (CONF)	4	10.53%
No (NO)	30	78.95%



Field Summary for G5Q9:		
Do you know how to Forward your extension to another number?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	10	26.32%
Yes, but find it difficult or confusing (CONF)	2	5.26%
No (NO)	26	68.42%



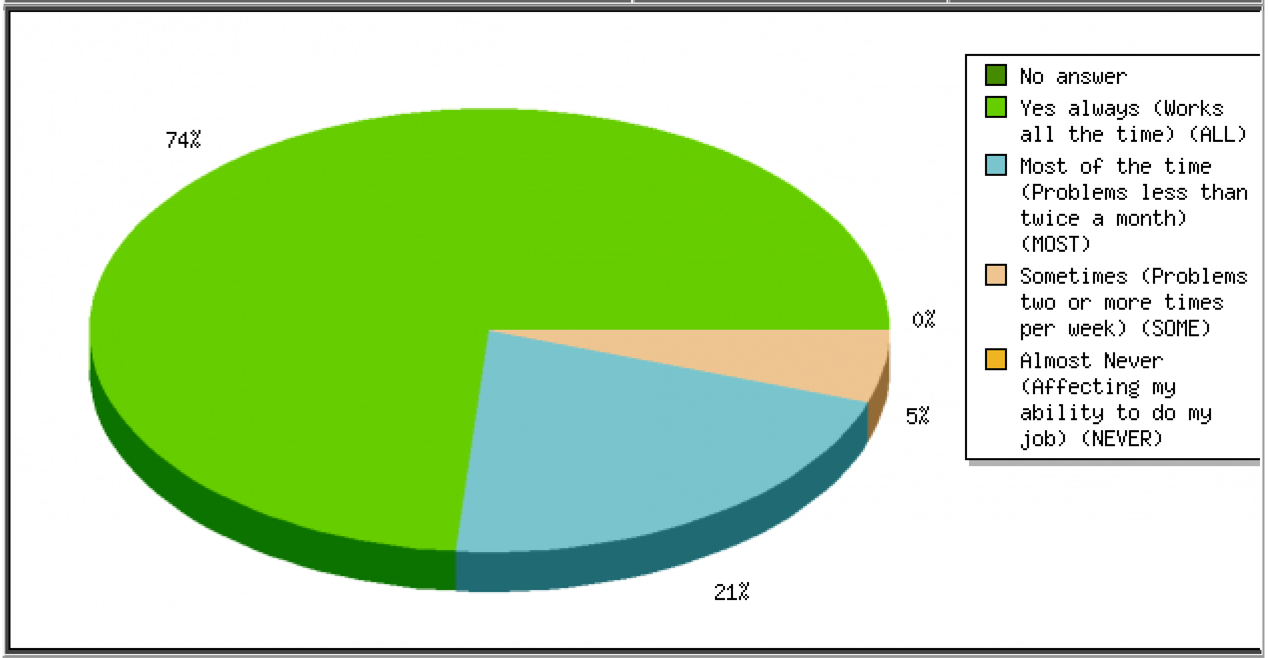
Field Summary for G5Q10:		
Do you know how to use the intercom feature?		
Answer	Count	Percentage
No answer	0	0.00%
Yes (YES)	18	47.37%
Yes, but find it difficult or confusing (CONF)	1	2.63%
No (NO)	19	50.00%



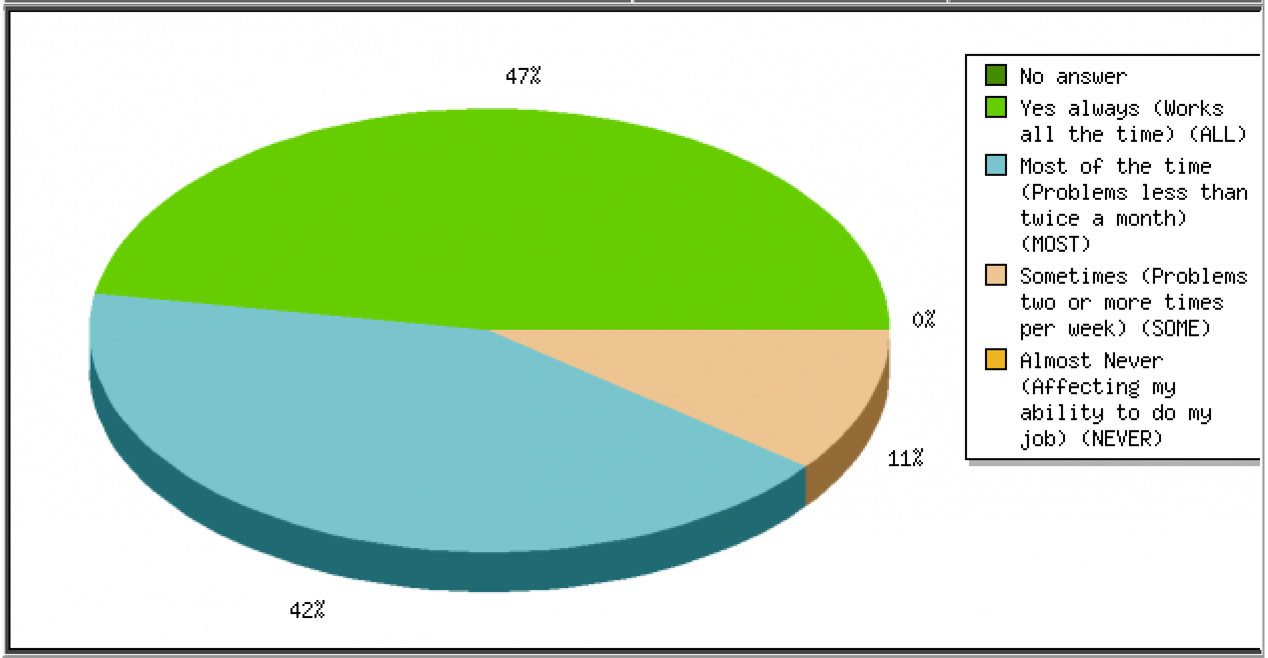
Field Summary for G5Q11:		
Do you have any other comments on topics not covered in this section?		
Answer	Count	Percentage
Answer	12	31.58%
No answer	0	0.00%

I do like the conference feature. I find it much easier to use than on the old phone. I have only used it to contact two other people but found that I only had to hit the conference button and it was done. I usually end up hanging up on people when I try to transfer so I usually just take a message and send an email to someone. Also, I thought that if we were on one line and someone else called their number was supposed to flash on the screen so we could decide whether this was a call we should take. When I have a second call coming in, their number does not appear on the screen. If there is a way to see who is calling me that would be great.
I have been unable to transfer calls using the instructions I have and would like to be able to transfer calls. As for the other features, I do not know how to use them because I don\'t have much use for them so I haven\'t tried to figure out how they work.
I don\'t know how to do some of these things right off the top of my head, but when I need to do them I pull out my instructions and can usually complete the task without any problem, except for call conferencing. I have never been able to do conferencing even by following the directions. I find conferencing to be the most confusing and frustrating thing about these phones.
I feel we need some training or one on one help from someone in the office who knows how to do this stuff. I don\'t need it enough to be familiar with how to do it.
need training in the above
What\'s park a call?
Some of the issues I have with the phone system is when I cover the front desk. This happens very seldom now that there are two people who do this on a regular basis.

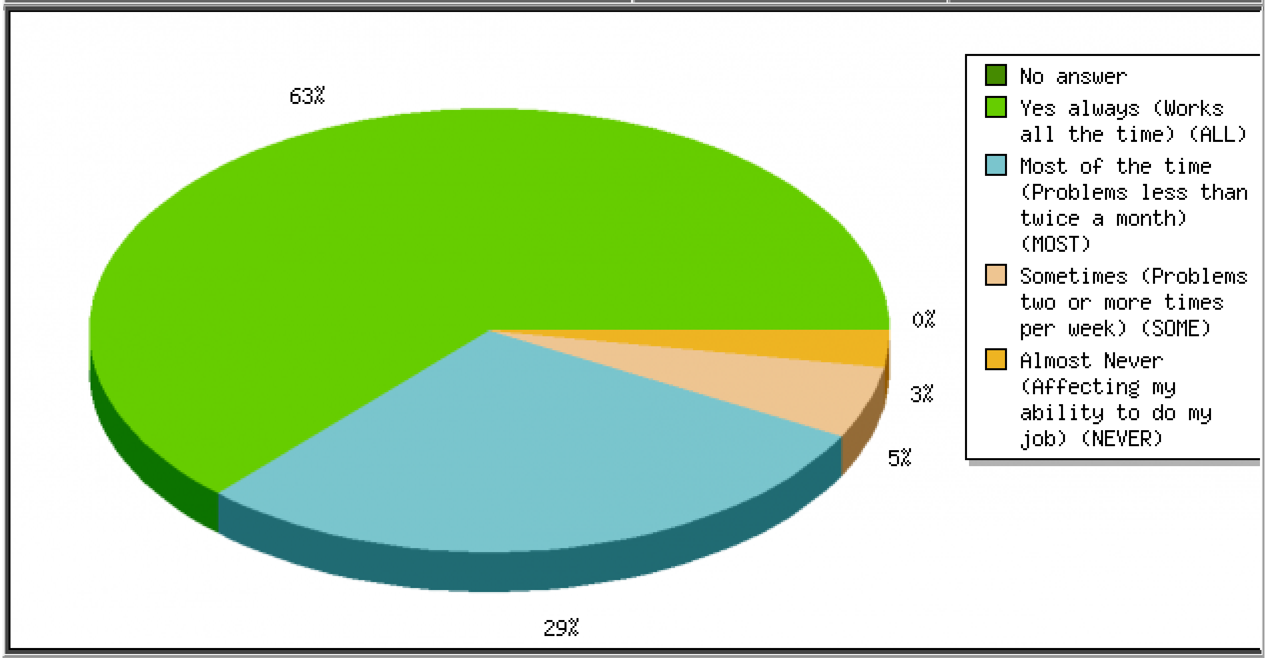
Field Summary for G6Q1:		
Do you receive your voicemails on time?		
Answer	Count	Percentage
No answer	0	0.00%
Yes always (Works all the time) (ALL)	28	73.68%
Most of the time (Problems less than twice a month) (MOST)	8	21.05%
Sometimes (Problems two or more times per week) (SOME)	2	5.26%
Almost Never (Affecting my ability to do my job) (NEVER)	0	0.00%



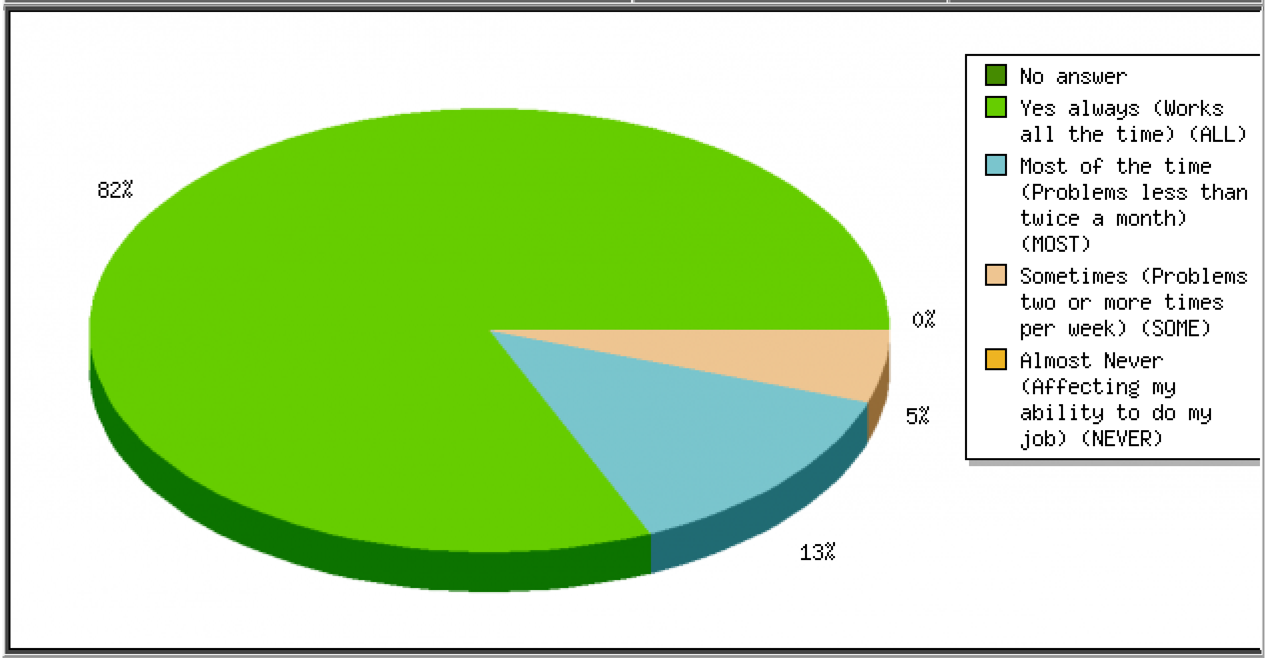
Field Summary for G6Q2:		
Are the message recordings audible and clear?		
Answer	Count	Percentage
No answer	0	0.00%
Yes always (Works all the time) (ALL)	18	47.37%
Most of the time (Problems less than twice a month) (MOST)	16	42.11%
Sometimes (Problems two or more times per week) (SOME)	4	10.53%
Almost Never (Affecting my ability to do my job) (NEVER)	0	0.00%



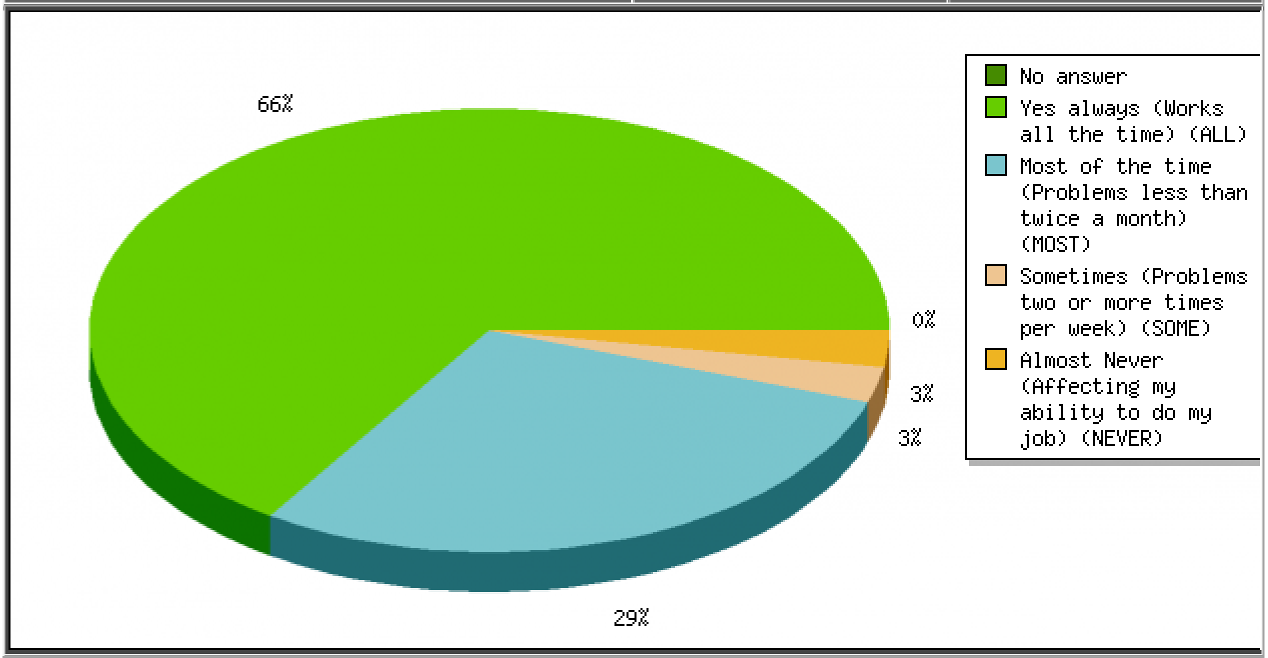
Field Summary for G6Q3:		
If you are receiving voice mail notices via email are they on time and easy to use?		
Answer	Count	Percentage
No answer	0	0.00%
Yes always (Works all the time) (ALL)	24	63.16%
Most of the time (Problems less than twice a month) (MOST)	11	28.95%
Sometimes (Problems two or more times per week) (SOME)	2	5.26%
Almost Never (Affecting my ability to do my job) (NEVER)	1	2.63%



Field Summary for G6Q4:		
Do you find the voice mail system easy to access and use?		
Answer	Count	Percentage
No answer	0	0.00%
Yes always (Works all the time) (ALL)	31	81.58%
Most of the time (Problems less than twice a month) (MOST)	5	13.16%
Sometimes (Problems two or more times per week) (SOME)	2	5.26%
Almost Never (Affecting my ability to do my job) (NEVER)	0	0.00%



Field Summary for G6Q5:		
Are the instructions provided for the system easy to follow?		
Answer	Count	Percentage
No answer	0	0.00%
Yes always (Works all the time) (ALL)	25	65.79%
Most of the time (Problems less than twice a month) (MOST)	11	28.95%
Sometimes (Problems two or more times per week) (SOME)	1	2.63%
Almost Never (Affecting my ability to do my job) (NEVER)	1	2.63%



Field Summary for G6Q6:		
Do you have any other voicemail related comments?		
Answer	Count	Percentage
Answer	14	36.84%
No answer	0	0.00%

The temporary message is a little bit confusing since there are two types of temporary messages - the one where you are on another line and then the one when you are on vacation. Also, it is a little difficult trying to delete the temporary message since it is about a 4 step process when you would think you could just hit one button to delete it. I also cannot seem to figure out how to listen to my new and old messages at the same time. I would have thought that once I went through my new messages that the voice would then ask if I want to go through the old messages but it doesn't, instead it just asks if I want to leave the program. I find it much easier (although time consuming) to call and get my new messages, hang up, and call again to then get through the old messages still on the machine.
I really like the ability to get voice mail from email . This helps greatly when I\'m in the Ashland office one day per week and can get my Morehead voicemail messages there .
I wish the voicemail messages that I receive were time and date stamped. It only gives me a prompt to tell me which message it is (first, second, etc.), but I never know when it was left.
No voicemail -- these questions are not applicable and answered only because I cannot exit the system without answering the questions.
When I press 7 to delete my voicemails after listening to them, they never actually get deleted, it stores them as saved messages and I have to choose \"advanced options\" and go in and delete them from there. Is there a way to make them delete automatically when you tell it to delete the first time (by hitting 7) instead of it saving them? This is kind of annoying, although it is a very small thing and not a big deal as far as the overall system goes.
I've not been taught to use the voicemail system ... thus the answers above.
Sometimes can't understand the messages, but this is probably due to the person speaking
Most of my telephone usage is listening to voice mail messages. I'm not sure what you mean by the "instructions provided."

Would like the date and time stamp on the voice mail.