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The *Asterisk* PBX

IP Telephony using Linux, Digium and *Asterisk*, the open source PBX

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Riseforth

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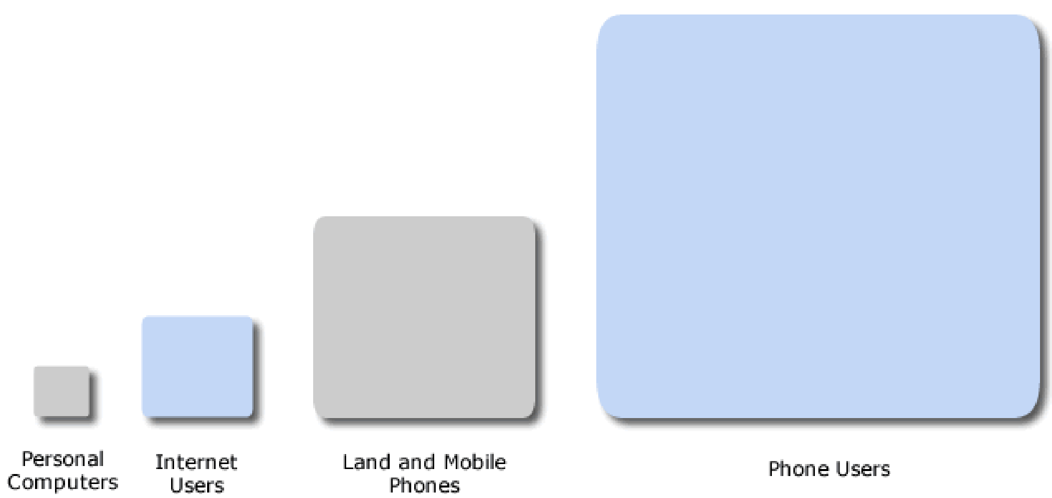
IP Telephony Innovation using Linux, Digium and *Asterisk*, the open source PBX

A Commanding Opportunity

Open source projects like *Asterisk* on the Linux platform present a new opportunity for computer-telephony integration (CTI) with a private branch exchange (PBX). In addition to PBX and interactive voice response (IVR) functionality, *Asterisk* is a softswitch, a protocol gateway, a media server, and a VoIP gateway. These and other services have long been dominated by proprietary hardware and telecommunications vendors. Potential for innovation exists now because of the broad availability of open source tools like *Asterisk* and open source databases running on powerful commodity Linux servers. They deliver an extremely low-cost platform for a developer to experiment with CTI with little risk. They can be used to improve employee productivity, improve customer service, provide new products and applications, and many other imaginative ends, taking advantage of the ubiquity of the telephone.



Businesses should question why the two-thirds of the population that does not use the browser, but does use the phone, continues to be underserved. Likewise, it is rare that a service organization delivers impeccable telephone service. Not only are automated systems frequently outdated, they are probably the single greatest point of customer dissatisfaction. To a business the cost of live telephone support is high, but the cost of inept telephone support is fatal – potentially causing the permanent loss of a customer, damage to customer loyalty at best.

Must this continue to be the case? Should we continue to force customers into confusing IVR menu systems? Improvements are possible. Take the language prompt for example. Most callers can be identified by their calling number. After selecting their language choice the first visit, the PBX database could retain this preference for application to future calls, just like a cookie saves a user's browser passwords and preferences. What a simple solution! But few companies use it, and there is a reason why.

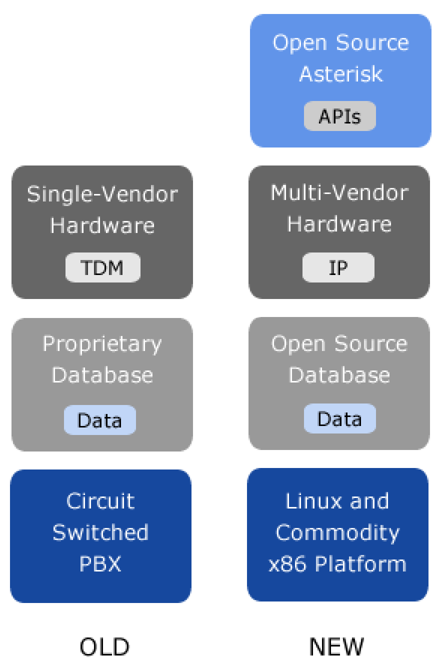
Giant Robots in a Sea of Databases

Even in the largest organization, where telephony budgets flow generously, there is little consistency within or across the customer facing telephony system. Customer callers jumping between service representatives often experience disconnects. The same customer issue must often be articulated over and over to each new subsequent representative because nothing is captured from prior calls. When a live representative is reached, too often vendors cannot answer their question or have no domain expertise. Customers are put on hold for extended periods without explanation or any idea about whether the call is over. Rarely is there a chance to navigate to a knowledgeable and familiar representative, because policies favor anonymous service people in unidentified places throughout the world. Instead, the customer with a question encounters the oxymoron called the "customer service 800 number".

The fact is, these perceptions represent a common failure of marketing and service executives to blueprint and implement excellent customer service experiences over the phone. Many of these problems are attributable to the fact that the traditional PBX is like a giant robot floating without an oar within sea of corporate databases, doing one assignment routinely, but unable to undertake trivial business duties within a reasonable timeframe or budget. Corporate executives, largely due to their own inattention, are now hostage to this very serious circumstance and the continuous threat to customer loyalty. Will they pick up a phone and call their own organization?

A Light on the Horizon

Perhaps there was once an excuse. The traditional telecom equipment manufacturers dominated the infrastructure with PBX systems that were designed primarily to be reliable, but were prohibitively difficult to program and extend. This is no longer an obstacle. Just as commodity hardware and software have borne new opportunities and innovation in data processing, there are ways for businesses to leverage commodity hardware and software to revolutionize voice systems. With the entrance of Linux-based commodity servers and Digium hardware running the open source *Asterisk* PBX, the curious CEO can now pose a new question to the builders and innovators: “Why are we failing to use this affordable technology?”



To expand their imaginations, the corporate service builders need to see both computer-telephony integration in the new light of the untapped possibilities that Linux, *Asterisk*, and standard hardware components represent. The fundamentals are in place today to leverage IP telephony at nominal cost. All the expensive heavy lifting in creating the infrastructure is complete. Access networks including packet routing, phones for millions of consumers, and the systems for connecting calls are available at little or sometimes no additional capital cost. The opportunity abounds for innovation based on commodity server hardware, Digium cards, Linux and open source *Asterisk*. The missing ingredient is creative thinking by the builders of businesses, products and services, and the proper motivation from executive management.

The SME and Large Enterprise Value Proposition

Linux, Digium, and open source *Asterisk* establish a new milestone in the evolution of computer-telephony integration, providing four specific benefits to the intelligent enterprise:

- Superior feature delivery and greater productivity
- Improved network utilization
- Lower telecommunications expense
- Enhanced operational responsiveness

Traditional telephony has underserved the small and medium enterprise (SME) market due to the high capital outlay and maintenance expense of on-premise PBX equipment and software. Until now the SME had to do without auto attendants, interactive voice response, intelligent call routing, call center screen popping, and other sophisticated PBX features. Large enterprises have been frozen in time with PBX systems that are difficult to program and costly to adapt to changing business requirements. Commodity hardware and open source software now radically change the cost – benefit equation.

A. Superior Services Delivery and Greater Productivity

Enterprises have the opportunity to greatly improve their service to customers thought the use of Linux-based PBX servers. But enterprise employees also benefit in many ways. By supporting voice and data over the same converged network, IP telephony enables a broad variety of features and services to improve employee productivity.

The IP PBX supports new desktop and system features that are unavailable with traditional circuit switched PBX systems. Some of the most common features and services are:

- Web Based Control over messaging including Voice Mail
- Call Center Administration including Presence, Forward and Follow
- Customer Relationship Management and Database Queries
- Conferencing Services

B. Improved Network Utilization

IP telephony offers better bandwidth utilization through silence elimination, redundancy reduction, and data throughput improvement. As a result, IP telephony uses only 10 to 20 percent of the bandwidth of traditional voice communications. The IP PBX is fundamentally more configurable and scalable than traditional circuit switched systems, because the office LAN infrastructure is normally in place and operating. An IP PBX can deliver WAN-based functionality as well, eliminating multiple PBX systems by supporting multiple branch locations through standard broadband Internet connections. *Asterisk* can be administered through a single interface from virtually anywhere. Underlying protocols are more accessible than traditional proprietary voice systems, and call control and traffic routing can be improved.

C. Lower Telecommunications Expense

Major IP telephony advantages are the ease of adding stations and the support of dispersed geographic locations. Traditionally it cost $100 or more just to move a single analog phone. With IP telephony, a business can merge voice, data and Internet services to a single broadband connection, reducing monthly access charges and lowering the cost of service to branch offices or home office workers. Because of voice compression and packet routing techniques, IP telephony trunk connections require less bandwidth, translating to reduced expenses. Free from state and country calling tariffs, Voice over IP also eliminates many or all long distance and international toll charges.

D. Enhanced Operational Responsiveness

Traditional voice systems use proprietary architectures, so adding third-party equipment or software can be costly or impossible. Not so with a Linux-based Digium and *Asterisk* solution. These commodity alternatives create vastly larger hardware choice, more software options and a larger field of professional support organizations, therefore delivering far more competitive pricing. The IP telephony PBX can support dozens of geographically dispersed locations, with applications provisioned across the entire organization at one time, and without the costly hardware redundancy.

Anatomy of an Asterisk PBX

The transition from circuit-switched PBX systems to server-based systems is possible because individual components are now modular, conformable and faster. Computer-telephony integration means that the same network of servers can process both data and voice – and the nature of *Asterisk's* modular design allows for extensive customization. For instance, dial plans for extensions can be configured to route traffic to either digital or analog endpoints, including phones.

Wildcard PCI hardware from Digium provides access to the telephone company and to analog endpoints such as traditional phones and networked devices such as the facsimile machine or printer. To protect customer investment and allow incremental migration, Digium provides PCI cards that interface with both analog and digital phones.

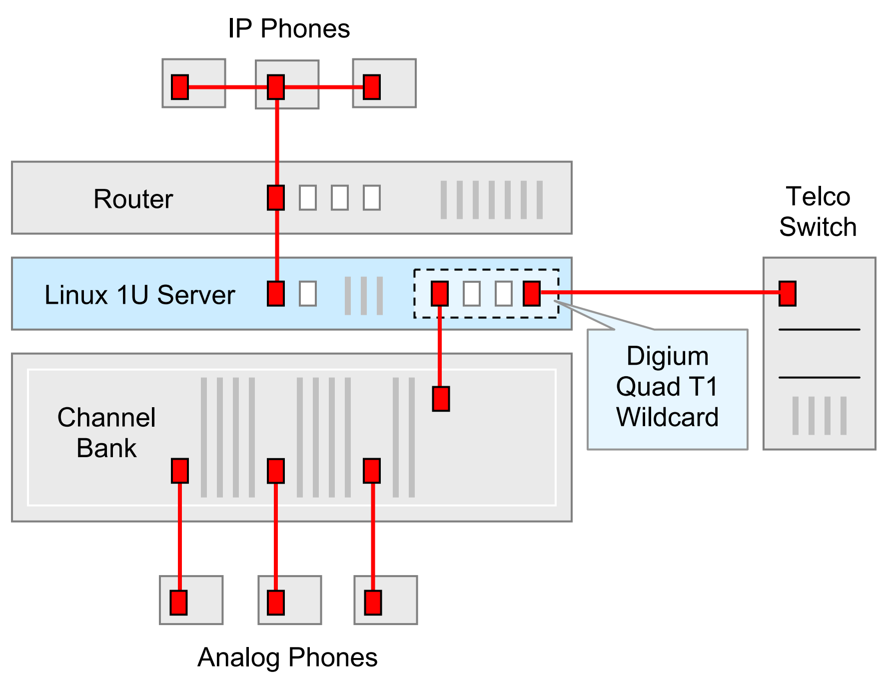
Asterisk also complements existing PBX systems by avoiding costly replacement of existing PBX investment while eliminating further vendor lock-in. In this fashion, *Asterisk* delivers hybrid PBX capability for purposes from simple additional lines to special features like branch-to-branch calling. Fully compatible with a wide range of IP and analog protocols and codecs, *Asterisk* translates between them on the fly.

An Asterisk-based Digium solution can support larger numbers of endpoints and services with little or no expense of adding additional PBX hardware. The key is having a Linux-capable administrator available to manage the computer telephony needs of the business. But a competent administrator working on the LAN or business applications has the same set of skills.

There are also companies that offer *Asterisk* system integration, administration consoles, and software packaging, support plans, and training.

The *Asterisk* application runs on the Linux server, which can also host a database. IP phones connect directly to the LAN router. Analog phones connect to a channel bank. The channel bank multiplexes a group of channels, both analog or digital, into a single higher bit-rate digital channel for transmission over a broadband trunk line. In the other direction, the channel bank demultiplexes received data into individual incoming channels. The 24 FXS channels found in channel banks aggregate to a T1 circuit.

A basic *Asterisk* PBX is illustrated below.   The PBX components shown can typically be purchased for about $5000, which is probably one-tenth the cost of other commercial alternatives. Actual performance of an *Asterisk*-based solution depends on traffic. CPU utilization will quickly increase when multiple calls are in progress, placing larger demands on the server.



Open Source Asterisk and Digium Hardware

Asterisk offers a large set of capabilities. In addition to traditional PBX features such as voice mail, call conferencing, automatic call distribution and interactive voice response, *Asterisk* delivers caller ID, call queuing, bridging and toll-bypass. For example, intelligent routing features can automatically redirect calls, making service representatives more accessible, responsive and productive – a win for both the company and the customer.

Asterisk provides video call support, message-waiting indicators, call parking and transfer, paging, and intercom, among others features. It includes over 500 recorded professional voice prompts. Incoming voice mails are recorded in Wave format for storage or forwarding to a user mailbox, or forwarding by email for playback on a personal computer.

Supported VoIP Protocols

Asterisk supports three VoIP protocols, two industry standards and one originally developed specifically for *Asterisk* and adopted by number of other hardware and software devices.

- *Inter-Asterisk Exchange (IAX):* IAX is the defacto standard VoIP protocol for *Asterisk* networking. IAX differentiates itself through transparent interoperation with NAT and PAT (IP masquerade) firewalls. This allows plug-and-play portability of PBXs and phones. IAX is extremely low-overhead (four bytes of header, as compared to at least 12 bytes of header for RTP based protocols like SIP and H.323).

- *Session Initiation Protocol (SIP):* SIP is the IETF standard for VoIP. Its call control syntax resembles SMTP, HTTP, FTP and other IETF protocols. SIP is widely regarded as the replacement standard for H.323 VoIP due to its relative simplicity and human-readability.

- *H.323:* H.323 is the ITU standard for VoIP.

Important advantages of *Asterisk*

The *Asterisk* software package delivers a number of unparalleled advantages:

Substantial cost reduction - Combined with low-cost Digium telephony hardware and a Linus PC server, *Asterisk* can be used to create a PBX at a fraction of the price of traditional PBX and key systems, while providing a level of functionality exceeding that of many of the most expensive systems available.

Control - *Asterisk* allows the user to take control of their phone system. Once a call is in a Linux server with *Asterisk*, *anything* can be done to it. In the same way that Apache gives the user fine-grained control over virtually every aspect of its operation (and configuration), the same applies to *Asterisk*.

Rapid deployment and development - **Asterisk** allows PBX's and interactive voice response (IVR) applications to be rapidly created and deployed. Its powerful command line interface (CLI) and text configuration files facilitate both rapid configuration and real-time diagnostics.

Customization - Through its support for internationalization, configuration files, and open source code, every aspect of *Asterisk* can be configured or modified. For example, codes for call features can be modified in order to accommodate proprietary protocols.

Dynamic content deployment - As web servers like Apache allow a developer to deploy dynamic content on the web, such as account information, *Asterisk* permits such dynamic content deployment over the telephone, with programming ease similar to the Common Gateway Interface (CGI), a vitally important web technology.

Flexible dialplan - Asterisk's unusually flexible dialplan allows seamless integration of IVR and PBX functionality. Many of *Asterisk's* existing features can be implemented simply with extension logic.

Myths Versus Reality

The transition to IP-based systems like *Asterisk* is unstoppable, but some myths persist. In reality, enterprise adoption is pushing aside the myths, technology is outpacing concerns, and first movers are demonstrating solid benefits.

Myth	Reality	Internal Use Strategy
Existing infrastructure cannot support the QoS standards necessary for real-time voice communications.	Ethernet switches support far more efficient voice traffic than traditional dedicated telephony, reducing operational cost.	Server and LAN infrastructure investment at most companies has largely outpaced traditional circuit switched voice communications systems, making this much newer than existing PBX systems. Opportunity exists for IP telephony to leverage these platforms.
The fundamental concern for IP telephony is voice quality.	Conversation degrades only when one-way delay exceeds 150 ms.	About half of normal voice interaction is silence. This means that 50% of the capacity of the traditional TDM network remains unused due to silence alone.
The traditional PBX has high reliability through stable components and built in manufacturer redundancy.	The traditional PBX deliver zero vendor redundancy and is inflexible and prohibitive to extend.	Eliminating the separate telecommunications network, and instead managing PBX servers in the same LAN cabinets and on the same operating systems as the data network, creates significant advantages in reliability, maintenance, and ease of administration.

Evaluating Customer Experience

In the personal computer world, some software developers have made careers out of graphical user interface (GUI) design. Companies like Apple and Intuit have developed brands around GUI design. But to the credit of Apple and Intuit, their user interface quality did not occur by accident. Rather, it is the result of a business philosophy to invest in and direct engineering attention towards the goals of simplicity and ease of use, combined with excellent aesthetics. A similar focus is needed for computer-telephony integration. To sharpen the perspective, a number of questions deserve management attention:

1. Survey the state of current voice services.

Companies should formally rate their own voice channel interactions. This action serves two purposes. First it helps identify areas of priority for potential improvement. Second, it provides a baseline for measuring future effectiveness. Customer satisfaction, time to resolution, input from survey groups, and other metrics can establish valuable parameters. Criteria for a passing score can be drawn, tested, and evaluated. Usage trends and satisfaction surveys can later validate progress.

2. Establish a matrix of problems and benefits.

The phone offers a primary customer interaction channel for most businesses and government organizations. But ask marketers or service providers to articulate their phone interaction objectives and the answers will be subjective generalizations like “deliver a better experience”. These metric-free goals offer no insight or direction in deploying or improving voice-enabled services and systems. Instead, ideas and experimentation should be based on an identified set of problems and benefits. Incremental solutions can be designed, tested and evaluated, with actual trial results tipping the decision to implement or not.

3. Create user personas to guide service development.

The world is not homogenous, and differences are far greater then simply language differences. Cultural expectations, levels of product or service familiarity, and other demographic influences can shape the needs and perceptions of customers. Without insight into these demographic differences, developers may incorrectly assume that all preferences are similar and identical to their own. Unfortunately subjective and anecdotal experiences are an unsound foundation on which to build for a target audience. Many companies are now using fictional personas to qualify the ways to address their unique customer requirements.

Conclusion

Linux on commodity servers, Digium cards and *Asterisk* provide a compelling opportunity for businesses to improve their telephony services, both from an operations perspective and from a service delivery perspective. Managers can take a number of steps to leverage the opportunity presented by Digium, *Asterisk*, and other open source and hardware solutions, when they:

- Identify productivity improvements possible using commodity telephony systems.
- Assess the business opportunity to raise customer satisfaction in voice services.
- Develop a vision and business objectives for applying telephony advances.
- Adopt tools like *Asterisk* and a development philosophy that encourages innovation.
- Implement standard hardware on Linux servers for unparalleled return on investment.

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