# White Paper

**Integrating Vocality with Asterisk** 

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#### INTRODUCTION

This white paper is intended to describe the different challenges, compatibility issues and physical and coverage limitations, that result from using legacy communications systems. It will show how Asterisk PBX or other IP-based PBXs solve these problems. As technology matures and budgets shrink, there is greater demand for more capabilities and ever lower capital outlay. The ability to establish interconnectivity between multiple legacy communication systems is always important, to offer more users the ability to communicate with one another.

Where achieving connectivity is crucial for daily operations, the ability to deliver reliable and high-quality communications is of the utmost importance. Providing this will result in significant operational improvements, with huge impact for individual businesses and sometimes even lives saved.

This white paper considers flexible open-source systems for implementing telephone private branch exchanges (PBX), like Asterisk, which is based on Session Initiation Protocol (SIP). These form the core of a solution that allows attached telephones to make calls to one another and to connect to other telephone services, such as the Public Switched Telephone Network (PSTN), Voice over Internet Protocol (VoIP) services and radio networks. Using Asterisk PBX to connect these technologies enables communication between more users, without the expense of a full technology refresh.



#### PROBLEM STATEMENT

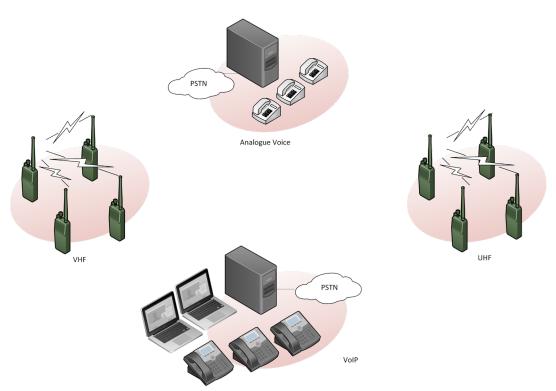
Through time, different types of technologies and infrastructure for communications have been designed with a specific purpose, as a solution to specific needs. Within the scope of voice-communication technologies, we find radio technologies (VHF, UHF, analog, digital, P25, and Tetra), conventional analog telephone systems (FXS, FXO) and Internet Protocol-based voice communication known as Voice over IP (VoIP).

These unique systems consist of different technologies and all work very well on their own. However, these disparate systems, combined with the lack of need to directly connect one another, has caused a 'natural' isolation of users. This situation has changed in recent decades, where timeliness in the dissemination of information has become critical for emergency responders, police and other businesses.

Communications infrastructures have experienced significant and dramatic improvements in technology and capabilities. Today they offer better quality, with higher speed and capacity, than their predecessors. These improvements make them more cost-effective, available and accessible to a larger population, in more places than before, thus allowing them to cope with current market demands. However, they are still perceived as a valuable, costly and limited resource, that has to be managed carefully.

In order to facilitate the high-quality, efficient and costeffective connectivity that the market demands, it is required that these isolated infrastructures interconnect directly, although they were not originally designed to do so. This includes retaining legacy equipment, that has been in operation a long time, and allowing it to interoperate with modern telephone systems. An agency might, for instance, have millions of dollars invested in radios that would otherwise have to be scrapped and replaced, in order to allow them to operate with other communications systems.

Figure 1 Isolated communication technologies and infrastructures





#### PROPOSED SOLUTION

#### Introduction of Solution

Bridging the gap between different and isolated technologies is essential. The augmentation of the communications infrastructure in this way adds new capabilities to existing equipment and reduces capital expenditure.

Developing this new infrastructure is feasible with standardsbased interoperability. In order to fulfill market demands, our solution observes these design principles:

- Use of versatile and portable equipment that is able to provide interoperability between existing isolated platforms.
- Scalable and flexible deployment allowing gradual expansion based on application needs.
- Based on open-source operating systems. This provides reliable operation, platform independence and adoption of new capabilities inherent in the operating system that results in the rapid development of new features.
- Cost-effective, lending itself to reuse of equipment.

#### Application of Solution

The proposed solution for this augmented communications infrastructure relies on the integration of Asterisk PBX with Vocality products; specifically BASICS Radio Relay, that is designed to provide radio interconnectivity via Radio over IP (RoIP) protocol and BASICS Voice that is designed to provide conventional analog voice circuits (FXS / FXO) via Voice over IP (VoIP) protocol.

The integration benefits from a unique characteristic that is inherent and common to both technologies: the use of the SIP protocol. This characteristic constitutes the key for facilitating the direct interconnection between the existing previously isolated technologies.

In this context, Vocality handles each isolated technology independently and for each, facilitates their signaling and audio transmission from analog to IP; whereas, Asterisk PBX handles the interconnection between Voice over IP (VoIP) based systems and phones.

Figure 2 shows the proposed solution for deployment of new communications infrastructure that allows interconnection of isolated communications technologies and devices.

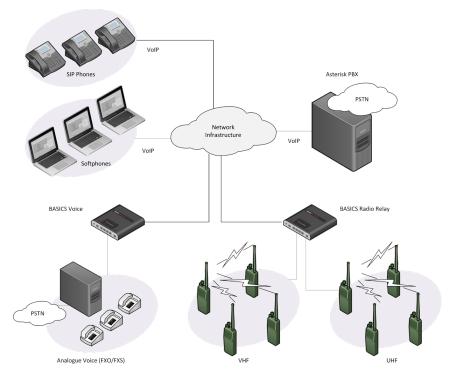


Figure 2 Integration of different communication technologies and infrastructures



Additional features and capabilities in the augmented architecture may be summarized as follows:

- Inherent to Asterisk PBX and now available to radio infrastructures (VHF, UHF, analog, digital, P25, and Tetra):
  - SIP phones and softphones direct interconnection with radios.
  - ▼ Direct connectivity with PSTN.
  - Ability for recording calls that involve radios, for quality control and fraud investigation purposes.
  - Ability for including radios in half-duplex tripartite conference calls.
  - Radios direct connection to Public Address (PA) systems.
- ▼ Inherent to Vocality BASICS Radio Relay and BASICS Voice:
  - Radio to radio connectivity, known as cross-banding. For instance VHF and UHF radios are now able to communicate with one another, in either the same or remote locations.
  - ▼ Radios able to call to either analog or IP-based phones.
  - Analog voice devices (FXS, FXO) direct connectivity with radios (VHF, UHF, etc).
  - Analog voice devices (FXS, FXO) direct connectivity to SIP phones and softphones, without the need of external Analog Telephone Adaptor devices and direct connection to Asterisk PBX.
  - New RTP filtering feature which allows G711 received packets containing 'silence to be discarded. This will lower bandwidth usage and make for easier RoIP interaction with radio systems, in some cases removing the need to use DTMF-PTT to activate/deactivate.

#### **DEPLOYMENT SCENARIOS**

The following list gives a brief overview of several scenarios and applications that involve integration of Asterisk PBX and Vocality products.

### **Customer Service Operation with Monitoring and Fraud Prevention**

This solution is designed for organizations whose core business relies on high-quality 'Dispatched Customer Service' operation, counting on success with the assurance that day-to-day transactions are conducted free of fraud. This includes organizations like taxi companies, insurance companies, home delivery (pizza or groceries), fuel delivery, mail and package delivery.

The integration of radio infrastructure to an Asterisk PBX system is achieved by using Vocality BASICS Radio Relay and the SIP protocol. It provides the following advantages:

- ▼ Dispatcher is able to use the same VoIP telephone for both users' call handling and radio dispatch operation. This reduces the number of radios required for operation.
- The audio conversation between the dispatcher and radio operators can now be recorded, for quality control purposes, in the very same way it is recorded for regular telephone users.
- Real-time monitoring is now also available for radio operators.
- Both phone and radio recordings are time-stamped and stored by Asterisk PBX, so they can be used as supporting documentation or evidence when conducting investigations on suspected fraud transactions.

#### Radio-to-Phone Interconnection

Organizations that require the ability to communicate from any phone extension to specialized groups, that regularly operate via radio infrastructure, are the 'typical' users for whom this solution has been designed.

In this case you interconnect Asterisk PBX with as many Vocality BASICS Radio Relay devices as needed, via the SIP protocol. In general, one Vocality BASICS Radio Relay port is associated with one radio group.

The advantage this integration provides is that the caller will get an immediate response from radio-based personnel. Some examples of organizations that operate with this solution are:

- Large retailers operations relay real-time communications between central and branch offices with their warehouses and stores, in order perform operations such as merchandise restock or relocation.
- Hotel chains relay their daily operations while having realtime communications between administration, security, maintenance, and housekeeping groups, in order to fulfill guests' needs. The added value depends on getting an immediate response from a specific group. Some larger hotel chains also extend this communication to other groups such as kitchen and restaurant.
- University campuses improve their operations by having the ability for faster and more reliable communication between the campus administration, security, emergency, and maintenance groups, in order to facilitate better service to staff and students throughout the campus.



#### **Public Address System (PA)**

This solution allows specialized groups in an organization, whose communication is primarily over radio infrastructure, to have direct access to Public Address (PA) systems within that organization.

This capability is achieved by interconnecting Asterisk PBX to Vocality BASICS Radio Relay, via the SIP protocol. PA function is then achieved either via 'PA over IP' that would deliver announcements through the telephones and/or through traditional PA systems, which have been integrated to the Asterisk PBX as well.

In addition to radio-direct-announcement functions, Asterisk PBX offers PA group management capabilities that allow announcement delivery in a selective way. For example, to specific users or groups of users only.

Oil and gas organizations have implemented this solution in their operations, benefiting from the ability to broadcast announcements directly to the field without intermediaries.



#### CONCLUSION

The integration of Vocality products with Asterisk PBX, or other IP PBXs, results in an augmented network that provides interoperability between disparate communication platforms and technologies. This network architecture is customizable and scalable according to the requirements and budgets of different businesses, in an efficient, easy to deploy and cost-effective way. Customers no longer need to refresh all of their equipment when they need additional capabilities to expand their closed communications infrastructure. Most importantly, voice communications are provided to more users, which provides greater situational awareness and streamlines activities leading to saving of money and sometimes lives.



#### ADDITIONAL INFORMATION

Please contact your Vocality representative for more information on how we can help solve your communication challenges.

About White Papers: White Papers are discussion starters or supplementary information written by Vocality technical experts. Should you have queries which are not answered by our current documentation, your local Vocality support team would be happy to hear from you. E-mail **support@vocality.com**.

