In the early eighties, a communications protocol was created that allowed the research community to send data anywhere in the world using packet switching networks. This protocol became known as TCP/IP and ultimately resulted in the creation of the revolutionary Internet. TCP/IP actually refers to a suite of different protocols but, at its core, there exists the fundamental protocol commonly referred to as IP.

Up until a few years ago, the majority of applications for “IP” were targeted at non-voice type traffic. This included e-mail and the transfer of files. However, more recently, the ability to use IP for the transmission of digitized voice has become so dominant that many industry observers are now calling this the “killer” application. Voice over IP, or VoIP as it is known, has revolutionized the telephony world and, now, many in the radio communications industry are realizing that this technology can power traditional two-way radio networks and breathe new life into existing systems.

The adaptation of VoIP for radio communications has been referred to as “Radio over IP”, although there is no actual standard, as such. Voice over IP or Radio over IP, whichever way you look at it, is driving the convergence of IT and communications. It can deliver several key benefits to fleet mobile operators, including public safety organizations. These include: lower costs, improved reliability and increased interoperability. All of these are derived from the ability of the technology to utilize existing IP infrastructure - the IP backbones that make up both local and wide area networks.

To help organizations reap the benefits of VoIP, Radio over IP gateways have been developed to enable two-way analog or digital transceivers to be interfaced with IP networks. The purpose of this paper is to explain how Radio over IP works and to describe the typical features and benefits of these adapters.

Figure 1: RoIP Gateways can connect a variety of Radio Types across Wide Geographic Areas
**RADIO VOIP VERSUS TELEPHONY VOIP**

Radio and telephony use the same mechanisms to transport voice through packet switched data networks. Voice is converted into digital data using a device called a CODEC; assembled into UDP packets; and transmitted at intervals of around 25ms. At the receiving end, the CODEC also performs the function of converting the digitized samples back into an analog signal.

Various digitization standards are available. The most basic standard (G.711) delivers the highest quality audio at 64Kbps. However, other standards, such as G.723.1 and G.729A, can deliver good quality audio at rates as low as 5.3Kbps, thereby minimizing the traffic impact on an existing network.

However, radio and telephony differ when it comes to signaling and control. Telephony is concerned with call setup, call monitoring and call control. For example, VoIP needs to be able to initiate a connection, pass dial tones, DTMF strings and call progress tones. It also needs to be able to perform echo cancellation. Radio, on the other hand, does not require these telephony features. But traditional radio functions still need to be controlled and monitored. This includes the PTT function which must be transported transparently through IP and in sync with the audio. Other functions include the COS from the radio which can be used to generate a busy signal or key-up another transceiver at the remote end. Channel change is also a function that may need to be transported through IP.

When it comes to radio signaling, some even more interesting problems need to be addressed. Firstly, selective calling tones, such as CCIR SELCAL and ANI, cannot be reliably transported using VoIP when digitization rates below 32kbps are used. This means they cannot be reasonably compressed. And since one of the goals of VoIP is to reduce IP bandwidth through high compression, VoIP and radio signaling are not compatible. Hence, Radio over IP must transport signaling tones as data messages and not as audio. Similarly, if CTCSS tones are to be transported over an IP network then they should also be converted into data messages. In both cases, the equipment that provides the interface between the radio and the IP network must be capable of decoding the CTCSS, SELCAL or ANI audio and converting these into data messages. When receiving a data message containing a signaling string, the same equipment must also be capable of encoding this into the appropriate audio tones.

The problem of passing signaling schemes can be further compounded by the inherent packet losses within IP networks. When passing voice, a low rate of packet loss is not really an issue. However, losses in packets containing tone signals can be detrimental, especially when using short tone bursts or when passing modulated tones. All of this is due to the fact that the transmission of digitized voice, whether telephony VoIP or Radio over IP, is achieved using a “send and forget” mechanism called UDP.

Ultimately, the most reliable way to transmit signaling schemes and tones through IP is to use a messaging protocol that is transparent to the end user equipment.
BENEFITS OF RADIO OVER IP

Two-way radio has always been used to provide mission critical communications across a broad spectrum of industries, ranging from emergency services and security, to utilities, resources and transportation. So far, radio is proven to be the most effective, reliable and cost efficient method of communications for day-to-day operations.

Now, Radio over IP offers a cost effective way to interconnect radio systems and operators together. It is a technology that enhances and adds value to radio communications networks. There are three distinct benefits of using Radio over IP:

1. lower costs
2. improved reliability
3. increased interoperability

The biggest benefit that IP brings to the radio community is **cost savings.** Firstly, new implementations can leverage from existing IP infrastructure. Many corporations and public safety organizations already maintain their own private IP LAN or WAN. Therefore, no extra cabling or communications paths are required for the installation of a new radio or console. Most areas also support public IP networks that can be used, via VPN’s, to augment private LANs. Once a system is in place, further upgrades or expansion is easy and inexpensive, since no significant wiring is required.

An indirect benefit of using IP networks is the availability and decreasing cost of equipment. Commercial off-the-shelf hardware can be used, in terms of the routers and switches that make up the networks. These can be sourced from a broad range of third party manufacturers, in an industry where costs are continually decreasing.

The biggest cost saving, however, comes from the technology’s ability to replace leased lines and expensive wireless links. Savings from the elimination of analog leased lines, alone, should result in capital payback within 6 months – at the most!

Radio over IP also provides **improved reliability.** The interconnections between radios and consoles become more reliable, since they form part of a mesh IP network. This provides an inherently resilient infrastructure that is not subject to a single point of failure.

Finally, there is the added benefit of **increased interoperability.** Once it’s in the IP domain, radio audio can be routed to virtually any type of radio system. This allows UHF, VHF and HF radios to be easily interconnected. However, the benefits of interoperability aren’t just related to the radio systems, they also apply to corporate communications, and this is made possible through the use of SIP technology.
INTRODUCTION OF SIP

SIP is an IETF driven protocol that has revolutionized telecommunications. It is a signaling protocol that controls multimedia communications sessions over IP. Its sole purpose is to negotiate, setup and tear down the sessions whilst the actual multimedia communications is carried-out by other protocols such as VoIP.

Historically, two-way radio was considered an adjunct to the corporate communications system and was treated as an independent system, isolated from the broader corporate communications infrastructure. To provide interoperability between users of the radio network and other communications systems (including other radio systems) required the installation of special bridging and patching equipment.

Whilst this works to a certain extent, it doesn’t make it easy for an organization to streamline the flow of communications through-out its workforce. To enable an organization to achieve maximum levels of collaboration and productivity, it needs to unify its communications mediums, regardless of whether they are digital or analog radio, PBX, mobile phones or PC’s.

However, just having “IP connected” radios doesn’t mean that an organization can achieve unified communications. That’s why SIP needs to be an integral part of VoIP gateways and adapters.

When used with land mobile radio, SIP compatible devices provide two significant additional benefits. Firstly, Radio over IP gateways/adapters that are SIP compliant enable the connected radio system to interact with a broader range of corporate communications mediums, without worrying about proprietary technology. In such an organization, subscribers on handheld portables or vehicle mounted radios can communicate with people who are not normally connected to the radio network. As an example, calls can be made between radio users and office staff with SIP compliant telephones. More recently, there has been a proliferation of soft-phones that are also SIP compliant. Omnitronics is starting to see applications where this level of unified communications eliminates the need for office staff to be desk bound and dramatically increases productivity.
The second key benefit happens at even the simplest level of SIP. Radio over IP devices that are SIP enabled provide greater flexibility within radio networks because SIP makes it possible to structure these networks so that they can be re-configured dynamically. This applies to both user calls and inter-site links. For example, radio calls can be made to specific destinations, on demand. This brings efficiency to a network from both the IP and radio traffic viewpoints. SIP also allows links between radio sites to be managed such that link paths can be re-configured to meet specific operating needs.
CONNECTION TYPES

There are three modes for connecting devices together at the IP level:

- Unicast
- Multicast
- Conferencing

ONE-TO-ONE CONNECTIONS

Unicast mode is commonly used for one-to-one or point-to-point communications. In this mode, each device is programmed with the destination IP address of the other device and packets are transmitted to the single IP address only.

ONE-TO-MANY CONNECTIONS

Although unicast mode can also be used for one-to-many or point-to-multipoint communications, a more efficient method involves the use of Multicast communications. With this mode, a device can transmit a single VoIP packet that will reach several destination devices at the same time.

Multicast works through a process of group affiliation, where each device transmits and listens to a group IP address.

Although Multicast is an efficient form of communications for point-to-multipoint applications, there are many implementation issues relating to this form of communications. Most often these issues occur when implementing Wide Area Networks and they can be difficult to resolve.

Therefore, a third method involves the ability to place groups of devices into conferences. When a device is part of a conference, it will transmit copies of the same VoIP packet to all of the other devices in the conference group as a burst. Although several copies of the same packet are transmitted at the same time, the nature of each individual transmission (Unicast) means that point-to-multipoint communications can take place over even complex networks.

There are various modes of conferencing that make the implementation of multi-point communications very practical. This includes mixing audio sources together before transmission and the use of designated “servers” within a conference group.

Whilst it’s not ideal for networks with large numbers of end-points, conferencing is well suited to groups of up to twelve end-points.
COMMON APPLICATIONS

REMOTE RADIO ACCESS
An operator can control and monitor a remote transceiver across a LAN or a WAN. The Transmit and Receive audio, along with the PTT and Busy/COS signals, are transported over the link transparently. SELCAL, ANI and DTMF are also transported reliably, regardless of the level of compression that is employed.

Multiple IP handsets and consoles can be multi-dropped to provide shared access to the transceiver by a number of operators.

More sophisticated applications can be supported using SIP compatible consoles. These consoles enable operators to select multiple remote channels for communications.

A SIP enabled RoIP Gateway provides connection to a private SIP PBX or to a public VoIP service provider. In the example below, the radio connected to the gateway becomes an extension on the SIP server’s database. It is given a phone number that enables calls to be routed between it and the PBX or the PSTN. Operators using the radio network are able to make PSTN calls by keying pre-defined DTMF or SELCALL strings.
This example demonstrates how organisations can extend radio network access to an office environment through a SIP PBX or even to users at home using third-party VoIP providers. Compliance with the industry standard protocol of SIP is what makes this radio-telephony application possible.

**POINT-TO-POINT RADIO INTERCONNECTION**

In this scenario, a pair of IP Gateways can be used to replace a leased line or a UHF/VHF link.

Two radios (for example, two repeaters) can be connected back-to-back over an IP link. This can typically be used to interconnect two sites over a Wide Area Network. The PTT and COS signals are transported over the link as data messages. The gateway will provide a configurable PTT output to the radio. It will also accept a configurable COS input from the radio. An active COS signal from the radio will enable the transmission of voice packets over the IP network and generate a PTT output at the opposite end.

![Figure 5: Point-to-Point Radio Interconnection](image)

Where the communications equipment is not able to provide a COS output, the VOX function of the IPR100 can be used. When a voice signal is detected at the radio port of the gateway, an internal COS signal is generated and transmitted to the destination gateway. This will also enable the transmission of voice packets over the IP network. A configurable hang period is automatically applied to the VOX function.
RADIO BRIDGING ACROSS IP

The third application scenario makes best use of the multicasting technique. The RoIP Gateway allows a number of transceivers to be interconnected over a LAN or WAN. Each gateway is linked to a common multicast group address. When one transceiver receives audio, voice packets are transmitted to the multicast address. Any other gateway that is linked to that address will accept those VoIP packets and re-transmit the audio to its respective radio.

![Diagram: Radio Bridging Across IP](image)

**Figure 6: Radio Bridging Across IP**

DIGITAL DONOR RADIO

There are two standard methods of connecting Digital Radios to a wider IP network. The first established is through a donor (or control) radio. The donor radio is connected to the server or soft console through a digital radio gateway. It provides an access point into the radio network and dispatch operators can then access the network from their workstations.

![Diagram: Access a MotoTRBO Network using a Donor Radio](image)

**Figure 7: Access a MotoTRBO Network using a Donor Radio**
DIGITAL WIRELINE

Another way to access the digital radio network is through a direct IP connection to the base radio or to some radio system controller. In this situation, the digital radio gateway provides the IP connection to both the radio network and, if applicable, the server. This configuration makes a broader range of features available to the dispatch operators.

Newer radio technologies such as DMR provide IP wire-line access for dispatch consoles.

CONCLUSION

Radio over IP (RoIP) provides organizations with a very flexible and reliable network infrastructure. By using the right tools, they can create cost effective networks that can grow and change as their organization grows and changes.