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White Paper

Understanding RoIP Networks

Revision 3.0

Interoperability Gateways

Radio Dispatch Systems

Location Services

Radio over IP

Introduction to Radio over IP (RoIP)

Radio over IP describes the application of Voice over Internet Protocol technology to the two-way radio. It is a generic term and does not describe any specific implementation or standard. This paper will focus on how RoIP is implemented using standard Internet Protocols for voice and what issues may be encountered. Some of these issues also apply to proprietary RoIP systems.

RoIP 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.

VoIP was designed to provide a telephone replacement. It allows a large service provider to connect many users for voice communications on demand. This is different to the typical ways that Radio over Internet Protocol is used.

The RoIP Difference

VoIP and RoIP use the same mechanism to transmit voice or audio data. However, radio and telephony differ when it comes to signalling and control.

Telephony	Radio
✓ Focus is Call Setup, Monitoring and Control	✓ Focus is on Instant Communication, monitoring and control
✓ Generally one-to-one	✓ Generally one-to-many
✓ Always full duplex	✓ Generally half duplex, although full duplex is becoming more common
✓ Typically a large Telco is the service provider	✓ Typically the organization itself manages their service
✓ Central Telco Server	✓ Peer-to-peer service
✓ Minimal customisation	✓ Specific to customer
✓ Capabilities required include passing dial tones, DTMF strings and call process tones	✓ Significant extra capabilities as listed below.

Additional Radio Capabilities:

General radio signalling information:

- Push to Talk (PTT): Indicates the radio should transmit and must be synchronized with the audio.
- Carrier Operated Switch (COS) from the radio: Used to generate a busy signal or key-up another transceiver at the remote end.

For Analog Radio:

- Tone Information: DTMF, Paging Tones, SELCALL, EIA Tones
- Continuous Tone Coded Squelch System (CTCSS)
- In-band Guard tones
- Radio channel frequency change

For Digital Radios

- Type of radio call in progress (Broadcast, Group or Individual calls)
- Radio Identification
- Advanced Radio Features: Stun, Revive, Remote Monitoring, Emergency
- Radio channel profile change (includes frequency and other operating parameters).

Applications

Radio over IP can be used in several ways to leverage radio systems. Some applications include:

- Access to remote radios
- Point-to-point radio interconnection, also point to multipoint and cross-banding
- Radio to Dispatch Console links
- Radio to Phone gateway
- Using existing infrastructure to reduce cabling.
- Remote Access

For further details on typical applications, see our other White Paper: VoIP for Radio Networks.

VoIP Standards

The two key VoIP standards defined by the Internet Engineering Task Force (IETF) are

1. Session Initiation Protocol (SIP) – described in RFC 3261
2. Real Time Protocol (RTP) – described in RFC 3550 / RFC 3551.

There are many other standards that are relevant to voice and multimedia communications. However, SIP and RTP are the two key protocols that do most of the heavy lifting.

Using existing VoIP standards for implementing RoIP provides several key advantages. These are:

- Improved Vendor-independent Interoperability
- Easy Integration with Existing Phone and Voice Systems
- Compatibility with off-the-shelf Voice Recorders
- Recognized by Routers, Firewalls and Network Tools.

The disadvantages to using standards are that they are more complicated and also do not support all of the radio information that is required.

Session Initiated Protocol (SIP)

SIP is a standard protocol used to set up VoIP calls. The purpose of SIP is to make communication possible. It:

- Determines the IP address of the Remote Device
- Determines the UDP Port Numbers to use for RTP
- Negotiates what Features can be used (e.g. the Audio Codecs)

It is only used to make and break connections. No audio is transmitted via SIP.

Types of SIP Devices

- *SIP User Agents*: End-user devices that create and manage a SIP session. A radio to VoIP gateway such as the Omnitronics IPR110Plus or a SIP VoIP phones are examples SIP.
- *SIP Servers*: There are various types of SIP servers that provide different functions. The two most important server types are:
 1. *Registrar Server*: Stores the devices registration information in a database. This can be local or provided by yet another server. This database is searched to find devices to which a connection may be made. It also authenticates the SIP device to ensure it supplies the correct credentials, namely user name and password.
 2. *Proxy Server*: Usually used to re-route requests or messages.

A SIP server may be both a registrar and a proxy server at the same time.

SIP is a text based protocol similar to the Hyper Text Transfer Protocol (HTTP) used for transferring data over the Internet/world wide web (www). The protocol consists of requests and responses.

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
```

Real Time Protocol (RTP)

RTP is a standard protocol for delivering audio and video over IP networks. RTP is used in communication and entertainment systems such as telephony, video teleconferences and television. It:

- Is designed for end-to-end, real-time transfer of stream data. The protocol includes features to compensate for problems that result from sending streams of data over unreliable packet switching networks.
- Prioritizes “Real-time” over Reliable: It is more important for the data to arrive on time than it is to guarantee that 100% of the data is delivered.
- Provides data transport *not* any mechanism to setup the link.
- Can send voice and radio information but when used in this manner it requires more system configuration and is less flexible.
- Can be used for point-to-point links (unicast) or point-to-multipoint (multicast).
- Includes facilities for detecting issues that may occur when sending data over packet switching networks such as:
 - Detection of last data packets
 - Coping with packets arriving in the wrong order
 - Coping with variable delays in packet delivery (Packet Jitter)

Common VoIP Issues

Delay

There are two types of delays that occur within packet switched networks:

- Constant Delays or Latency
- Variable Delays resulting from the Nature of Packet Switching. Usually described as a “Jitter Delay”

To compensate for Jitter Delay, the end device needs to keep a buffer of audio data so that it can continue to play audio even if the next packet is a little late to the party. This in itself can be another source of delay.

When looking at the total delay the following sources need to be considered:

1. Digital to Analogue Conversions
2. Framing Delay (Audio Data is collected into a fix duration packet, typically 20 milliseconds or some multiple of 20 milliseconds)
3. Software Processing which time taken to Encode or Decode the Audio
4. Fixed Network Delay
5. Variable Network Delay (Jitter)
6. Jitter buffering delay

Portable mobile radio systems are usually half-duplex with a Push to Talk (PTT). This means that radio system can tolerate delay *better* than telephone systems. However, the RoIP system must ensure that the Push to Talk signalling is synchronized with audio so that no leading or trailing syllables get cut off.

In Telephone systems, most callers notice round-trip delays when they exceed 250 milliseconds, so the one-way latency budget would typically be 150 milliseconds. 150 milliseconds is specified in ITU-T G.114 recommendation as the maximum desired one-way.

However, in some applications, such as radio set-ups with multiple chained repeaters, the audio delay can actually be desirable. By delaying the audio relative to the Push to Talk signal, the first spoken syllable is not lost.

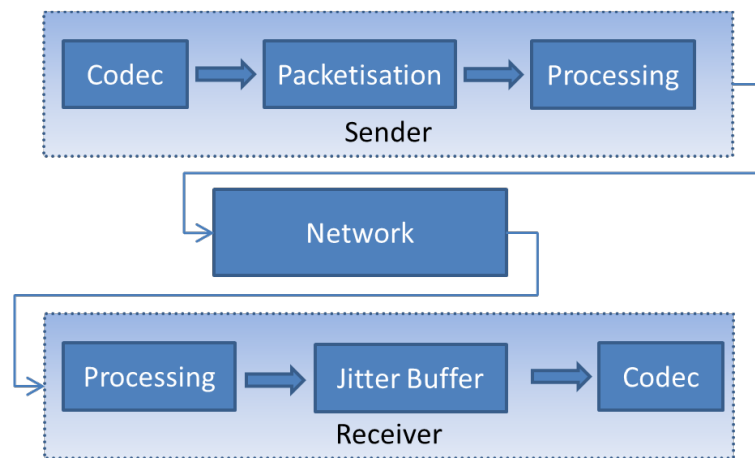


Figure 1: Sources of delay in VoIP

Echo

Without delay, echo is not usually an issue. However, once a delay has been added, the echo becomes perceptible to system users. This can be an issue in two common scenarios:

- *Telephone 4 wire to 2 wire hybrids*: When interfacing to a 2-wire phone system the hybrid circuits can produce significant echo
- *Operator Crosstalk* in an operations centre where multiple operators are communicating with common radio channels. The operators may be able to hear the audio from other operators with added network delay.

Eliminating Echo

Echo can be eliminated in some cases using Digital Signal Processors (DSP) to process the audio. Echo cancellation algorithms use an adaptive filtering algorithm to remove the echo signal from the received audio. The DSP algorithm uses an adaptive filter to remove the echo signal and can take a perceptible time to converge and cancel the echo. DSP based echo cancelling can affect audio quality during “double talk” (when both parties talking at the same time on a full-duplex link).

Packet Loss

On real world networks packets sometimes don't make it through. However, voice is generally intelligible even with quite high levels of packet loss as VoIP systems incorporate Packet Loss Concealment (PLC) algorithms to compensate.

On wired networks such as LANs or WANs, packet loss generally only occurs as a result of systems being overloaded or congested. However, other links are subject to loss of individual packet, such as WIFI networks or Microwave links.

Generally, well designed RoIP systems should provide intelligible audio with very high levels of packet loss. Packet loss of 10% or less should still provide acceptable audio quality. However, effective communications may be dependent on other mechanisms (such as TCP/IP based mechanisms) that may not be as tolerant of packet loss.

Coder Decoders (CODECS)

Radio and Telephony use the same mechanisms to transport voice through packet switched data networks.

At the transmitting end voice is converted into digital data, assembled into data packets and transmitted at regular intervals. Typically, these intervals are 20 milliseconds or some multiple of it.

At the receiving end the digital data is collated and converted back to into an analog signal. Generally, VoIP / RoIP systems use a sample rate of 8000 samples per second and single channel (mono) audio.

Software codecs take digitized audio and encode it into a form that is efficient to send over the network. They compress the audio so less data is required to transmit it. Generally, the codecs used for VoIP are lossy (i.e. the compression is achieved by discarding some of the audio information meaning a loss in the quality of the audio).

The codecs used for real time communications *must be* low latency, they need to be able to process the audio in real time (a codec only has 20 milliseconds to encode or decode 20 milliseconds of audio). Codecs such as MP3 don't have to encode the audio in real-time so they can achieve better compression and audio quality. However, they still need to decode in real time.

Widely used Codecs for VoIP and RoIP

Codec	Bits per Second	Notes
G.711	64 000 (8 bit, 8kHz)	Most widely used Codec
G.726	48 000, 32 000, 24 000 or 16 000	Adaptive Differential Pulse Code Modulation (ADPCM)
GSM	13 200	2G Mobile Phone Standard
G7.29	8 000	Used in a lot of VoIP Systems
IMBE	7 200	"Improved Multi Band Excitation"
AMBE	3 600	"Advanced Multi Band Excitation". Widely used in digital radio standards. Includes Error Correction.
ACELP		Algebraic Code Excide Linear Prediction
TETRA	7 2000	

The more modern codecs (G.729, IMBE, AMBE and the ACELP) used in digital radio are covered extensively by patents. The licensing costs associated with using these technologies are significant.

Recently there have been several low bandwidth narrowband codecs developed that are not covered by patents and being made available royalty free. Examples of these are BroadVoice16 (ITU J.161) and ITU G.718.

Network Protocol Overhead

If RTP is used to transmit the audio, there is an overhead caused by the network protocols used to transmit the data. If the data is sent over an Ethernet LAN the overhead typically consists of:

Protocol	Overhead (in bytes)	Overhead (in bytes)
Real Time Protocol (RTP)	12	12
User Datagram Protocol (UDP)	8	8
Internet Protocol (IP)	20	20
Ethernet	-	38
Total	40	78

With a packet size of 20 milliseconds, the overhead for the network layers becomes significant. In the case of the high compression codecs, the overhead is actually larger than the payload.

Codecs and Tone Transport

One issue for analog radio system and the high compression codecs is that the codecs have been optimized for speech signals. As a result, they can affect the fidelity of tones passed through them.

Codecs that compress the audio can result in tones being distorted. 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 signalling are not compatible. Hence, Radio over IP must transport signalling 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 signalling string, the same equipment must also be capable of encoding this into the appropriate audio tones.

The problem of passing signalling schemes can be further compounded by the inherent packet losses within IP networks. When passing voice, a low rate of packet loss is not usually an issue. However, when lost packets contain tone signals, it can result in breaks in the continuous tone, which will cause tone detectors to stop recognising the tone.

Ultimately, the most reliable way to transmit signalling schemes and tones through IP is to use a messaging protocol that is transparent to the end user equipment. Most of the radio functions that require tones have been replaced by digital functionality. The exception is audible alarms – such as a site wide emergency tone.

IP Connectivity Methods

There are a number of methods of connecting Radios over IP and this will depend on your IP network.

User Datagram Protocol

Typically, RTP is sent using UDP as the transport protocol. UDP is the simplest of the IP protocols as it provides for simple, stateless delivery of data packets. However, as it does not provide any mechanism for detecting and resending lost packets it can be unreliable. The RTP layer must detect and cope with information or audio data loss.

Transmission Control Panel

Another widely used transport layer Internet Protocol is TCP which is a reliable delivery guaranteed protocol. For RTP, however, TCP is not appropriate. The main problem with TCP is that if a packet is lost the TCP layer will stop delivering data until that lost packet has been successfully resent, resulting in large delays. In other words, TCP prioritizes reliability over real-time.

Multicasting

In computer networking; multicasting is a mechanism to send information to a group of hosts simultaneously. Multicast uses network infrastructure efficiently by requiring the source to send a packet only once, even if it needs to be delivered to multiple receivers. The routers in the network take care of replicating the packet to reach multiple receivers only when necessary. Ultimately this means greater bandwidth efficiency.

However, there are some issues that can mean multicasting can be difficult to implement:

- Multicasting does not work on the general Internet as ADSL routers do not support multicasting.
- In a WAN, all the routers in the network have multicasting enabled for multicast to work
- Configuring network routers to use multicasting is a complex task.

Conferencing

In situations where multicasting is not practical, a similar effect can be used by creating multiple unicast links to create a conference or party line.

By using Omnitronics' proprietary conferencing mode multiple RoIP adaptors can communicate with all other adaptors within the conference. Unlike multicast addressing which uses a one-to-many relationship, conferencing is peer-to-peer. In essence, this is a simulated multicast mode. Delay is also minimized by simply passing packets through rather than decoding them.

Whilst this peer-to-peer arrangement may be simple to implement, it can become impractical for larger network configurations. This is because the number of links that need to be configured grows exponentially with the number of radios.

To avoid creating such complex configuration scenarios, the RoIP adaptor purposely limits the number of remote radios that can be used in this conferencing mode. In a case where more than 5 radios need to be linked, one of the RoIP adaptors can be setup as a bridge server linking two conference groups.

Comparison Table

Connection Method	Advantages	Disadvantages
Unicast Real Time Protocol (RTP)	<ul style="list-style-type: none"> • <i>Constantly connected</i> - if the IP link fails at any time, it can be immediately detected by the console software and warn the user. • <i>Can be used over the Internet</i> - as it can be configured to work via Network Address Translation (NAT). 	<ul style="list-style-type: none"> • <i>Requires static IP addresses</i> for all parties - there could be additional cost from IT service providers as Dynamic Addressing is not used • <i>Limited Operator Positions</i> - due to the point-to-point nature of the protocol • <i>Greater Configuration of End Devices</i> -especially when multiple transmit destinations are required.
Multicast RTP	<ul style="list-style-type: none"> • <i>Constantly connected</i> • <i>Bandwidth efficiency</i> - more Voice Paths can be transported the same IP network than other. • <i>Simplified Configuration</i> - through the use of Dynamic IP (DHCP) • <i>Supports More Consoles & End Points</i> - on the same IP Network 	<ul style="list-style-type: none"> • <i>Won't work over the Internet</i> - LAN/WAN Only • <i>Complex Setup</i> - Many IT administrators will not allow multicasting schemes over the LAN or WAN. • <i>All Network Devices need to Support Multicast</i> - This could be costly and time consuming • <i>Difficult to Identify Potential Problems</i> - when used with more than a single unmanaged switch on a LAN

Seession initiated Protocol (SIP)	<ul style="list-style-type: none"> • <i>Connected only when required</i> - the IP network is freed up to be used by other devices thereby reducing overall bandwidth requirements. • <i>Simplified Configuration</i> – through the use of Dynamic IP (DHCP) & there is no need to configure the destination addresses of the receiving units. 	<ul style="list-style-type: none"> • <i>Doesn't work well where there is Bad Packet Loss</i> - if the IP link fails at any time, it cannot be immediately detected by the console software & warn the user, without considerable overhead. • <i>Can have issues with NAT and Firewalls.</i> • <i>Many IP networks are not 'naturally configured for SIP</i> - Firewalls will prevent the SIP packets passing through the router and stopping a link being made.
Omnitronics Unique Conferencing Mode	<ul style="list-style-type: none"> • <i>Enables Point-to-Multipoint Comms</i> - without the need for multicast support. • <i>Reduced Router Configuration</i> - to achieve the same result as multicast. • <i>Can be used over the Internet</i> - as it can be configured to work via Network Address Translation (NAT). 	<ul style="list-style-type: none"> • <i>Requires static IP addresses</i> • <i>Restricted to a Smaller Number of Devices</i> • <i>Greater Configuration at the End Devices</i> - as compared to SIP or Multicast. • <i>Less bandwidth efficient than multicasting.</i>

A Final Note on Digital Radio

Digital radios with IP connectivity are types of Radio over IP. In fact some of these standards that are based on the SIP and RTP protocols this paper has described. In particular DMR-AIS, APCO P25 DFSI, ISSI and CSSI standards use of both RTP and SIP.

However, there are several problems with connecting digital radio systems together over IP.

1. Firstly there is the problem that there are several standards and none of these standards are interoperable.
2. None of the current Digital radio protocols use widely implemented or freely available Codecs (even those that use RTP and SIP).

One solution to the above is to continue to use gateways to aid this interconnection over IP, providing this higher network functionality. However you must ensure that those gateways have been designed to transmit the added digital radio data and functionality.

This gateway approach also provides a number of other benefits. This includes:

- Digital Radios with Point-to-Point Protocols can be Shared Amongst a Number of Operator Positions
- Only ONE vocoder is required (and located in the gateway) and shared amongst a number of users
- Gateways allow the Radio to be accessed simultaneously by a mixture of dispatch consoles

Gateways provide organizations with the flexibility to change protocols to meet future demands

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