



FAQ

How much IP network bandwidth do I require for VoIP Audio Channels?

With radio now commonly being used over IP networks, Radio transmissions are now sharing bandwidth with the many IT devices an organisation maintains. In order to ensure that no packets are lost, it is important that enough bandwidth is available.

Calculating Bandwidth

The IPR series of Radio over IP (RoIP) interfaces use the Real Time Protocol (RTP) for transferring audio across a network. Each VoIP audio channel requires a certain amount of bandwidth.

The below calculates the worst case bandwidth for a *single channel* in *one direction*. This is the worst case as no channel should be on constantly.

The formula is:

Bandwidth Required = (Overhead kbps + Audio Requirement kbps) * Number of Channels

Overhead

In the IPR's, voice is converted to a packet every 40 milliseconds and 25 packets are sent per second.

Each packet carries a IP/UDP/RTP header overhead of 320 bits.

Therefore, the overhead is **8000bps**.

Audio

The VoIP audio bandwidth requirement depends on the compression used to encode the audio (codec). Use the following figures (in kilobits per second) to determine the bandwidth required for each VoIP channel:

G.711 codec	72kbps
G.726 codec	40kbps
GSM codec	22kbps



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Although each of the codecs will work on the IPR devices, it is recommended to use the G.711 codec as this will provide the highest quality. As the compression is increased the quality decreases.

Other Considerations

In addition to voice packets, RTCP (Real Time Control Protocol) packets are also sent. However, these are only sent once every 5 second and hence the amount of bandwidth used by these packets is negligible. For a SIP connection no data is sent at all when a connection has not been made.

When multiple connections are active, either from multiple SIP connections or from conferencing mode, then each VoIP channel will consume the amount of bandwidth shown – If a 960SIP console has 12 active SIP channels and is transmitting to all of them (all 12 channels switched to the on position) at the same time, then the bandwidth used will be, worst case, $12 \times 72\text{kbps} = 864\text{kbps}$.

Notes:

All of these calculations or requirements are in bits per second (bps), not bytes per second (Bps).

This is the bandwidth consumed while transmitting one half duplex channel. Channels will only transmit when the radio is busy and/or Voice Activity Detection is active, so actual bandwidth usage will be lower than this figure.

