# Intercom over IP

## The Communications Engineer's Guide to Integrate IP into Comms

**Convergence is one of the biggest topics in multimedia applications. Moving from analog production to a fully integrated digital production environment was the task in the 90s. The new century moves convergence even further to combine digitally integrated production facilities with seamless networking systems. This degree of integration is quite new to our industry and presents a challenge not only for the management of such complex interacting systems, but also for staff members having to cope with a fundamental change in production paradigms.**

**This white paper is designed to identify the possibilities and the limits of seamlessly integrating intercom and IP based services. It describes the available technologies under the premise of professional intercom parameters like: max call delay, audio quality, resilience and cost effectiveness.**

Engineers that work in media initially pose questions that reflect their environment. For example, audio related engineers may ask:

Why use IP for real-time applications?
How about system latency?
Which audio-coding is used? (reg. audio quality)

A potential Network administrator may ask:

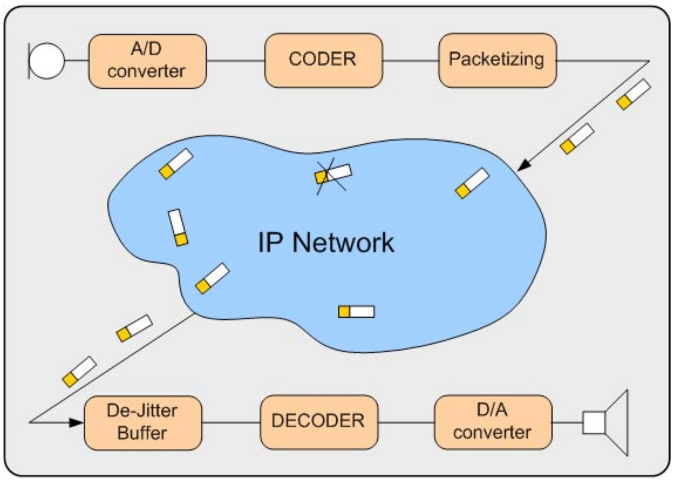
Why not use IP, our IP pipe is big enough?
How about packet sizes?
Which audio-coding is used? (reg. data rate)

Evidently different responibilities generate different sets of questions. So let's explore the possibilities and discover what is required.

### What is VoIP?

Voice over IP, a technology that uses data network services to carry voice data, dates back to the early 1980s although it wasn't really accepted until the 1990s shortly after the development of the World Wide Web, html and http. Until then the public Internet was not prepared to handle real-time voice data. The data rate and the quality of service was simply not consistently available. Early VoIP pioneers had to handle a variety of difficulties such as audio delays of several seconds, dropouts and a low audio quality.

Today high bandwidth data links are available for almost everybody. The problem of availability of data rate is solved by simply increasing the backbone and access data rate. Is the problem of real-time capability now history?




### VoIP functional principle.

If you compare VoIP and standard telephony, you will find some similarities. The main difference is that VoIP does not allocate real telephone lines, but uses a local network or the internet to transport voice data. [See the illustration above for functional blocks used for VoIP.] The incoming voice has to be first digitized by an ADC (analog-digital-converter). A coder is used to compress the native audio data rate by using one of several possible coding algorithms. Such algorithms differ in the way the audio is coded and offer various quality and data rate models. The

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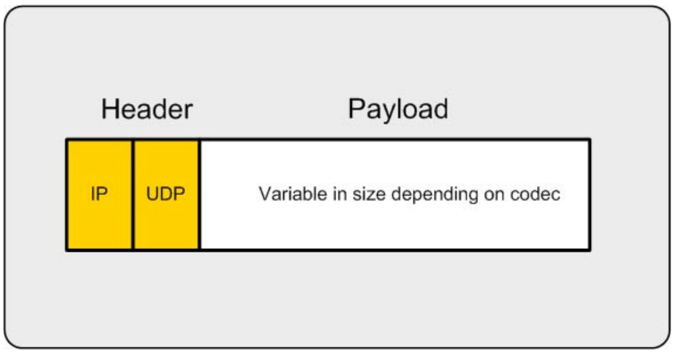
data compressed audio signal is now packetized to enable the transport over IP. A packet consists of a header and the payload.

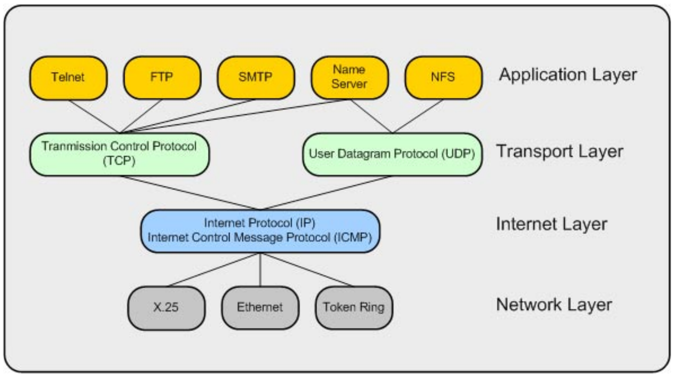



The header contains all the information to identify and address the destination. The payload carries the coded audio samples. At the destination, the incoming packets are stored in a buffer to cope with variations in the packet transport delay. After decoding the incoming data and reconverting it to an analog signal by using a DAC (digital-analog-converter), the signal can be sent to a speaker. Sound easy?
But to recognize the limitations, we have to dig deeper into the principles of IP data transport.

## How is the data transported through a network?

The transport of electronic data through a network requires some formal rules of behavior. Those rules are collected in protocols. The complexity of data traffic requires not only one protocol, but sub protocols, to manage several sub actions. These sub protocols are arranged in layers. The protocol's main tasks are to:

- address the destination
- control the data traffic
- supply a secure transport service




Which protocol is the ideal one for communication applications? The simplest answer is: There is no ideal one!

## TCP

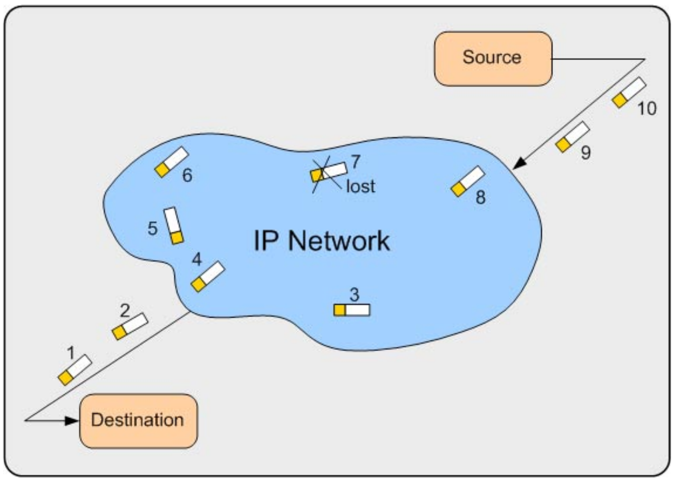
The TCP/IP protocol is quite popular, but what drives this protocol, and could it be used for communication applications?
Allow me to define the elements of the TCP and point out advantages and disadvantages.
TCP (transport control protocol) describes the method of data transport over a network. TCP is used for file transfer or email applications. The TCP protocol ensures a reliable, connection orientated bit stream. If you send a packet over TCP, every packet will trigger a positive acknowledgement message to the source if a packet is received. If the source does not receive this message, the source will retransmit the packet. As packet losses can occur in networks, this mechanism makes sure that the overall packet loss is minimized.

## UDP

The UDP protocol is a connectionless protocol. This protocol has no mechanism to handle lost or defected packets. Compared to TCP, a UDP header contains less information and is much smaller.

## Which protocol is the best choice for real-time voice transport?

It appears, that TCP is the best protocol. It's much safer and handles lost or defected packets. What does that mean in real-time voice applications? Let's do this step by step:



A source sends packets to a destination over TCP. The packets will, due to the IP network definition, not appear in the right order, nor will all packets  be received initially.

At the destination a buffer stores the packets to analyze the headers, send acknowledgements to the source and waits for retransmitted packets. This is fine for file transfer or email applications because they are not time critical. In real-time voice applications, however, this is a nightmare because the buffer delays will exceed a level to allow bidirectional communication. Sadly, voice applications have to use the UDP protocol. Using the same scenario as above with UDP:  the destination could not wait for a retransmitted packet because there is no mechanism to tell the source to resend. In reality it‘s more practical to live with lost packets than with long buffer delays. Keep in mind that these buffer delays have to be added to the transmission, coding, and decoding delay. You can begin to understand what „real-time“ in networks really means.

Now that we have a protocol to use let‘s look at the audio coding.

### Coding Algorithms

Searching for the right coding algorithm means leveling a balance between audio quality, data rate and processing delay. The higher the data compression, the higher the audio delay (not true for all cases, but generally speaking). How to balance? To get a feeling of data rates in industry coding processes see the following table:

| Algorithm | Audio BW | MOS | Bit rate |
|---|---|---|---|
| G.711 | 3.5kHz | 4.1 | 64kBit/s |
| G.722 | 7.5kHz | 4.5 | 64kBit/s |
| G.723 | 3.0kHz | 3.5-4 | 5,33/6,4kBit/s |
| G.729AB | 3.2kHz | 4 | 8kBit/s |

MOS is a commonly used subjective method to measure the quality of audio links. The measurement ranges from 1 (not acceptable) to 5 (very good). The above mentioned bitrate is the amount of data we have to transport per second, so after packetizing the network data rate will be higher than the coding data rate because of the packet headers.

### Coding data rate vs. network bandwidth

Network traffic is an important parameter for network administrators. Let‘s take G.711 as a given audio coder with a given coding bitrate of 64kBit/s. A fatal fault wil result from only calculating the coding bitrate, and this causes a lot of confusion.. Remember, this data has to be filled into packets and these packets need to add a packet-header. This issue would not be worth a individual chapter, if there was just a header (data-rate) to add. As the packets are variable in size, and the header is fixed, the ratio between payload size and header is variable. This means:

Short packets -> high “header overhead” per payload
Long packets -> low “header overhead” per payload

Because packet sizes are limited through router and network definitions there is another variable parameter to balance. Keep in mind that the network data rates double in bidirectional communications.

### Ethernet / Internet

I think we all can agree that Ethernet as the most established and most dominant network standard.
The main problem with any real-time transport over Ethernet is its unpredictability. You cannot forecast the time a packet needs from source to destination. The only solution is to increase the data rate significantly. Try to imagine the situation for the Internet. Real-time audio over IP requires at least a fraction of a leased line with guaranteed data rates. Intercom over Internet will become possible with higher data throughout.

### Resilience?

Is VoIP an alternative for producing or timing critical applications? Well the only true answer is: No, but! The „but“ means in most installations, there are a percentage of really critical subscribers and a percentage of subscribers not that critical. For the non-critical ones, VoIP is an ideal method to save significantly on connection costs. VoIP users must accept and tolerate delays, hiss glitches, and dropouts because this is in the nature of the transport mechanism - not VoIP itself.

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## Bringing it all together

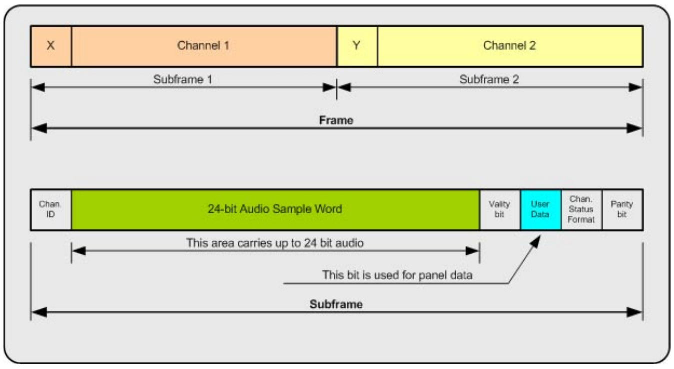
We at Riedel decided that the first step was to develop an external VoIP Codec. The unit is called Connect IP and comes in a 19"/1RU box. The coding and packetizing engine is quite powerful and allows for selecting a wide range of audio coding algorithms to adapt to the various types of applications, including:

- pure audio link
- audio link and trunk data (for trunked systems)
- remote panels

The supported coding engine supports the following audio bandwidths:

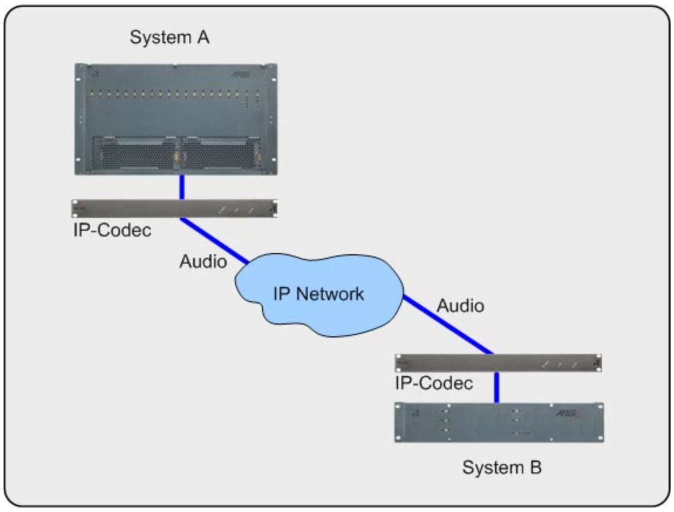
| Audio Bandwidths | | Network Traffic |
|---|---|---|
| 3,5kHz | | 40kBit/s |
| 3,5kHz | (2-channel mode) | 80kBit/s |
| 7kHz | | 80kBit/s |
| 7kHz | (2-channel mode) | 150kBit/s |
| 15kHz | | 150kBit/s |
| 15kHz | (2-channel mode) | 300kBit/s |
| 20kHz | | 230kBit/s |
| 20KHz | (2-channel mode) | 460kBit/s |

The unit is a standalone box, equipped with an audio I/O port and a network port. The interface comes with a configuration tool to adjust the network and IP settings. The I/O port is a digital AES3 compatible audio port. In Artist systems, the link between matrix and panel is a bidirectional AES link. The audio part of the AES stream is used for audio transportation and the AES user bits carry the panel data.



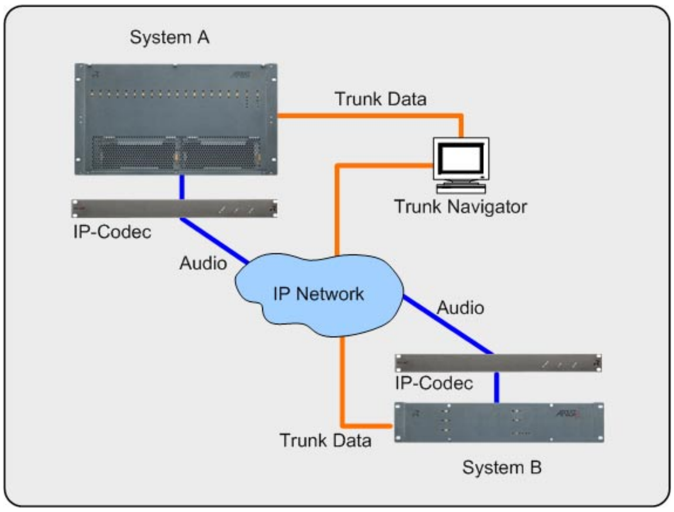

## Application: 4-wire over IP

This application makes use of the VoIP technology simply to transport audio between two sites. The Connect IP units provide an AES3 I/O and could be used to create a 4-wire over IP.



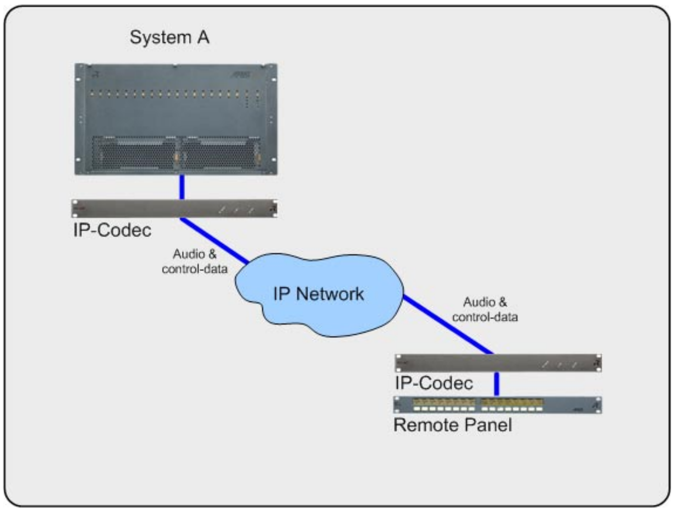

## Application: Trunking over IP

IP could also be an application for trunked intercom systems. IP codec is used for audio transport and connecting the intercom systems to a central Trunk Navigator to manage the dynamic port allocation.



## Application: Remote Panel over IP

If complex trunking is not needed, a remote panel could be an alternative. The panel receives the audio and control data through the network. The remote panel is fully configurable from the host matrix and behaves just as if it was connected directly to the matrix.




## Conclusion

Despite the fact that IP-based real-time audio applications still face some limitations, the acceptance of these solutions is growing consistently. Driving factors include IP's cost-effectiveness  and the ability of the quickly growing networks to meet the needs of real-time audio transport in the near future. Audio-over-IP is still a highly dynamic field, and for intercom manufacturers this offers two possibilities of coping with this reality: taking a compromise-afflicted codec and concentrating on the implementation in the manufacturer's intercom architecture, or concentrating on the development of the appropriate codec solution which will become an integrated solution of the intercom system. Both methods contain advantages and disadvantages. At the end of the day it's a question about quality vs. time-to-market.

In the first step Riedel concentrated on developing the appropriate solution regarding protocols and algorithms. The current outcome is an external device. Now that this solution is available Riedel is working on the second step: integrating this solution in the Artist platform's modular card structure.