CASE STUDY

PugaNET: How IP can simplify systems and reduce cost.

“The challenge was to build a 256x256 audio routing network to serve 25 remote stations across an IP WAN with the possibility to route any source from any location to any destination. Without IP, this would have been crazy expensive. And probably racks full of stuff. With IP, we delivered a compact, impressive system at a remarkably low cost.”

– Igor Zukina, Director of Engineering, Streamcom, New Zealand.

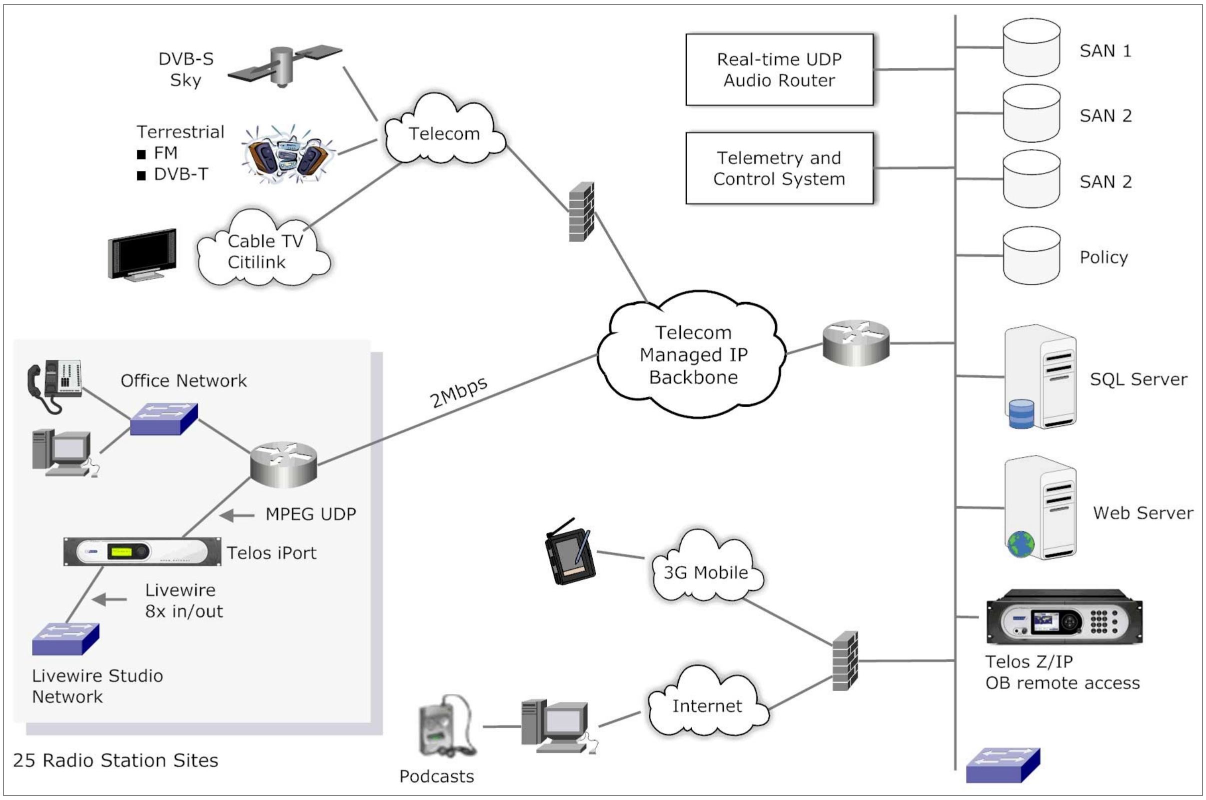
Igor describes the system:

PungaNET, commissioned at the beginning of 2009, is a real-time national audio distribution and contribution network that connects 25 radio stations across New Zealand. Each station has eight audio inputs and outputs that can be cross-connected system-wide in any needed configuration. The network also provides TCP/IP connectivity for standard office applications such as file sharing, VoIP, and Internet access. A public website provides Internet streaming of programs for all member stations.

Routing and management (booking and scheduling) is provided by a central management system, which is a software application with a Web user interface that is accessible across the network. Audio equipment is provided by Telos/Axia and the central switching equipment by New Zealand-based XI-Audio. Cisco routers are used at the network edges.

A requirement for the system design was that it be based on a standard private IP networking service available from Telecom New Zealand. This is a product made to provide secure IP service for businesses between their office locations. These IP networks are much lower cost than the old synchronous circuits which were used in the past to interconnect audio codecs. The IP network can provide QoS, with various levels being available at a range of price points. We chose an “interactive class network” which is a product sitting between the cheapest “business data” and the most expensive “real-time” class of service networks. The central site has a 60Mbps circuit and each station has a symmetrical 2Mbps circuit. The latency of the data network from the sites to the center is around 10-15ms, although for one of the most remote rural sites it is 45ms. Network jitter is constantly monitored and is around 5ms. No lost packets have been detected so far.

Telecom NZ has a basic network configuration that they offer at a good price, but when you request custom configurations, the cost skyrockets. For PungaNET we required multiple levels of queue priorities, which would be custom. We had in the past ordered custom-configured circuits for a number of customers and these become a support nightmare. For example, the special settings were lost during every major Telco network maintenance cycle. This may not be a problem with all providers, but it is something to consider during system design: keep it simple where you have no direct control. To avoid this problem, we decided to install our own IP routers at the network edges and use a single class-of-service network from Telecom NZ. This provides us with full packet tagging and queue management under our own control.



Audio Coding

With 2Mbps links at the stations and the need for eight stereo channels, audio coding is required. We use a few different codecs:

- Audio distribution : Discrete stereo AAC-LC @128 kbps
- Audio contribution: Discrete stereo AAC-LD @128 kbps
- Intercom: Mono AAC-LD @64 kbps
- Internet stream: Parametric stereo AAC-HE V2 (AAC+) @28kbps

The MPEG AAC family offers a range of bitrate and delay trade-offs. For on-air audio, we went with standard AAC for its high fidelity. For the intercom system, low audio latency is more important than fidelity, so we chose AAC-LD for that. For Internet streaming, we use AAC-HE for its very high bitrate efficiency.

Routing of Audio and GPIO

We didn't want to use IP multicast in PungaNET's WAN due to our desire to stay with a standard Telco IP product. Therefore, we contracted with local firm XI-Audio to make a unicast UDP audio stream router that gives the same result but keeps the routing entirely within our hands. The router runs on off-the-shelf PC hardware, so cost was much lower than a traditional audio router would have been. It is located at the central site.

Each station has a Telos iPort, which encodes and decodes eight channels of audio. The iPort outputs are UDP/IP streams. Each of these is mapped to a port on the UDP router. The router receives the stream and sends it to the required destination – or copies of the stream to multiple destinations, if needed. The outputs from the router are mapped to the iPort decoder inputs at each station. The router has 256 inputs and 256 outputs, all carried over one RJ-45. From a user perspective, it works just like a traditional audio cross-point router.

The UDP switch is controlled by Axia's PathFinder router control software, which provides a user interface and automatic management of route changes.

GPIOs are provided by Axia devices at the stations. These are switched along with the audio in the router and managed by PathFinder.

Mixing of the network to network and network to local streams

The PungaNET provides scheduling live-to-air content. A master control-like facility is required which can provide cross-mix transition from local audio to network audio, from one network program to another, and from the network back to local. This mixing is performed by a V-mix instance (supplementary audio mixer function) with the station iPorts.

Common Resource management

The Central Router Management System provides a Web-based management tool. Users at each station can log on to the CRM over the network to manage and schedule network and local resources. The systems database and business logic provides a large set of resource scheduling and controlling mechanisms. Users can share studio resource to selected members of the network, configure intercom routes, or simply schedule on-air programming. Program providers can publish their schedules to the program guide.

There is a Pathfinder server at each station. The CRM system interrogates each individual PathFinder at all the remote locations and collects information about all audio and GPIO devices, ports, and PathFinder router organizations.

When a user schedules channels and resources in the CRM System, event scripts are created and loaded to each PathFinder scripting engine for execution. Schedule events are downloaded to each PathFinder server 48 hrs in advance so that a fault or maintenance of the CRM system will not affect system routing.

Audio Interfacing

All stations have Axia Livewire-based studios. The Telos iPort is has a direct RJ-45 connection to the studio Livewire network that conveys all eight in/out audio channels. Thus, there is no analog or AES3 anywhere in the system.