Principles of the Radioscape DAB Emergency Tunnel System (DETS II)

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Introduction

The DETS II system is intended to provide emergency voice break-in to DAB broadcast radio services within a localised area. As the name implies, the original use of the system was in road tunnels, but the principle can be applied to other situations where a local broadcast override is required.

Theory of operation

In an FM or AM broadcast system, voice break-in is simple to implement; the broadcast transmission is simply replaced with one carrying the announcement. For DAB and other digital services, this simple approach will not work, because the receiver has to actively acquire and track the frequency and timing of the broadcast signal. If the signal is simply replaced without preserving the frequency and timing characteristics of the original, the receiver's tracking is disrupted and it has to then re-acquire the signal. Receivers vary in their ability to cope with this, but many can take a minute or more to complete the process; approaching an emergency situation in a vehicle travelling at 100km/h or more, a minute can be a matter of life or death. The emergency announcement signal must therefore be time- and frequency-aligned to the original broadcast.

An additional complication arises out of the ensemble structure of DAB and related systems. Each radio channel carries a multiplex of a number of different services – typically 8 or more – each carrying different data. This structure must be preserved during voice break-in; if it is not, the receiver will simply regard the break-in channel as a new service and will add it to its internal database but not re-tune to it. The aim of the system must therefore be to create an ensemble identical in layout to the original broadcast, but with the audio data replaced by the emergency announcement.

Implementation

A very common misconception is to assume that the DETS system re-modulates and re-broadcasts the original signal under non-emergency conditions. Whilst it is capable of operating in this mode, in practice this is rarely useful; the inevitable processing delays would result in a zone of interference where the original and rebroadcast signals overlapped, and the time misalignment would result in an audible loss of a section of the signal at the transition between ambient and rebroadcast regions. To avoid this, re-broadcast is done directly at RF, with the off-air signal being amplified, filtered and re-radiated locally. Obviously, for this to be successful, there must be sufficient isolation between the receiving and re-radiating antennas to prevent the onset of RF instability; in a tunnel environment, where re-broadcast is typically achieved using a radiating cable ("leaky feeder") system and the receiving antenna is sited some distance from the tunnel entrance, this condition is not difficult to satisfy.

When an announcement is required, an RF switch is used to replace the conditioned off-air signal with the locally-generated emergency broadcast. In situations where RF instability is likely, the system can be used in an "override" mode. Here, the off-air signal is not rebroadcast but is assumed to be present in the area of interest. When an announcement is required, the RF output of the DETS system is enabled at such a level as to swamp the ambient broadcast. In either case, *the DETS installation produces a suitably formatted output continuously; the switching is done at RF. The system itself does not know or care whether its output is currently being utilised as the transmission source.*

As already mentioned, the system must produce a signal with the same configuration as the off-air broadcast and time- and frequency-aligned to it. DETS II uses a number of components to achieve this – the Subchannel Replacer, Audio Encoder Controller, Timing Controller, Audio Encoders, Audio Multicaster and COFDM. For clarity, it is first assumed that the system is operating with a single ensemble and a single language.

Configuration tracking

The off-air signal is demodulated by a full-ensemble receiver, and the FIC and subchannel data extracted. The FIC data is processed to provide configuration data to the rest of the system, as well as being combined with the subchannel data streams to produce an ETI stream. This ETI stream will be to all intents and purposes identical to the one that was used to create the original broadcast.

The configuration data derived from the FIC provides information about the bit-rates and formats of any audio channels that may be contained in the ensemble. This data is passed to the Audio Encoder Controller, which attempts to configure its available Audio Encoders to produce data streams of compatible bit rate and format.

Audio Encoding

The Audio Encoders derive their input from the Audio Multicaster. This is used to allow a single external input from an audio interface to be duplicated to all of the encoders. Thus, each encoder produces a compressed version of the same audio content (the emergency announcement), but at a different bit rate and if necessary, in a different format (i.e. MPEG or AAC).

The compressed audio data from the Audio Encoders is fed to the Subchannel Replacer. Here, each audio subchannel in the off-air derived ETI stream has its audio data replaced with an encoded stream of a suitable bit rate and format. If for some reason a subchannel cannot be replaced, for instance because no encoded stream is available, then the system provides the option of either leaving the stream unchanged or replacing it with silence. Non-audio subchannels such as packet or stream data are not affected and are passed through unchanged.

Modulation and Timing

The resulting modified ETI stream is passed to the COFDM, which uses it to produce a modulated RF signal with the same configuration as the off-air signal but with the content replaced. Timing of the output signal is locked to a GPS-derived time and frequency standard.

The output RF signal is inevitably somewhat delayed with respect to the off-air broadcast, due to the time deinterleaving and reinterleaving processes in the receiver and COFDM respectively. The exact magnitude of this delay must be precisely controlled such that the transmitted Phase Reference Symbols of the off-air and replacement signals coincide to within less than half of the system guard interval (i.e. +/- 123us in Mode 1). In practice, the alignment should be rather better than this, and is typically maintained to within +/- 10us or less. Since delay variations of this order are common due to multipath, receivers will cope quite happily with this without having to re-tune.

Time alignment is measured at RF. The off-air and output signals are down-converted and the null symbols detected by the Signal Alignment subassembly. The results from this are fed to the Subchannel Replacer, which then adjusts the timestamp data in its output ETI stream to minimise the offset. Note that there will still be a relative delay between the off-air and replacement signals, but it will be an integral number of transmission frames. This does not matter, as the audio content is different anyway. The delay means that there will be a glitch in the CIF count seen by the receiver, but in practice the receiver is unaffected by this and continues to demodulate its selected subchannel(s) as normal; it is simply that the audio in this subchannel(s) has been replaced with the emergency announcement. It is commonly assumed that some sort of reconfiguration is required to achieve this, but in fact this is not the case; the multi-rate audio encoding architecture means that each subchannel keeps its original configuration.

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Multi-Ensemble Operation

The system is easily extended to cover more ensembles by adding more Subchannel Replacer Units (SCRU) (which include the receiver) and COFDMs. Each SCRU-COFDM pair provides coverage of a single ensemble, with all SCRUs connected to a common Audio Encoder Controller. The Timing Controller, including the Signal Alignment measurement hardware, is incorporated within the Audio Encoder Controller Unit (AECU). Since the timing references for the replacement and off-air signals are derived from stable frequency standards, a single Timing Controller can control the timing alignment for a large number of ensembles; the Signal Alignment hardware tunes to each frequency in turn, and updates the SCRU for that frequency with the measured timing information.

The output data from the Audio Encoders is streamed using a multicast protocol, so each encoder can feed more than one SCRU.

Multi-Language Operation

To operate with more than one announcement language, the encoders are operated in groups, each group taking its input from a separate audio source. Each subchannel in the SCRU has a language set either automatically, from the service or ensemble language, or manually, and the SCRU will choose an encoder from the appropriate language group to provide the source data for that subchannel. The Audio Multicaster is capable of handling inputs from more than one source and distributing them to the appropriate groups.

Number of Encoders Required

Whilst a multi-ensemble system may have a very large number of subchannels, in general relatively few audio encoders are needed as most broadcasters tend to standardise on the use of a small number of bit rates. Fundamentally, one encoder is needed for each unique combination of bit rate, data format (MPEG or AAC) and language.

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Equipment Layout

A minimum system for a single ensemble consists of 3 units:

AECU:

- Hardware components
  - o Signal Alignment subassembly
  - o Audio Input Card
- Software Components
  - o Timing Controller
  - o Audio Encoder Controller
  - o Audio Multicaster
  - o Up to 8 Audio Encoder instances
  - o The Access Gateway service, which provides the interface to the remote control and monitoring client.

SCRU:

- Hardware Components
  - o Receiver subassembly
  - o ETIC – COFDM interface, GPS receiver and time/frequency generation
- Software Components
  - o RF to ETI interface
  - o Subchannel Replacer

COFDM:

- Hardware Components
  - o Modulator assembly
- Software Components
  - o COFDM Controller

Each additional ensemble requires one extra SCRU and one extra COFDM. The AECU is common to all ensembles. The AECU may be fitted with up to 4 inputs, allowing it to cater for a quad-language system. If more than 8 data rate/language/format combinations are required, additional Multiple Audio Encoder Units (MAEU) may be added. These connect to the system via the IP network, and provide extra processing power to allow for more encoder instances.