A new mobile digital broadcasting system is needed for the FM band

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Introduction

In Ofcom's "Future of Radio" consultation document, it forecast that 90% of all radio listening will be via digital platforms in 10 years' time, and it will then probably be feasible to look at switching FM off. In the same document, Ofcom has also suggested some possible uses of the FM band after FM has been switched off:

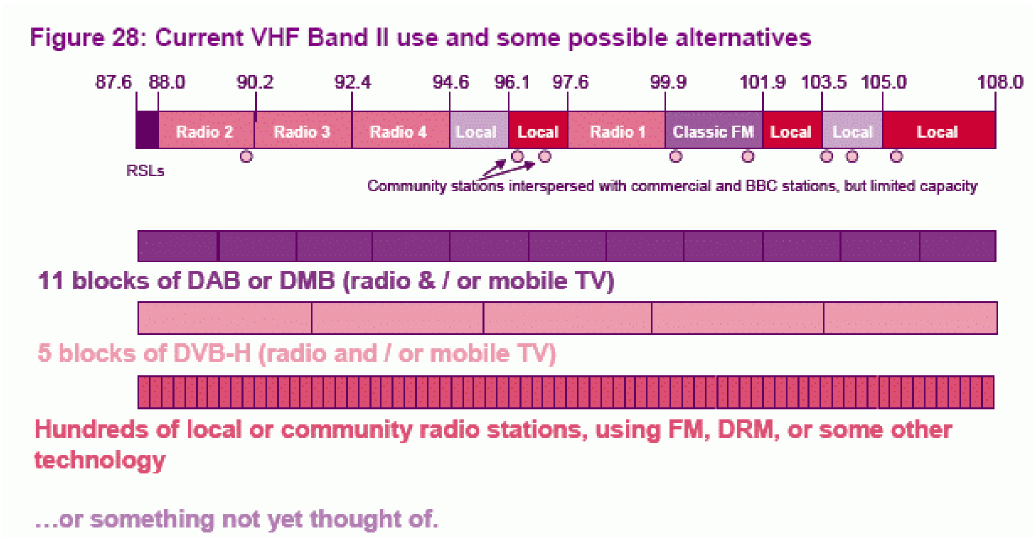

Figure 1. Current and possible future uses of the FM band

I would like to propose that a new mobile digital broadcasting system should be designed from the ground up, because mobile digital communications technology has advanced dramatically since the design of the DAB and DVB-T systems. The more recently introduced DMB/DAB+/DAB-IP and DVB-H systems are simply extensions of the DAB and DVB-T systems, respectively, and their design was held back by backward compatibility constraints, and any newly-designed system could provide a huge increase in both spectral and power efficiency relative to the existing mobile broadcasting systems.

As FM will not be switched off for at least 10 years, there is plenty of time to design a new system from the ground up so that once FM is switched off the spectrum can be used as efficiently as possible. In the longer term, such a system could also eventually replace the various DAB systems transmitting in Band III.

If such a system were developed, then it could be added to receivers silently, just as the ability to receive DAB+ will be added to DAB receivers without consumers being aware of this happening; and if development of the new standard takes place sooner rather than later, the vast majority of DAB+ receivers would be able to receive the standard by the time that FM is switched off and new transmissions commence on Band II.

Certain mobile digital communications technologies now exist that did not when the current systems were designed, and some of these technologies now perform so close to the optimum level that large improvements in spectral efficiency above and beyond what is possible today simply wouldn't be possible.

Possibly the most important technology that is now feasible is that of over-the-air software upgrades, which would effectively mean that whatever new system is designed will never become obsolete – the only limitation is the computing power of the receivers, which is itself a moving target.

Furthermore, if we assume that Moore's Law continues unabated for the next 10 years – the experts predict that they will be able to match Moore's Law for a further 15 years – computing power will have doubled five times, so in theory computing power will be 32-times greater than today; so raw computing power is hardly going to be a limitation.

The issue of power consumption should also improve over time, because the further shrinking of components should allow lower on-chip voltage levels to be used, and the inventiveness of the microelectronics industry will no doubt find other ways of reducing power consumption over the next decade.

The most important lesson that should be learnt from the design and subsequent implementation of DAB is that digital broadcasting systems should be designed to be as efficient as possible – the designers of the DAB system decided to go for the least computationally complex technologies, which resulted in DAB being very inefficient, and it subsequently had to be upgraded 4 years after the BBC began advertising it on TV, which is obviously far from ideal.

The implications of using a high spectrally and power efficient system are that:

- far more channels can be transmitted, therefore maximising choice for consumers
- transmission costs will be far lower
- the likelihood that higher audio and picture quality will be provided is far higher, because the transmission costs would be far lower and the available capacity would be far higher
- because there would be more channels, this increases competition, which should, in theory, drive up the quality of both content and technical quality
- there is more capacity available for innovative new services
- more spectrum would be freed up for other uses

If the huge proliferation of TV channels on Sky is anything to go by, then it could easily be envisaged that the number of mobile TV channels and radio stations – and new data services as yet unthought of – that would want to transmit in 10 years' time could be far higher than many might expect today. So in this respect we're at a similar position to where the designers of the DAB system were at in 1990, where we don't actually know what the demand for spectrum in the FM band will be, but it could well be very high.

It therefore makes sense to assume high demand will materialise, and try to ensure that it can be accommodated rather than taking a wait-and-see approach and then reacting once the demand becomes high (and it is too late).

In any case, using more advanced technology will still drastically reduce transmission costs, because modern technologies are far more power efficient than the technologies used in current systems, which is obviously desirable for broadcasters.

Furthermore, if Ofcom is unwilling to regulate to ensure that good quality is provided, and the commercial radio and TV industries don't want to spend more money than is absolutely necessary, then using a very efficient system to make transmission costs as low as possible gives consumers the best chance of actually being provided with the quality they would like.

There will be people who would argue that with the mobile Internet likely to become ubiquitous, or at least widely available in the next 10 years then there is no need for a new mobile broadcasting system to be designed. But as Figure 2 shows below, there will always be a need for more popular channels to be broadcast, because the alternative is to distribute them using unicasting, which is a grossly inefficient means of distribution, because each user receives his or her own individual transmission.

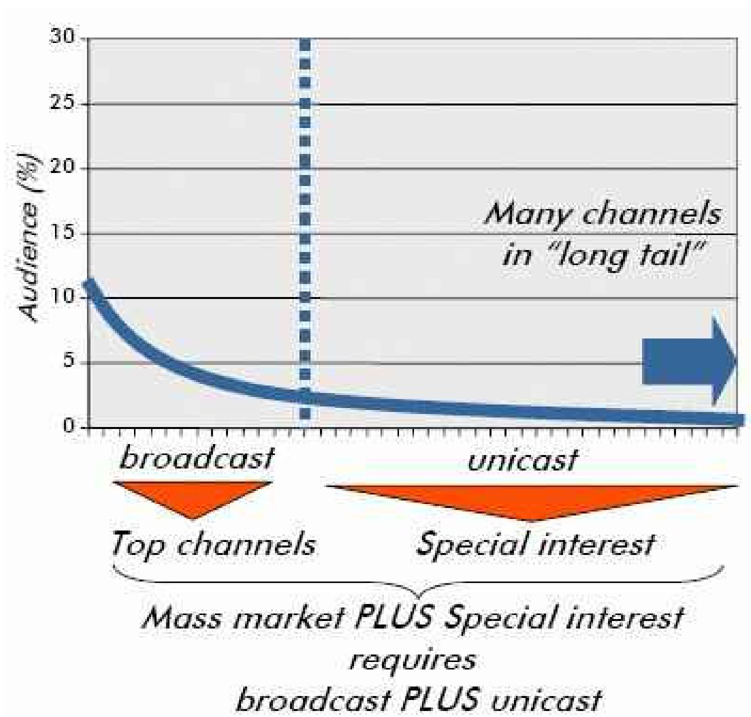

Figure 2. The requirement for broadcast plus unicast

Furthermore, Figure 2 also suggests that it would be more efficient to broadcast the more popular on-demand content to be stored in the receivers' memory rather than dealing with large volumes of unicast requests.

Ultimately, no more can be done in terms of the technology used than to provide a system that is as spectrally and power efficient as possible (within reason).

1. Problems with the existing systems

1.1 The old DAB system

In 10 years' time, the MP2 services will all have been switched off, so there is no point in discussing the old DAB system here.

1.2 DAB+/DMB/DAB-IP

The old DAB system has recently been upgraded to produce three new variants, which have adopted modern audio and video codecs and stronger error correction coding in the form of Reed-Solomon coding:

- DAB+ – the upgraded version of DAB for digital radio (to differentiate it from the variants meant for mobile TV below)
- DMB – used for mobile TV
- DAB-IP – used for mobile TV

However, although the AAC+ audio codec has now been adopted for DAB+, the core DAB transmission scheme, which is used by all of the above variants, is still using technologies that date back to the early 1990s:

1. DAB uses differential modulation in the form of DQPSK, which has the dual problems of severely limiting the spectral efficiency of the transmission scheme, and it carries a 3 dB SNR penalty – which means literally that the transmitter powers are double what they would be if synchronous modulation were used.
2. The concatenated convolutional coding with Reed-Solomon coding that DAB has *just* adopted was state-of-the-art in the early 1990s, and since then turbo codes have appeared which provide vastly superior performance.

Basically, the new DAB variants are extensions of an out-of-date system, and they are thus unable to solve the underlying problems that exist in the core DAB transmission scheme.

The result is that the new DAB variants are nowhere near state-of-the-art in terms of spectral and power efficiency, so if they were used on the FM band then that spectrum will be used very inefficiently.

1.3 DVB-H & MediaFLO

DVB-H and MediaFLO use much better transmission scheme than the DAB variants due to the use of stronger error correction coding schemes. However, DVB-H and MediaFLO have the following problems with respect to transmission on the FM band:

1. The minimum DVB-H channel bandwidth is 5 MHz, which would only allow 4 channels to be used on the 20 MHz-wide FM band (Figure 1 is incorrect, because it says that there can be 5 DVB-H channels in parallel on the FM band). This makes DVB-H inflexible in terms of spectrum planning for multi-frequency networks (i.e. you cannot transmit on the same channel in both, say, Liverpool and Manchester due to co-channel interference issues), which I would suggest makes DVB-H unsuitable for use on the FM band, and it is especially unsuited for use with local multiplexes because it will consume spectrum far too quickly.
2. Wide channel bandwidths also imply high-capacity multiplexes, which means that they can carry a very large number of radio stations due to the low bit rates used with modern audio codecs. Narrower-bandwidth channels would therefore be better suited to carrying radio stations – or lower bit rate data services – because this decreases the likelihood that multiplexes are not fully utilised.

1.4 DRM+

DRM+ also uses a better transmission scheme than the DAB variants, but it has one main drawback:

1. DRM+ uses very narrow channel bandwidths, and it is thus prone to flat-fading (where the whole channel bandwidth is significantly attenuated), which can only be mitigated to a certain extent, and cannot be solved completely. Therefore, it doesn't make sense to use DRM+ when other systems can be used that don't suffer from flat-fading.

DRM+ does have one advantage over systems that use wider channels, however, in that because its channel bandwidth is so narrow (typically 50 kHz) it will always be the most suitable system for small-scale radio stations with a small, or uniquely-shaped, coverage area.

2. Far lower transmission costs for broadcasters

2.1 The FM band's inherent power advantage

One inherent advantage of transmitting on the FM band is the lower path loss. Quoting Friis' free-space path loss equation in decibel form:

Free space loss (dB) = 32.44 + 20 log f (MHz) + 20 log d (km)

This shows that the path loss varies according to 20 log f (MHz), which in turn means that the difference in path loss due to transmitting at frequencies 1 and 2 is:

Power difference (dB) = 20 log (f_2 / f_1)

Frequency band	Typical frequency MHz	Reduction in path loss due to transmitting in the FM band at 100 MHz
Band III	200	6.0 dB
UHF	600	15.6 dB

Table 1. Reduction in path loss due to transmitting in the FM band

2.2 Lower required SNR for modern systems

Most newly designed mobile systems nowadays use near-optimal error correction coding schemes such as turbo coding, which results in the required SNR being much lower than that required by systems that use weaker error correction coding, such as all of the new and old variants of the DAB system.

For example, Table 2 shows the required C/N (carrier-to-noise ratio – the only difference between SNR and C/N is where it is measured in the receiver) for DMB/DAB-IP and for the best-performing mobile TV system, MediaFLO, which uses turbo coding along with RS coding:

System/ Transmission mode	Modulation and error correction code rates (CR[1])	Spectral efficiency bits/s/Hz	Required C/N dB
DMB/DAB-IP PL3A	QPSK CR ½ RS(204,188)	0.64	15[2]
MediaFLO Mode 1	QPSK CR ½ RS(16,12)	0.525	6.6
MediaFLO Mode 2	16-QAM CR ½ RS(16,12)	0.7	8.6

1 – CR stands for code rate
2 – Based on findings in Radioscape white paper – DAB+ and DAB-IP both use the same error correction coding as DMB, which was tested in Radioscape's white paper, therefore their performance should be identical.
3 – See reference [6]

Table 2. Comparison of required C/N for different mobile TV systems

Interpolating between MediaFLO Modes 1 and 2 to the 0.64 bits/s/Hz spectral efficiency of the DAB-based systems results in a required C/N figure of 7.9 dB for MediaFLO; or in other words, for the same level of spectral efficiency, MediaFLO requires 7.1 dB lower C/N than the newer DAB-based systems for mobile TV reception.

2.3 Combination of lower SNR & transmission frequency

With all else being equal, the reduction in transmitter powers due to the combined effect of using a modern system that allows the use of low SNR levels along with the reduction in transmission frequency from 200 MHz to 100 MHz is simply the sum of the above two reductions:

Reduction in power (dB) = reduction due to frequency + reduction due to SNR

Reduction in power (dB) = 6.0 + 7.1 = 13.1 dB

In linear units, this is:

Reduction in power (linear) = $10^{(13.1 / 10)}$ = 20.4

That is, with all else being equal, the transmitter power levels could theoretically be reduced by a factor of 20 by using a modern mobile broadcasting system in the FM band in comparison to using DMB/DAB-IP in Band III.

The above example used mobile TV systems simply because there are no advanced digital radio systems, and also because C/N figures are available for the mobile TV systems, but a similar theoretical reduction in transmitter power levels would be expected for digital radio stations as well.

And borrowing an assumption made in an EBU Technical Review article that the total transmitter powers are proportional to transmission costs, this would mean that a 20-fold reduction in transmitter powers would lead to a 20-fold reduction in transmission costs in comparison to transmitting DAB in Band III.

In reality, when the transmission powers are reduced by such a large amount, the proportional relationship between power and cost would not hold, so a 20-fold reduction in transmission costs would not be achievable. However, it would be fair to say that with such a large reduction in transmission powers then the transmission costs would also be reduced by many times over – which is the desired goal of such a new system, as stated in the Introduction of this document.

3. Technologies for a new mobile broadcasting system

The goal in the design of a new mobile broadcasting system, and indeed any communication system, is to simultaneously minimise the SNR and maximise the spectral efficiency of the system, and to choose the most efficient component technologies available, as this will lead to the most efficient system with the lowest transmitter powers, and hence the lowest transmission costs.

As Figure 3 shows, the main way to simultaneously minimise the SNR and maximise the spectral efficiency is to target increasing the robustness of the signal. This stresses the importance of using strong error correction coding, and most recently designed systems have opted to use near-optimal error correction coding schemes for this reason. An example of increasing the efficiency of the component technologies used is DAB's recent adoption of the AAC+ audio codec.

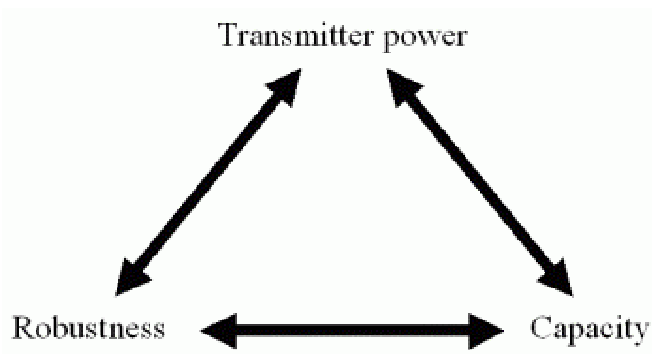

Figure 3. Communication systems' fundamental three-way trade-off

The following sub-sections discuss individual technologies that could be combined to form a highly spectrally efficient and power efficient mobile broadcasting system that would be cheap to transmit.

3.1 Full over-the-air (OTA) software upgradeability

Although this provides no improvement in efficiency in itself, allowing full OTA software upgradeability allows advances in technology to be easily incorporated.

An obvious example of why such a feature would be so useful is the DAB system, because I dare say that the processors in all current DAB receivers could handle MP3 and Reed-Solomon decoding, so as the problem with the DAB system has been the MP2 codec, all receivers could have been upgraded over the air, and this would have vastly improved the DAB system. Then with the emergence of AAC+, WorldDAB could have specified that the computational complexity of receivers should increase in order to support AAC+ in future.

Another benefit is that new media formats can be supported on the fly. For example, if a receiver has a colour screen, the receiver could download the software necessary to display some image format on the screen, which it doesn't already have the software for.

Basically, software upgradeability makes the system far more flexible, and it allows continual improvements to be made rather than one-off large-scale improvements followed by a long period without any improvements.

And far from being pie-in-the-sky, OTA software downloads are expected to become the norm in the mobile phone industry, and it is envisaged that in future, if you travel to a country which uses a different mobile phone standard to the UK, you will be able to download the software required for the phone to receive the other country's standard.

3.2 Transmit diversity

You will probably have heard of MIMO technology, which is where multiple transmit and multiple receive antennas are used to improve capacity and/or range and/or the robustness of wireless transmissions – one example is the "Pre-N" Wi-Fi equipment (which stands for the fact that it is proprietary equipment that performs similarly to the forthcoming IEEE 802.11n Wi-Fi standard) you can buy in computer peripheral retailers that use multiple antennas at both ends of the link in order to improve capacity and range in comparison to the 802.11 b/g Wi-Fi standards.

Unfortunately, MIMO (Multiple-In, Multiple-Out) technology cannot be used on a mobile broadcasting system designed for use on the FM band for reasons such as FM aerials being long (half-wave dipoles are approximately 1.5m long); car owners wouldn't want to install multiple aerials on their car roofs; and especially because a new mobile broadcasting system would need to cater for mobile phone reception. At low frequencies such as Band II (the FM band) and Band III (used for DAB), the headphone lead has to be used for the aerial for reception on mobile phones or personal radios, so it would be impractical to ask users to have two headphone leads flapping around as they walk, not to mention the fact that when multiple antennas are used they should ideally be spaced at least half a wavelength apart – which for FM is about 1.5m!

However, it is possible to use multiple antennas at one end rather than both, and for the case of a mobile broadcasting system for the FM band the appropriate solution is to use multiple transmitting antennas and one receiving antenna, which is referred to as 'transmit diversity', or MISO (Multiple-In, Single-Out).

MIMO and transmit diversity take advantage of "spatial diversity", which means that signals travelling over different spatial paths "see" a different channel (even though the antennas are only typically spaced half a wavelength apart), and thus the signals received via the spatial paths are themselves different; the benefit of having signals travelling through different spatial channels is that there is a lower probability that the paths between all of the antennas (there are N^2 spatial channels when there are N transmitting and N receiving antennas) are simultaneously faded, thus the probability that the signal can be received reliably increases.

Engineers working on the DRM+ system have had positive results from experimenting with transmit diversity in order to try and overcome the flat-fading problem that is inherent to narrowband wireless systems. The transmit diversity scheme they have implemented is called Cyclic Delay Diversity (CDD), which has a block diagram shown in Figure 4 [1]:

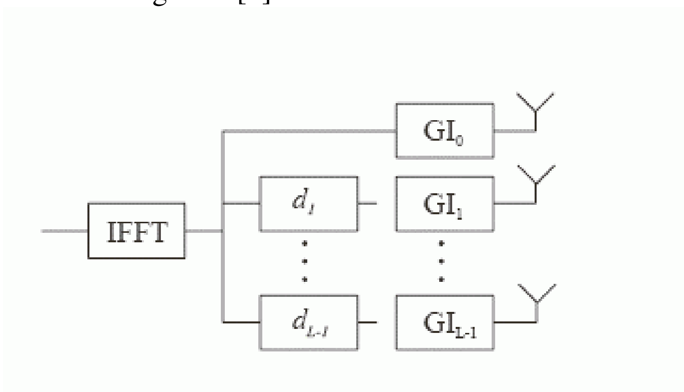

Figure 4. CDD transmit diversity scheme

The IFFT acts as the OFDM modulator and generates the useful part of the OFDM symbol in the time-domain; this is then cyclically-shifted by different delays for each antenna by the ‘d’ delay elements in the block diagram; the respective guard intervals are added (which consist of copies of the end of the respective cyclically-shifted useful part of the OFDM time-domain symbol); and the resulting symbol is transmitted on each antenna.

This method generates artificial echoes, and so increases the frequency selectivity of the received signal, which is especially useful for attempting to overcome flat-fading. Figure 5 shows the BER vs E_b/N_0 curves with and without the use of CDD transmit diversity on DRM+ [1].

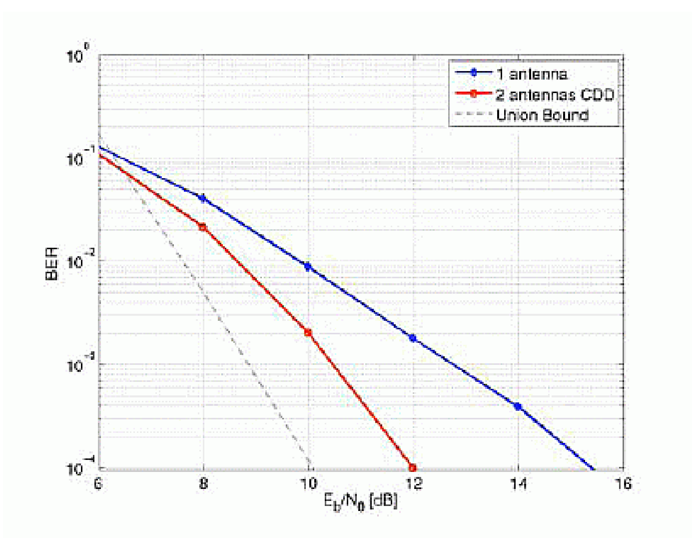

Figure 5. BER vs E_b/N_0 with and without CDD transmit diversity

Figure 5 shows that CDD provides a 3 dB SNR advantage at a BER of 10^{-4} in comparison to using one transmit antenna; and using more transmit antennas would provide further improvements in performance due to increased spatial diversity, although the rule of diminishing returns applies.

One significant advantage that CDD has over other some of the other transmit diversity schemes is that the receiver doesn't have to carry out any additional processing compared to when a single transmitting antenna is used.

Although transmit diversity is especially useful for narrowband channels that are subject to flat-fading, the addition of spatial diversity will also improve the reliability of reception for wideband systems – part of the so-called "network gain" that results from receiving signals from different transmitters in an SFN will be due to increased spatial and delay diversity.

3.3 Error correction coding

The importance of error correction coding seems to have been vastly underestimated by the engineers that worked on both versions of DAB and on DRM. Stronger error correction coding makes signals far more robust, and this simultaneously maximises the spectral efficiency and minimises the required SNR, which leads to lower transmitter power levels thus making the system cheaper to transmit. Furthermore, lower required transmitter powers reduce the frequency re-use factor, which increases the capacity per unit area – i.e. more multiplexes can be transmitted per country.

3.3.1 Turbo + RS coding

Turbo codes were the first error correction codes to perform within a fraction of a dB of the Shannon limit, which is the theoretical information rate limit beyond which no communication system can pass.

Because of their near-optimal performance, turbo codes are being employed by an increasing number of mobile communication systems, including systems such as the new HSDPA upgrade to the 3G mobile phone system, the MediaFLO mobile TV system, the MBMS broadcast/multicast variant of the 3G system and the forthcoming DVB-SH (DVB-Satellite services for Handheld) satellite system.

However, turbo codes only perform within a fraction of a dB of the Shannon limit on the Additive White Gaussian Noise (AWGN) channel, and transmission over the mobile (Rayleigh) channel is more difficult because of multipath propagation. Research has shown that concatenating turbo coding with an outer layer of Reed-Solomon (RS) coding performs better than turbo coding alone over the mobile channel [2].

The reason why RS coding improves performance is that multipath channels inherently cause bursts of errors to occur, and RS coding is well suited to correcting short bursts of errors.

CRC checks can also be used to improve the error correcting capability of RS codes and/or reduce receiver power consumption. The way they can improve the error correcting capability of RS codes is by using "erasures", as used on DVB-H's MPE-FEC, where the CRC checks instruct the decoder where the errored packets are in a grid, and this doubles the number of bytes that the RS decoder can correct (because half of the error correction capability of a block code is consumed by locating the errors). And power consumption can be reduced with CRC checks because an RS codeword that is known to be correct (because it hasn't failed the CRC check) doesn't need any further processing to correct the errors.

3.3.2 Turbo - Multi-level coded modulation (T-MLCM) + RS coding

A possible alternative to using turbo + RS coding as mentioned above would be to use turbo-multi-level coded modulation (T-MLCM), which is the combination of Multi-level coded modulation (as used on the DRM system) with turbo coding, where the component encoders and decoders used in a multi-level scheme are replaced by turbo encoders and decoders, respectively, in order to improve performance.

Figure 6 below shows that T-MLCM performs far better than ordinary MLCM [3], especially as the required BER (bit error rate) decreases. The near-vertical drop in BER for T-MLCM suggests that – which is also provided by turbo + RS coding – it would perform far better for carrying mobile TV channels than using ordinary MLCM, because video requires a lower BER for robust reception than audio does.

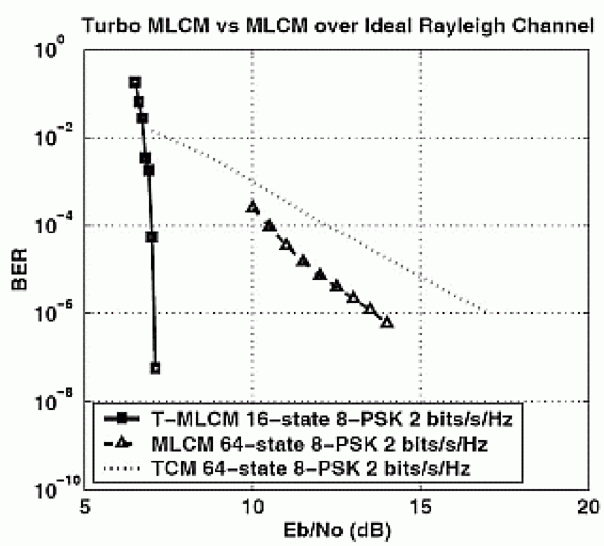

Figure 4.30: Comparison of Performance of T-MLCM vs Conventional MLCM for 8-PSK, 2 Bits/s/Hz Schemes over Ideal Rayleigh Channel.

Figure 6. Performance of T-MLCM vs MLCM over the Rayleigh (mobile) channel

As for the case of ordinary turbo coding discussed in the previous section, a T-MLCM code could be concatenated with an outer layer of RS coding to improve burst error performance and thus reduce the required SNR further.

3.3.3 Low Density Parity Check (LDPC) codes

LDPC codes are the other main type of error correction codes that perform close to the Shannon limit due to the use of the “turbo principle”, where the error correction decoder iterates towards the desired solution, which is the key to the optimal performance of both LDPC and turbo codes.

LDPC codes are included in the mobile WiMAX 802.16e system specification and they have been combined with a BCH code for the new DVB-S2 satellite system.

One advantage LDPC codes have over turbo codes is their lack of patent protection.

3.4 Statistical multiplexing

Statistical multiplexing (stat-muxing) has been used on digital TV systems for many years in order to fit more channels into the fixed capacity terrestrial multiplexes or satellite transponders due to the fact that the video bit rate is inherently variable (VBR – variable bit rate). Stat-muxing has also been implemented on the MediaFLO system, and it has recently been trialed on the DVB-H system, where initial results have shown that it provides a 35% gain in terms of the number of mobile TV channels that can be carried in a multiplex.

The bit rate of audio is also inherently variable, although engineers working on DAB have ignored this fact. Although the bit rate of audio doesn’t vary to the same extent as video’s does, a very significant gain would still be achieved if stat-muxing were used for multiplexing radio stations together. Furthermore, the gain from stat-muxing increases with the number of channels being transmitted – because it is statistically far less likely that all channels require a high bit rate at any instant in time.

I would estimate that a 20% gain in efficiency could be expected from stat-muxing radio stations together.

3.5 Layered modulation & source coding

Layered modulation and source coding is a method that is similar to what hierarchical modulation attempts to achieve, whereby there is a high priority base layer which has a low required SNR level, and a lower priority enhancement layer which has a higher required SNR level. This is advantageous, because when the SNR falls below the minimum SNR level required for the lower priority layer, the receiver falls back to using the high priority base layer and reception is not lost completely.

This method is similar to that used on the FM system, where receivers revert back to mono when the SNR is too low to receive stereo adequately.

The MediaFLO system offers layered modulation and source coding for video transmissions where the video frame rate is reduced from 30 frames/s to 15 frames/s when the SNR is too low to decode the higher bit rate stream, but to the best of my knowledge (MediaFLO is a proprietary system, and Qualcomm, who designed the system, has not released much information about how the system was designed) MediaFLO does not use this technique for protecting audio.

However, MPEG-4 Audio has specified Scalable AAC, which allows for mono-to-stereo scalability; an example of which is shown in Figure 7 for a low bit rate HE-AAC channel [4]:

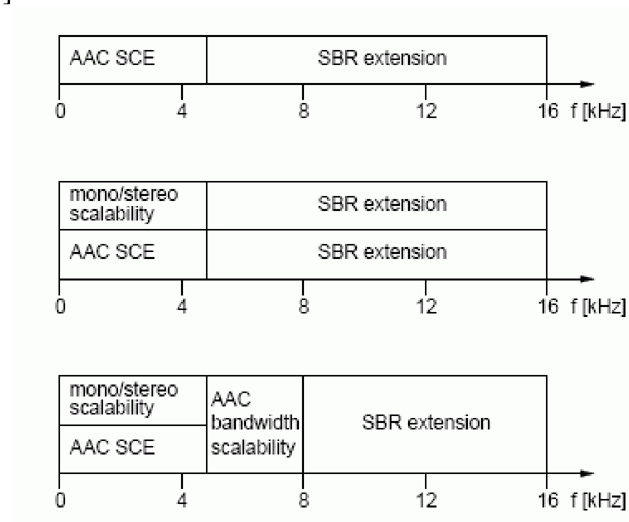

Figure 7. Mono-to-stereo scalability via Scalable AAC

There are two possible ways to allow mono-to-stereo scalability to be implemented: use hierarchical modulation and use QPSK modulation for the high-priority mono base layer and 16-QAM for the low-priority enhancement layer; or a more efficient way would be to use UEP (unequal error protection) and place the mono signal in the highly-protected part of the UEP profile and the mono-to-stereo enhancement layer and possibly the bandwidth scalability sections in the lower protected part.

3.6 MPEG-4 Audio Version 2 Error Resilience Tools

MPEG-4 Audio Version 2 specified two Error Resilience tools to improve the error performance of MPEG-4 audio being transmitted over noisy or multipath channels:

- Error Robustness (ER) tools
- Error Protection (EP) tools

3.6.1 Error Robustness (ER) tools

The ER tools perform things such as reordering the bit-stream in order to avoid error propagation [5], and they therefore allow AAC/AAC+ audio to perform at higher BER levels than would otherwise be possible – i.e. as the name suggests, the audio becomes more robust to errors in the bit-stream.

The ER tools are already used on the DRM system, and they are available for use on DMB; bizarrely, however, when I enquired with WorldDAB about whether the ER tools would be used on the new DAB+ system, they said that they hadn't made up their mind about whether to use these (obviously beneficial) tools or not.

The DVB-H system and the MediaFLO systems don't seem to use the ER tools either, which is presumably because these systems were designed for mobile TV use, and with video requiring a significantly lower BER than audio, the audio would inherently be received more robustly than the video on the same multiplex. However, on any newly-designed mobile digital broadcasting system that is meant to carry both radio stations and mobile TV channels, it would obviously be beneficial to use the ER tools.

3.6.2 Error Protection (EP) tools

As shown in Figure 8 below, the EP tools allow UEP (unequal error protection) to be used to protect the audio. The EP tool rearranges the AAC/AAC+ audio bit-stream into groups according to how sensitive the bits are to errors, and the more sensitive data is protected more strongly than the less sensitive data so as to achieve a lower probability of errors producing audible disturbances.

UEP (not the MPEG-4 EP tool) is used on the DAB and DRM systems to protect the MP2 and AAC/AAC+ audio, respectively, but the new DAB+ system and the DMB, DAB-IP, DVB-H and MediaFLO systems don't use it – despite the fact that the DAB specification states that UEP profiles can be designed for non-MP2 audio.

However, rather than implement the whole of the MPEG-4 EP tool – for example, there would be no point in applying the RS coding for a second time – what the existence of the EP tool shows is that it would be possible to protect AAC/AAC+ audio with UEP by simple rearrangement of the audio data into groups, and it would therefore be beneficial to implement it into any new mobile digital broadcasting system.

And as already mentioned, UEP could be used in conjunction with Scalable AAC to implement mono-to-stereo layered modulation and source coding, which could vastly reduce the likelihood that an audio signal drops out entirely, and this would therefore improve the quality of service and increase the effective transmitter coverage area, albeit that the fringe areas would only receive a mono signal.

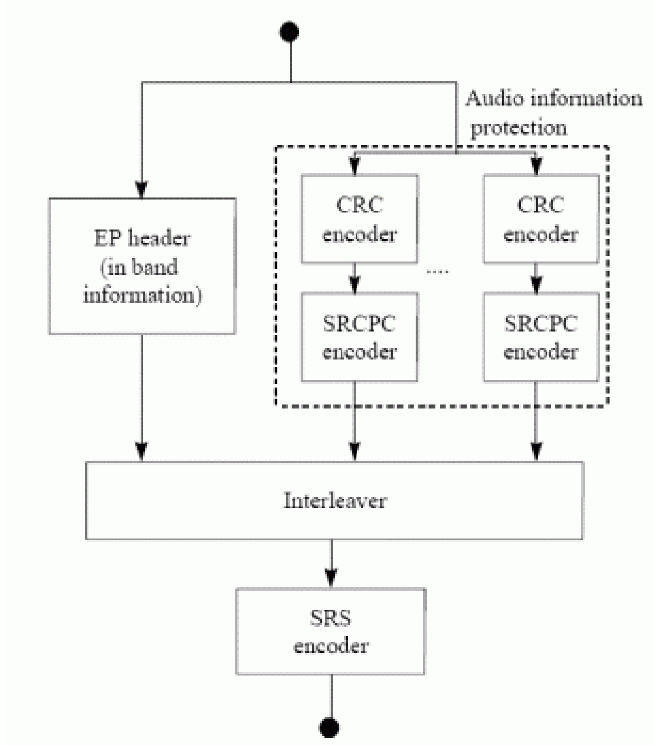

Figure 8. MPEG-4 Audio Version 2 Error Protection tool [5]

(SRCPC stands for systematic rate-compatible punctured convolutional code, CRC stands for cyclic redundancy check and SRS stands for shortened Reed-Solomon code.)

3.7 New variants of OFDM modulation

OFDM technology has moved on since the standardisation of DAB, DVB-T and DRM, and VSF-OFCDM and TDS-OFDM are two of the newer flavours of OFDM that would be prime candidates for a new mobile digital broadcasting system.

3.7.1 VSF-OFCDM

VSF-OFCDM stands for Variable Spreading Factor - Orthogonal Frequency Code-Divison Multiplexing, and it is the modulation scheme used on the NTT DoCoMo 4G prototype system. VSF-OFCDM is the combination of OFDM and CDMA (code-division multiple access), where the latter is the modulation scheme used by the UMTS 3G mobile phone system.

With CDMA, users are allocated an individual ‘PN’ (pseudo-noise) code (a pseudo-random sequence of +1’s and –1’s) to differentiate the user’s signal from the signals of all other users. The user’s data is multiplied by this PN code in order to ‘spread’ the signal over the channel bandwidth, and the use of unique codes for the different users allows a large number of transmissions to be overlaid on top of one another within the same channel bandwidth. Once synchronised, the receiver will multiply the received signal by the same code to retrieve the user data.

VSF-OFCDM uses 2-dimensional spreading, which means that it spreads the signal in both the time- and frequency-domains – i.e. the data on the OFDM subcarriers in the frequency-domain is multiplied by a PN code and the time-domain signal generated by the OFDM modulation is spread by another PN code.

CDMA systems provide a "processing gain" due to their spreading of the data signal over the channel bandwidth, which is given by:

Processing gain = channel bandwidth / data bandwidth

This processing gain is equal to the "spreading factor", which in turn equals the number of "chips" used in the PN code to multiply the user data. The spreading factor for 2-dimensional spreading equals the product of the time- and frequency-domain spreading factors.

In a broadcasting system, you would substitute radio stations or TV channels for users in the above discussion, and each channel would be assigned its own PN code – the PN code acts like an individual channel in CDMA systems.

It may prove that combining OFDM with CDMA offers no benefit in comparison to using OFDM alone, but CDMA is good at interference rejection, whereas OFDM is not, therefore it might prove that certain combinations of time- and/or frequency-domain spreading factors provide superior performance to OFDM alone. For example, it might be the case that frequency-domain spreading combined with OFDM (referred to as MC-CDMA, which stands for multicarrier CDMA) provides better co-channel interference rejection than OFDM alone, which leads to a lower frequency re-use distance, thus the capacity per unit area could be increased.

3.7.2 TDS-OFDM

TDS-OFDM stands for time-domain-synchronous OFDM, and it is the modulation scheme adopted for the forthcoming Chinese DMB-T/H terrestrial TV and mobile TV system. The TDS part of its name refers to the fact that it inserts a pseudo-noise sequence into the OFDM guard interval (the guard interval only consists of a repetition of the end of the OFDM symbol, so it doesn't carry any useful information) in order to carry out OFDM synchronisation and channel estimation. In comparison, systems such as DVB-T/H that use synchronous modulation (DAB/DAB+/DMB/DAB-IP don't even use synchronous modulation) transmit known pilot signals on a certain number of subcarriers per OFDM symbol in order to allow the receiver to synchronise and estimate the channel, but these pilot cells consist of redundant information, so they eat into the useful bit rate. China claims that their TDS-OFDM achieves a 10% increase in capacity – not a huge amount, but better than nothing – in comparison to DVB-T and the time-domain synchronisation is also faster.

3.8 Other desirable features

The following features are not critical in terms of the spectral efficiency of a new system, but they are nonetheless desirable features for a new mobile digital broadcasting system to have.

3.8.1 A wider range of channel bandwidths

As already mentioned, the wide channel bandwidth used by the DVB-H and MediaFLO systems makes them inflexible in terms of frequency planning for multi-frequency network environments; in particular, they are unsuitable for use as local multiplexes.

The wider channel bandwidth systems also make them less suitable for carrying radio stations alone, because each multiplex can carry a very high number of radio stations due to the high multiplex capacity and the relatively low bit rates used for radio stations that use modern high-efficiency audio codecs.

Therefore, in order to improve flexibility, it would be advantageous if narrower channels were allowed at least as an option. One possible range of channel bandwidths would be 1, 2, 3, 4 and 5 MHz.

If it was deemed that only one channel bandwidth should be used due to RF design issues, then for flexibility of frequency planning I would suggest that a channel bandwidth similar to that used by DAB would be a far better solution than using wider channel bandwidths.

3.8.2 Per-channel parameters

One drawback with the DVB-H system is that the same error correction code rate has to be used for all of the channels on the multiplex, which stops DVB-H multiplexes that are configured for mobile TV use from taking advantage of the higher BER levels that audio channels can handle. Therefore, it would be beneficial for error correction code rates to be allocated on a per-channel basis.

3.8.3 Time-slicing & fast channel change time

Time-slicing, as used on the DVB-H and MediaFLO systems, is an important technique to drastically reduce the RF front-end power consumption by switching the RF front-end on only when the desired channel is being transmitted. DVB-H achieves around 95% to 98% reduction in RF front-end power consumption for typical channel and multiplex bit rate levels.

However, time-slicing as implemented on DVB-H can lead to high average channel change times for low bit rate channels due to the off-time between time slices being approximately (ignoring the short burst duration) given by:

off-time ≈ burst size / channel bit rate

so, for a constant burst size, as the channel bit rate goes down, the off-time goes up, leading to an increase in the average channel change time.

The MediaFLO system displays a significantly lower average channel change time than DVB-H due to 4 channel bursts being transmitted per second, which avoids the long off-time situation with low bit rate channels on DVB-H. It would therefore be advisable to follow MediaFLO's time-slicing method for any new system.

4. MediaFLO – currently the best system

MediaFLO is currently the best mobile digital broadcasting system, because it already offers the following features that some or all of the other systems do not:

- Turbo + RS error correction coding for high spectral and power efficiency
- Statistical multiplexing
- Layered modulation & source coding for video (although not for audio)
- Per-channel transmission parameters
- Time-slicing with a short channel change time

However, its main drawback that, in my opinion, makes it unsuitable for use on the FM band is its wide channel bandwidths, which has the following problems:

- Wide channel bandwidths are inflexible for spectrum planning
- Wide channel bandwidth multiplexes have too high a capacity for them to be used exclusively for radio stations or low bit rate data services

Another drawback is that it is a proprietary system whereas the other existing systems are open systems, and manufacturers shouldn't have to pay a single company for the right to use technologies that it didn't invent when WorldDAB or DVB could easily design a system themselves that outperforms MediaFLO.

Also, similar to all of the other systems, MediaFLO lacks the more advanced features that could be used to improve the error robustness – and hence the power efficiency – for audio transmission; in particular, it lacks layered modulation and source coding that could be provided by the combination of MPEG-4 Scalable AAC and the use of UEP.

5. Conclusions & Discussion

As FM is going to be around for another decade or so, there exists an opportunity to plan ahead and to design a highly efficient system that can be used on the FM band.

Technologies exist today that would allow a huge increase in spectral and power efficiency relative to what existing mobile digital broadcasting systems achieve. Such large increases in spectral and power efficiency would proportionately reduce transmission costs, which would obviously be desirable for broadcasters, but it would also maximise choice for consumers and maximise the probability that good audio and picture quality is provided.

It could easily be envisaged that there will be a high number of mobile TV channels that would want to transmit; more radio stations and other audio services; more popular on-demand content would be more efficiently distributed via broadcast than unicast; and existing and as yet unthought of data services will also need to be carried. Therefore, it is likely that there will be a high demand for broadcast capacity, so it makes sense to assume that this high demand will materialise and ensure that it can be accommodated rather than to do nothing.

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