Digital Audio Broadcasting

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SC5 - b
Digital Audio Broadcasting

• DAB Broadcasting
  – OFDM, SFN, Transmission frames
  – UK ensembles, System Parameters

• Source Coding – MP2

• Channel Coding
  – Convolution, Puncturing, Freq & Time interleaving

• Receiver front end

• Channel decoding

• Synchronization
Main References

1. ETSI. “Radio Broadcasting Systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers”. EN 300 401, European Telecommunications Standards Institute, April 2000.


History of DAB

• 1986 – DAB consortium formed
  – France, Germany, Netherlands, UK
  – Eureka 147 development project

• 1990 – First trial broadcasts

• 1993 – Public demonstration system in UK

• 1995 – Network broadcasts in UK

• 1997 – World DAB forum formed
Problems with AM and FM

- **Multipath fading**
  - Reflections from aircraft, vehicles, buildings
  - very large variations in signal strength over distances of ~1 m

![Graph showing multipath fading](image1)

- **Interference**
  - from equipment, vehicles and other radio stations
OFDM

- Orthogonal Frequency Division Multiplexing
- 1536 carriers at 1 kHz spacing
  - symbol length: 0.246 + 1.0 = 1.246 ms
  - 2 bits per carrier per symbol (DQPSK)
Cyclic Prefix

- Convolution with channel impulse response
  \[= \text{sum of delayed, scaled copies}\]

- If channel impulse response < 0.246 ms:
  - No inter-symbol interference
  - DFT gives input spectrum of symbol multiplied by channel response:
    - frequency dependent amplitude and phase shift
Frequency Domain Orthogonality

- Taking DFT of 1 ms segment is equal to:
  1. convolving spectrum with sinc (FT of 1 ms window)
  2. sampling at multiples of 1 kHz

- Component frequencies are orthogonal and do not interfere.

- Doppler spread damages orthogonality:

  \[
  \text{ICI power} \approx 0.5 \left( \frac{v f_0 T}{c} \right)^2 P_0 = -14 \text{ dB at } 190 \text{ km/h}
  \]
Single Frequency Network

• All transmitters send an identical signal
• Interference-free if delay + multipath < 0.246 ms relative to nearest transmitter
• Optimal spacing $\approx c \times 0.246 \text{ ms} = 74 \text{ km}$

• Transmitters further than $\approx 1.2 c \times 0.246 \text{ ms}$ do more harm than good
Fill-in transmitter

- Can have a low-power fill-in transmitter to solve a local reception problem
- Add delay to synchronize with main Tx
UK DAB Multiplexes/Ensembles

- Each DAB multiplex: 1.536 MHz bandwidth
  - 0.176 MHz gap between multiplexes
- Four multiplexes per 7 MHz TV channel
  - 2 National ensembles
  - Regional + Local (3 for London)
Spectral Efficiency

• Existing FM transmissions
  - Each transmitter has a bandwidth of 0.2 MHz
  - Nearby transmitters must be 0.4 MHz apart
  - 2.2 MHz needed for a network covering entire country

• DAB
  - 1.5 MHz for 10 services covering entire country using a single frequency network
  - 15 times more efficient!
Frame organization

Audio

MP2 frames (24 ms)

DAB frames (96 ms)

Synchronization and header

ClF: Common Interleaved Frame (24 ms of data)
DAB Transmission Frame

- CIF: (2.3+55.3) kbits/24 ms = 2.4 Mbps total
  - FIC: 96 kbps for multiplex config and service names
  - MSC: 2.304 Mbps for audio + data
    - Services: Radio 2 = 256 kbps, Radio 7 = 155 kbps
    - Only need to decode the wanted portion of the MSC
System Parameters

- **Centre frequency**, \( f_0 \approx 220 \text{ MHz} \) (Band III)
  - Wavelength \( > 1 \text{ m} \) \( \Rightarrow \) diffraction around objects
  - Lower frequencies are full up
- **Total bandwidth** = 1.537 MHz
  - Needs to be \( > 1.5 \text{ MHz} \) for fading to be frequency selective
  - \( < 1.6 \text{ MHz} \) to fit four into a 7 MHz TV channel
- **Cyclic Prefix** = 0.246 ms
  - Needs to be \( > \) (transmitter spacing)/1.2c to allow SFN
  - Wasteful if long compared to useful symbol length
- **Carrier Spacing** = 1 kHz \( \Leftrightarrow \) **Useful symbol length** = 1 ms
  - Symbol length \( < 0.4/(\text{Doppler spread}) \approx 0.4c/(f_0v) = 10 \text{ ms} \)
  - Symbol length \( < \) cyclic prefix for efficiency
- **Transmission frame** = 76 symbols = 96 ms
  - Long for efficiency, short for ease of synchronization
Source Coding

• Based on MP2 (MPEG 1 Layer 2)
  – Simpler than MP3 but less good

• Masking – Psycho-acoustic model
  – Loud sounds make quieter sounds inaudible at nearby frequencies and times

• Sub-band Processing
  – Input @ 48 kHz sample rate
  – Divide into 32 subbands of 750 Hz @ 1.5 kHz
    • 36 samples/subband in each 24 ms CIF frame
    • Only low 27 subbands are used (0 to 20.25 kHz)
Masking

- Normal hearing threshold is A
- Threshold is changed to B because of tone C
- Higher quantization noise allowed in bands near tone C
- Band D can be completely eliminated
- Threshold calculated from FFT spectrum + Psycho-Acoustic model
Subband Processing

- **Sample Rates: Input @ 48 kHz, Subbands @ 1.5 kHz**
  - Total number of samples stays the same
- **Noise and speech spectra are roughly flat within a sideband**
- **All bandpass filters are 750 Hz wide**
  - Efficient to implement
Use FFT to calculate a masking level for each subband
Subband Coding

• Scale Factor calculated for every 8 ms
  - Scale factor = max absolute signal value
  - Samples are divided by scale factor before quantization
  - 3 scale factors per 24 ms - quantized to 6 bits each
  - omit scale factors 2 and/or 3 according to how similar they are
    • need 2 bits to say what the choice is.

• Bit Allocation determined for entire 24 ms
  - Choose bits per sample for each subband:
    • < 2.25 kHz: 16 choices: 0, 1.7, 3, 4, ..., 14, 15, 16
    • < 8.25 kHz: 16 choices: 0, 1.7, 2.3, 3, 3.3, 4, 5, ..., 12, 13, 16
    • < 17.25 kHz: 8 choices: 0, 1.7, 2.3, 3, 3.3, 4, 5, 16
    • < 20.25 kHz: 4 choices: 0, 1.7, 2.3, 16
  - \( n \) bits gives SNR of \( 6n + 1.6 \) dB
  - Subbands with 0 bits need no scale factors (save up to 18 bits)
Bit Allocation Procedure

• Aim: Maximize the minimum (over all subbands) mask – to – quantization noise ratio
  – If this ratio is > 1 then quantization noise inaudible

• Method
  1. Initialize bit allocation to 0 for each subband
  2. Find the worst subband
  3. Give it an extra bit (or fraction of a bit)
  4. Go back to step 2.
  5. Stop when all available bits are used up
Channel Coding

- Energy Dispersal randomizes carrier phases
- Convolution Code adds protection
- Time interleaving combats burst errors
- Freq interleaving combats freq selective fading
- Inverse DFT converts phases into a waveform
Energy Dispersal

- If carrier phase changes linearly with frequency then IDFT gives a single impulse
  - Bad news for the transmitter

- Solution:
  - XOR data bits with a pseudo-random sequence
  - Generator polynomial: \( P(X) = X^9 + X^5 + 1 \)
  - Reset shift register at start of each 24 ms frame

9-bit shift register + two XNOR gates
Convolution Coding

- Four separate convolution codes with constraint length of 7
  - $M$ bits $\rightarrow 4(M+6)$ bits

- Mother code has rate $\frac{1}{4}$
- Extra 6 bits from emptying the shift register
- Reset Shift-Reg every 24 ms

Output: $x_{0,1} \ x_{1,1} \ x_{2,1} \ x_{3,1} \ x_{0,2} \ x_{1,2} \ \cdots \ x_{3,M+5} \ x_{0,M+6} \ x_{1,M+6} \ x_{2,M+6} \ x_{3,M+6}$
Puncturing

- Not all \(4M+6\) bits are transmitted
- Predefined puncturing patterns. **Examples:**
  - **Rate 1/3 code**
    - 8 input bits → 32 mother → 24 transmitted
    - Transmit: ✓✓✓✓ ✓✓✓✓ ✓✓✓✓ ✓✓✓✓ ✓✓✓✓ ✓✓✓✓
  - **Rate 2/3 code**
    - 8 input bits → 32 mother → 12 transmitted
    - Transmit: ✓✓✓✓ ✓✓✓✓ ✓✓✓✓ ✓✓✓✓ ✓✓✓✓ ✓✓✓✓
- Code for each service defined in FIC
  - FIC itself always uses rate 1/3 code
Unequal Error Protection

• Some audio code bits are much more critical than others
  – e.g. bit allocation, scale factors, samples, text

• Predefined unequal protection rates

• Example: 128kbps UEP level 3

3072 bits / 24 ms = 128 kbps

352 | 704 | 1920 | 96

1056 (0.33) | 1408 (0.5) | 3360 (0.57) | 216 (0.44)

6040 bits / 24 ms (0.51)
**Time Interleaving**

- Makes data robust to burst errors
- Delay each bit by between 0 and 15 CIF frames
  - Delay between 0 and 360 ms
  - Imposes a coding delay of at least 360 ms
  - Requires memory in the receiver
- Delays of adjacent bits differ by $\geq 4 \times 24$ ms
  - Adjacent bits are always in different transmission frames
- Not used for Fast Information Channel
Frequency Interleaving

- The first 1536 bits of the symbol are assigned to carriers in a pseudo random sequence (same for all symbols).
- The next 1536 bits use the same sequence.
- Each carrier gets 2 bits (0 Hz carrier is not used).
- Prevents fading causing burst errors.

2 sync + 75 data symbols = 96 ms

Data for one symbol = 3072 bits

1536 carriers (excluding 0 Hz)

-768 kHz 0 +768 kHz
DQPSK Modulation

• If $x_k, y_k \in \{0,1\}$ are the bits that map onto carrier $k$, then the complex amplitude for symbol $n$ of a transmission frame is:

$$a_k(n) = a_k(n-1) \times (1 - 2x_k + (1 - 2y_k)j) / \sqrt{2}$$

• All carriers have constant $|a_k(n)|=1$

• Called $\frac{1}{4}\pi$ DQPSK because phase increment is an odd multiple of $\frac{1}{4}\pi$

• Worst case discontinuity at symbol boundary is 1.71 (instead of 2 for plain DQPSK without the $\frac{1}{4}\pi$)
Transmitter Output

- 1537 carriers padded with zeros to 2048 for efficient IFFT
- cyclic prefix added to complex IFFT output
- Real/Imag parts modulate $\cos(\omega t)$ and $\sin(\omega t)$
- Bandpass filtered to remove sidelobes
  - $-71$ dB bandwidth = 1.94 MHz
- DC carrier unused – difficult to control phase
DAB Receiver

Bandpass filters have 1.537 MHz bandwidth

Bosch D-FIRE design
Receiver Downsampling

- ADC sample freq of 24.6 MHz aliases 30.72 MHz to 6.144 MHz
- Complex band-pass filter passes 6.144 MHz ±768 kHz DAB channel
- Downsample by 12 to give sample rate of 2.048 MHz with DAB channel aliased down to DC
- Lowpass filter energy to detect Null symbol every 96 ms
Downsampling

- Analog filter removes images at 12 MHz spacing
- Digital sampling @ 24 MHz $\Rightarrow$ 30 MHz aliased to 6 MHz
- Complex digital filter removes images at 2 MHz spacing
  - negative frequencies removed completely
  - use polyphase filter and combine with downsampling
- Downsampling to 2 MHz $\Rightarrow$ 6 MHz aliased to DC
Receiver Demodulation

- **Time-sync** selects a 1 ms window for the FFT
- **Freq-sync** multiplies complex signal by $e^{j\omega_{\text{offset}}t}$ to correct for tuning errors and doppler shifts
- **CORDIC** block calculates phase and amplitude for DQPSK demodulation
Channel Decoding

• Must reverse the channel coding:

- FFT (and later processing) need only process symbols that contain the wanted service
- Time de-interleaving requires time delays of up to 15 CIF frames (24 ms each) – needs memory
Viterbi Decoding

- Mother code is $\frac{1}{4}$ rate
  - Punctured before transmission
- Receiver must unpuncture to restore original mother code but with “unknown” bits
- Constraint length of 7 $\Rightarrow$ trellis has $2^6 = 64$ states
- Branch metric compares input 4 bit sequence with “correct” value:
  - Cost for each bit = -1 if correct, +1 if wrong, 0 if punctured
- Delay decisions for $\approx 32$ bits
- Re-encode and compare with input to estimate BER

See Data Comms lecture 19
Soft Decisions

• “hard decision” decoder uses branch metric of $\pm 1$
• Ideal Branch metric is $\log \left( prob(z | x) \right)$
  – $z$ is observed bit, $x$ is “correct” bit
  – adding and/or multiplying by a constant makes no difference
• Can calculate ideal metric if you know the noise characteristics:
  – Flat Rayleigh fading with complex FFT output $s_n$
  – Ideal branch metrics for the two QPSK bits are
    $$\pm \mathcal{R}\left(s_n s_{n-1}^*\right) \text{ and } \pm \mathcal{I}\left(s_n s_{n-1}^*\right)$$
  – Use a 4-bit signed number to represent this
Error Concealment

- Errors in MP2 bit allocation bits or high bits of scale factors are catastrophic
  - CRC check words are included in the MP2 bit stream
  - If these are wrong then sound is muted
Synchronization Requirements

• The 48 kHz audio sample clocks must be identical in transmitter and receiver (long term average)
  – otherwise receiver will have too many/few samples
• At the input to the FFT, the carrier frequencies must be integer multiples of 1 kHz (= sample freq/2048)
  – otherwise the carriers will not be orthogonal
  – carrier frequencies are altered by doppler shifts
• The FFT processing window must be timed to make the most constructive use of multipaths
  – In practice the FFT window aims to start at the end of the cyclic prefix of the strongest received signal
Frame Synchronization

- Detect Null symbols by low energy in 1 ms
- \((96\text{ ms})^{-1} = 10.417\text{ Hz} \times 2359296 = 24.576\text{ MHz}\)
  - Exact frequency multiplication is done using a phase lock loop
  - 24.576 MHz is the ADC sample clock
- \(24.576\text{ MHz} \div 512 = 48\text{ kHz}\) audio sample clock
- Also finds approximate start of first symbol
Effect of Frequency Offsets

- FFT samples spectrum at multiples of 1 kHz
- If carrier frequencies have an offset, $\Delta f$, then they will no longer be orthogonal
  - must measure $\Delta f$ and compensate
  - compensation can be combined with the FFT
- Divide $\Delta f$ into integer and fractional multiple of 1 kHz
  - Integer part $\Rightarrow$ wrong carriers
  - Fractional part $\Rightarrow$ inter carrier interference
Fine AFC

- Frequency error of $\Delta f \Rightarrow$ additional phase shift between successive symbols of $\Delta \phi = 2\pi \Delta f T$
  - $T$ is symbol period = 1.246 ms
- Phase shift for each carrier should be $(\frac{1}{4} + \frac{1}{2}k)\pi$ in the absence of noise
  - Find deviation from nearest correct value
  - Form energy-weighted average of phase error over all carriers
  - calculate $\Delta f$
  - apply correction before or during FFT calculation by multiplying input signal by $\exp(-2\pi j \Delta f)$

- Only works to within a multiple of $\frac{1}{4}\pi$
Coarse AFC

- Transmitted phases of phase reference symbol carriers are known:
  - Subtract transmitted phases from FFT output and do inverse FFT
  - Try lots of values of $\Delta f$ in the range $\pm 8$ kHz or so
    - Subtract phases due to $\Delta f$
    - Result is impulse response of channel
  - Pick $\Delta f$ that gives the highest peak
    - Position of peak indicates where to put the end of the cyclic prefix
Benefits of DAB

- CD quality
- Mobile reception
- Spectral Efficiency
- European Standardisation
- Data as well as Audio
- Lower transmitter power
- Receiver features
  - easy tuning
  - pause