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.
2. W. Hoeg and Thomas Lauterbach. "Digital Audio Broadcasting: Principles and Applications of Digital Radio". John Wiley, 2003.
3. C. Gandy. "DAB: an introduction to the Eureka DAB System and a guide to how it works". Technical Report WHP-061, British Broadcasting Corp, June 2003.
4. M. Bolle, D. Clawin, K. Gieske, F. Hofmann, T. Mlasko, M.J. Ruf, and G. Spreitz. "The receiver engine chip-set for digital audio broadcasting". In Intl Symp on Signals Systems and Electronics, pages 338–342, October 1998.
5. K. Taura, M. Tsujishita, M. Takeda, H. Kato, M. Ishida, and Y. Ishida. "A digital audio broadcasting (DAB) receiver". IEEE Trans Consumer Electronics, 42(3):322–327, August 1996.

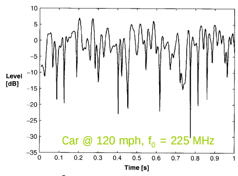
History of DAB

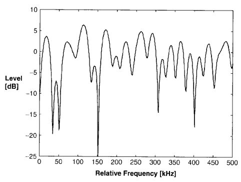
- 1986 – DAB consortium formed
  - France, Germany, Netherlands, UK
  - Eureka 147 development project
- 1990 – First trial broadcasts

- 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

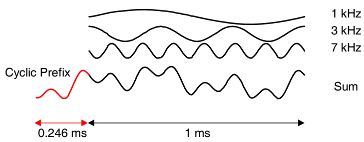




- Interference
  - from equipment, vehicles and other radio stations

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)



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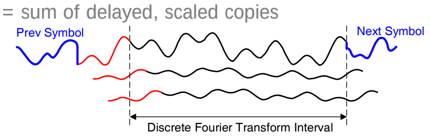
Cyclic Prefix

- Convolution with channel impulse response
  = 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

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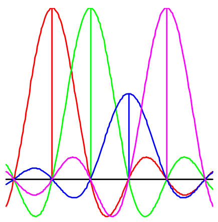
Frequency Domain Orthogonality

- 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:

$$\text{ICI power} \approx 0.5\left(\frac{vf_0T}{c}\right)^2 P_0 = -14\ \text{dB at } 190\ \text{km/h}$$

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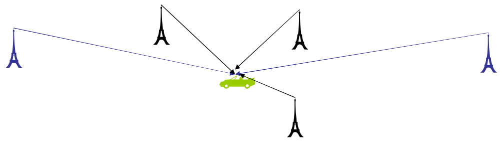
Single Frequency Network

- All transmitters send an identical signal
- Interference-free if delay + multipath < 0.246 ms relative to nearest transmitter
- Optimal spacing ≈ c × 0.246 ms = 74 km



- Transmitters further than ≈ 1.2 c × 0.246 ms do more harm than good

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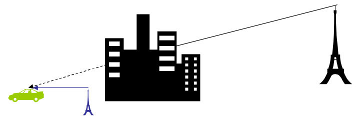
Fill-in transmitter

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



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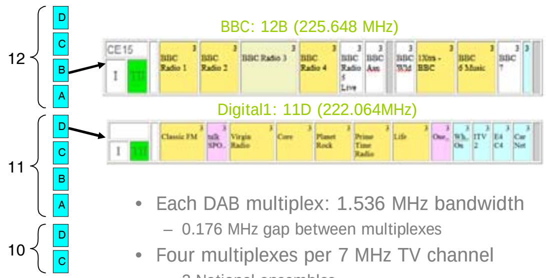
UK DAB Multiplexes/Ensembles



- 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)

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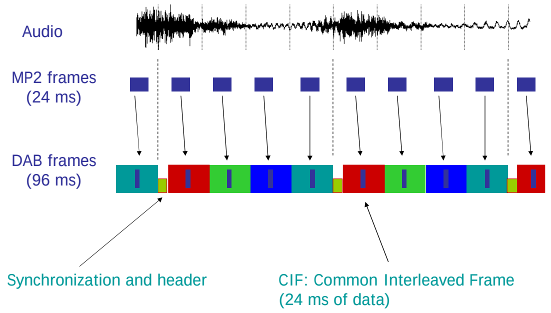
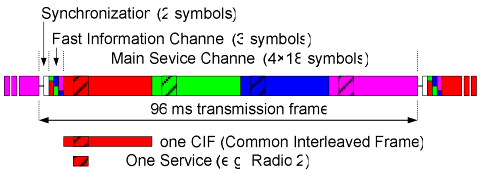
Spectral Efficiency

- 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



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

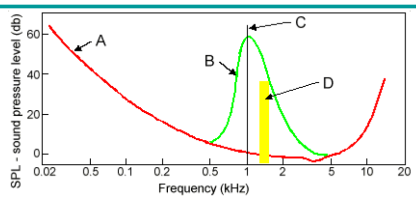
System Parameters

- Centre frequency, $f_0 \approx 220$ MHz (Band III)
  - Wavelength > 1 m ⇒ diffraction around objects
  - Lower frequencies are full up
- Total bandwidth = 1.537 MHz
  - Needs to be > 1.5 MHz for fading to be frequency selective
  - < 1.6 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 ⇔ Useful symbol length = 1 ms
  - Symbol length < 0.4/(Doppler spread) $\approx 0.4c/(f_0 v)$ = 10 ms
  - Symbol length < cyclic prefix for efficiency
- Transmission frame = 76 symbols = 96 ms
  - Long for efficiency, short for ease of synchronization

Source Coding

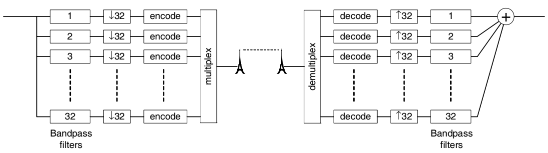
- Based on MP2 (MPEG 1 Layer 2)
  - Simpler than MP3 but less good
- Masking – Psycho-acoustic model
  - loud sounds make quieter sounds inaudible at nearby frequencies and times
- Sub-band Processing
  - Input @ 48 kHz sample rate
  - Divide into 32 subbands of 750 Hz @ 1.5 kHz
    - 36 samples/subband in each 24 ms CIF frame
    - Only low 27 subbands are used (0 to 20.25 kHz)

Masking



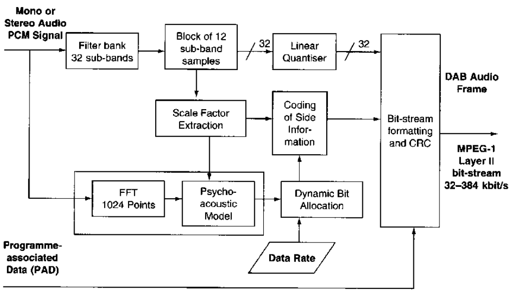
- Normal hearing threshold is A
- Threshold is changed to B because of tone C
- Higher quantization noise allowed in bands near tone C
- Band D can be completely eliminated
- Threshold calculated from FFT spectrum + Psycho-Acoustic model

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
  - efficent to implement

Encoder Block Diagram



Use FFT to calculate a masking level for each subband

Subband Coding

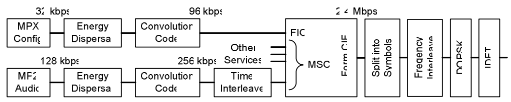
- Scale Factor calculated for every 8 ms
  - Scale factor = max absolute signal value
  - Samples are divided by scale factor before quantization
  - 3 scale factors per 24 ms - quantized to 6 bits each
  - omit scale factors 2 and/or 3 according to how similar they are
    - need 2 bits to say what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

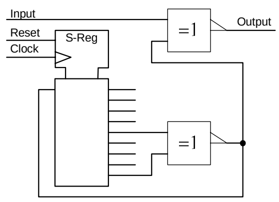
Channel Coding

- 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

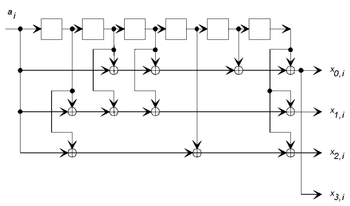


9-bit shift register + two XNOR gates

Convolution Coding

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

- Mother code has rate ¼
- Extra 6 bits from emptying the shift register
- Reset Shift-Reg every 24 ms



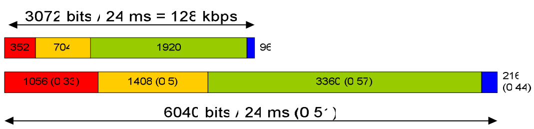
Output: $x_{0,1}\ x_{1,1}\ x_{2,1}\ x_{3,1}\ x_{0,2}\ x_{1,2} \cdots x_{3,M+5}\ x_{0,M+6}\ x_{1,M+6}\ x_{2,M+6}\ x_{3,M+6}$

Puncturing

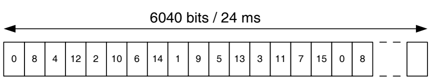
- Not all $4M+6$ bits are transmitted
- Predefined puncturing patterns. Examples:
  - Rate 1/3 code
    - 8 input bits → 32 mother → 24 transmitted
    - Transmit: ✓✓✓× ✓✓✓× ✓✓✓× ✓✓✓× ✓✓✓× ✓✓✓× ✓✓✓× ✓✓✓×
  - Rate 2/3 code
    - 8 input bits → 32 mother → 12 transmitted
    - Transmit: ✓✓×× ✓××× ✓✓×× ✓××× ✓✓×× ✓×××
- 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

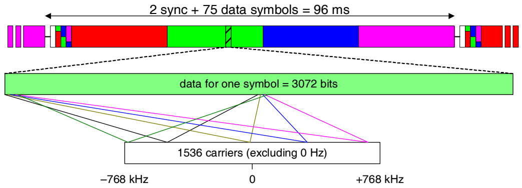


Time Interleaving



- 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 o 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

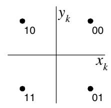


- The first 1536 bits of the symbol are assigned to carriers in a pseudo random sequence (same for all symbols).
- The next 1536 bits use the same sequence.
- Each carrier gets 2 bits (0 Hz carrier is not used)
- Prevents fading causing burst errors

DQPSK Modulation

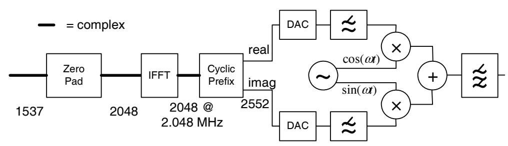
- If $x_k, y_k \in \{0,1\}$ are the bits that map onto carrier k, then the complex amplitude for symbol n of a transmission frame is:

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

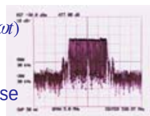


- All carriers have constant $|a_k(n)|=1$
- Called ¼π DQPSK because phase increment is an odd multiple of ¼π
- Worst case discontinuity at symbol boundary is 1.71 (instead of 2 for plain DQPSK without the ¼π)

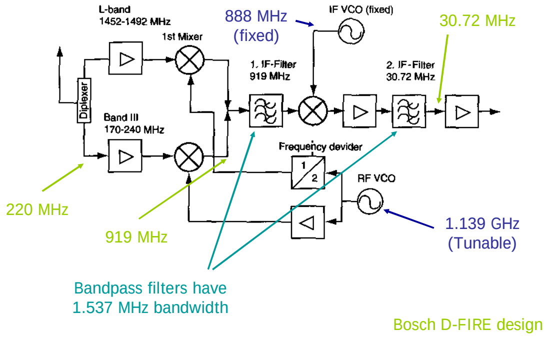
Transmitter Output



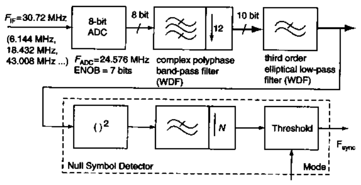
- 1537 carriers padded with zeros to 2048 for efficient IFFT
- cyclic prefix added to complex IFFT output
- Real/Imag parts modulate $\cos(\omega t)$ and $\sin(\omega t)$
- Bandpass filtered to remove sidelobes
  - –71 dB bandwidth = 1.94 MHz
- DC carrier unused – difficult to control phase



DAB Receiver

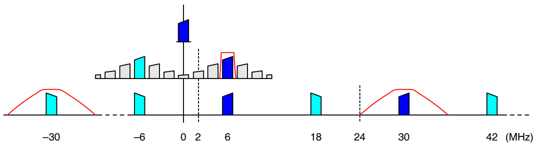


Receiver Downsampling



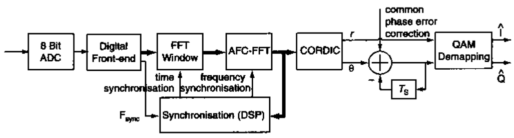
- ADC sample freq of 24.6 MHz aliases 30.72 MHz to 6.144 MHz
- Complex band-pass filter passes 6.144 MHz $\pm$768 kHz DAB channel
- Downsample by 12 to give sample rate of 2.048 MHz with DAB channel aliased down to DC
- Lowpass filter energy to detect Null symbol every 96 ms

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
- Downsample to 2 MHz $\Rightarrow$ 6 MHz aliased to DC

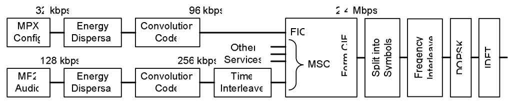
Receiver Demodulation



- 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

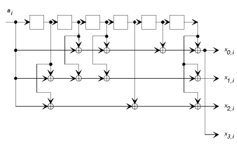
Channel Decoding

- 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 ¼ 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

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Soft Decisions

- "hard decision" decoder uses branch metric of ±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

$$\pm\Re\left(s_n s_{n-1}^*\right) \text{ and } \pm\Im\left(s_n s_{n-1}^*\right)$$

  - Use a 4-bit signed number to represent this

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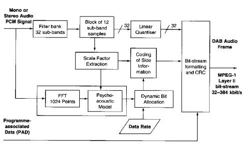
Error Concealment

- 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



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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

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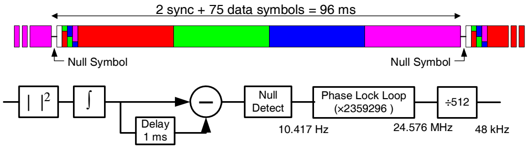
Frame Synchronization



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

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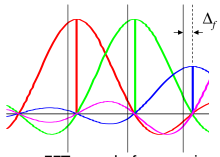
Effect of Frequency Offsets

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



FFT sample frequencies

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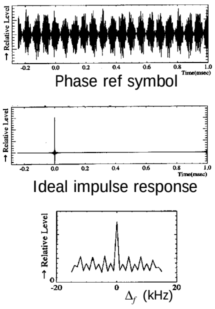
Fine AFC

- 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 $(¼+½k)\pi$ in the absence of noise
  - Find deviation from nearest correct value
  - Form energy-weighted average of phase error over all carriers
  - calculate Δf
  - apply correction before or during FFT calculation by multiplying input signal by by $\exp(-2\pi j\Delta_f t)$
- Only works to within a multiple of $¼\pi$

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 Δ_f in the range ±8 kHz or so
    - Subtract phases due to Δ_f
    - Result is impulse response of channel
  - Pick Δ_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