DAB

Digital Audio Broadcasting

DAB history

DAB has been under development since 1981 at the Institut für Rundfunktechnik (IRT).

In 1985 the first DAB demonstrations were held at the WARC-ORB in Geneva and in 1988 the first DAB transmissions were made in Germany.
Later DAB (or Eureka-147) was developed as a research project for the European Union (Eureka project number EU147), which started in 1987 on initiative by a consortium formed in 1986. The MP2 (MPEG-1 layer-2) audio coding technique was created as part of the EU147 project. DAB was the first standard based on orthogonal frequency division multiplexing (OFDM) modulation technique, which since then has become one of the most popular transmission schemes for modern wideband digital communication systems.

A choice of audio codec, modulation and error-correction coding schemes and first trial broadcasts were made in 1990. Public demonstrations were made in 1993 in the United Kingdom. The protocol specification was finalized in 1993 and adopted by the ITU-R standardization body in 1994, the European community in 1995 and by ETSI in 1997. Pilot broadcasts were launched in several countries in 1995.

The UK was the first country to receive a wide range of radio stations via DAB. Commercial DAB receivers began to be sold in 1999 and over 50 commercial and BBC services were available in London by 2001. The UK has to date been the most successful market for DAB and is being projected to be in 40% of homes by 2009.[5]

By 2006, 500 million people worldwide were in the coverage area of DAB broadcasts, although by this time sales had only taken off in the UK and Denmark. In 2006 there are approximately 1,000 DAB stations in operation world wide.[6]

The standard was coordinated by the European DAB forum, formed in 1995 and reconstituted to the World DAB Forum in 1997, which represents more than 30 countries. In 2006, World DMB Forum took over the coordination.

MP2

- MP2 is a sub-band audio encoder, which means that compression takes place in the time domain. By comparison, MP3 is a transform audio encoder, which means that compression takes place in the frequency domain after transformation from the time domain.

- The MP2 encoder does not exploit interchannel redundancies. This makes MP2 less efficient than MP3 on low bitrates (lower than 256 kbit/s). For example, a 128 kbit/s MP3 encoded audio usually sounds, to the human ear, truer to the original source than the same audio encoded as 192 kbit/s MP2.

- MP2 performs better than MP3 on high bitrates (256 to 384 kbit/s) and is generally more error resilient than MP3, so MP2 is considered optimal, and is the de facto standard, for broadcast applications. Typically, private broadcasters worldwide compress their material at 256kb/s while their counterparts in public broadcasting use 384kb/s.

- Like MP3, MP2 is a perceptual codec, which means that it removes information that the human auditory system will not be able to perceive. To choose which information to remove, the audio signal is analyzed according to a psychoacoustic model, which takes into account the parameters of the human auditory system. Research into psychoacoustics has shown that if there is a strong signal on a certain frequency, then weaker signals at frequencies close to the strong signal's frequency cannot be perceived by the human auditory system. This is called frequency masking. Perceptual audio codecs take advantage of this frequency masking by ignoring information at frequencies that are deemed to be imperceptible, thus allowing more data to be allocated to the reproduction of perceptible frequencies.

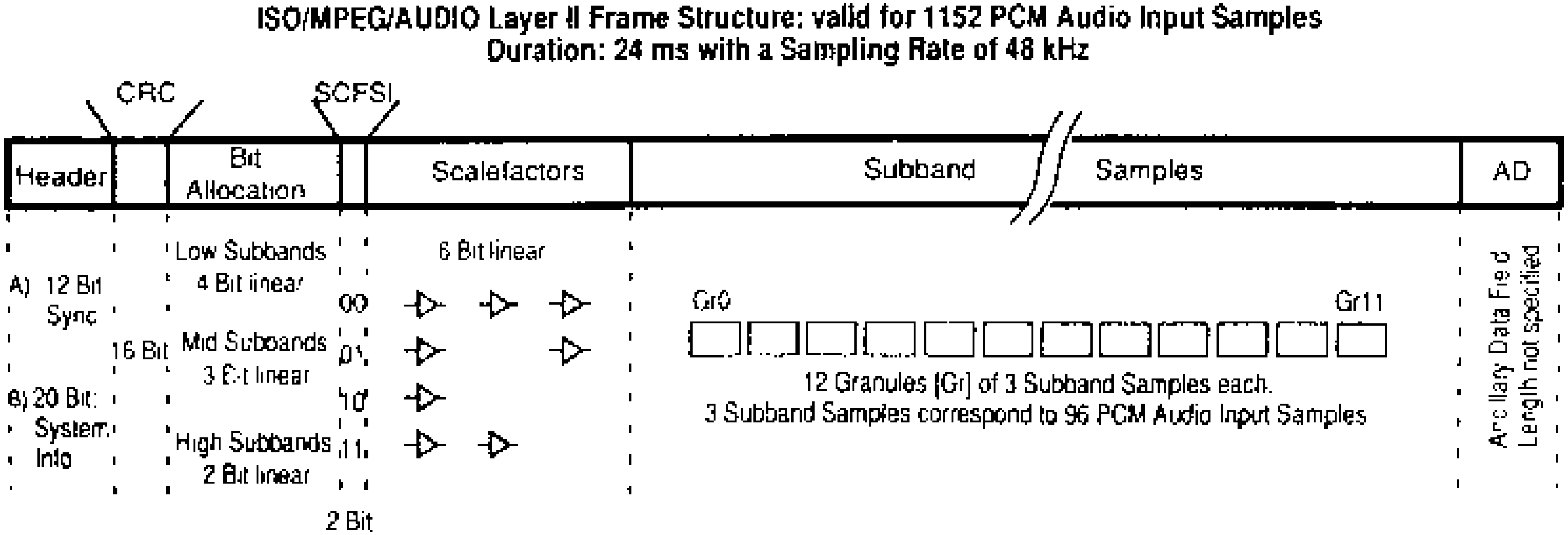
- MP2 splits the input audio signal into 32 sub-bands, and if the audio in a sub-band is deemed to be imperceptible then that sub-band is not transmitted. MP3, on the other hand, transforms the input audio signal to the frequency domain in 576 frequency components. Therefore, MP3 has a far higher frequency resolution than MP2, which allows the psychoacoustic model to be applied more accurately than for MP2. Because the psychoacoustic model can be applied more accurately, MP3 has greater scope to reduce the bit rate, which is why MP3 doesn't need as high a bit rate as MP2 to get an acceptable audio quality.

The Layer 2 time-frequency mapping is a polyphase filter bank with 32 subbands, the subbands are equally spaced in frequency.

The Layer 2 psychoacoustic model uses a 1024-point FFT to get detailed spectral information about the signal. The output of the FFT is used to find both tonal (sinusoidal) and nontonal (noise) maskers in the signal. Each masker produces a masking threshold depending on its frequency, intensity, and tonality. For each subband, the individual masking thresholds are combined to form a global masking threshold. The masking threshold is compared to the maximum signal level for the subband, producing a signal-to-masker ratio (SMR) which is the input to the quantizer.

The Layer 2 quantizer/encoder first examines each subband's samples, finds the maximum absolute value of these samples, and quantizes it to 6 bits. This is called the scale factor for the subband. Then it determines the bit allocation for each subband by minimizing the total noise-to-mask ratio with respect to the bits allocated to each subband. (It's possible for heavily masked subbands to end up with zero bits, so that no samples are encoded.) Finally, the subband samples are linearly quantized to the bit allocation for that subband. Layer 2 allows each subband a sequence of three successive scale factors, and the encoder uses one, two, or all three, depending on how much they differ from each other.

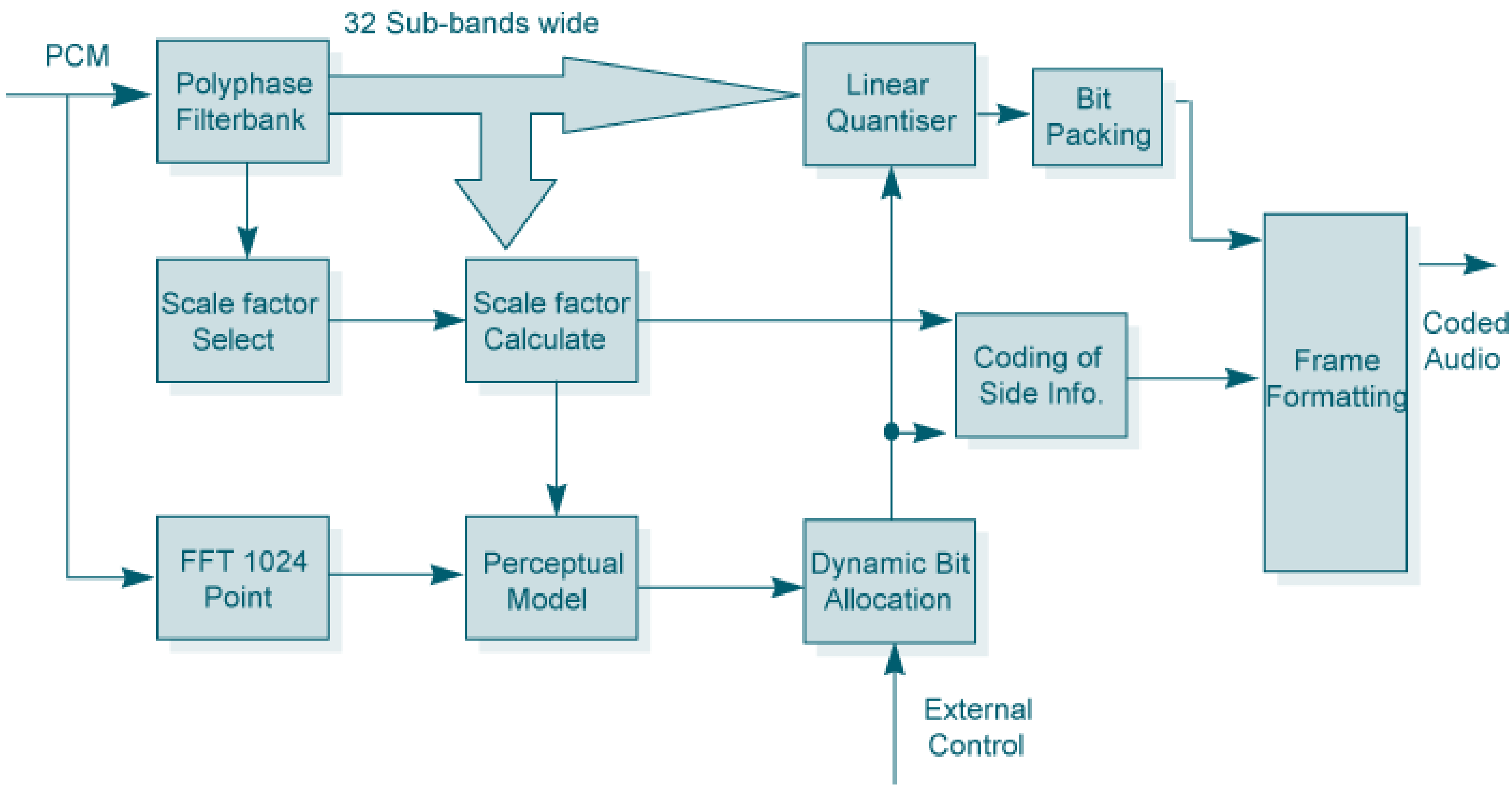
The Layer 2 frame packer includes header and CRC structure. The number of bits used to describe bit allocations varies with subband: 4 bits for the low subbands, 3 bits for the middle subbands, and 2 bits for the high subbands (this follows critical bandwidths). The scale factors (one, two or three depending on the data) are encoded along with a 2-bit code describing which combination of scale factors is being used. The subband samples are quantized according to bit allocation, and then combined into groups of three (called granules). Each granule is encoded with one code word.



Layer 2 processes the input signal in frames of 1152 PCM samples. At 48 kHz, each frame carries 24 ms of sound. Highest quality is achieved with a bit rate of 256k bps, but quality is often good down to 64k bps.

In audio context, MP2 stands for MPEG-1/2 Layer II (see also the MPEG-2 FAQ of the MPEG Audio Subgroup), but this abbreviation is also used for MPEG-2 video files.

This format is currently used in broadcast like satellite, DVB and DAB; and also in legacy VCD and SVCD discs.



Generation of the DAB Signal

You will see in Figure 1 how each service signal is coded individually at source level, error protected and time interleaved in the channel coder. Then the services are multiplexed in the Main Service Channel (MSC), according to a pre-determined, but adjustable, multiplex configuration. The multiplexer output is combined with Multiplex Control and Service information, which travel in the fast Information Channel (FIC), to form the transmission frames in the Transmission Multiplexer FIG 1 Finally, Orthogonal Frequency Division Multiplexing (OFDM) is applied to shape the DAB signal, which consists of a large number of carriers. The signal is then transposed to the appropriate radio frequency band, amplified and transmitted.

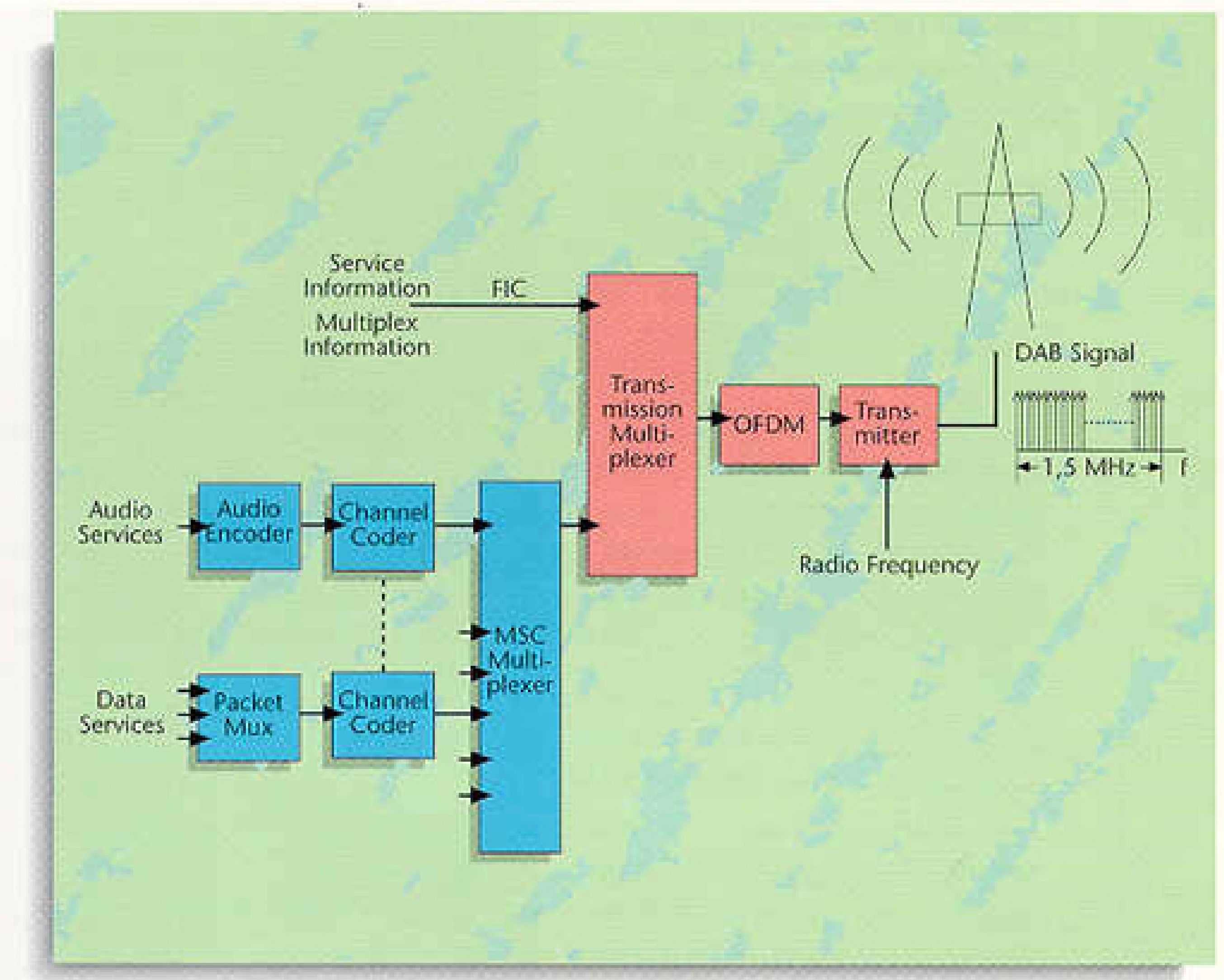


Fig. 1: Generation of the DAB Signal

Reception of a DAB signal

Figure 2 demonstrates a conceptual DAB receiver. The DAB ensemble is selected in the analogue tuner., the digitised output of which is fed to the OFDM demodulator and channel decoder to eliminate transmission errors. The information contained in the FIC is passed to the user interface for service selection and is used to set p the receiver appropriately. The MSC data is further processed in an audio decoder to produce the left and right audio signals or in a data decoder (Packet Demux) as appropriate.

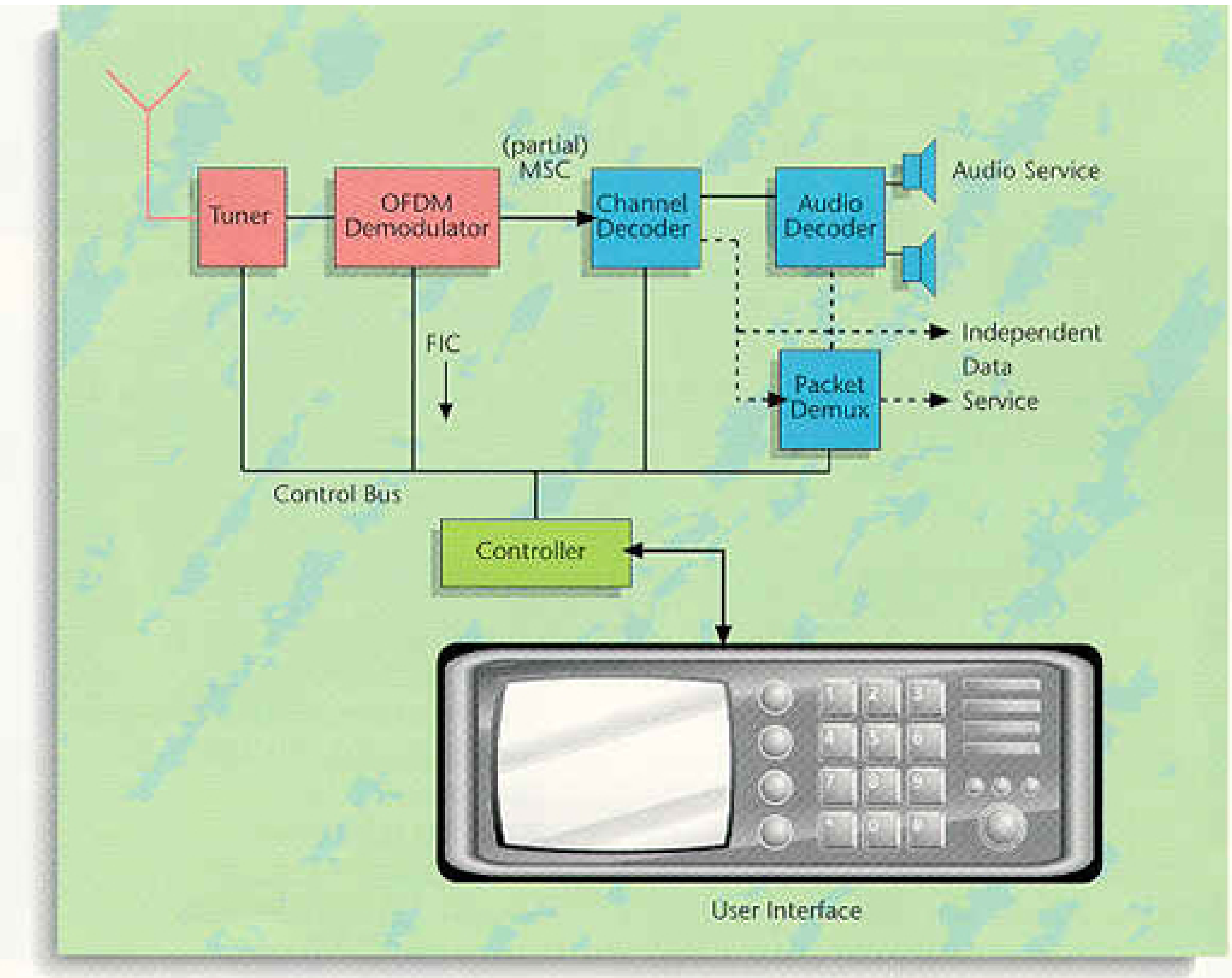


Fig. 2 : Conceptual DAB Receiver

Guard interval for elimination of inter-symbol interference

One key principle of OFDM is that since low symbol rate modulation schemes (i.e. where the symbols are relatively long compared to the channel time characteristics) suffer less from intersymbol interference caused by multipath, it is advantageous to transmit a number of low-rate streams in parallel instead of a single high-rate stream. Since the duration of each symbol is long, it is feasible to insert a guard interval between the OFDM symbols, thus eliminating the intersymbol interference.

The guard-interval also reduces the sensitivity to time synchronization problems.

The cyclic prefix, which is transmitted during the guard interval, consists of the end of the OFDM symbol copied into the guard interval, and the guard interval is transmitted followed by the OFDM symbol. The reason that the guard interval consists of a copy of the end of the OFDM symbol is so that the receiver will integrate over an integer number of sinusoid cycles for each of the multipaths when it performs OFDM demodulation with the FFT.

Although the guard interval only contains redundant data, which means that it reduces the capacity, some OFDM-based systems, such as some of the broadcasting systems, deliberately use a long guard interval in order to allow the transmitters to be spaced farther apart in an SFN, and longer guard intervals allow larger SFN cell-sizes. A rule of thumb for the maximum distance between transmitters in an SFN is equal to the distance a signal travels during the guard interval — for instance, a guard interval of 200 microseconds would allow transmitters to be spaced 60 km apart.

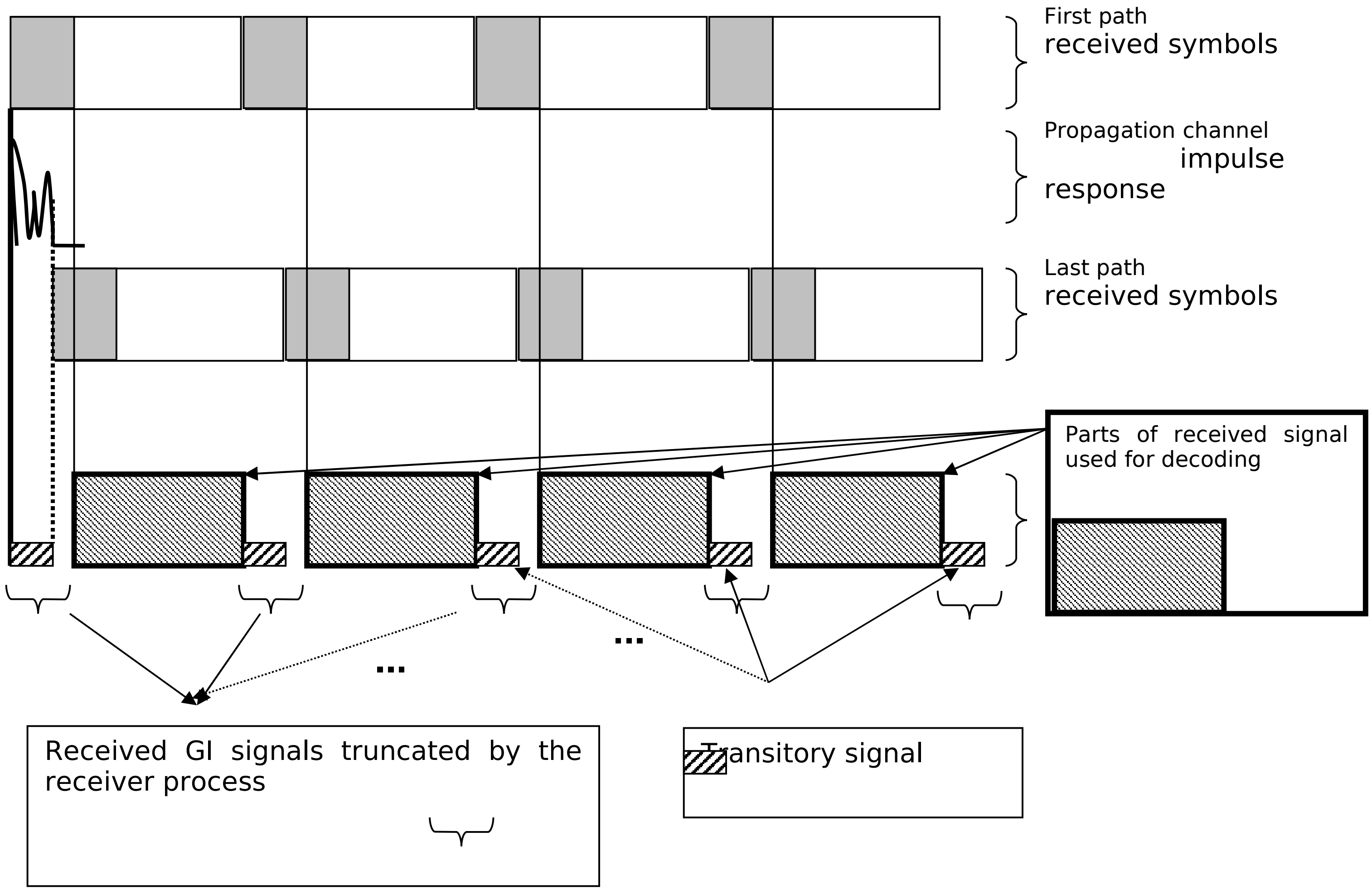
DAB standard: 170 MHz … 230 MHz

L-band.

- 200MHz BAND. Mode I:
- symbols of 1ms useful duration with a guard interval of 0.246 ms.
- 1536 sub-carriers transmitted simultaneously per symbol
- a QPSK code for each sub-carrier
- symbols are organised into frames of 77 symbols.
- the first symbol is a null one (with no-frequency transmitted or only the centre frequency)
- the second symbol is the reference one where all the sub-carriers are transmitted with reference code elements. This symbol is used for the propagation channel estimation.

- A “typical” radiated power for DAB transmitters 1 kW.
- 1.5MHz bandwidth (1536 orthogonal sub-carriers of 1kHz bandwidth each).
- White spectrum !

DAB signal structure



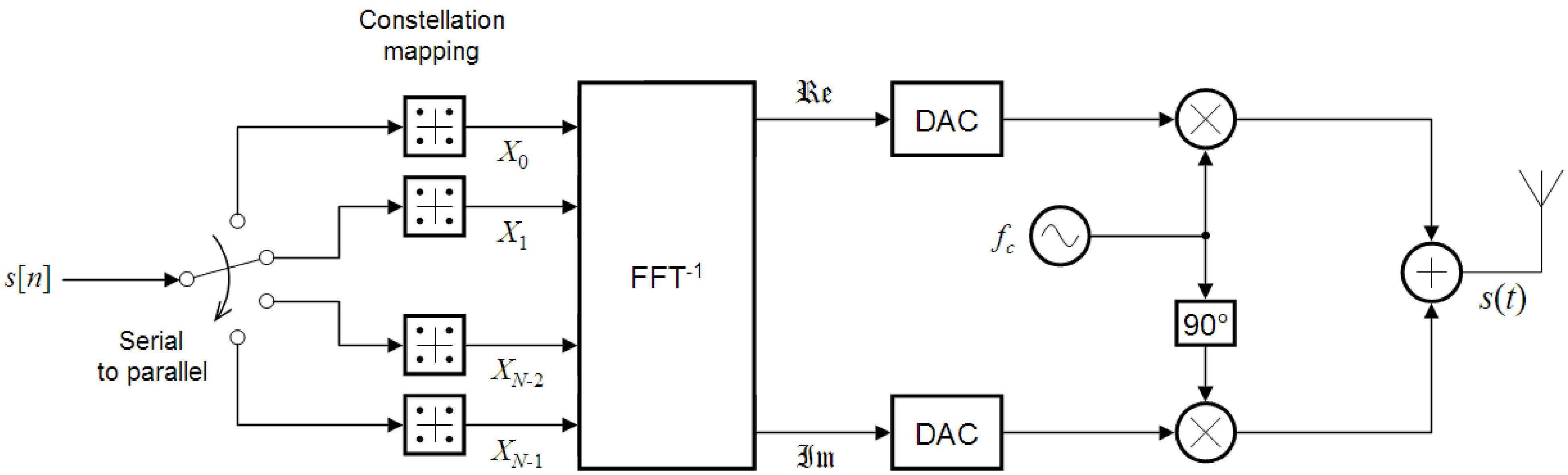
OFDM maths

$$\nu(t) = \sum_{k=0}^{N-1} X_k e^{i2\pi kt/T}, \quad 0 \le t < T,$$

Xk – symbol
k – subcarrier index

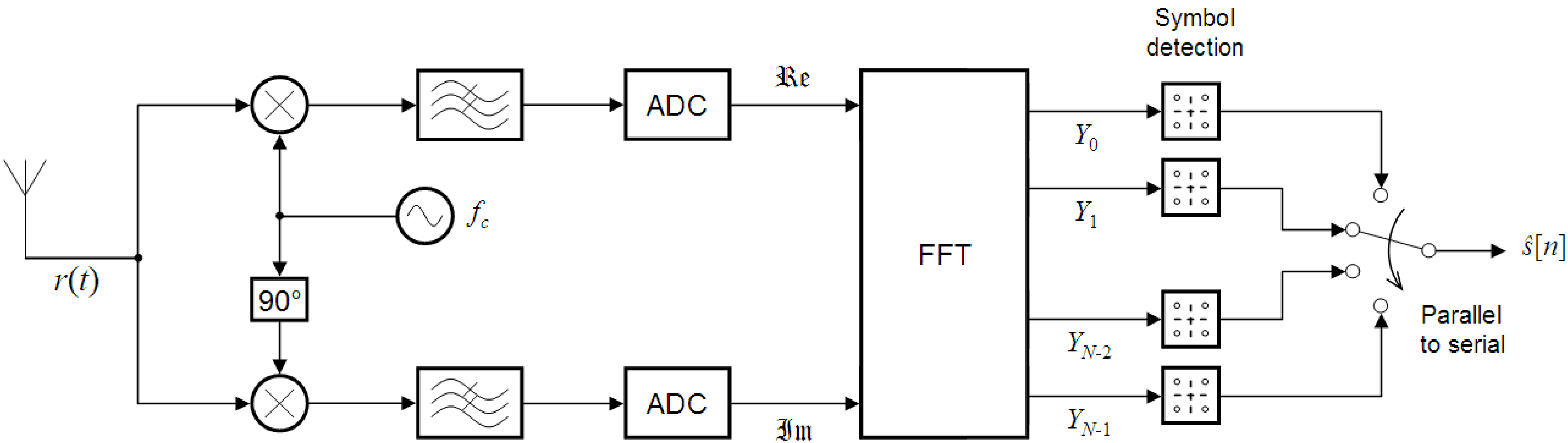
- just a Discrete Fourier Transform....
- subcarriers are orthogonal (recall DFT...) - no interference between subcarriers!

OFDM coder



Constellation mapping
$s[n]$
Serial to parallel
X_0
X_1
X_{N-2}
X_{N-1}
FFT^{-1}
$\Re$e
$\Im$m
DAC
DAC
f_c
90°
$s(t)$

OFDM decoder



Digital radio summary

- MPEG 1 layer II (“MP2”)
- OFDM (det by FFT)
- DPSK (det with conj())
- Multipath
- SFN (single frequency networks)– inteference – fading (not all freqs)
- fading assumed flat over subcarrier
- some subcarriers fading – correction codes