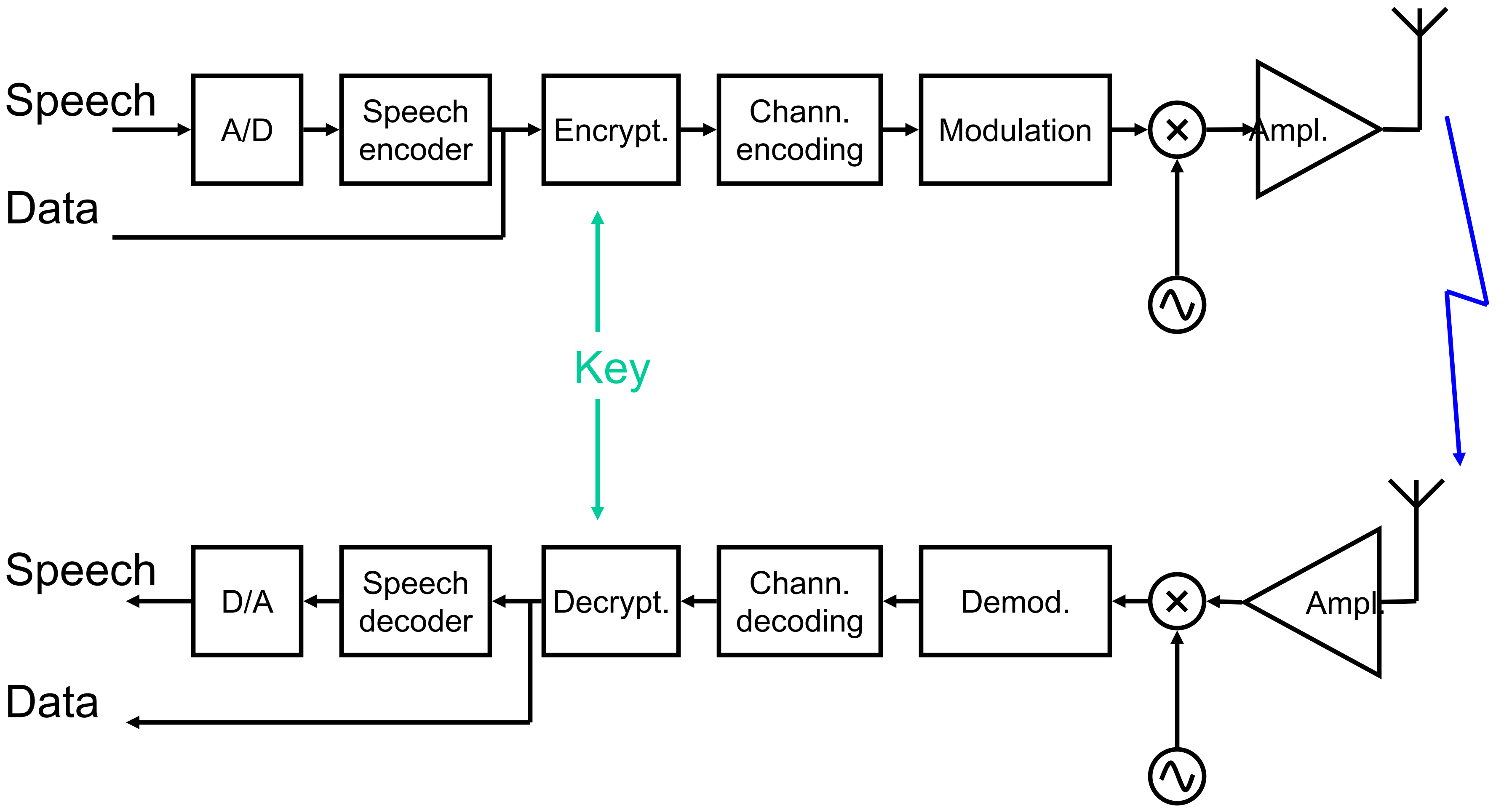
Chapter 10

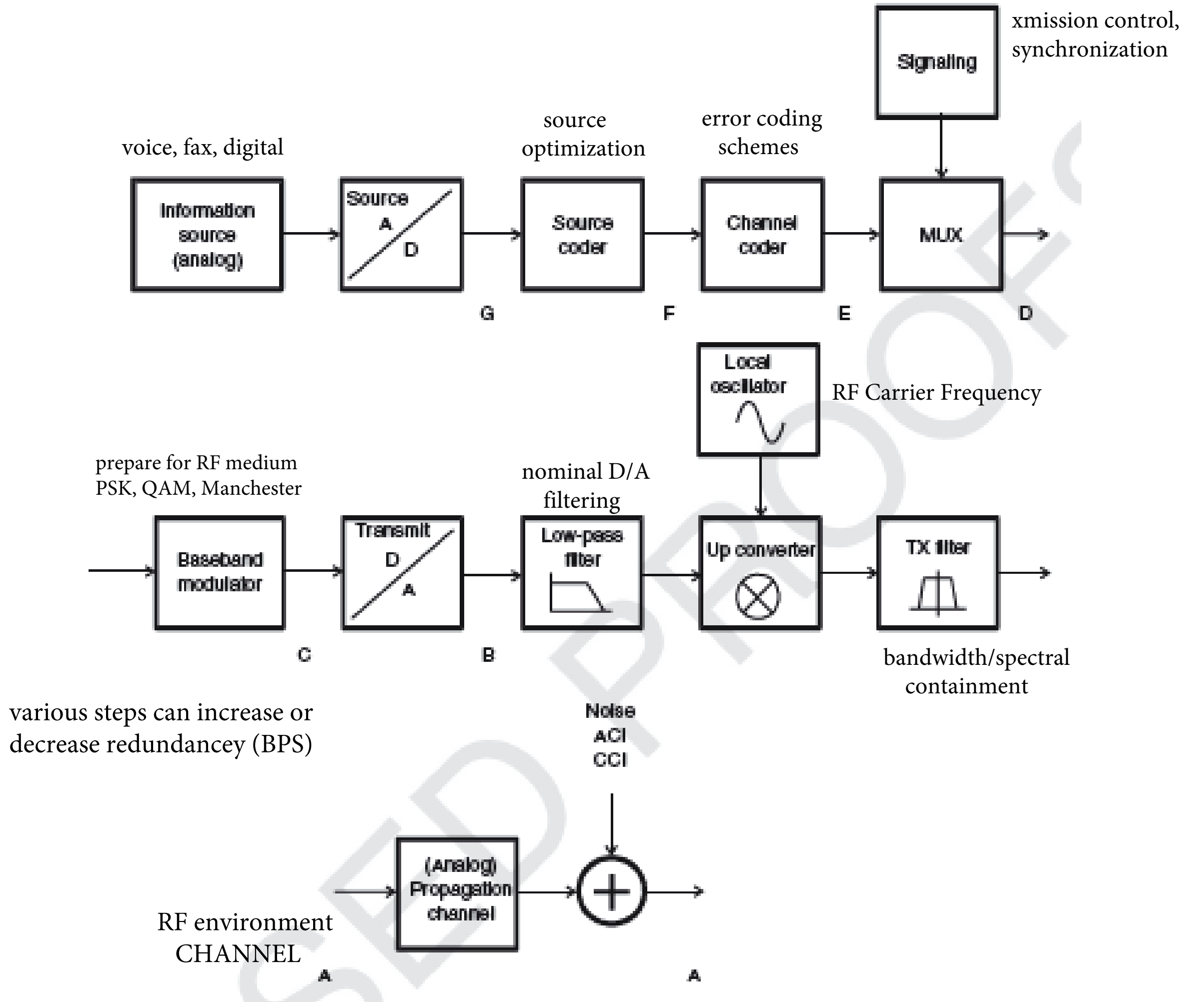
Structure of a wireless communications link

Primary Wireless Problems: Fading (best handled by diversity), Delay Dispersion (multipath handled by equalizers) and BER (handled by coding schemes)

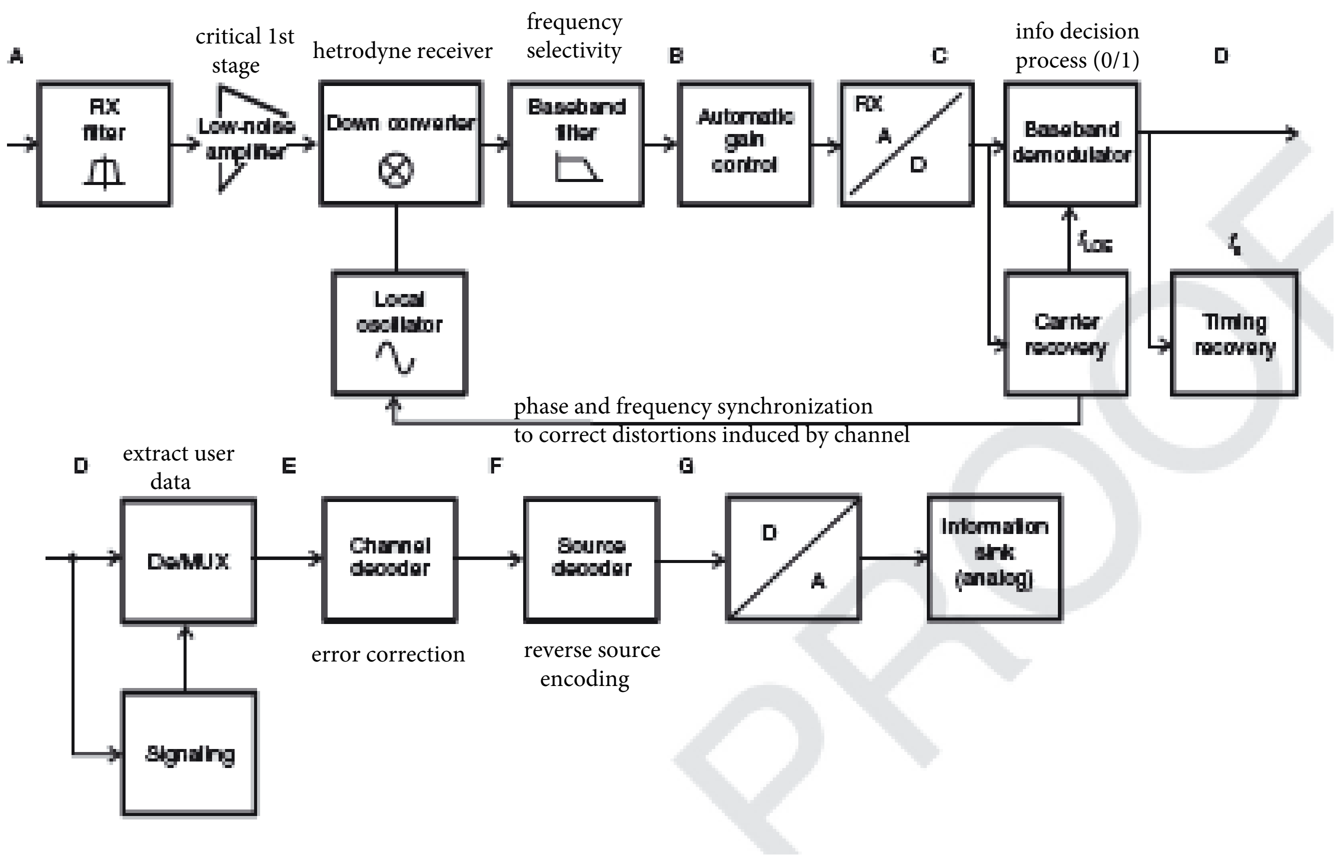
Block diagram



Block diagram transmitter



Block diagram receiver



- The *information source* provides an analog source signal and feeds it into the *source ADC* (Analog to Digital Converter). This ADC first band limits the signal from the analog information source (if necessary), and then converts the signal into a stream of digital data at a certain sampling rate and resolution (number of bits per sample). For example, speech would typically be sampled at 8 ksamples/s, with 8-bit resolution, resulting in a datastream at 64 kbit/s. For the transmission of digital data, these steps can be omitted, and the digital source directly provides the input to interface "G" in Figure 10.2.
- The *source coder* uses a priori information on the properties of the source data in order to reduce redundancy in the source signal. This reduces the amount of source data to be transmitted, and thus the required transmission time and/or bandwidth. For example, the Global System for Mobile communications (GSM) speech coder reduces the source data rate from 64 kbit/s mentioned above to 13 kbit/s. Similar reductions are possible for music and video (MPEG standards). Also, fax information can be compressed significantly. One thousand subsequent symbols "00" (representing "white" color), which have to be represented by 2,000 bits, can be replaced by the statement: "what follows now are 1,000 symbols 00," which requires only 12 bits. For a typical fax, compression by a factor of 10 can be achieved. The source coder increases the entropy (information per bit) of the data at interface F; as a consequence, bit errors have greater impact. For some applications, source data are *encrypted* in order to prevent unauthorized listening in.
- The *channel coder* adds redundancy in order to protect data against transmission errors. This increases the data rate that has to be transmitted at interface E – e.g., GSM channel coding increases the data rate from 13 to 22.8 kbit/s. Channel coders often use information about the statistics of error sources in the channel (noise power, interference statistics) to design codes that are especially well suited for certain types of channels (e.g., Reed–Solomon codes protect especially well against burst errors). Data can be sorted according to importance; more important bits then get stronger protection. Furthermore, it is possible to use interleaving to break up error bursts; note that interleaving is mainly effective if it is combined with channel coding.
- *Signaling* adds control information for the establishing and ending of connections, for associating information with the correct users, synchronization, etc. Signaling information is usually strongly protected by error correction codes.
- The *multiplexer* combines user data and signaling information, and combines the data from multiple users.[2] If this is done by time multiplexing, the multiplexing requires some time compression.

In GSM, multiaccess multiplexing increases the data rate from 22.8 to 182.4 kbit/s ($8 \cdot 22.8$) for the standard case of eight participants. The addition of signaling information increases the data rate to 271 kbit/s.

- The *baseband modulator* assigns the gross data bits (user data and signaling at interface D) to complex transmit symbols in the baseband. Spectral properties, intersymbol interference, peak-to-average ratio, and other properties of the transmit signal are determined by this step. The output from the baseband modulator (interface C) provides the transmit symbols in oversampled form, discrete in time and amplitude.

  Oversampling and quantization determine the aliasing and quantization noise. Therefore, high resolution is desirable, and the data rate at the output of the baseband modulator should be much higher than at the input. For a GSM system, an oversampling factor of 16 and 8-bit amplitude resolution result in a data rate of about 70 Mbit/s.

- The *TX Digital to Analog Converter* (DAC) generates a pair of analog, discrete amplitude voltages corresponding to the real and imaginary part of the transmit symbols, respectively.
- The *analog low-pass filter* in the TX eliminates the (inevitable) spectral components outside the desired transmission bandwidth. These components are created by the out-of-band emission of an (ideal) baseband modulator, which stem from the properties of the chosen modulation format. Furthermore, imperfections of the baseband modulator and imperfections of the DAC lead to additional spurious emissions that have to be suppressed by the TX filter.
- The *TX Local Oscillator* (LO) provides an unmodulated sinusoidal signal, corresponding to one of the admissible center frequencies of the considered system. The requirements for frequency stability, phase noise, and switching speed between different frequencies depend on the modulation and multiaccess method.
- The *upconverter* converts the analog, filtered baseband signal to a passband signal by mixing it with the LO signal. Upconversion can occur in a single step, or in several steps. Finally, amplification in the Radio Frequency (RF) domain is required.
- The *RF TX filter* eliminates out-of-band emissions in the RF domain. Even if the low-pass filter succeeded in eliminating all out-of-band emissions, upconversion can lead to the creation of additional out-of-band components. Especially, nonlinearities of mixers and amplifiers lead to intermodulation products and "spectral regrowth" – i.e., creation of additional out-of-band emissions.

- The *(analog) propagation channel* attenuates the signal, and leads to delay and frequency dispersion. Furthermore, the environment adds noise (Additive White Gaussian Noise – AWGN) and co-channel interference.
- The *RX filter* performs a rough selection of the received band. The bandwidth of the filter corresponds to the total bandwidth assigned to a specific service, and can thus cover multiple communications channels belonging to the same service.
- The *low-noise amplifier* amplifies the signal, so that the noise added by later components of the RX chain has less effect on the Signal-to-Noise Ratio (SNR). Further amplification occurs in the subsequent steps of downconversion.
- The *RX LO* provides sinusoidal signals corresponding to possible signals at the TX LO. The frequency of the LO can be fine-tuned by a carrier recovery algorithm (see below), to make sure that the LOs at the TX and the RX produce oscillations with the same frequency and phase.
- The *RX downconverter* converts the received signal (in one or several steps) into baseband. In baseband, the signal is thus available as a complex analog signal.
- The *RX low-pass filter* provides a selection of desired frequency bands for one specific user (in contrast to the RX bandpass filter that selects the frequency range in which the service operates). It eliminates adjacent channel interference as well as noise. The filter should influence the desired signal as little as possible.
- The *Automatic Gain Control* (AGC) amplifies the signal such that its level is well adjusted to the quantization at the subsequent ADC.
- The RX *ADC* converts the analog signal into values that are discrete in time and amplitude. The required resolution of the ADC is determined essentially by the dynamics of the subsequent signal processing. The sampling rate is of limited importance as long as the conditions of the sampling theorem are fulfilled. Oversampling increases the requirements for the ADC, but simplifies subsequent signal processing.
- *Carrier recovery* determines the frequency and phase of the carrier of the received signal, and uses it to adjust the RX LO.
- The *baseband demodulator* obtains *soft-decision* data from digitized baseband data, and hands them over to the decoder. The baseband demodulator can be an optimum, coherent demodulator, or a simpler differential or incoherent demodulator. This stage can also include further signal processing like equalization.
- If there are *multiple antennas*, then the RX either selects the signal from one of them for further processing or the signals from all of the antennas have to be processed (filtering, amplification, downconversion). In the latter case, those baseband signals are then either combined before being fed into a conventional baseband demodulator or they are fed directly into a "joint" demodulator that can make use of information from the different antenna elements.
- *Symbol-timing recovery* uses demodulated data to determine an estimate of the duration of symbols, and uses it to fine-tune sampling intervals.
- The *decoder* uses soft estimates from the demodulator to find the original (digital) source data. In the most simple case of an uncoded system, the decoder is just a hard-decision (threshold) device. For convolutional codes, *Maximum Likelihood Sequence Estimators* (MLSEs, such as the Viterbi decoder) are used. Recently, iterative RXs that perform joint demodulation and decoding have been proposed. Remaining errors are either taken care of by repetition of a data packet (*Automatic Repeat reQuest* – ARQ) or are ignored. The latter solution is usually resorted to for speech communications, where the delay entailed by retransmission is unacceptable.
- *Signaling recovery* identifies the parts of the data that represent signaling information and controls the subsequent demultiplexer.
- The *demultiplexer* separates the user data and signaling information and reverses possible time compression of the TX multiplexer. Note that the demultiplexer can also be placed earlier

in the transmission scheme; its optimum placement depends on the specific multiplexing and multiaccess scheme.

- The *source decoder* reconstructs the source signal from the rules of source coding. If the source data are digital, the output signal is transferred to the data sink. Otherwise, the data are transferred to the DAC, which converts the transmitted information into an analog signal, and hands it over to the information sink.