Flexible Multi-Standard Terminals for 2nd and 3rd Generation Mobile Systems

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Abstract: *The principle of software radio architecture is investigated and applied for the implementation of a multi-standard terminal (MST) for operation with both current 2nd and future 3rd generation mobile radio systems. A key strategy for the design of the transceiver is to place the analog-to-digital converter (ADC) and digital-to-analog converter (DAC) as close as possible to the antenna. In addition to reducing the analog front-end, this would allow more transceiver operations to be performed by programmable digital signal processing devices. Flexible DSP techniques are also ideally suited for the implementation of modem functions like channel estimation, synchronisation, equalization, interleaving, source and channel coding for a variety of different transmission technologies and multiple access schemes. This also enables the concept of over-the-air software download for the reconfiguration of the MST. The radio receiver front-end is realized with two intermediate frequencies (IF). The design must satisfy requirements like sensitivity, dynamic range, selectivity and resistance against interfering and blocking signals. Digital down-conversion is performed by sub-sampling at low IF, followed by multirate techniques for decimation and digital filtering. Several trade-offs between the analog front-end and the digital demodulation must be balanced carefully with respect to the available technology, e.g. more digital processing power can relax the requirements for analog filtering. This work has been performed in the context of the FIRST project (Flexible Integrated Radio System Technology) as part of the ACTS mobile line.*

1. Introduction: The Benefits of Multi-Standard Terminals

The mobile communications scene has witnessed the parallel emergence of a wide variety of radio standards throughout the world [8, 9]. This phenomenon has resulted in a growing interest in the field of reconfigurable multi-standard terminals (MST) within the telecommunications industry and the concept is gaining prominence amongst operators, equipment manufacturers and technology suppliers.

A MST may be formally defined as a subscriber unit which is capable of operation according to a variety of different mobile radio standards. Such a capability, coupled with a mechanism for reconfiguration of the terminal over the air-interface, represents a crucial element in the development of future mobile communication systems. Some of the key benefits of a reconfigurable MST may be described as follows:

Economies of Scale. The proliferation of new digital cellular standards in the US, Europe and Japan has resulted in the adoption of diverse subscriber (and base) terminal architectures in different geographical areas. The ability to develop a single reconfigurable transceiver which can be adapted via software to operate according to arbitrary radio standards clearly represents immense potential savings on the part of equipment manufacturers.

Transparent Roaming of Users. A key advantage of MSTs would be their ability to allow users to roam across networks transparently, regardless of the diverse multi-band and multi-standard environments. The need for such *inter-standard* roaming is particularly strong in the fractured technical market of the US, where a plethora of systems based on GSM, IS-54, IS-95, DECT etc. coexist in the PCS band.

Teleservice Adaptation by the Network Operator. The ability to reconfigure subscriber terminals over the air-interface would present the network operator with the ability to create and provide a wide range of services and features suited to the needs of individual customers.

Evolution Towards UMTS. The concept of UMTS currently pursued in Europe [2, 9] is intended to provide a *revolutionary* improvement in the quality of cellular systems and services. However, the huge investment in GSM infrastructure within Europe, and the phased roll-out of any future 3rd generation system, mean that UMTS will only be viable if an *evolutionary* path is also identified. UMTS mobile transceivers capable of also implementing the GSM air-interface represent a powerful solution to the problem of backward compatibility.

Hybrid and Adaptive Transport Modes in UMTS. Any transmission mode that assumes worst-case channel conditions inevitably makes inefficient use of the radio resource. The channel can be used more efficiently by adapting the features of the transmitted signals to the channel conditions. Implementations of this concept of *intra-standard* adaptation have been investigated in the RACE-II ATDMA [11] and CODIT [1] proposals for UMTS. Dynamic adaptation of burst structures, modulation, channel and source coding, as well as the use of hybrid multiple-access schemes are likely to be key features of the UMTS air-interface. As such, future UMTS transceivers will inherently be based on what amounts to multi-standard technology.

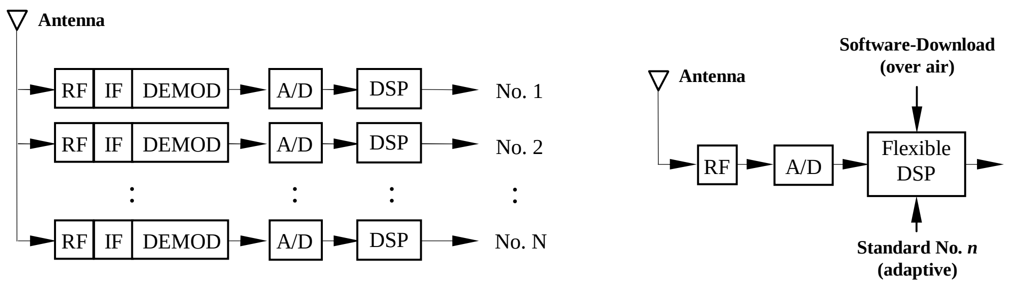
2. Operational Requirements and Basic Software Radio Architecture

The operational characteristics of an ideal MST may be categorized as follows:

(1) Software-Definable Operation. A true MST must be capable of accommodating future undefined air-interfaces, signalling protocols and teleservices. Such a high degree of flexibility can only be achieved through a mixture of reconfigurable technologies such as programmable DSPs and FPGAs [4, 5]. Furthermore, such technology would enable the process of over-the-air software-based adaptation of the MST by the local infrastructure.

(2) Multi-Band Operation. The ability to process signals corresponding to a wide range of frequency bands and channel bandwidths is a critical feature of a MST and impacts heavily on the RF (Radio Frequency) segments of the terminal. The left part of Figure 1 depicts a traditional approach to the implementation of a MST receiver, illustrating the use of separate transceiver chains for each individual radio standard. While such a realization may be viable for dual-standard terminals, the analog component-count prohibits the use of this approach for the support of multiple standards. Moreover, note that the channel selection and demodulation processes are performed completely in the analog domain followed by digitization at baseband. The logic behind this type of radio architecture is to minimize the digital signal processing effort at the expense of a rather complex analog part. The majority of today's single- and dual-mode terminals are still based upon this traditional approach.

The logic behind the concept of a digital software radio [7], on the other hand, is to simplify the analog segment of the transceiver by shifting some of the signal processing effort into the digital domain. This is shown in the right part of Figure 1 which illustrates the process of passband digitization followed by digital down-conversion. However, sampling at RF in the 2 GHz range is still too optimistic with today's technology [12]. But digital signal processing at IF reduces the analog component-count and provides significant flexibility by allowing programmable digital channel-selection filtering in the receiver. This, however, is achieved at the expense of additional DSP power (due to the high sampling rates) and excessive demands on the ADC.



Traditional MST with *N* distinct transceivers **FIRST's vision of a software-definable MST**

Figure 1. Alternative multi-standard terminal architectures

(3) Multi-Mode Operation. The ability to transmit and receive signals corresponding to different modulation and error protection schemes, burst structures, compression algorithms and signalling protocols is another essential feature of a MST. Given that many of the above baseband operations are already performed by programmable DSPs and microprocessors in current commercial products, multi-mode operation can be envisaged to be readily feasible in the very near future.

Figure 2 shows a logical diagram of the overall structure of a MST with emphasis on the baseband signal processing aspects. The MST is subdivided into five main parts. The analog RF part (front-end) contains the multiband antenna, PA (Power Amplifier), LNA (Low-Noise Amplifier), DAC and ADC. The processes of modulation and demodulation are split between the analog and digital parts. The baseband signal processing part includes operations for synchronisation in conjunction with channel estimation, channel equalization, interleaving and error-control coding. Modulation, interleaving and channel coding appear as toolboxes to underline the flexible nature of the MST. Adaptation metric computation refers to the evaluation of criteria for dynamic adaptation of the MST transport mode. The multimedia application part consists mainly of the source encoding and decoding processes, including the MPEG4 algorithms currently under study. The management and control part controls the overall MST functionality and is responsible for software re-configuration of the terminal. This part also implements all signalling protocols required for the operation of a mobile terminal.

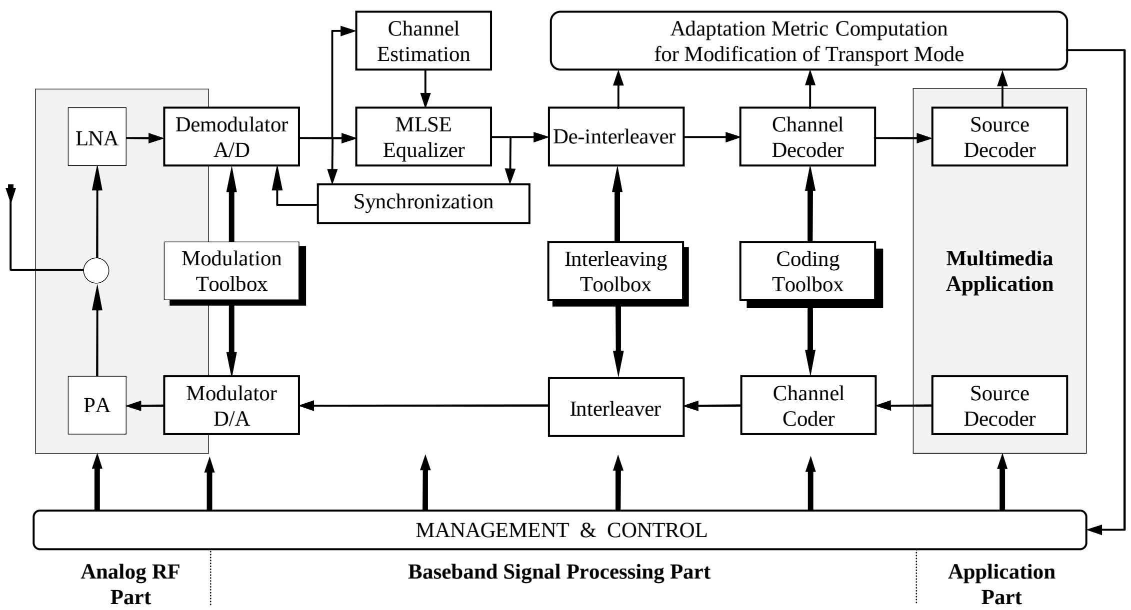


Figure 2. Structure of a flexible multi-standard terminal

This paper focuses on the architecture of a MST for the support of three radio standards operating in the 2 GHz band, namely DCS1800, DECT (representing 2nd generation cellular and cordless systems) and RACE-II ATDMA (as a candidate for UMTS). A great deal of further research is still required on different aspects of the software radio architecture for MSTs, but this paper addresses some of the most important questions and gives an overview of potential solutions.

3 Air-Interfaces of ATDMA, DCS/GSM and DECT

Table 1 contains some basic information about the transmission schemes of the mobile radio standards to be considered for the MST. Note that the differences between these air-interfaces can *at least in principle* be achieved by switching between the different air-interface parameters.

Table 1. Air-interface parameters of ATDMA, DCS/GSM and DECT

	ATDMA				DCS/GSM	DECT
Cell Type	Long Macro	Short Macro	Micro	Pico		
Cell radius [km]	35	20	1	0.1	8 / 35	0.3
Uplink band [MHz] Downlink band [MHz]	1920 – 1980 2110 – 2170				1710 – 1785[1] 1805 – 1880[1]	1880 – 1900 1800 – 1900
Duplex scheme	FDD (and TDD)				FDD (TDD)	TDD
Carrier spacing [kHz]	276.92		1107.69		200	1728
Symbol. rate per carrier [ksymbols/s]	360	225	900		270.833	1152
Modulation scheme Bits per complex symbol	GMSK 1	offset 4- or 16-QAM 2 or 4			GMSK 1	GFSK 1
Spectral rate[2] [Symbols/s/Hz] [Bits/s/Hz]	 1.30 1.30	 0.81 1.625 or 3.250			 1.35 1.35	 0.67 (×2) 0.67 (×2)
Frame duration [ms]	5				4.615	10
Slots per frame	15	9	36		8	12 (×2)
Slot duration [μs]	333	556	139		577	416
Data rate/slot[3] [kbits/s]	13.2	26.4[5]		34.4[5]	22.8 (full) 11.4 (half)	32.0
Data rate/carrier[4] [kbits/s]	198	237.6[5]	950.4[5]	1238.4[5]	182.4	384

[1]) DCS values. For GSM: UL 880-915 MHz, DL 925-960 MHz.
[2]) With respect to carrier spacing. [3]) 1 slot allocated. [4]) All slots allocated. [5]) 4-QAM assumed.

The ATDMA project has defined four main cell types [11], whilst the 2nd generation standards DCS/GSM and DECT are designed for a single cell type, although certain kinds of extensions are currently under discussion for DCS/GSM [10]. The access schemes of all transport modes shown in Table 1 are based on combinations of TDMA and FDMA. The duplex scheme is performed both by FDD (except for DECT) and TDD (except for high data rates where more than 50% of the slots are allocated in one direction). The carrier spacing varies over a wide range and can cause severe problems for both analog and digital filtering as well as for the ADC.

The preferred modulation scheme for large cells is GMSK where the constant signal envelope allows the use of highly-efficient PAs. Higher data rates in the smaller cells of ATDMA are supported by 4- or 16-QAM, with additional requirements on the PA linearity. The two-sided bandwidth is denoted by $(1+\alpha)/T$, where α denotes the excess bandwidth and $1/T$ the symbol rate. The two-sided bandwidth will be limited to a reasonable value of 1.5 MHz in the analog part of the MST, thus allowing $\alpha \leq 66\%$ for all ATDMA modes and $\alpha \leq 30\%$ for DECT. In the case of DCS/GSM the broad 1.5 MHz spectrum would contain not only the desired narrowband signal of 0.3…0.4 MHz bandwidth, but also adjacent channel interferers and blocking signals.

4 Transceiver Front-End

The transceiver front-end including all RF and IF processing stages from antenna to ADC at IF must satisfy requirements such as sensitivity, dynamic range, selectivity and resistance against interfering and blocking signals. Figure 3 gives an overview of the transceiver in terms of a frequency plan. Given the complicated arrangement of the frequency bands to be covered, it seems unrealistic to use a diplexer for separation between transmit and receive frequencies. In case of TDD operation, transmit and receive will never occur simultaneously in time, and hence the separation can be achieved by a simple switch at the antenna.

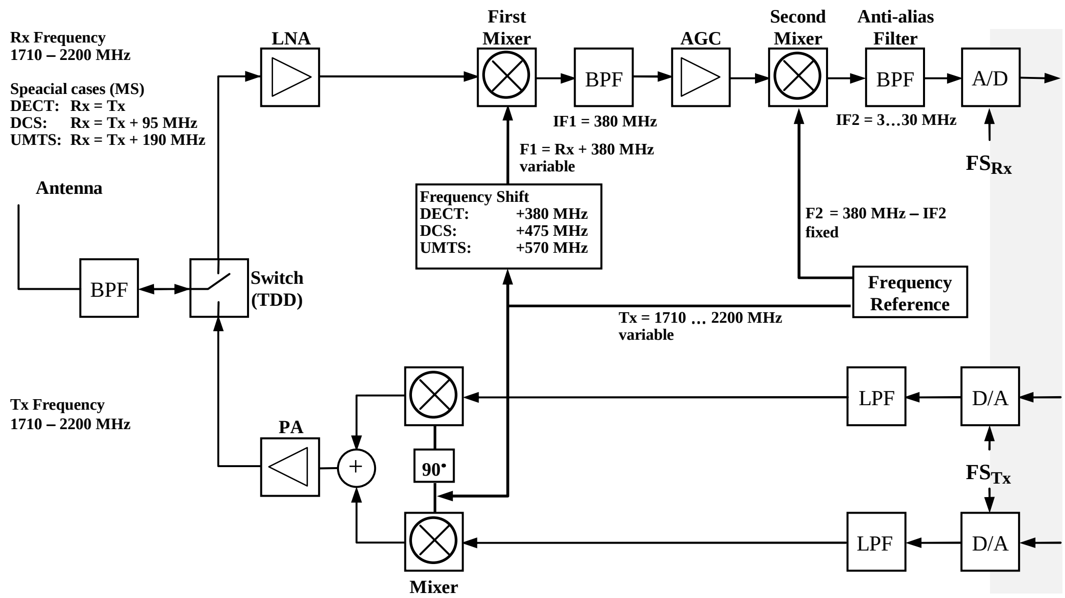


Figure 3. Transceiver Frequency Diagram.

The superheterodyne principle with two fixed IFs is the preferred solution for the receiver with a first high IF1 for image rejection and a second low IF2 to perform highly-selective filtering and to ensure appropriate further processing. The alternative of only one IF would imply problems with image rejection and the required steepness of the analog filter frequency responses. The transmit and receive frequencies are denoted as Tx and Rx, respectively. Given today's technology, the maximum sampling rate for the ADC is restricted to about 100 MHz or even lower [12]. This prohibits ADC at RF rates of 2 GHz, but digital processing at lower IF2 rates is actually feasible.

The transmitter is realized by direct up-conversion in order to meet requirements of high spectral purity, i.e. to limit spurious frequencies which typically occur due to mixers and IF processing. The pulse shaped baseband I and Q components of the transmitted signal are available at an oversampled rate, so that simple lowpass filtering after the DAC would be sufficient. The DAC must operate at a maximum sampling rate of ≈10 MHz for the case of 8-times oversampling. A resolution of 12 bits will achieve the requirements of the frequency mask [3].

According to most mobile specifications, Tx must be extremely precise and cannot be achieved with the usual synthesizers. Consequently, Rx has to be estimated in the receive branch with a high degree of accuracy to generate the transmit reference Tx from Rx.

The receiver path with its two intermediate frequencies is illustrated in more detail in Figure 4. Due to the wide operational bandwidth and the potentially large number of signals appearing within this bandwidth, adjacent channel interference and blocking signals can generate large intermodulation products depending on the non-linearity of the mixers and amplifiers. The overall dynamic range is extremely high, cf. Figure 5. The antenna filter must have a bandwidth of 500 MHz, if not operated as tracking filter. The LNA has a fixed amplification whereas the gain control is performed at a later stage at IF1.

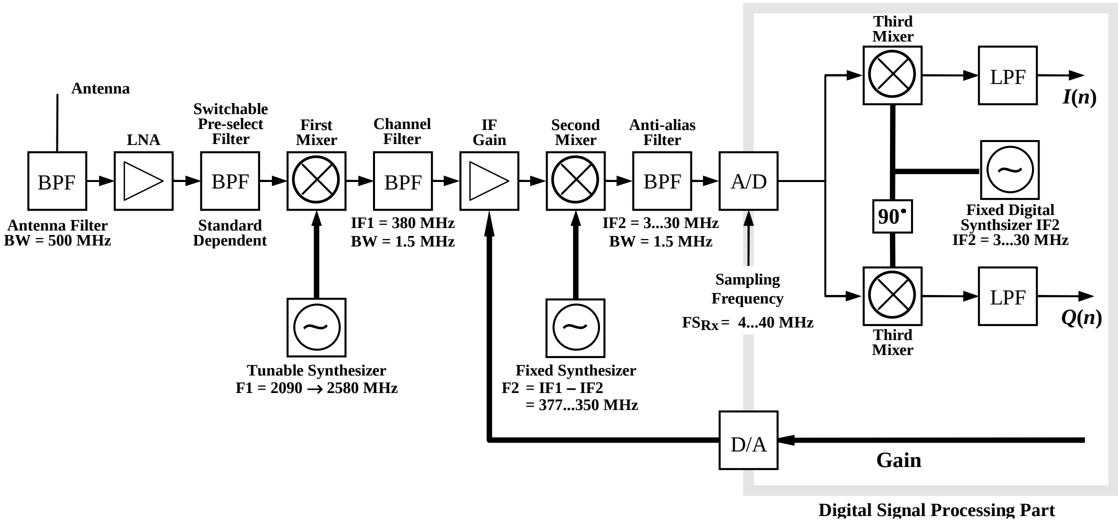


Figure 4. Block diagram of receiver path

The bandwidth of the bandpass (BP) channel filter is 1.5 MHz as already mentioned. For narrow carrier spacing like DCS/GSM, more than one carrier and also blocking signals will be received within the 1.5 MHz band. The final filtering is therefore performed at baseband.

The choice of IF2 depends highly on the interrelation with the baseband signal part, since there are two contradictory requirements: While a large value of IF2 would simplify the design of the anti-alias filter, a low value is required to simplify the ADC and the digital processing complexity. An ADC with FS = 40 MHz and a resolution of 12 bits is a reasonable maximum for current technology. The highest reasonable value for IF2 would be around 30 MHz and the lowest possible value is about 3 MHz. The digital down-conversion simplifies drastically if the relation $FS/IF2 = 4/3$ holds. Hence the concept of IF sub-sampling [4,5,6] is used here, where FS is less than twice the frequency of the highest-frequency component in the signal to be sampled.

Let $BW = 1.5$ MHz be the passband width of the analog filters. The width of the transition band is then $(2\times IF2)-BW$ for the BP channel filter and $(FS/2)-BW = (2/3\times IF2)-BW$ for the anti-alias filter. An increase in IF2 relaxes the required steepness of the analog filter frequency responses, but requires a higher FS and thus more effort for digital signal processing.

5 Analog-to-Digital Converter (ADC) and Automatic Gain Control (AGC)

According to the DCS/GSM specifications [3], receiver operation must still be possible in the presence of cochannel interference with a signal level of 9 dB below the wanted signal level. Furthermore, the receiver must cope with adjacent channel interference with signal levels of 41 dB above the wanted signal level for a frequency distance of only 400 kHz. Finally, blocking signals of more than 60 dB within the 1.5 MHz bandwidth are specified. The stopband of the anti-alias filter in combination with the BP channel filter must therefore achieve an attenuation of approximately 75…80 dB. In general, two major fading effects have to be distinguished:

Shadowing effects and free-space transmission losses (according to MS-BS distance) are slow compared to the frame duration (~5 ms). The receiver can average the received signal power over a certain number of frames. Moreover, the variations of the received signal level due to these effects can be tracked and predicted by Kalman filter techniques. The gain control information is based on this long-term tracked/predicted value which will be referred to as *mean power*. The dynamics of these effects may extend to about 80 dB.

Fading effects due to frequency-selective propagation are *fast compared to the frame duration*, i.e. the fading effects are statistically independent and are not predictable from slot to slot. But frequency-selective fading is typically slow compared to the slot duration. Hence the radio channel is assumed to be nearly constant during a slot. The dynamics of frequency-selective fading are typically restricted to 40 dB, i.e. +10…–30 dB around the mean power.

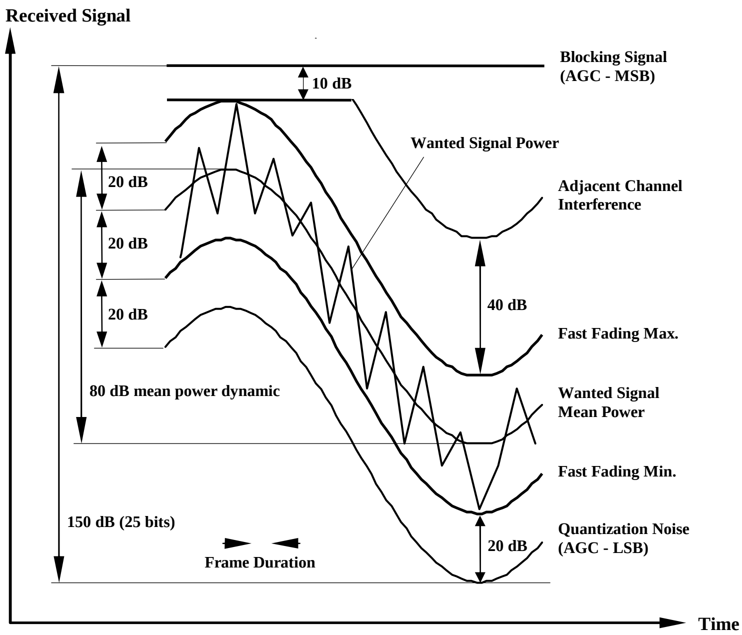


Figure 5. Power diagram for typical mobile radio channel

Figure 5 shows a diagram for the received signal power as a function over time for typical worst case mobile radio conditions. The figure does not correspond exactly to the specifications of a particular standard such as DCS or GSM. The zigzag curve displays the received power of the wanted signal averaged over a single slot within a frame, i.e. one sample in the zigzag curve corresponds to one frame. The mean power of the wanted signal is averaged over several frames showing slow variations of about 80 dB. The quantization noise after ADC should be 8…20 dB smaller than the wanted signal, depending on the modulation scheme and the partition with the thermal noise. The adjacent channel interference is assumed as 40 dB above the wanted signal. The blocking signal could be caused by any other system and hence it seems reasonable to assume that it is about 10 dB higher than the maximum signal occurring in the wanted mobile network. Figure 5 does not reflect the fact that a sporadic total loss of slots due to fast deep fading effects would be acceptable if the transport mode is based on interleaving schemes with sufficient depth and appropriate forward error correction mechanisms.

If proper MST operation is required with blocking signals of *maximum level* which are not rejected by the analog filters of 1.5 MHz bandwidth, then an AGC makes no sense at all and the ADC must have a resolution of 20…25 bits for a 120…150 dB power level range. This seems to be challenging even for reduced FS. If the blocking specifications are relaxed and scalable analog filters are used, then the slow fading mean power can be tracked with an AGC as indicated in Figure 4, so that adjacent channel interference levels of 10...40 dB would become acceptable with an ADC resolution of 12…17 bits. A further reduction of 5 bit is possible with an appropriate interleaving scheme. If the ADC is not subjected to systematic errors, then a further slight relaxation of its resolution is possible due to averaging of dither effects in case of high oversampling (see below).

6. Synchronisation and Analog/Digital Interface

While the synchronisation control information is usually derived in the digital domain in conjunction with channel estimation and equalization, the adjustment of the sampling clock and carrier could be performed either in the analog or the digital domain. Each of these two solutions has specific advantages and drawbacks, and again represent trade-offs between the complexity of analog and digital parts of the receiver.

In principle, carrier control is not required at all since the phase rotation could also be achieved with the equalizer. But as already mentioned, an accurate carrier reference Rx is required for Tx and hence carrier control outside the equalizer is preferred. Clock control is also not required in principle, since the sampling phase variations during the short time slots are negligible. Carrier and clock frequencies are coupled together for all mobile radio standards, hence knowledge of carrier frequency gives knowledge of clock frequency and vice versa.

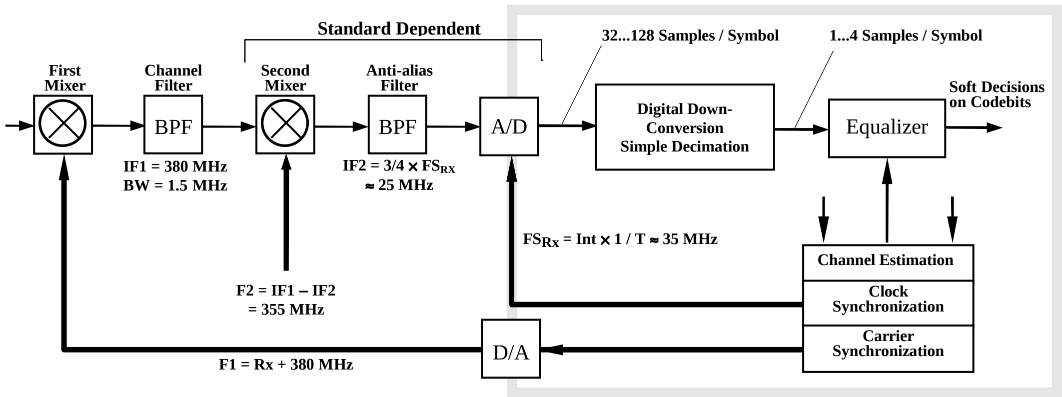


Figure 6. Analog Synchronisation

Analog synchronisation is depicted in Figure 6, where the first mixer and the ADC are controlled by the digital part. The control of the first mixer has two drawbacks: A DAC is required to send the carrier control information from the digital to the analog part, and the control loop has a considerably long delay due to the digital decimation and lowpass filtering. The control of the ADC has also a severe drawback: The symbol rates for different standards are quite distinct, and they are usually related by irrational factors. Distinct FSs would require distinct IF2s in order to maintain the relation $FS/IF2$ = 4/3 (see below), thereby impacting on the location of the passband of the anti-alias filter and on the second mixer frequency F2. Such a dependence of the analog part on the specific standards is contradictory to the philosophy of software radio.

However, for the triplet of ATDMA, DCS and DECT it is possible as shown in Figure 6 to choose integer factors *Int* as powers of 2 so that $FS = Int/T \approx 35$ MHz holds and the impact on the anti-alias filter is negligible. Note that *Int* = 32...128 equals the number of samples per symbol.

Figure 7 shows the situation with digital clock and carrier control. A distinct feature of this architecture is that sampling is controlled by a *fixed* clock whose timing is not adjusted by the digital part and is asynchronous with respect to the received data stream. Clock adjustment is performed completely in the digital domain via interpolation in conjunction with the decimation process. Carrier adjustment is performed by means of a complex phase rotation also within the decimation process. The decimation and associated lowpass filtering operations required for digital synchronisation can be very demanding in terms of processing power, especially since the decimation factors are no longer convenient integers as was the case with analog synchronisation.

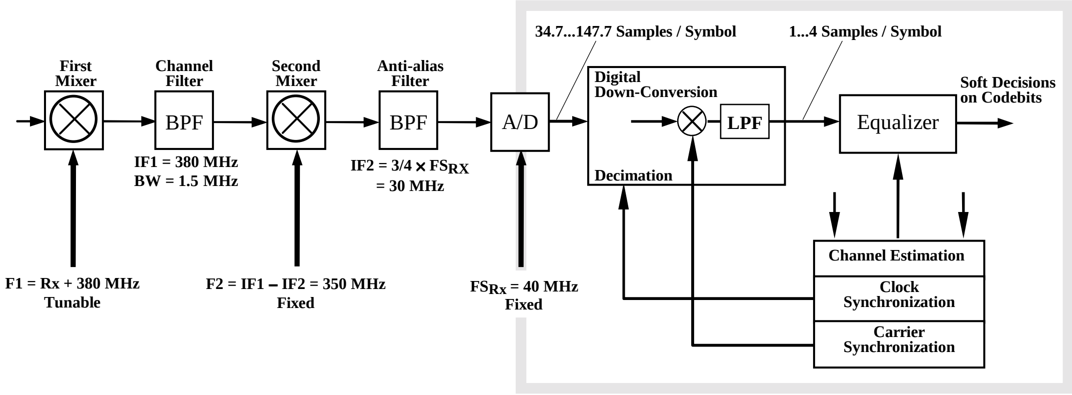


Figure 7. Digital Synchronisation

After ADC the data rate is fixed between 4 and 40 Msample/s. Prior to equalization, channel estimation and synchronisation, the data rate has to be reduced to 2 samples per symbol. Figure 8 shows as an example a chain of decimation filters designed for transport modes based on ATDMA rates. We assume FS = 28.8 MHz, a symbol rate per carrier of 900/450/225 ksymbol/s and L_{max} = 36/18/9 slots per frame. The final digital lowpass channel filter (LPF) has to separate the wanted signal with a spectral bandwidth of between 0.3 and 1.5 MHz from the total baseband signal of 1.5 MHz constant width. Therefore this filter effectively performs variable decimation. Figure 8 also indicates the required filter lengths and number of operations under the assumption of polyphase half-band filter structures.

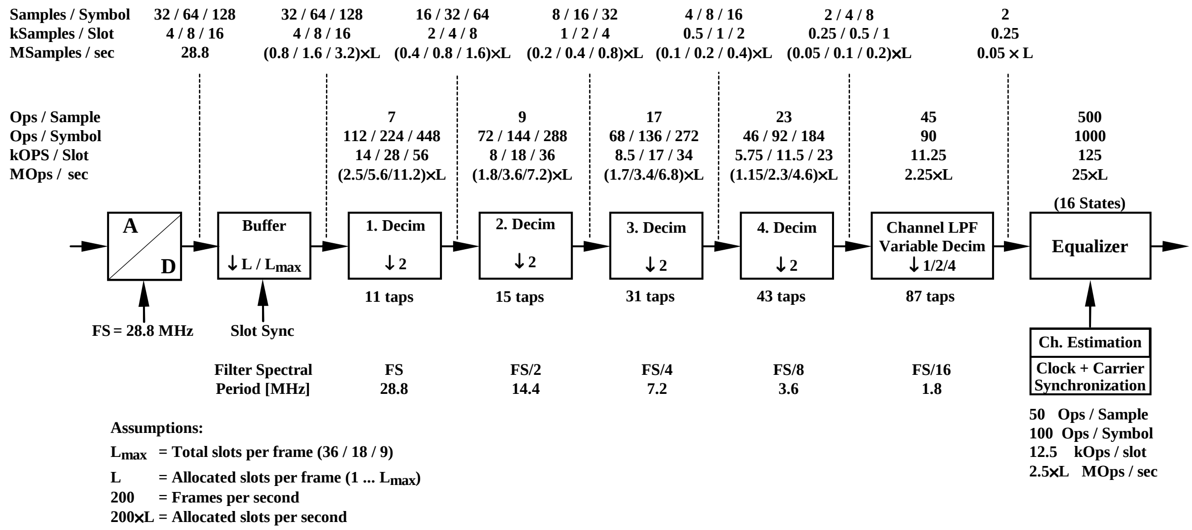


Figure 8. Number of operations required for decimation, filtering and equalization

Let $L \leq L_{max}$ be the number of allocated slots per frame. A buffer memory is introduced between the ADC and the decimation process, which is fed by the ADC output at a high data rate, but only during L slots per frame. The memory shifts data out at a reduced speed so that the data is spread over the whole frame. The buffer achieves a data rate reduction by a factor of L_{max} /L prior to the first decimation stage. The total number of operations for the decimation chain and LPF sums up to (9.7/17.15/24.6)×L million operations per second. The maximum value is ≈500 MOps/s for L=L_{max} (together with equalization). It is interesting to note that the number of operations must be *increased* for *decreasing* symbol rates. Figure 8 shows also the required number of operations for synchronisation and channel estimation as well as for the Viterbi equalization assuming an add-compare-select accelerator.

7. Conclusions

The use of a common wideband analog front-end in conjunction with IF sub-sampling prior to digital down-conversion is a major characteristic of software radio architectures as applied to MSTs. However, it has been shown that this can have serious ramifications when dealing with narrowband standards. Due to the fact that the blocking signals in narrowband standards are not suppressed by the wideband analog receiver, the ADC must have a very high resolution and AGC cannot be applied. But relaxed specs for blocking allow for certain reductions of the ADC requirements, depending on modulation, interleaving, oversampling and the required noise figure. Blocking signals also demand high stopband attenuations both for analog anti-alias and digital lowpass channel filtering.

The ADC sampling frequency FS should be high to simplify the analog filters, but a high signal processing load for digital filtering results as a drawback. A reasonable compromise for this trade-off would be in the range FS = 7...30 MHz. By means of digital clock and carrier synchronisation, variations in the analog part (due to different standards) and the need for additional DACs are both avoided, but at the expense of highly-complex digital decimation filtering. This, in addition to transceiver linearity, ADC design, intermodulation performance and digital signal processing for modem functions, represents the main bottlenecks in the realisation of software radio MSTs.

Further research with regards to the feasibility of the concept of software radio MSTs will be continued within the FIRST project.

Acknowledgements: The contributions by our colleagues in the FIRST project are gratefully acknowledged.

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