A SOFTWARE RADIO ARCHITECTURE FOR MULTI-CHANNEL DIGITAL UPCONVERSION AND DOWNCONVERSION USING GENERALIZED POLYPHASE FILTERBANKS WITH FREQUENCY OFFSET CORRECTION

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ABSTRACT

Polyphase FilterBanks (PFBs) provide a computationally efficient approach to extracting channels of arbitrary bandwidth from a wideband signal. This technique, know as digital downconversion, would find use in software radio applications. Conversely, PFBs can also be used to perform digital upconversion, a process in which a wideband signal is constructed from several narrowband channels. In this paper, we apply the PFB technique to the IS-136 standard (North American Digital TDMA). Since the IS-136 standard requires non-overlapping adjacent channel filter masks, we show in this paper that the Generalized PFB is the optimal architecture given the non-integer sampling rate conversion that is needed to go from IF to baseband. Since frequency offset correction is an important consideration for radio receivers, we also present an augmented GPFB architecture that intrinsically performs frequency offset correction.

1. INTRODUCTION

There are several mobile communication standards currently in service in various parts of the world, which do not interoperate with each other at the physical layer. Moreover, the development cost for these standards remains high since each standard requires a different radio front-end. Recognizing these problems, there is currently a strong interest in designing hardware platforms that can support multiple standards simultaneously, resulting in what is known as a software radio [3]. Since each wireless standard specifies a different channel bandwidth, traditional single-carrier techniques are no longer applicable, where the radio functions are optimized around a fixed channel bandwidth. For example, the channel bandwidths for the three most popular mobile communications standards are: GSM = 200kHz, IS-136 = 30kHz, and IS-95 = 1250kHz.

To help achieve the goal of catering to several channel bandwidths, the channel selection process should be performed digitally (cf. [1][2]). To this end, the radio front-end captures a broadband signal (containing multiple channels) rather than a single narrowband channel. Such an architecture is ideally suited for a wireless base station that needs to process multiple channels simultaneously. Thus, at the base station, we replace multiple costly single-channel radios with a single broadband radio where the rest of the channel selection and filtering is performed digitally via Digital Upconversion (DUC) and Digital Downconversion (DDC).

Polyphase Filterbanks (PFB) are known to provide an efficient architecture for DDC and DUC [2]. There are several flavors to the PFB: (a) maximally decimated (critically sampled) [4], (b) oversampled [4], and (c) fractionally-sampled (Generalized PFB) [7]. In this paper, we show that GPFB is the most efficient architecture for implementing the IS-136 standard due to the fractional change in the sampling rate. The GPFB presented in [7] considers an even-type architecture. In the paper, we present the mathematical framework for the Odd-Type GPFB, which allows us to perform frequency offset compensation as well.

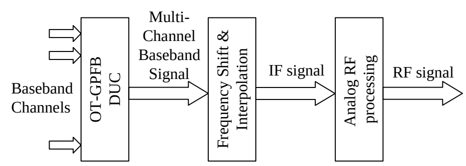

Fig. 1: Proposed DUC architecture

Fig. 1 presents an architecture for DUC using GPFB (Generalized Polyphase Filterbanks). Multiple IS-136 channels at baseband are processed by the GPFB to create a broadband signal at baseband, which is then upconverted to an intermediate frequency (IF) via interpolation and digital frequency upconversion. The digital-IF signal is then converted to an analog signal via a D/A, followed by quadrate upconversion to RF, band-pass filtering, and power amplification prior to transmission.

Fig. 2 presents an architecuture for DDC using GPFBs. The RF signal is downconverted to IF, and then sub-sampled to created a broadband digital-IF signal. This signal

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is quadrature-downconverted to baseband. The GPFB then extracts the narrowband IS-136 channels from the broadband signal. The preamble training sequence helps in determining the frequency offset, which is then fed back to the GPFB to perform frequency offset correction.

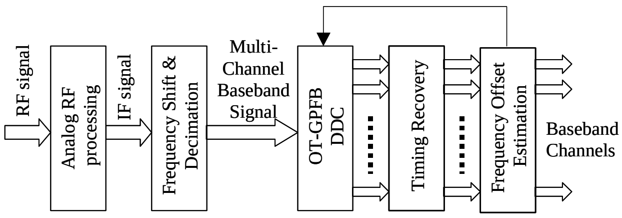

Fig. 2: Proposed DDC architecture

2. GENERALIZED POLYPHASE FILTERBANK

The generalized polyphase filterbank is an efficient digital signal processing structure for peforming channel filtering and frequency translation [8]. By applying a polyphase decomposition to the channelization filter, the filtering operation can be shared across all the sub-channels. Simultaneously, the frequency shift to the center frequency is achieved via an *Inverse Discrete Fourier Transform* (IDFT), which can be implemented very efficiently using *Fast Fourier Transform* (FFT) algorithms [5][6]. In this work, the *synthesis* filterbank structure performs DUC, and the *analysis* filterbank structure performs DDC.

2.1. Odd- and even-type filterbanks

The channelization process for a single channel is depicted in Fig. 3. The baseband signal $y_k(m)$ – transported on channel k – is interpolated by a factor M to yield the signal $w_k(n)$. Images that appear in the frequency domain due to the interpolation process are removed by the FIR filter f(n), which may also perform pulse-shaping. The signal $z_k(n)$ is shifted to the center frequency $2\pi k/Q$, where the frequency shifting factor is defined as

$$W_Q = e^{j(2\pi/Q)} . \qquad (1)$$

Q is the total number of channels processed by the filterbank.

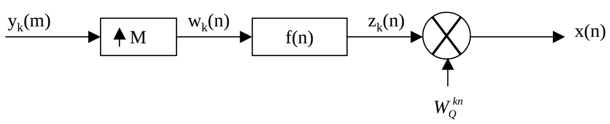

Fig. 3: Single channel processing

The equally spaced center frequencies of the Q channels for the **even-type** filterbank are given by:

$$\omega_k = \frac{2\pi}{Q} k . \qquad (2)$$

If channel 0 is not centered at the origin, but instead is offset such that $\omega_0 = \frac{2\pi}{Q}\varepsilon$, the filterbank is referred to as an odd-type. One application where ω_0=0 is difficult to achieve is in the case of radio receivers, where variable frequency offsets due to oscillator instabilities are quite common. In such applications, the odd-type filterbank provides a computationally efficient solution since an additional frequency shifting stage is not needed.

The mathematical framework for the even-type Generalized Polyphase Filterbank has been presented in [7]. We extend this analysis to arrive at the mathematical formulation for the odd-type GPFB (OT-GPFB), that could find use in multi-standard multi-channel Software Radios. The center frequencies in the **odd-type** filterbank are given by:

$$\omega_k = \frac{2\pi}{Q}(k+\varepsilon) = \frac{2\pi}{Q}k' , \qquad (3)$$

where we introduce the index

$$k' = k + \varepsilon . \qquad (4)$$

2.2. Digital Up Conversion (DUC)

The up-conversion process of the Q baseband channels $y_k(m), k = 0,1,\ldots,Q-1$ is depicted in Fig. 4. The sampling rate for a single channel is S_{SC}; S_{MC} is the sampling rate of the multi-channel signal x(n). The relationship between these sampling rates is

$$S_{MC} = M \cdot S_{SC} = Q \cdot B_{SC} , \qquad (5)$$

where B_{SC} is the channel spacing in the frequency domain. Hence, we get:

$$Q/M = S_{SC}/B_{SC} . \qquad (6)$$

It should be emphasized that (6) places some important constraints on the structure of the filterbank and the output sampling rate, since the channel sampling rate S_{SC} and the channel spacing B_{SC} are parameters that are defined by the communications system standard.

If the total frequency bandwidth is B_K , the sampling theorem introduces the constraint that:

$$M \geq \frac{B_K}{S_{SC}} . \qquad (7)$$

Furthermore, from (5) the number of channels is given by:

$$Q = \frac{S_{SC}}{B_{SC}} M \qquad (8)$$

has to be an integer. The design of the filterbank thus needs to fulfill equations (7) and (8).

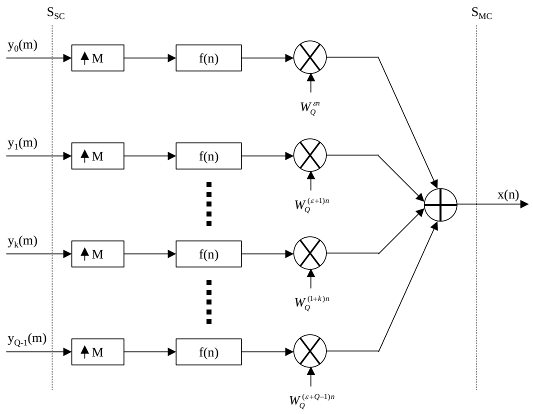

Fig. 4: Synthesis of Q channels

The output x(n) of the odd-type filterbank shown in Fig. 4 is given by [1][7]:

$$x(n) = \frac{1}{Q} \sum_{k=0}^{Q-1} W_Q^{k'n} \sum_{m=-\infty}^{\infty} y_k(m) f(n - Mm) . \qquad (9)$$

To convert the structure shown in Fig. 4 to a canonical filterbank representation, we define several variables. If the output signal x(n) is grouped into sets containing Q samples each, then the absolute time index n can be represented as:

$$n = Qr + \rho, \quad \rho = 0, 1, \ldots, Q-1 \qquad (10)$$

where r is the index to the set, and ρ is the index within the set. Also, we define R as the *greatest common divisor* (gcd) of the sampling rate change M and the multiplicity of channels Q

$$R = \gcd(M, Q) . \qquad (11)$$

Furthermore, we introduce two other integers η_i and η_d

$$\eta_i = \frac{Q}{R} \qquad \eta_d = \frac{M}{R} . \qquad (12)$$

Equation (9) can now be re-expressed as:

$$x(Qr + \rho) = \sum_{m=-\infty}^{\infty} f'_\rho(\eta_i r - \eta_d m) v'_\rho(m) W_Q^{\varepsilon Mm} , \qquad (13)$$

where we introduce a modified IDFT

$$v'_\rho(m) = \frac{1}{Q} \sum_{k=0}^{Q-1} y_k(m) W_Q^{k'\rho} \qquad (14)$$

and a set of modified polyphase resampling filters:

$$f'_\rho(s) = f(Rs + \rho) W_Q^{\varepsilon Rs} . \qquad (15)$$

The resulting structure is based on a counterclockwise commutator model, as shown in Fig. 5. The baseband data of the input channels is first processed in a multiplication stage and then a modified Q-point IDFT is applied (the sequence of operations can be swapped). The odd-type differs from the even-type GPFB in three aspects: (a) the multiplication stage is absent in the even-type structure, (b) the odd-type incorporates a modified IDFT and (c) the branch resampling filters get modified. It should also be noted that Fast Fourier Transform algorithms can be used for IDFT computation even when the number of inputs is not a power of 2 (cf. [5][6]).

Data coming from the IDFT is resampled in each branch with the branch filter by a resampling factor η_d/η_i. Finally, the branch data is combined in the commutator, where the branch output sampling rate is increased by a factor Q.

There are η_i groups of R branch filters each. In each group, there are just delayed versions (by one tap) of the branch filters in the predecessor group. This can easily be seen in equation (15). When ρ reaches the value R, exactly the same filter taps as for ρ=0 are selected for the branch filters. Due to the fact that the global filterbank filter f(n) has a finite length, the length of the branch filters decreases by one tap from one group to the next. The special structure of the branch filters may be exploited to simplify the hardware implementation.

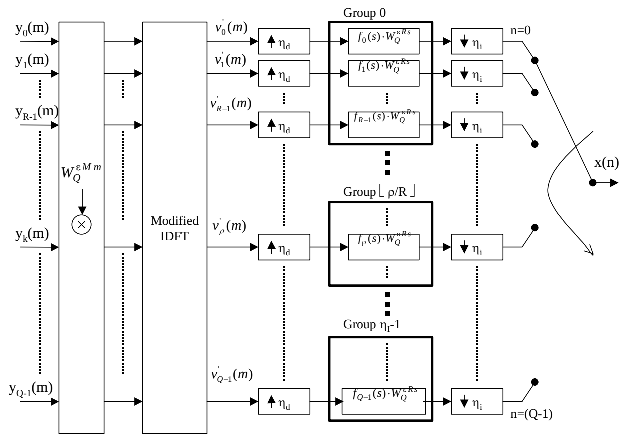

Fig. 5: Odd-type synthesis filterbank

The GPFB requires resampling in each of the Q branches. It is therefore important to have efficient resampling algorithms in the filterbank structure (cf. [1]).

Reduced number of input channels

In terms of necessary operations per channel, the GPFB is most efficient if there are exactly Q input channels, as the filter processing is shared between the channels. If there are only Z input channels, the number of polyphase branches is still Q and the computational requirements for the filtering process does not decrease. Only the size of the modified

IDFT changes from Q×Q to Z×Q. It should be emphasized that the subset of input channels is arbitrary, i.e. the channels do not have to be adjacent, but may be selected from the entire range k=0, 1, ..., Q-1.

A basic constituent of the filterbank structure with Z input channels is the Z×Q-point IDFT. A brute force implementation would require Z×Q complex multiplications. However, there are efficient algorithms that reduce the required number of operations significantly (cf. [5][6]).

2.3. Digital Down Conversion (DDC)

The polyphase implementation of the generalized *analysis* filterbank can be derived in a manner similar to that of the *synthesis* filterbank. For a single channel, the required signal processing is depicted in Fig. 6. The incoming multichannel signal x(n) is frequency shifted to bring the channel spectrum of channel k to baseband. After that, the spectrum of channel k is extracted and optionally match filtered by h(n). Finally, the single channel baseband signal is decimated by a factor of M.

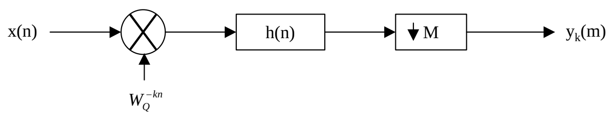

Fig. 6: Single channel down-conversion

The single channel output in the odd-type case is [1][7]:

$$y_k(m) = \sum_{l=-\infty}^{\infty} h(l)x(Mm-l)W_Q^{-k'(Mm-l)} \ . \quad (16)$$

Again, we introduce the resampling filters

$$h'_\rho(s) = h_\rho(s) \cdot W_Q^{\varepsilon Rs} \quad (17)$$

and the resampler outputs

$$v_\rho(m) = \sum_{r'=-\infty}^{\infty} h'_\rho(\eta_d m - \eta_i r) x_\rho(r) W_Q^{-\varepsilon Mm} \ . \quad (18)$$

With a modified IDFT the overall filterbank output for the odd-type channel arrangement becomes

$$y_k(m) = \sum_{\rho'=0}^{Q-1} v_\rho(m) W_Q^{k'\rho} \ . \quad (19)$$

It should be mentioned that this is also a counterclockwise commutator model using a modified IDFT. Compared to the synthesis filterbank, the sequence of operations in the analysis filterbank is simply reversed.

3. OVERSAMPLED POLYPHASE FILTERBANK

If the interpolation factor of the synthesis filterbank (or conversely the decimation factor for the analysis filterbank) is restricted to an integer value of the number of channels Q, the *Oversampled Polyphase Filterbank* (OPFB) structure results [4]. OPFB does not require resamplers in the polyphase branches and thus simplifies the filterbank implementation. On the other hand, the OPFB is less flexible than the GPFB, and will in certain applications require an additional resampling stage (cf. Section 4). As mentioned above, for the OPFB we have

$$M = Q/I \ , \quad (20)$$

where I is the integer oversampling factor (setting I=1 leads to the totally decimated filterbank structure) [4],[8]. Now we arrive at [cf. (6)]

$$I = Q/M = S_{SC}/B_{SC} \quad (21)$$

with the restriction that I is an integer. We thus get

$$R = \gcd(M, Q) = M \quad (22)$$

and the integer numbers η_i and η_d are

$$\begin{aligned} \eta_i &= Q/R = I \cdot M/R = I \\ \eta_d &= M/R = 1 \end{aligned} \ , \quad (23)$$

i.e. the resamplers in the polyphase branches are now replaced by decimators (DUC) and interpolators (DDC), respectively.

4. DUC AND DDC FOR IS-136

We study the application of the GPFB and its derivative, the OPFB, for designing a multi-channel IS-136 transmitter and receiver system. The IS-136 channel spacing is B_{SC}=30 kHz, the channel baseband sampling rate is S_{SC}=24.3 ksps, and the maximum total width of a frequency band is B_k=15 MHz (max. 500 channels).

Given these parameters, we arrive via (7) and (8) at minimum values of M=700 and Q=567, resulting in S_{MC}=17.01 Msps in case of the GPFB. The filterbank parameters for this case are R=7, η_I=81 and η_d=100. It should be mentioned that the next possible parameter pair would be M=800 and Q=648 with S_{MC}=19.44 Msps, i.e. the filterbank output sampling rate is not very flexible and has to be chosen accordingly. The branch filter in the GPFB simultaneously serves as the pulse-shaping square- root-raised-cosine and image-rejection filter in the interpolation process. It thus has to fulfill the IS-136 emission mask requirements.

If an OPFB is used, then (21) is not fulfilled for the example given above. An additional resampling stage is required to adjust the sampling rates and simultaneously perform pulse shaping. Care has to be taken to fulfill the Nyquist criteria. The root raised cosine filter for IS-136 has a roll-off factor of 0.35, resulting in a baseband channel bandwidth of 16.4 kHz after pulse shaping. It should be noted that the totally decimated filterbank cannot be applied in this case and a minimum oversampling factor of I=2 with S_{SC}=60 ksps is needed, which requires resampling by 200/81. S_{MC}≥15 MHz can now be chosen easily, e.g. S_{MC}=16.5 MHz with M=275 and Q=550. Ideally, the filterbank filter for the OPFB is now a brick-wall image rejection lowpass filter f(n). The overall structure is shown in Fig. 7.

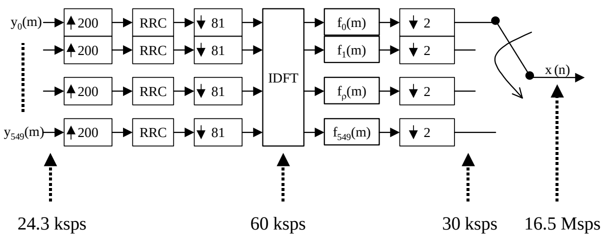

Fig. 7: OPFB in IS-136 DUC

Let us consider the Digital Upconversion (DUC) of 48 IS-136 channels to a 15MHz band. Then, Table I presents the complexity comparison between GPFB and OPFB, measured in terms of millions of complex multiplications per second. IDFT is the dominant computation, and it scales linearly with the sampling rate. IDFT processing for the OPFB operates at 60ksps, whereas IDFT processing for GPFB operates at 24.3ksps. Hence, GPFB is a more computationally efficient scheme than OPFB because the IDFT processing in the GPFB is performed prior to any sampling rate changes (see Fig. 5), whereas in the case of OPFB, IDFT is performed after an increase in sampling rate (see Fig. 7).

Table I

	Resamp. Stage	Mult. Stage	IDFT	Branch Filtering	Total
GPFB	-	1.16	661.34	153.09	**815**
OPFB	23.04	2.88	1584.0	132.00	**1741**

Moreover, it should be noted that the computational complexity of the filterbank can further be reduced, if the base station is part of a cellular network with regular frequency reuse pattern, where e.g. only every 7th channel is used (cf. [2]).

5. CONCLUSIONS

We have presented the mathematical framework for the odd-type Generalized Polyphase Filterbank (OT-GPFB), which can handle frequency offset correction in addition to frequency translation, channel filtering, and re-sampling. Polyphase Filter Banks (PFBs) offer a versatile and flexible approach to realizing multi-channel, multi-standard software-defined radios. Furthermore, we have applied the OT-GPFB to the IS-136 standard, and shown that compared to the Oversampled PFB (OPFB), the OT-GPFB is twice as computationally efficient.

6. REFERENCES

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