POLYPHASE ANALYSIS FILTER BANK DOWN-CONVERTS UNEQUAL CHANNEL BANDWIDTHS WITH ARBITRARY CENTER FREQUENCIES DESIGN I

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ABSTRACT

Cognitive radios respond to the demand to optimize the use of accessible radio spectrum. On the transmitter side a cognitive radio has to be able to detect the available white spaces in the spanned frequency range and to use them to send signals at randomly located center frequencies with variable bandwidths. On the receiver side the radio has to be able to filter and down-convert these same signals. In this paper we present a novel analysis channelizer for cognitive radios based on a variant of the standard channelizer engine. We modify the standard structure of the polyphase M-to-1 down sampling channelizer to perform M/2-to-1 down sampling. We post process and combine the output ports of the modified channelizer with small synthesis channelizers to assemble multiple wide band signals spanning variable bandwidths with arbitrary center frequencies. Perfect reconstruction filters in the synthesis channelizers guarantee seamless assembly of channel segments forming the resulting super channels.

1. INTRODUCTION

A digital receiver has to perform the first tier tasks of filtering, spectral translation and signal conversion to reverse the first tier processing tasks performed at the transmitter.

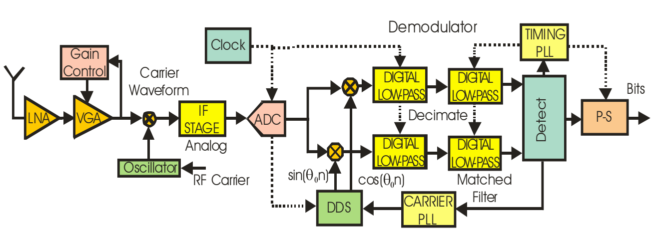

Figure 1. Block Diagram of Primary Signal Processing in a Typical Digital Receiver.

The receiver must also perform a number of second tier tasks not present in the transmitter. These tasks are performed to estimate unknown parameters of the received signal such as amplitude, frequency, and timing alignment. Figure 1 shows the block diagram of first and second tier processing of a typical digital receiver. The digital receiver samples the output of the analog intermediate frequency (IF) filter and down converts the intermediate frequency centered signal to base-band with a digital down converter. The base-band signal is down sampled by a decimating filter and finally processed in the matched filter to maximize the signal-to-noise ratio of the samples presented to the detector. The digital signal processing (DSP) portion of this receiver also includes carrier alignment, timing recovery, channel equalization, automatic gain control, signal-to-noise estimation, signal detection and interference suppression blocks.

Because the receiver contains analog hardware components the receiver incorporates a number of third tier digital signal processing compensating blocks to suppress the undesired artifacts formed by the analog blocks. To simultaneously down convert multiple signals to baseband using the receiver structure shown in Figure 1, it is necessary to replicate its sampled data processing section. Here is the main limitation: the more signals we have to simultaneously down convert, the more sampled data sections we need to implement!

In this paper, we propose a novel analysis channelizer for the receiver of a cognitive radio. By using it we avoid the need to replicate the sampled data section of the receiver. It is sufficient to use only a single partitioned prototype filter followed by an IFFT block to simultaneously demodulate all the received signals. Its structure is based on the polyphase channelizer [1] [2] and it is able to process all channels in the collected spectral span. Post processing will assemble, from the channelizer base-line channels, arbitrary bandwidths randomly located in the spanned frequency range.

In Section 2 of this paper, after a brief review of the standard M-to-1 polyphase channelizer down converter, the variation that makes forms novel channelizer receiver structure is proposed. In Section 3 we introduce the first of two signal processing tasks. This task is performed by an M/2-to-1 polyphase down converter channelizer. In this sub-section we also specify the Nyquist low-pass prototype filter required to support the second task. In Section 4 we present the block diagram of the post process synthesis block explaining the signal up sampling task along with the frequency shifting and reassembling processes. In Section 5 we show simulation results demonstrating the effectiveness of our proposed channelizer variant while in Section 6 we present our conclusions.

2. ANALYSIS CHANNELIZER DESIGN

In its most common incarnation, a polyphase down sampling channelizer simultaneously down converts and down samples M equally spaced, fixed bandwidth signals. Figure 2 shows the complete structure formed by an M-port commutator, an M-path partitioned low-pass prototype filter and an M-point inverse discrete Fourier transform (IDFT) block. For computational efficiency the IDFT is implemented with the IFFT algorithm.

In this engine, the commutator delivers M consecutive samples to the M input ports of the M-path filter performing the signal sample rate reduction which causes M spectral folds in the frequency domain. With an output sample rate of f_S/M, all M multiples of the output sample rate alias to base band (DC). The alias terms in each arm of the M-path filter exhibit unique phase profiles due to their distinct center frequencies and to the time offsets of the different down sampled time series delivered to each commutator port. In particular, each of the aliased terms exhibits a phase shift equal to the product of its center frequency k with its path time delay r. These phase shifts are shown in Eq. (1)

$$\varphi(r,k) = -\omega_k \Delta T_r = -2\pi \frac{f_s}{M} k r T_s = -\frac{2\pi}{M} rk \qquad (1)$$

where f_S is the sample rate at the input to the polyphase down converter. The partitioned M-path filter aligns the time origin of the sampled data sequences delivered by the input commutator to a single common output time origin. This task is accomplished by the all-pass characteristics of the M-path partitioned filter that apply the required differential time delay to the individual input time series. Finally the IFFT block performs the equivalent of a beam-forming operation; the coherent summation of the time aligned signals at each output port with selected phase profile.

The phase coherent summation of the outputs of the M-path filters separates the various aliases residing in each path by constructively summing the selected aliased frequency components located in each path, while simultaneously

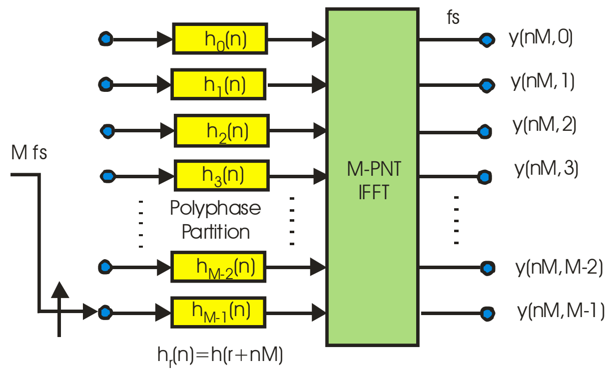

Figure 2. Standard M-Path Polyphase Demodulator: M-Port Commutator, M-Path Polyphase Filter and M-Point IFFT.

destructively canceling the remaining aliased spectral components. The IFFT block extracts, in each arm, from the myriad of aliased signals only the alias with the particular matching phase profile. Summarizing, we can describe the three basic operations performed by a standard polyphase down converter as: sample rate change, due to the input commutator; bandwidth reduction, due to the M-path partitioned filter weights and Nyquist zone selection, due to the IFFT block. These three operations are completely independent of each other and they can be modified to achieve different goals for both of the receiver and transmitter. The standard channelizer is critically sampled with channel spacing, channel bandwidth, and output sample rate equal to f_S/M. This causes the transition band edges of the channelizer filters to alias onto itself which would prevent use of the next processing step. Our first change to the basic channelizer is to double the output sample rate without changing the channel bandwidth or channel spacing. In doing so the sample rate will be twice the bandwidth of the prototype filters as well as twice the channel spacing [5].

Figure 3 shows the complete block diagram of the proposed analysis channelizer. It is composed by an M/2-to-1 down sampler channelizer using a Nyquist prototype filter, followed by N synthesizer blocks. These are 2-to-P_n up converter or synthesis channelizers. The IFFT size, P_n, is chosen to span the bandwidth of the n-th received signal spectrum. It has to span enough input channel bands to form an output sample rate of two samples per assembled bandwidth. We explain the reason for including the synthesizers in our design in this way: at the output of the M-to-2 down converter channelizer all the signals have been down sampled and their frequency bands have been translated to base-band by the aliasing channelizing process.

In order to assemble the channel segments into a wider bandwidth super channel the time series from each segment must be up sampled and frequency shifted to their appropri ate position and then added to form the time series corresponding to the wider bandwidth assembled channel.

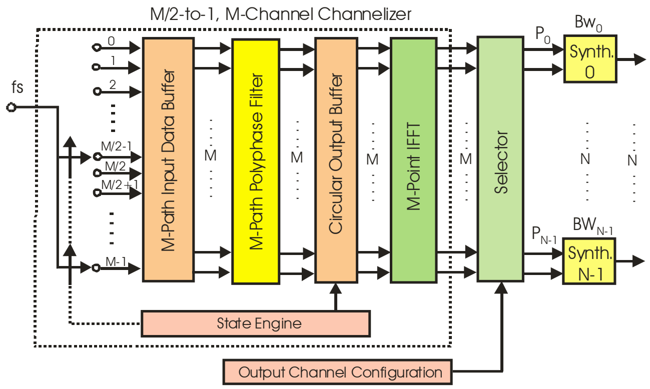

Figure 3. Proposed Modified Analysis Channelizer.

These are the functions performed by the synthesizers following the first down sampler channelizer. A channel configuration and sub-channel alignment block provides the necessary information to a selector block that is connected to the synthesizer output series. The selector routes all the segments required to assemble the wider bandwidth channel to the synthesizer that performs their reassembly.

Depending on the center frequency of the assembled spectrum a further filtering and frequency shifting block may be placed after the synthesizers to shape and properly frequency align and band limit the assembled signal.

3. M-to-2 CHANNELIZER DOWN CONVERTER

In this section we present the reconfigured version of the standard M-path polyphase down converter channelizer. It is capable of performing the sample rate change from the input rate fs to the output rate 2fs/M. The choice to use an M-to-2 down sampler offers the advantage that it avoids the spectral folding at the channel band edge for channel widths equal to the channel spacing. We have derived the form of the M-to-2 down sampler channelizer in an earlier paper [7]. In this paper we briefly explain its block diagram that is shown in the first processing block of Figure 3. More detailed references are found in [7]. The input data buffer performs the correct data loading of the M/2 input samples into the M-path filter while the circular output buffer performs the time alignment of the shifting time origin of the input samples in the M-path filter with the non-shifting time origin of the phase rotator outputs of the IFFT. The M/2 time sample shift of the input time series redefines the time origin and causes sinusoids with an odd number of cycles in the length M array to alternate sign on successive shifts. The alternating sign is the reason that the odd indexed frequency bins alias to the half sample rate while the even indexed frequency bins to alias to DC. Rather than reverse phase of alternate output samples from the odd indexed bins we perform an M/2 point circular shift of alternate M-length vectors before presenting the vector to the IFFT. The circular shift applies the correct phase alignment to all frequencies simultaneously. Of course, there is not a circular shift of the buffer but rather an alternate data load into the IFFT input buffer. Particular attention is paid in designing the low pass prototype filter used in the modified analysis channelizer. The received spectra are randomly located in the frequency domain and their bandwidths can easily span and occupy more than one base-line channel. We need to collect and process all spanned channels corresponding to a single input bandwidth and assemble them without processing artifacts at the output of the synthesizer block. The SQRT Nyquist filter presents the interesting property that its band edge gain is 0.707 (or −3 dB). By using this filter as the low pass prototype in our channelizer, we place M of them across the whole spanned spectrum with each filter centered on k fs/M. All adjacent filters exhibit −3dB overlap at their band-edges. The channelizer working under this configuration is able to collect all the signals energy across its full operating spectrum range even if signals occupy more than one adjacent channel and resides in the channel's overlapping transition bandwidths Note that the SQRT Nyquist prototype low-pass filter has to be designed with its two sided 3 dB bandwidth equal to 1/M-th of the channelizer input sampling frequency. This is equivalent to the filter impulse response having approximately M samples between its peak and first zero crossing and having approximately M samples between its zero crossings. The integer M is also the size of the IFFT as well as the number of channels in the M-to-2 channelizer. The prototype filter must also exhibit reasonable transition bandwidth and sufficient out of band attenuation or stop-band level. We designed our system for a dynamic range of 80 dB, the dynamic range of a 16-bit processor. To obtain a reasonable filter length that satisfied the design requirements we used the *Remez* algorithm rather than MATLAB's *rcosine* routine to design the low-pass prototype SQRT Nyquist filter.

The choice of M determines the spectral resolution of the channelizer. If two or more spectra fall in the same channel bandwidth, a post processing task, by a second pass channelizer, is necessary to separate them. By designing the channel spacing to accommodate the most likely expected channel width we minimize this problem. The optimum choice of M depends on the channelizer's application.

4. SYNTHESIZER CHANNELIZER

At the output of the first M-to-2 channelizer all the signal bandwidths less than or equal to the channel spacing have been aliased to base band. For signals with bandwidths less than channelizer channel bandwidth we have to filter and resample with arbitrary interpolators to obtain desired output sampling rate of 2-samples per symbol bandwidth. We expect that many of the spectra we presented to the first channelizer in the previous section have bandwidths which are wider than the channel spacing. The channelizer parti-

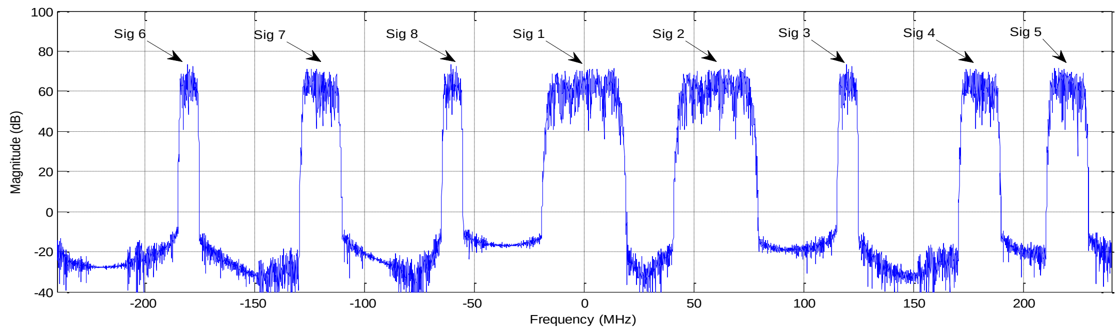

Figure 5. Received Signal Spectrum.

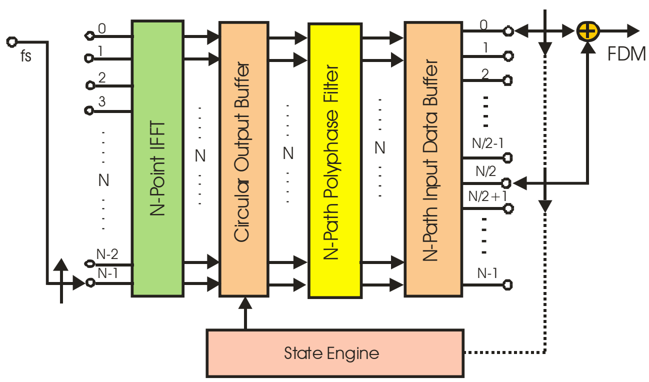

Figure 4. Synthesis Channelizer Structure.

tioned their bandwidths into several fragments and aliased every segment to baseband. In order to recombine the fragments we must first up sample each input time series and second translate them to their proper spectral region. We can then form the sum to obtain the super channel representation of the original signal bandwidth. This is exactly the task performed by the synthesis channelizers. It is designed as a 2-to-P_n up converter channelizer where P_n is approximately twice the number of base-line channels spanned by the wideband spectrum. Figure 4 shows the structure of the channelizer. We note that it is a smaller version of the dual processing tasks performed by the input channelizer. The circular buffer block performs the phase rotation alignment of the IFFT block with the N-path polyphase filter while the N-path data buffer is necessary for the correct data delivering and merging from the N-path polyphase filter output. More details on this structure are in [6] and [7]. The IFFT size for the synthesizer is P_n, this is also the number of the filter paths and it needs to be carefully chosen. In order to recombine the segmented signal components we have to satisfy the Nyquist criteria for their sum. Since this is a receiver channelizer we elect to have the output sample rate near two times the output symbol rate. We can achieve this by selecting P_n to be approximately twice the number of channels being merged in the synthesizer. An actual system may have a few standard size IFFT's to be used for the synthesis channelizer and the user may have to choose from the small list of available block size. The block sizes must be even to perform the 2-to-M resampling by the technique described here. The channel selector placed between the IFFT output and the synthesizer bank also connected with the channel control and sub-channel alignment block will provide the correct zero extended base-line inputs to the synthesis channelizer.

5. ANALYSIS-SYNTHESIS DEMONSTRATION

In the simulation results shown in this section we are considering a composite input signal to the M-to-2 down converter channelizer that contains eight randomly located QPSK signals with three different bandwidths as shown in figure. 5. The signal is sampled at 480 MHz. The bandwidths chosen for these signals are 7.5∗1.25MHz, 15∗1.25 MHz, and 30∗1.25 MHz. The signals are shaped by SQRT Nyquist filters with 25% excess bandwidth. The IFFT transform size of the first M-to-2 down converter channelizer is M = 48 with an output sample rate per channel of 20 MHz. We have 48 inputs to the selector block that, controlled by a channel control block, routes them to the selected synthesis channelizers. Because in this example we are using three different bandwidths we could have three different synthesizers. However, because one of the processed bandwidth is smaller than the channel spacing that for this example is 10 MHz, we have no need to deliver these signals to a synthesizer. The narrowest of three bandwidths of the eight shown in figure 5 are already adequately channelized at the output of the first channelizer; the only task to perform is to resample them in order to achieve the desired 15 MHz output sampling rate. We do have need to process the five remaining bands. The IFFT sizes P_n with n = 0 and 1, we selected to process the other two bandwidths are: 4-PNT for the biggest one and 6-PNT

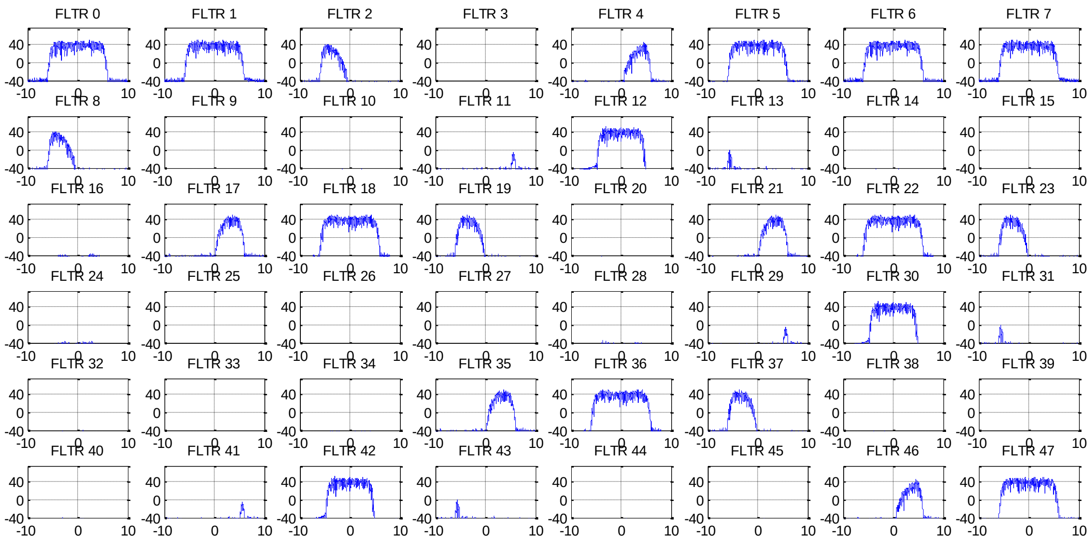

Figure 7. M-to-2 Down Converter Channelizer Outputs.

for the other one. These sizes are the minimum possible that provide at least two samples per symbol. Note that for the 15∗1.25 MHz bandwidth signals, the minimum IFFT size to get two samples per symbol is 3 but, because of the structure of the synthesizers, we can only have an even number of IFFT points [6],[7]. For that reason we choose this number to be 4. It is the smallest possible even index giving us at least two samples per symbol. The sampling rate at the output of this synthesizer is 40 MHz so we simply resample the signals to achieve desired 30 MHz output rate.

Figure 6 shows the impulse response and the magnitude response of the designed prototype low-pass SQRT Nyquist filter. It is designed to be 48 samples per symbol. Its length is 1200 taps while its stop band attenuation is -80dB. Note that, since this filter is M-path partitioned, the length of each filter in the M-path bank is only 25 taps.

Figure 7 shows all the spectra of the output time series from the 48 channels of the polyphase M-to-2 analysis channelizer engine. The down converter polyphase channelizer has partitioned the entire spanned frequency range into 48 segments. It is easy to recognize in this figure the different spectra composing the received signal. Here we see that filters 4-through-8 are segments of a single frequency band. At this point, we up sample, frequency shift, and recombine the time series from coupled channel outputs using the synthesis channelizers.

The spectra of the eight channels, including six that have been de-fragmented are plotted in Figure 8. This is the Humpty-Dumpty story with a happy ending: we can put all the pieces back together again! We also matched-filtered each of the eight channelized and reconstructed time signals

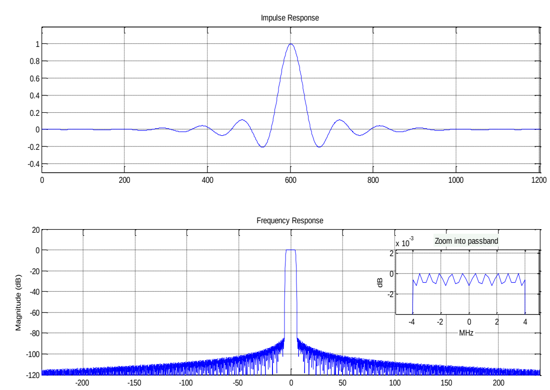

Figure 6. Nyquist Low-Pass Prototype Filter.

and present their constellations in Figure 9; here we see that all of the QPSK constellations shrink to one point demonstrating the correct functionality of the proposed channelizer.

6. CLOSING COMMENTS

In this paper, we presented a novel analysis channelizer to channelize multiple signals showing arbitrary bandwidths and randomly located center frequencies. The core of the proposed structure is a polyphase channelizer engine. It is used, at first time, to down sample and down convert the composite received spectrum. A channel configuration block communicates with the selector that delivers the right inputs to the synthesizers that follow it. These synthesizers upsample, translate and reassemble previously fragmented spectral segments. A Nyquist prototype filter is used to al-

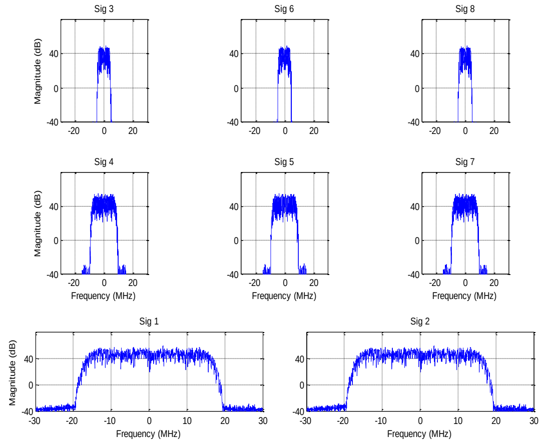

Figure 8. Synthesizer Outputs: Demodulated and Reconstructed Spectra.

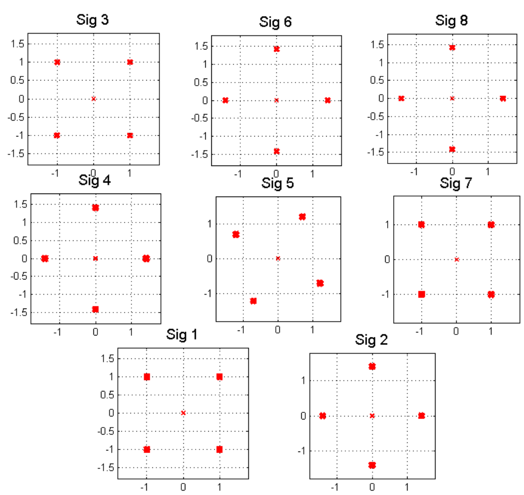

Figure 9. Synthesizer Outputs: Demodulated and Reconstructed Constellations.

low the perfect reconstruction of the assembled signals. The size of the IFFT in the synthesizers is the smallest possible that gives us at least two samples per symbol. Theoretical reasoning and simulation results show that all the received signals are perfectly demodulated.

Note that another structure capable of simultaneously demodulating signals with arbitrary bandwidths and randomly located center frequencies has been presented in [9]. It is based on an M-to-4 polyphase down converter channelizer and the synthesizers are simply adders. We can decide to use either option depending on the application. The channelizer receiver shown in [9] can be applied most efficiently for signals which are composed of spectra having similar bandwidths because in that structure we are using the same under-sampling factor N for all the spectra. It has been chosen in order to give at least two samples per symbol for the largest bandwidth. In the structure presented in this paper, we have a synthesizer for every bandwidth possible and their IFFT sizes are chosen in order to give us at least two samples per symbol for the processed bandwidth. This is the reason why it is very efficient for signals composed of different bandwidths.

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