POLYPHASE SYNTHESIS FILTER BANK UP-CONVERTS UNEQUAL CHANNEL BANDWIDTHS WITH ARBITRARY CENTER FREQUENCIES DESIGN II

Xiaofei Chen (San Diego State University, San Diego, CA, USA; chenxiaofei_sdsu@yahoo.com); Elettra Venosa (Seconda Università degli Studi di Napoli, Aversa, CE, Italy; elettra.venosa@unina2.it); and fred harris (San Diego State University, San Diego, CA, USA; fred.harris@sdsu.edu).

ABSTRACT

Cognitive radios (CR) are a response to the demand to optimize the use of accessible radio spectrum. The CR transmitter must 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. The CR receiver must be able to filter and down-convert these same signals. In this paper we present a novel synthesis channelizer for cognitive radios based on a variant of the standard channelizer engine. We modify the standard structure of the polyphase 1-to-M up sampling channelizer to perform 1-to-M/2 up sampling. We pre process and pre-process the input ports of the modified channelizer with small analysis channelizers to disassemble multiple wide band signals spanning variable bandwidths with arbitrary offset center frequencies. Perfect reconstruction filters in the analysis channelizers guarantee seamless assembly of channel segments forming the resulting synthesized super channels.

1. INTRODUCTION

A digital transmitter has to perform the first tier tasks of filtering, spectral translation and signal conversion which will be matched and reversed by first tier processing performed at the receiver. Figure 1 shows the block diagram of first tier processing of a typical digital transmitter.

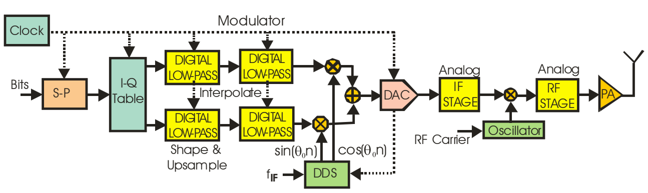

Figure 1. Block Diagram of Primary Signal Processing in a Typical Digital Transmitter.

The digital transmitter accepts binary input sequences and outputs RF amplitude and phase modulated wave shapes. The DSP part of this process starts by accepting b-bit words from a binary source at input symbol rate. These words address a look-up table that outputs gray coded ordered pairs, I-Q constellation points that control the amplitude of phase of the modulated carrier. The I-Q pair is input to DSP based shaping filters that form 1-to-4 up-sampled time series designed to control the wave shape and limit the baseband modulation bandwidth. The time series from the shaping filter are further up-sampled by a pair of interpolating filters to obtain a wider spectral interval between sampled data spectral replicates. The interpolated data is then heterodyned by a digital up-converter to a convenient digital intermediate frequency (IF) and then moved from the sampled data domain to the continuous analog domain by a digital to analog converter (DAC) and analog IF filter. Further analog processing performs spectral transformations required to couple the wave shape to the channel. To simultaneously up convert multiple signals from baseband using the transmitter structure shown in Fig. 1, it is necessary to replicate its sampled data processing section. Here is the main limitation: the more signals we have to simultaneously up convert, the more sampled data sections we need to implement!

In this paper, we propose a novel synthesis channelizer for the transmitter of a cognitive radio. By using it we avoid the need to replicate the sampled data section of the transmitter. It is sufficient to use only a single partitioned prototype filter preceded by an IFFT block to simultaneously interpolate and up-convert all the transmitter signals. Its structure is based on the polyphase channelizer [1] and it is able to process all channels in the output spectral span. Input signals with wider bandwidth than the channelizer bandwidth are easily accommodated. Pre processing by analysis channelizers will disassemble arbitrary input bandwidths with randomly positioned frequency offsets into reduced bandwidth sub-channels matched to the base-line channelizer bandwidths which are reassembled by the synthesis channelizer.

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

2. SYNTHESIS CHANNELIZER DESIGN

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

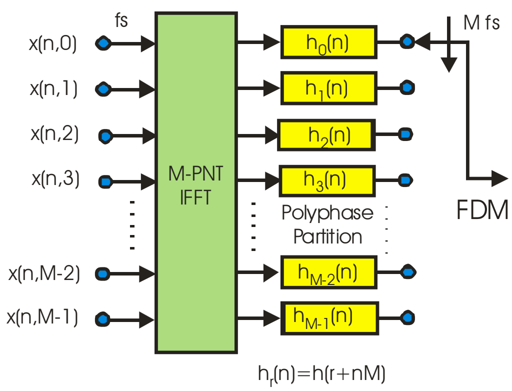

Figure 2. Standard M-Path Polyphase Modulator: M-Point IFFT, M-Path Polyphase Filter and M-Port Commutator.

In this engine, M-point IFFT performs two simultaneous tasks; an initial up sample of 1-to-M, forming an M-length vector for each input sample x(n,k) and further imparts a complex phase rotation of k cycles in M-samples on the up sampled output vector. The IFFT generates a weighted sum of complex vectors containing integer number of cycles per M-length vector. The polyphase filter forms a sequence of column coefficient weighted, MATLAB's dot-multiply, versions of these complex spinning vectors. The sum of these columns, formed by the set of inner products in the polyphase partitioned filter, is the shaped version of the up-converted M-length vector output from the IFFT. The M-port commutator delivers M consecutive samples from the output ports of the M-path filter to deliver the 1-to-M interpolated, up converted and shaped time series formed by the synthesizer channelizer.

Summarizing, we can describe the three basic operations performed by a standard polyphase up converter as: digital up conversion to selected Nyquist zones by the IFFT,

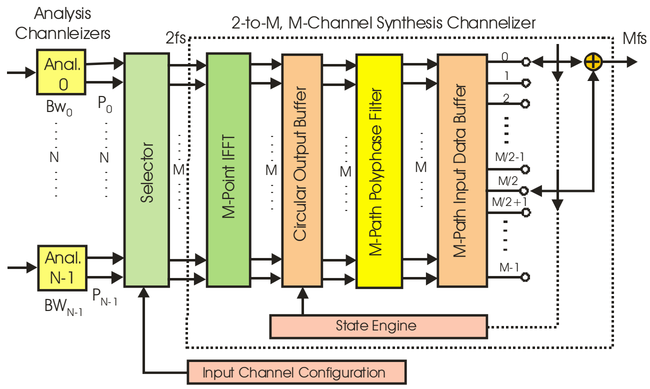

Figure 3. Proposed Modified Synthesis Channelizer.

spectral shaping and filtering due to the M-path partitioned filter weights, sample rate change due to the M-port output commutator. 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 input sample rate equal to f_S. The input signals to the up converter channelizer are shaped external to the channelizer. The sample rate of the shaped signal must of course satisfy the Nyquist criterion. We often chose to operate shaping filters at 2-samples per symbol for ease of signal generation while satisfying the Nyquist criterion. The minimum band center of a channelizer should be matched to the 2-sided bandwidth of the input signal, not to its symbol rate or twice its symbol rate.

To more easily accommodate the channel spacing requirement our first change to the basic channelizer is to require the input sample rate to be twice the 2-sided bandwidth of the input signal rate without changing the channel bandwidth or channel spacing. In doing, so the input sample rate will be twice the bandwidth of the prototype filters as well as twice the channel spacing [5].

Fig. 3 shows the complete block diagram of the proposed synthesis channelizer. It is composed by a 1–to-M/2 up sampler channelizer using a SQRT Nyquist prototype filter, preceded if necessary by N pre-processing analysis blocks. These are P_n-to-2 down converter or analysis channelizers. The IFFT size, P_n, is chosen to span the bandwidth of the n-th input signal spectrum. It has to decompose a wider input spectrum into P_n channels matching the bandwidth and sample rate of the base-line synthesis channelizer which will coherently be recombined in the synthesis channelizer. We explain the reason for including the analyzer in our design in this way: all the signals at the input of the 1-to-M/2 up converter channelizer with bandwidths smaller than channel spacing are to be up sampled and their frequency bands translated from base-band to the selected center frequency by the channelizing process. In order to insert a wider bandwidth signal into the synthesis channelizer we first have to disassemble the wider input channel into segments with sample rate and bandwidth expected by the synthesis channelizer. These segments are reassembled into the wider bandwidth super channel by the synthesizer. A channel configuration block provides the necessary information to a selector block that is connected to the analysis input 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 disassembled spectrum an earlier filtering and frequency shifting block may be placed before the analyzers to frequency shift the about to be disassembled signal.

3. 2-to-M CHANNELIZER UP CONVERTER

In this section we present the reconfigured version of the standard M-path polyphase up converter channelizer. It is capable of performing the sample rate change from the input rate 2fs to the output rate Mfs. The choice to use a 2-to-M up sampler offers the advantage that it permits the input signal to be oversampled by 2 and avoids the difficulty of having the sample rate precisely match two sided bandwidth of the input signals as well as permitting a shorter length prototype channelizer filter due to an allowable wider transition bandwidth. We have derived the form of the 2-to-M up sampler channelizer in an earlier paper [5]. In this paper we briefly explain its block diagram that is shown in the second processing block of Fig. 3. More detailed references are found in [5]. The M-point IFFT applies the complex phase rotation to the separate baseband input signals as well as performs the initial 1-to-M up sampling of the input samples. The circular buffer following the IFFT performs the correct data offset of the two M/2 point halves of the IFFT output vector to maintain phase alignment with the M/2 channelizer output vector. The complex sinusoids output by the IFFT always defines its time origin as the initial sample of its output vector. The output of the polyphase filter exhibits a time origin that shifts due to the M/2 time sample shift embedded in the output commutator. The M/2 time sample shift of the output time series 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 up convert to a frequency k+N/2 rather than frequency index k. Rather than reverse phase alternate input samples to the odd indexed IFFT bins we perform an M/2 point circular shift of alternate M-length vectors from 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 from the IFFT output buffer into the polyphase filter.

Particular attention is paid in designing the low pass prototype filter used in the modified synthesis channelizer. The input spectra are to be placed in randomly located positions in the frequency domain and their bandwidths can easily span and occupy more than one base-line channel. When an input signal bandwidth extends over multiple base-line channel widths it is decomposed by an input analysis filter that spans the number of input channels required to accommodate the wider bandwidth. These segmented channels are reassembled without processing artifacts at the output of the synthesizer block. The SQRT Nyquist filter presents the interesting property that it has a band edge gain of 0.707 (or −3dB). By using this filter as the low pass prototype in our synthesis 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 reside in the channel's overlapping transition bandwidths. Note that the SQRT Nyquist prototype low-pass filter has to be designed with its two sided 3dB 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 and as well as the number of channels in the M-to-2 channelizer. The prototype filter must also exhibit a 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 multiple narrow band signals fall in the same channel bandwidth second synthesis channelizer may be required to combine them. By designing the channel spacing to accommodate the most likely expected channel width we minimize the need for the second signal combiner. The optimum choice of M depends on the channelizer's application.

4. ANALYSIS CHANNELIZER

All the input signals to the 2-to-M channelizer with bandwidths less than or equal to the channel spacing are to be up sampled and up converted from base band. For signals with bandwidths less than channel bandwidth we may have to filter and resample with a second synthesis channelizers to obtain the desired input sampling rate of 2-samples per channel bandwidth. We expect that many of the spectra we presented to the second channelizer in the previous section have bandwidths that are wider than the channel spacing. The analysis channelizers partition their bandwidths into several fragments and aliased every segment to baseband. In order to obtain the proper bandwidth and sample rate fragments we must bandwidth limit each spectral segment, translate the spectral segment to baseband, and reduce sample rate to twice the channel bandwidth. This is exactly the

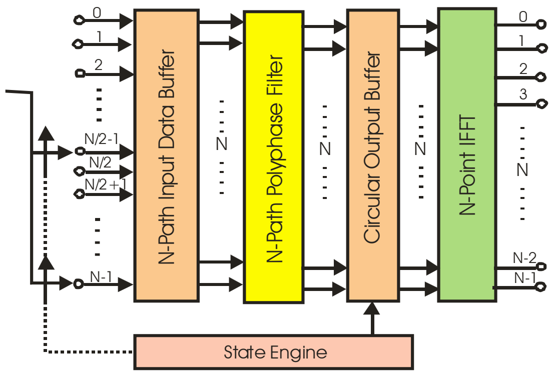

Figure 4. N-to-2 Analysis Channelizer Structure.

task performed by the analysis channelizers. They are designed as a P_n–to-2 down 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 previously described up sampling channelizer. This system accepts N/2 input samples and outputs time samples from N output channels. The N-point input buffer is fed by a dual input commutator with an N/2 sample offset. The N-path filter contains polynomials of the form $H_r(Z^2)$ and $Z^{-1} H_{r+N/2}(Z^2)$ in its upper and lower halves respectively. The circular output buffer performs the phase rotation alignment of the IFFT block with the N/2 stride shifting time origin of the N-path polyphase filter. More details on this structure are in [4], [5].

The IFFT size for the synthesizer is P_n, this is also the number of the filter output channels. In order to recombine the segmented signal components we have to satisfy the Nyquist criteria for their sum. Since this is a pre-processor analysis channelizer that feed the M-path synthesis channelizer we must have its output sample rate two times the channel bandwidth symbol rate. We can achieve this by selecting P_n to be approximately twice the number of channels being merged in the synthesizer and setting its input sample rate to be $2fsP_n$ so that the preprocessor output rate per channel is the required 2fs. An actual system may have a few standard size IFFT's to be used for the analysis channelizer and the user may have to choose from the small list of available block sizes. The block sizes must be even to perform the 2-to-N resampling by the technique described here. The channel selector placed between the analysis bank and the IFFT also connected with the input channel control will provide the correct outputs from the pre processor analysis channelizer to the output synthesizer.

5. SYNTHESIS-ANALYSIS DEMONSTRATION

In the simulation results shown in this section we consider a set of distinct input signals delivered to the 2-to-M up converter channelizer. These are QPSK signals with three different bandwidths as shown in figure 5. The symbol rates chosen for the three signals denoted 1, 2, and 3, are 7.5, 15.0 and 30.0 MHz. The signals are shaped by SQRT Nyquist filters with 25% excess bandwidth, hence the two sided bandwidths are 7.5·1.25, 15.0·1.25, and 30.0·1.25 MHz respectively. The IFFT transform size of the base-line 2-to-M up converter channelizer is M = 48 with 10 MHz channel spacing for which the required input sample rate per input signal is 20 MHz and for which the output sample rate will be 480 MHz.

For ease of signal generation, all three signals were shaped and up-sampled to 60 MHz sample rates with shaping filters designed for 8, 4, and 2 samples per symbol respectively. Signal 1, the first signal, is down sampled 3-to-1 to obtain the desired 20 MHz sample rate for the up converter channelizer. Signals 2 and 3, the second and third signals are down sampled 6-to-2 in 6 point IFFT analysis channelizers which form 10 MHz channels at 20 MHz sample rate. Signal 2 is spanned by three 10 MHz channels which will feed 3 input ports of the 48 point IFFT and signal 3 is spanned by five 10 MHz channels which will feed 5 input ports of 48 point IFFT controlled by a channel control block that routes them to the selected synthesis channelizer ports.

Any of the signals presented to the analysis channelizers can be baseband heterodyned or frequency shifted prior to the down sampling channelizer. The output channels that span the offset input bandwidth of the analysis channelizer are the channels passed on the synthesizer and these change due to a frequency offset. The inserted baseband frequency offset will survive the synthesis channelizer.

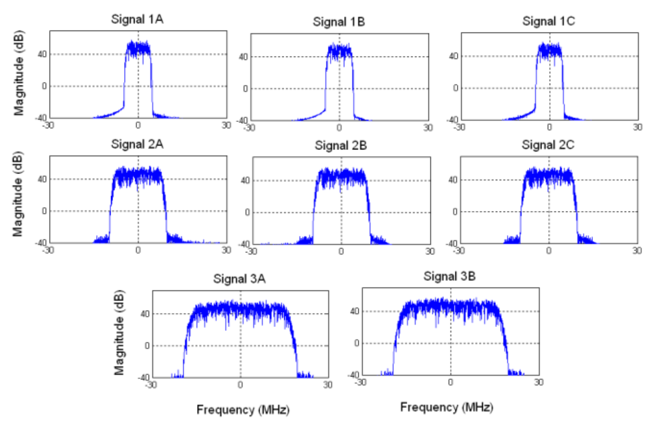

Figure 5. Baseband Signal Spectra.

Figure 6 shows all the spectra of the output time series from the 6 channel polyphase 6-to-2 analysis channelizer engine processing signal 3, the wideband signal. Also seen is the spectrum of signal 3 and the frequency response of the 6 channels formed by the analyzer. Note the spectra in the upper subplots have been filtered by the channelizer 10 MHz SQRT Nyquist pass band frequency response, have been translated to baseband and have been sampled at 20 MHz. Five of these segments are presented to the five input ports of the 2-to-48 synthesis channelizer centered on the desired frequency translation index. The synthesizer filters are also SQRT Nyquist filters which finish shaping the segmented spectral intervals so they have been Nyquist filtered as they are up converted and reassembled by the synthesizer.

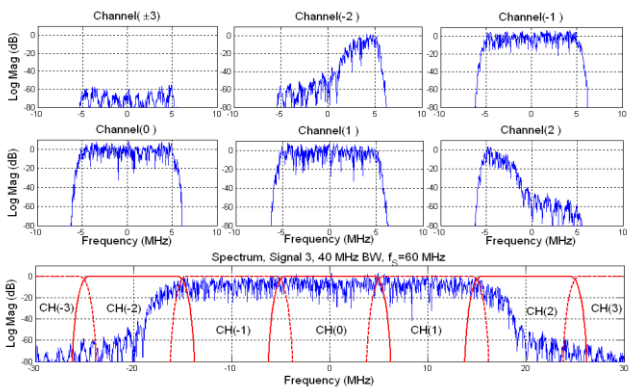

Figure 6. Spectral Fragments Formed by 6-Channel 6-to-2 Down Sample Analysis Channelizer Processing Wide Bandwidth Signal 3.

The up converter polyphase synthesis channelizer accepts time sequences sample at 20 MHz with bandwidths less than 10 MHz. We have delivered three signals that satisfy these constraints along with four signals that were conditioned by analysis channelizers that portioned their bandwidths into segments that also satisfied the input signal constraints. The spectra of the separate components delivered to the synthesis channelizer are shown in Figure 7. The composite signal assembled by this process is shown in Figure 8. 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 fragmented in Figure 6. At this point, we up sample, frequency shift, and recombine the time series from coupled channel outputs using the synthesis channelizer. The spectra of the eight channels, including six that have been fragmented in analysis channelizers and then defragmented in the synthesis channelizer are plotted in Figure 8. This is the Humpty-Dumpty story with a happy ending: we can put all the pieces back together again!

6. CLOSING COMMENTS

In this paper, we presented a novel synthesis channelizer to channelize multiple signals showing arbitrary bandwidths and randomly located center frequencies. The core of the proposed structure is a polyphase channelizer engine.

The engine up samples and up converts separate components with input bandwidth constrained to the channelizer bandwidth. Wider bandwidth signals are pre-processed to fragment their bandwidths to segments acceptable to the synthesis channelizer. The synthesis channelizer up samples, translates, and reassembles the fragmented spectral segments. Use of SQRT Nyquist prototype filters in the analysis and synthesis channelizer assures perfect reconstruction of the assembled signals.

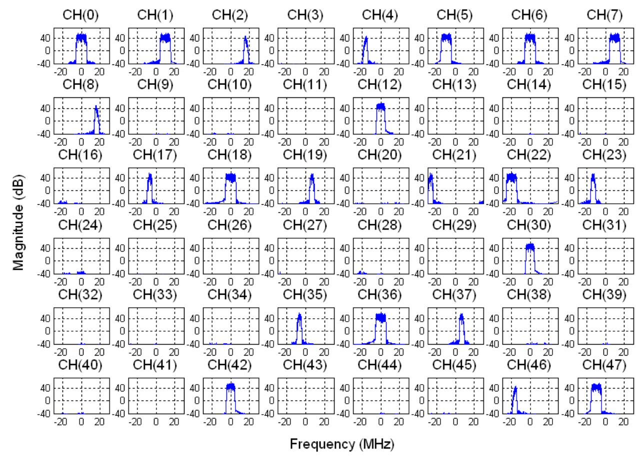

Figure 7. 2-to-M Up Converter Channelizer Inputs.

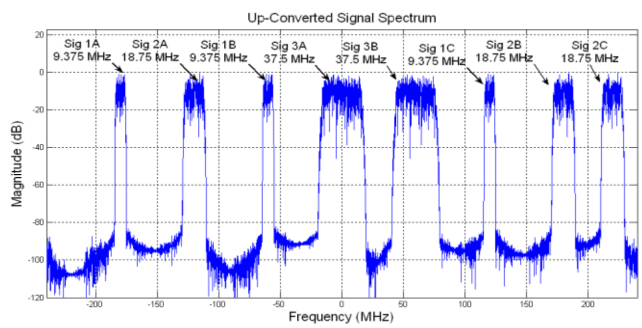

Figure 8. Up-Converter Synthesized Spectrum with Unequal Bandwidth Fragmented and Defragmented Spectral Components.

7. REFERENCES

[1] f.j. harris, "*Multirate Signal Processing for Communication systems*", Prentice Hall, 2004.

[2] f. harris, C. Dick, and M. Rice, "Digital Receivers and Transmitters Using Polyphase Filter Banks for Wireless Communications", Special Issue of *Microwave Theory and Techniques, MTT*, Vol. 51, No. 4, , pp. 1395-1412, April 2003.

[3] f. harris, "Software Defined Radio: A Tutorial," *IEEE Instrumentation and Measurement Magazine*, February 2010.

[4] f. harris, C. Dick, X. Chen, and E. Venosa, "Wideband 160 Channel Polyphase Filter Bank Cable TV Channelizer," Submitted to *IET Signal Processing*, January 2010.

[5] f. harris, C. Dick, X. Chen, and E. Venosa, "M-Path Channelizer with Arbitrary Center Frequency Assignments," *WPMC 2010*, March 2010.

[6] E. Venosa, X. Chen and f. harris, "Polyphase Analysis Filter Bank Down Converts Unequal Channel Bandwidths with Arbitrary Center Frequencies," *SDR Conference*, Washington DC, 30 Nov. - 3 Dec. 2010.