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High Resolution Seamless Linking Between Digital And Analog Radio Reception

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Abstract—**Radio communication has reached to a point where both Digital radios and the conventional Analog radios are sophisticatedly placed in the radio spectrum to coexist under the guidelines of ITU. In many cases, Digital services are the simulcast of existing FM services. Therefore, a moving car experiences a constantly varying coverage area. When the reception quality of the signal from one transmitter becomes weak, the receiver must be able to identify and intelligently switch to another transmitted signal and continue to play the same station with good reception quality. Ordinary digital receivers that are able to do the switching, usually make transient echo like noise or "clicks". Any Digital system introduces a significant processing delay in the receiver due to digital processing. The signal is delayed by 2-4 seconds depending on the decoding circuitry. FM signals are available without this delay. The existing technique for the time delay estimation has achieved the resolution of 1ms. This paper gives the method to accurately estimate this time difference between the signals with the resolution of 0.0625ms in order to obtain perfect synchronization while moving from digital radio reception to analog radio reception.**

Keywords— Discrete Fourier Transform (DFT), Decimation, Low-pass Filtering, Resolution.

I. INTRODUCTION

In the migration to Digital Radio Broadcasting, it may be initially difficult to ensure adequate availability of digital radio receivers that to affordable cost to listeners. In such a scenario, a radio broadcaster will have to incur additional expenditure in order to distribute its programs in parallel on a digital transmitter. Before migrating to the Digital radio broadcasting completely, a well-designed ecosystem and roadmap is necessary for smooth transition from analog to digital broadcasting. When the car travels from Digital Radio reception to analog reception, the switching of audio may not be seamless. In general, the signals from two transmitters may transmit information of same station, but they might have different transmission parameters (like signal delay etc.) or they might use different broadcast systems. Some countries have digital radio broadcasters as a simulcast for existing FM services.

So, in effect, a moving car can experience any of the below situations:

a. Digital Broadcast Area broadcasting one particular station → FM with RDS Broadcast Area broadcasting the same station

b. Digital Broadcast Area broadcasting one particular station → Another Digital Broadcast Area broadcasting the same station

In these above situations, the direct linking between transmitters may lead to non-seamless linking. Even though, broadcasters compensate for the propagation delay between simulcast Digital service and FM service, there will still be a significant delay present between the Digital decoded output and FM decoded output. It is essential to accurately find this time difference between these two signals to obtain perfect synchronization. Therefore, when the car travels from Digital Radio reception to analog reception, the switching of audio is not seamless due to the signal delay between the two transmitters.

II. EXISTING TECHNIQUE

The Seamless linking technique that is already existing is designed to estimate the time delay difference at the resolution of 1 millisecond. When the linking command is received, the audio linking module performs the time delay compensation between Source 1 and Source 2 based on the signal delay measurement and switches the audio output from Source 1 to Source 2. For linking back command, the audio linking module performs the time delay compensation between Source 2 and Source 1 based on the signal delay measurement and switches the audio output from Source 2 to Source 1. There are different modes available in the existing technique for the estimation of time difference between the two sources. Each mode has its own parameters required to find out the actual delay.

Direct linking and linking back between the sources might produce the "pop" or "click" noise. To avoid this noise, cross fade of audio streams technique is applied. Cross fade of audio streams refers to mixing of the end of the main audio source and the start of the alternate audio source to produce seamless audio.

III. SIGNAL DELAY MEASUREMENT

The Time delay measurement between two source signals can be determined by using cross correlation. Most of the Signal Processing literature gives the details on correlation between two signals either in time domain or frequency domain. The choice of time domain or frequency domain is based on the measurement length of the signal. If the measurement signal length is small, time domain method will give faster result over frequency domain method and vice versa. Consider Source1 as *x(n)* and Source2 as *y(n)*:

Cross-correlation of *x(n)* and *y(n)* is a sequence ($r_{xy}(l)$),

$$r_{xy}(l) = \sum_{n=-\infty}^{\infty} x(n)y(n-l) \,, \qquad \text{where } l = 0, \pm 1, \pm 2, K$$

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Correlation is similar to convolution except for time reversal process. So, the above cross correlation equation can be expressed as the following:

$$r_{xy}(l) = x(l) * y(-l)$$

According to properties of convolution, convolution in time domain is equal to multiplication of these signals in frequency domain. Based on this, the cross correlation of two discrete signals x and y in time domain is same as multiplication of Fourier Transform of X with conjugate of Fourier transform of Y.

Cross correlation output=IFFT (FFT(X) x Conjugate FFT(Y))

In this paper, cross correlation in frequency domain is used to measure the time delay between the both signals.

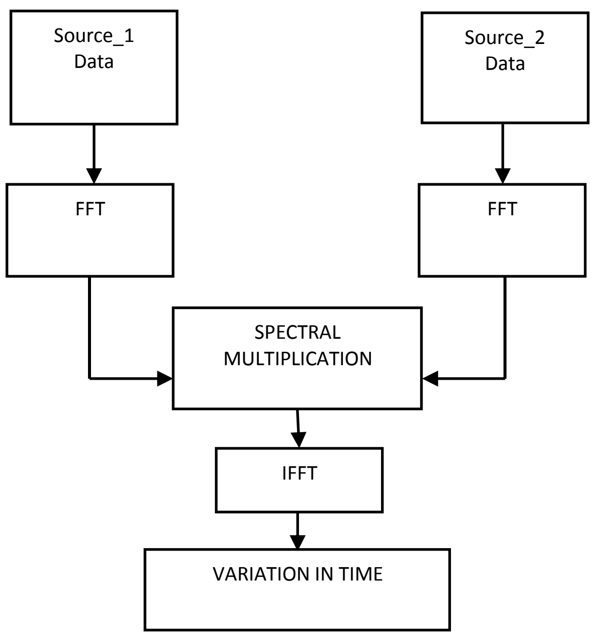

Figure 1: Block Diagram for Time Delay Measurement

IV. PROPOSED TECHNIQUE

Two audio sources namely **DABstream_A.wav** and **DABstream_E.wav** were taken as test inputs and analyzed with the DFT size 8K (i.e.) 8192 for the estimation of time delay between them. The two audio files (say source A and source B) taken as inputs have the following specifications:

- Sampling rate of source A and source B: 48 kHz
- DFT size considered: 8K

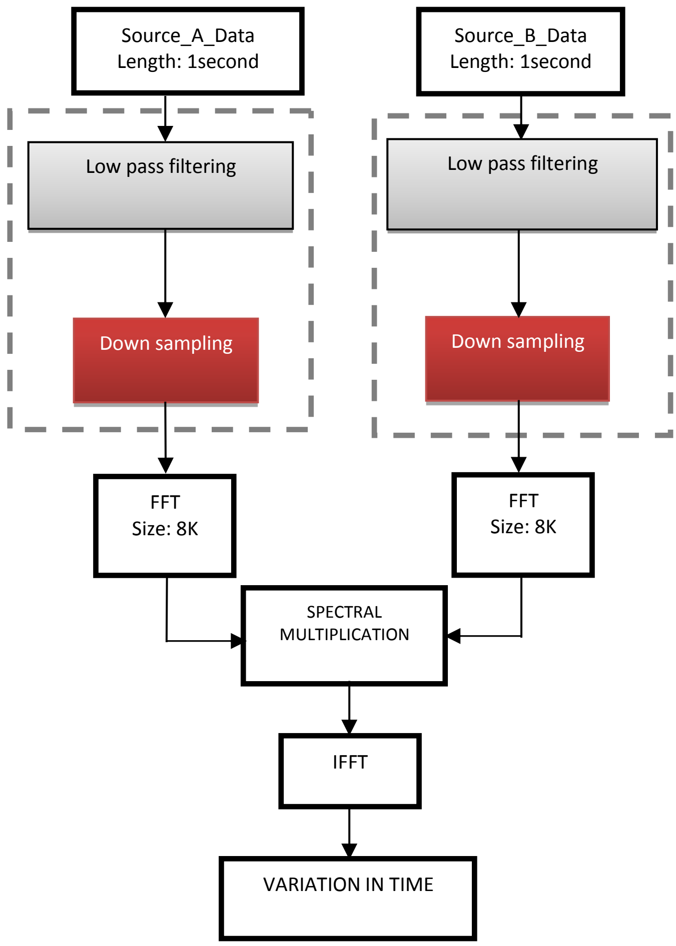

Figure 2: Block Diagram for Time Delay Measurement Between two test signals

A. DECIMATION

Decimation is the process of reducing the sampling rate. This implies low pass filtering a signal, then reducing the number of samples. Signal's highest frequency must be less than half the post decimation sampling rate. So, to remove the high frequency components prior to down sampling, low pass filtering is performed. Decimation reduces the number of samples thus reduces the resolution. Resolution is the smallest unit of measurement or change that can be indicated by an instrument. Here, Resolution refers to minimum delay that can be measured. For any DFT size, resolution is measured at the 1st index. Thus, resolution depends only on the sampling rate.

Resolution = 1st index / sampling rate

For the given two audio sources with sampling rate of 48000 samples per second, if we measure the delay without decimation, the resolution will be 1/48000 (i.e.) 0.020833ms. If the delay between the sources increase (say 3 seconds), DFT size should also be increased, because 8K DFT will not be able to measure the larger delay. To overcome this problem and to achieve the required resolution of 0.0625ms, we go for decimation. So, in order to achieve this resolution, the decimation factor has to be chosen in such a way to satisfy the required resolution. To obtain the required resolution, the decimation factor has to be chosen as 3. If the decimation factor is chosen as 3, the number of samples per second gets reduced to 16000 and thus, the resolution will be 0.0625ms (i.e.) 1/16000.

B. DELAY CALCULATION

After performing the decimation by the factor of 3, the index of the maximum peak occurred at the index 907 as shown in the figure 3. Without decimation, the peak should have occurred at the index 2721 (i.e.) 907xdecimation factor. The output of IFFT may result in many correlation peaks, of which the peak with best correlation has to be chosen for the time delay estimation.

Delay = Index of maximum peak / Sampling rate
= (907/ 16000)
= 0.0566875 second

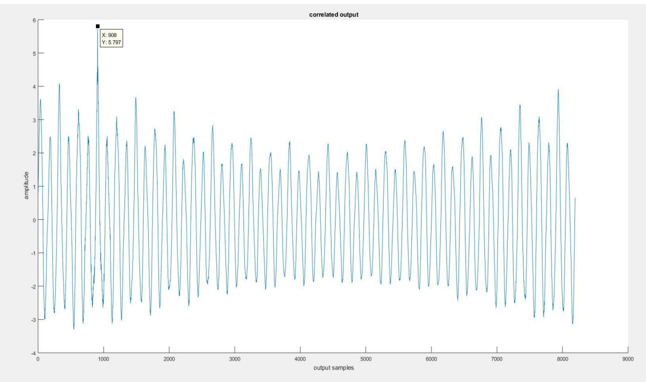

Figure 3: Correlation output between two audio sources

C. MEASUREMENT FOR HIGHER DELAY

If the length of the source is taken as 48000 samples (i.e.) 1 second of data, the length after the decimation will become 16000 (i.e.) 48000 / 3. This length is more than sufficient for 8K DFT, as it takes only the first 8K samples. Therefore, the length of the source taken for measurement is 24000 samples (i.e.) half a second and the length after the decimation is 8000 samples, which can be processed with 8K DFT. The maximum delay that can be measured with the half a second of data as input is 250 milliseconds (i.e.,) 24000 / (2*48000) = 0.25 second. Therefore, if the delay is greater than 250ms, this technique will not be able to estimate the delay. Introducing offsets to the sources will also help us to find out the correct delay between the sources.
The offset value is considered as the timing instance from where the Scanning operation is started.

To overcome this problem, the proposed technique is integrated with the existing package which can measure higher delay between the sources.

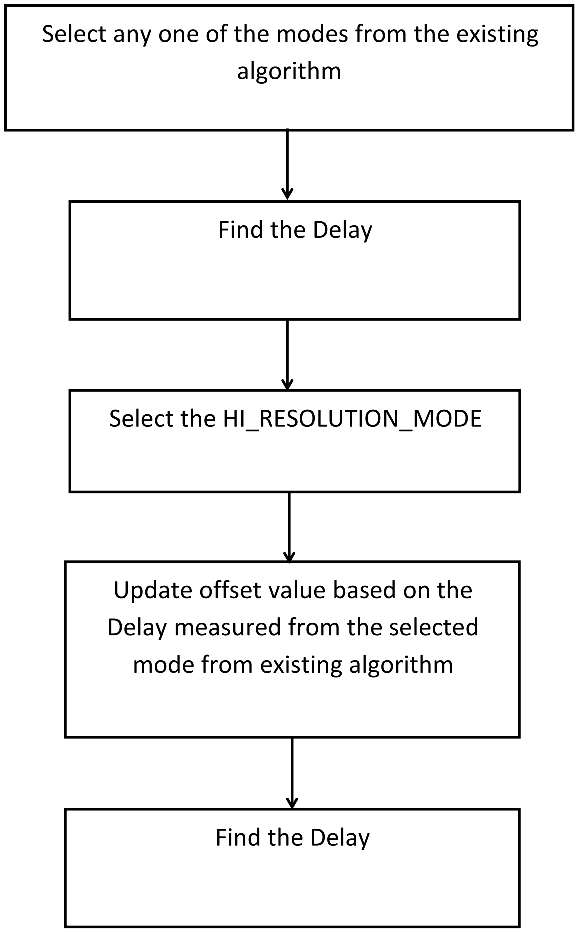

Figure 4: Work flow of HI_RESOLUTION_MODE to measure higher delay

V. RESULTS

The proposed methodology is compiled and executed in C language using Microsoft Visual Studio 2015 and integrated with the existing algorithm.

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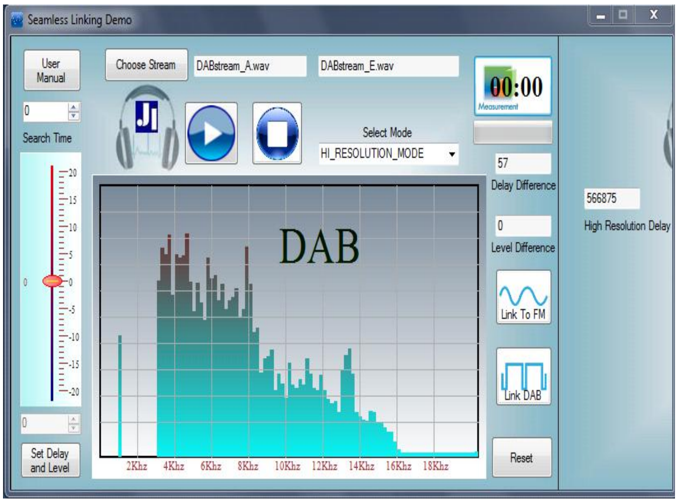

Figure 5: Results obtained from both existing and proposed technique in GUI

In the Figure 5, the Delay Difference box in the GUI gives the time difference calculated with existing technique (i.e.) 1ms resolution and the High Resolution Delay box gives the time difference calculated with proposed technique with the resolution of 0.0625ms.

TABLE I. COMPARISON BETWEEN THE EXISTING MODE AND THE PROPOSED MODE

Test Inputs	Delay measured with Existing Mode	Delay measured with Proposed Mode
DABstream_A.wav and DABstream_E.wav	57ms	56.6875ms
DABstream_A.wav and DABstream_B.wav	4159ms	4158.6875ms
DABstream_A.wav and DABstream_C.wav	1550ms	1549.5625ms
DABstream_A.wav and DABstream_D.wav	1010ms	1010.0625ms

The proposed algorithm is tested with different sets of inputs and the above Table I. shows the comparison of the estimated delay between the existing and the proposed mode in the algorithm.

VI. CONCLUSION

To achieve seamless linking between analog and digital radio reception, it is required to estimate the time delay between the sources and compensate the delay on the alternate source before linking.

Our requirement is to estimate the delay with the resolution of 0.0625ms. An algorithm was proposed that estimates the time delay with the required resolution. This proposed algorithm is integrated with the existing Seamless linking package which has the time difference resolution of 1ms. This technique depends on the output from one of the modes from the existing technique to measure the higher delay with the required resolution.

Thus, High resolution Time delay estimation for seamless linking between digital and analog radio reception is achieved at the resolution of 0.0625ms.

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