Chapter 2

Literature Review

The field of time-frequency methods is changing very rapidly with the introduction of many new ideas for a wide range of applications such as speech, biomedical, communication etc. Like all fields and particularly emerging ones, it has a plethora of different motivations. Many applications are reported in the field of radar also, but very few in under water applications. In this chapter, first, some important references for sonar signal processing are presented. These references describe the important processing techniques employed in current sonars. The rest of the chapter gives a detailed account of the previous work done in five types of time-frequency methods, namely Short Time Fourier Transform, Wavelet Transform, Wigner Ville Distribution, Fractional Fourier Transform and Ambiguity Function. For each of these methods, references for their basic theory are first given, followed by applications in allied areas like speech, communication, radar etc. A few references for applications of these TFMs in sonars are also given. In the last section, references for Mellin transform and its application are given.
2.1 Sonar Signal Processing

DSP has revolutionized sonar processing in many ways. It is only during the last twenty to twenty-five years that the impact of modern high-speed digital electronics has been felt in the military sonar manufacturing industry. The result has been a steady transition from analog processing to digital processing and from separately implemented subsystems to a more integrated computer-controlled combat system. Sonar signal processing has been heavily influenced by the commercial availability of powerful processors and algorithmic developments. Cross-fertilization of ideas from DSP applications in radar, speech, communications, seismology and other related fields have greatly benefited sonar processing techniques. These applications require a good deal more than simple borrowing of techniques from other disciplines, due to the unwieldiness of the undersea propagation medium.

References by Knight et al[1], Urick[2] and Winder[3] describe “mainstream” sonar digital signal processing functions along with associated implementation considerations. These, along with additional references by Neilson[4], Burdic[5], Waite[6], Baggeroer[7] and Leon Camp[8] form a good basis to understand sonar digital signal processing. Radar related references by Simon Kingsley[9], Benjamin[10] and Levanon[11] have analogous applications to sonar processing. The matched filtering technique is described in detail by Glisson et al in their 1969 papers[12,13]. All these references talk about the important sonar functions such as Detection, Localization, Classification, Tracking, Parameter estimation, Communications and Countermeasures. Present day sonar systems are extensively based on the methods given in these references. The performances of the new methods developed in this thesis work are compared with some of these conventional methods.

2.2 Time Frequency Representations

The topic of time-frequency methods is one of the modern DSP tools for non-stationary signal processing. Like all fields and particularly emerging ones, it has a plethora of different motivations. Many applications are reported in the fields of speech and image processing, communications, radar etc. These applications are enumerated in the following pages.

The references by Shie Qian and Dapang Chen[14] and Leon Cohen[15] form a good basis to understand joint time-frequency and space-spatial frequency distributions. The
Uncertainty principle is explained by Cohen in his paper[144]. Following sections contain reference for five time-frequency methods, namely Short Time Fourier Transform, Wavelet Transform, Wigner Ville Distribution, Fractional Fourier Transform and Ambiguity Function. STFT is very much part of the evolution of TFM. So, even though STFT has not been explored for sonar applications in this dissertation, its references are also included here.

2.2.1 Short-Time Fourier Transform (STFT) and Gabor Transform

Frequency analysis of the instantaneous rms values of periodic components of rotating machine vibrations as a function of rotational speed is usually referred to as order tracking. The component under test is either run up in speed or coasted down, the latter often for electrical equipment that cannot be run at continuously increasing or decreasing rpm. Order tracking is an important tool in development and diagnostics of many components such as gear boxes, reciprocating engines, exhaust systems, electrical generators and paper mill rollers. Automotive and machinery reliability engineers rely heavily on order analysis for examining rotating machinery. The methods traditionally used were FFT based methods and Vold-Kalman technology. Dennis Gabor developed the technique for STFT, in which he adapted the Fourier transform to analyze only a small section of the signal at a time by windowing[24]. Compared to these existing order analysis methods, TFM s like STFT and Gabor transform are found to be more intuitive and powerful, especially when time-varying harmonic analysis is required[16, 17].

Speech coders and automatic speech recognition systems are designed to act on clean speech signals. Therefore, corrupted speech signals must be enhanced before their processing. Time-frequency methods like Gabor transform show superior speech enhancement performance over traditional speech enhancement method such as spectral subtraction[18].

STFT is used in speech processing to estimate signal parameters, such as group delay of a transmission channel, speech formant frequencies, excitation time, vocal tract group delay and channel group delay[19]. Owen’s and Murphy’s paper describes a method of using STFT with a family of windows for speech processing[20]. The shift-invariance and rotation-invariance properties of STFT are studied in Arikan’s paper[21]. Sean et al have explained the time-corrected instantaneous spectrogram and its applications in their paper[22]. In radar signal processing, high resolution and high velocity are conflicting requirements and signals
must be carefully designed according to the particular needs. Many different methods for signal analysis are clearly desirable in order to understand the properties of the signals and build appropriate receivers. It is found that Gaussian signals enable to improve range resolution at the expense of the velocity resolution or vice versa[25]. STFT can give better performance in ISAR imaging also[26].

2.2.2 Wavelet Transform (WT)

Basic theory of Wavelet Transforms is given in the references 27,28,29,30 and 31. The following references bring out the potential of using Wavelet transforms for transient detection. Athina[32] has proposed a new scheme for detecting transient signal of unknown waveform and arrival time, using Discrete Wavelet Transform, where the presence of a transient is indicated by a peak. By choosing the dilation and translation parameters appropriately, we can control the sharpness of this peak. Studies of Mordechai et al[33] have shown that when prior information regarding the relative bandwidth and time-bandwidth product of the signal to be detected is incorporated into the problem, Wavelet Transform outperforms other methods. Zhen and Peter[34] have done a comparative study of different methods to evaluate their performances in the detection of unknown transients in white Gaussian noise.

TFMs have found application for signal denoising, where the aim is to estimate unknown signals embedded in Gaussian noise. Papers by Donoho and Johnstone explain signal denoising using Wavelet transform[56,35]. They have formalized the Wavelet coefficient thresholding for removal of additive Gaussian noise from deterministic signals. The discrimination between noise and signal is achieved by choosing an orthogonal basis which efficiently approximates the signal with few non-zero coefficients. A given Wavelet function may not necessarily be best adapted to an underlying signal of an observed random process. Furthermore, the reconstruction performance is dependent upon the noise realization. This indicates that a universal Wavelet basis is practically beyond reach, and that further optimization is required. Then, it is necessary to adaptively select an appropriate best basis for denoising. A Das et al have designed 2-band optimal Wavelets for denoising[36]. Weiss and Dixon[57] presents a multi-resolution approach for denoising underwater acoustic signals. Stephen et al[47] have presented the theory of improved transient detection using Wavepacket
transform. Philippe et al[48] have tried to combine two powerful techniques WT and Higher order statistics for detection of transients. The use of both these techniques makes detection possible in low SNR conditions. Next, Christoph et al have used DFT and time-invariant Wavepacket transform for the classification of transients[49]. Chen Xi et al[50] have proposed a Doppler ultrasound analysis method based on Wavepacket transform for embolic detection.

The Wavelet transform has proved to be a useful tool in oceanography and meteorology. This is demonstrated by Meyers et al[37] in examining the dispersion of Yanai waves. The transform modulus clearly reveals the propagation of the different wavelengths across the basin. The narrowing of the range of wavelengths in the western region observed supports the hypothesis that the narrow range of frequencies observed in the western equatorial ocean is a consequence of Yanai wave dispersion. These results could not be obtained using standard Fourier techniques. Wave elevation is an essential parameter in ocean wave mechanics, naval architecture and ocean engineering. The conventional practice is to measure wave height using wave gauges. However, a wave gauge can only give time histories of wave profiles at one location. To measure wave height at many points, we need to employ many wave gauges. These wave gauges themselves disturb the field. Instead, Lee and Kwon[38] talks of a new technique for measuring wave profiles by Wavelet transform using Mexican hat Wavelet which is proving to be a promising technique for detecting 2-D profiles of waves. The Wavelet transform was applied to the video images of the waves. This technique has the potential to provide low cost, high resolution field measurements of wave profiles in the laboratory. One great advantage is that the measuring process does not disturb the wave field at all. Christopher and Compo[39] describes how Wavelet analysis is also being used for numerous studies in geophysics like tropical convection, El-nino Southern oscillation, atmospheric cold fronts, dispersion of ocean waves, wave growth and breaking, and coherent structures in turbulent flows. It has become a common tool for analyzing localized variations of power within a time series. By decomposing a time-series into time-frequency space, one is able to determine both the dominant modes of variability and how these modes vary in time.
Wavelet analysis is also used in image compression, where better energy compaction, multi-resolution analysis and many other features make it superior to the existing DCT systems like JPEG. The new JPEG2000 compression standard uses the Wavelet transform and achieves higher compression rates with less perceptible artifacts and other advanced features. In image compression applications, the conventional DCT-based techniques have been superceded by Wavelet and Wavepacket transform based techniques due to advantages like computational complexity, performance etc[40,41,42,44,45,46].

Cancellation of harmonics interferences in circuits uses notch filters or ANC techniques. But Lijun Xu[51] have shown that Wavepacket decomposition can be used for the same purpose with some added advantages like zero signal distortion etc. For image compression applications, Thomas et al[52] have shown that the conventional DCT-based techniques have been superceded by Wavelet and Wavepacket transform based techniques due to advantages like computational complexity, performance etc. It is perhaps reasonable to claim that the modern era of spectral estimation began with the 1958 book by Blackman and Tukey[53], where they give details of estimating the spectral density function by different methods. Remarkably, these estimators are infact Wavelet-based estimators, using the Haar Wavelet filter. By applying DWT and Wavepacket techniques, better spectrum estimators can be arrived at. Practical signal compression schemes for image and video coding standards have been using signal-independent approximations like FT and DCT. But Wavelets and Wavepacket transforms are now proving to give better results[54]. Sungwook et al[55] have proposed a speech enhancement method based on spectral entropy using adapted Wavepacket transform.

Dragana[58] describes an underwater detection method using Wavelets with adaptive window-length. Zhang et al[59] have succeeded in extracting the main components in helicopter noise using Wavelet analysis. Prior to transient analysis, the transient has to be detected. The onset time and duration of the transient event is estimated by a less complex algorithm. Page Test(or Cusum Test) proposed by Douglas.A.Abraham[60] is a well established procedure for transient detection. Page Test is a parametric signal test for detecting change in probability distribution of a random process with minimum number of
samples of the random process. In order to make this test distribution independent and thus more robust, nonparametric version of Page Test is better[61].

Chunhua et al[62] have developed improved schemes for target detection in sonars using preprocessing by sub-band adaptive filtering using WT, followed by higher order correlation techniques in post-processing. Vignaud’s paper[63] on a Wavelet transform based Relax algorithm is found to be promising in automatic target recognition in radars. A novel approach to complex target recognition using Wavelet decomposition of the radar cross section is introduced by Delise et al. The recognition levels achieved by this method are much better than with conventional methods[64]. The design of the Wavelet transform based frequency classifier by Francisco et al[65] relies on properties of the Wavelet transform, namely the capability of nearly describing a band pass transient signal by a small set of Wavelet decomposition scales, giving rise to a computationally efficient classification scheme. Ground-penetrating radars are being increasingly used for near-surface studies. Currently for noise suppression, τ-p transform is being used. But DWT based methods in Luigia’s paper are found to give better results[66]. Comparison of Fourier and Wavelet expansions in Passive Acoustic Thermal Tomography by Bograchev[67] shows that the Wavelet scheme is more compact. This compactness reduces the number of unknowns in the inverse problem and increases the accuracy of the reconstruction.

Target extraction and estimation are long-standing problems in radar and sonar signal processing. A signal dependent transform is needed to concentrically represent signal components, while the projection of clutter or noise on the transform space is dispersed. Shi Zhuo et al[68] have developed a method using Wavepacket transform to meet this demand. Andre et al[69] have used Wavepacket transform for multipath channel identification problem with very promising results. Wong et al[70] have proposed the use of Wavepacket decomposition in a radar multi-sensor tracking system, in order to reduce data rate and thereby the communication cost.

The paper by Ingrid Daubechies and Wim Sweldon[71] is a tutorial on lifting scheme. They have shown that DWT can be decomposed into a finite sequence of simple filtering steps called lifting steps. This decomposition asymptotically reduces the computational
complexity of the transform. Compact hardware architectures for implementing lifting-based DWT are proposed by these papers[72,73,74].

2.2.3 Fractional Fourier Transform (FrFT)

Namias introduced Fractional Fourier Transform in the field of quantum mechanics for solving some classes of differential equations efficiently[75]. Since then, a number of applications of Fractional Fourier Transform have been developed, mostly in the field of optics. The motivation behind the proposed method is the ability of FrFT to process chirp signals better than the conventional Fourier Transform. FrFT is basically a time-frequency distribution, a parameterized transform with parameter \( \alpha \), related to the chirp rate. It provides us with an additional degree of freedom (order of the transform), which in most cases results in significant gains over the classical Fourier transform. It is well known that in sonar systems, chirp processing can be applied in a number of areas. Some FrFT applications are reported in radars. However, it remains relatively unknown in acoustics. Given below are some applications in different areas, including radars.

Ozaktas et al[76,77] have come up with a discrete implementation of Fractional Fourier Transform. Like Cooley-Tukey’s FFT, this efficient algorithm computes FrFT in \( O(N\log N) \) time which is about the same time as the ordinary FFT. Hence, in applications where FrFT replaces ordinary Fourier transform for performance improvement, no additional implemention cost will occur. A satisfactory definition of the discrete FrFT that is fully consistent with the continuous transform is given by Cagatay et al[78]. This definition has the same relation with the DFT as the continuous FrFT has with the ordinary continuous Fourier Transform.

Luis Almeida[79] has interpreted FrFT as a rotation in the time-frequency plane. This paper describes its relationship with other TFMs such as WVD, AF, STFT and spectrogram, which support’s the FrFT’s interpretation as a rotational operator. Filtering in fractional Fourier domains may enable significant reduction of MSE compared to ordinary Fourier domain filtering. This reduction comes at essentially no additional computational cost because of the availability of the efficient algorithm for computing FrFT developed by Ozaktas et al[80]. New beam forming techniques are essential to increase the spectral efficiency of wireless communication systems. FrFT based beam forming is better than conventional
methods, especially in moving and accelerated source problem, in performance as well as computational complexity[81,82].

Ran Tao et al[83] have explored the properties and applications of periodic non-uniformly sampled signals in FrFT domain. Their results can be used to estimate the chirp rate and the sampling offsets. Jozef et al[84] have developed an original method for constructing the TFM from the squared magnitudes of their FrFT outputs, using alpha-norm minimization by Renyi entropy maximization.

In radar target identification problems, the target is assumed to have rigid body motion. But in real-world situations, a target may have rotating part beside the main body, like a helicopter with a rotor or a ship with scanning radar. Then, it is difficult to extract motion information(Doppler) using conventional techniques. Another scenario is maneuvering targets, such as aircrafts and missiles, where the Doppler frequencies are time-varying. TFMs like adaptive Chirplet representation have shown potential in these two radar applications. References 85 and 86 address the problem of feature extraction from inverse SAR data collected from targets with rotational parts using Chirplet transforms.

Chris Capus[87,88] et al have proposed the short-time implementation of FrFT. STFT variants of FrFT can be implemented in two ways, depending on how the optimum alpha is chosen. The optimum alpha can be selected for the whole data block, or one for each processing block length. These implementations show improvements in time-frequency resolutions with bat signals, linear and non-linear chirps. Individual chirps in a mixture of chirps can be extracted using FrFT by a filtering and reconstruction technique. Both linear as well as non-linear chirps can be extracted by this method.

Hong-Bo Sun et al[89] have employed FrFT in radar signal processing. FrFT is applied in airborne SAR for detection of slow moving ground targets. For airborne SAR, the echo from a ground moving target can be regarded approximately as a chirp signal and FrFT is a way to concentrate the energy of a chirp signal. Unlike WVD, FrFT is a linear operator and do not suffer from cross terms. Moreover, to solve the problem whereby weak targets are shadowed by the side lobes of strong ones, a new filtering technique called clean is used, thereby detecting strong and weak moving targets iteratively.
In complex undersea environments, where a multitude of simultaneous sonar transmissions may exist, it is desirable to identify a received sonar echo based on its point of origin, platform and mission. This capability can help distinguish friendly sonar sources from counter fraudulent transmissions intended to confuse or mislead. The solution done for this problem is to secure embedding of a robust digital watermark in sonar transmissions. The ideal framework for embedding information is the time-frequency transform. Bijan Mobasseri et al[90] have shown the watermark can be recovered from a single ping. Moreover, the watermark survives various channel impairments including noise, seabed clutter and multipath. The same concept can be used for covert undersea communications using biologically occurring signals as cover.

The time delay estimation (TDE) between the reference signal and its delayed version is an important problem in many areas such as radar, sonar, geophysics, biomedicine and ultrasonic imaging. The conventional method of TDE uses the cross-correlation between the reference and the delayed signal, and estimates the time delay by finding the extremum of this cross-correlation. Various other estimators are also proposed in the literature. However, these estimators suffer from severe degradation in performance at low SNRs. These estimators also need some kind of interpolation to obtain sub sample resolution of the time estimate. By using FrFT, an additional degree of freedom is added and it can be exploited to obtain multiple estimates of the time delay, each corresponding to the different angle of the FrFT. The multiple estimates can be averaged to obtain more robust estimates or the estimate corresponding to least error can be chosen if optimum alpha is known a priori[91].

2.2.4 Wigner Ville Distribution (WVD)

Among all the TFMs, the Wigner Ville Distribution, is the most efficient representation, in giving the best resolution in both time and frequency and is independent of any analysis width. However, it has a unique drawback. With multi component signals, WVD exhibits cross terms because of the product terms in its definition, which clutters the distribution, thus making it difficult to interpret. Given below are some references for cross-term removal and applications of WVD.

Ljubisa proposes PWVD for cross term reduction[92]. Pan et al[93] have proposed a signal classification method based on WVD in combination with cross-correlation technique
for the time-frequency representation of vibration signature. This technique is found to be better than the traditional SID technique.

Analytic signal based spectral estimators present no phase dependence for mono-components, but contrary to previous claims, they are not phase invariant for multi-component signals, and perform worse than their real signal counterparts in high noise. Edgar and Richard proposes an analysis of Wigner-Ville spectra of continuous and discrete signals with time-limited windows demonstrating a better frequency concentration and less phase dependence than real and analytic signal Fourier Spectra. The WVD presents accurate frequency estimates for multi-component stationary signals, where cross term interference is attenuated by smoothing the WVD in time (SWVD). It also has an excellent performance in the presence of noise, making it a good alternative to classical spectral estimation approaches. Furthermore, it is especially appropriate for the case of non-stationary multi-component signals due to the good WVD temporal resolution, thus representing a superior spectral estimation technique suitable for the analysis of a variety of physical processes[94]. Wolfgang and Flandrin[95] propose pseudo-WVD for spectral analysis. In general, the corruptive noise is assumed to be additive and Gaussian and is addressed in many works. However, in some situations, the Gaussian assumption of the noise is not valid and therefore alternative analysis techniques are needed. This paper talks of a novel technique to analyze a sinusoid contaminated by additive noise having unknown heavy-tailed distribution. Examples of heavy tailed distributions include Laplace, Cauchy and alpha-stable distributions with alpha less than 2. Conventional Spectrogram method suffers from the low resolution in the time-frequency domain, while the basic WVD suffers from the presence of artifacts for non-linear FM signals and cross terms for multi component signals. But, the robust polynomial WVD (r-PWVD) outperforms the other two methods in terms of artifact suppression and time-frequency resolution for this class of signals[96,97].

Daniela[98] introduces the Wigner Distribution Function (WDF) and its most important properties as a mathematical tool in several areas of signal processing that include signal retrieval, image recognition, characterization of signals and optical systems, and coupling coefficient estimation in phase space. The mathematical formalism can be applied to spatial, temporal or spatio-temporal phase spaces, to coherent, partially coherent or digital
signals, offering a unified view for the analysis of field propagation through various optical systems. The WDF is thus a universal tool. Andrew et al[99] have proposed different implementation methods for generating analytic signals from real signals.

Time-frequency signal analysis methods, including the WVD and spectrogram for time-varying spectral measurement, the cross-WVD for time-varying coherence characterization, and complex demodulation and instantaneous frequency for time-dependent frequency analysis, have been applied to the neuro-physiological signal analysis, including the EEG, the evoked potential, and the blood flow signals. Traditionally, these signals were either interpreted by physicians who were experienced in the observations of many types of Waveforms, or by classical spectral analysis approaches based on the stationary or piecewise stationary techniques. These techniques frequently fail to characterize these Waveforms since the assumption of stationarity was invalid, resulting in unreliable results. However, the time-frequency analysis methods do not assume stationarity and the results are consistent with physician's interpretation. Thus, a bridge has been constructed between modern signal processing techniques and problems in neurosciences. Researchers from both the medical and engineering fields are making joint efforts to solve these problems (100-chapter 23).

Marriage of the ambiguity function and Fourier transform is the foundation of the field of time-frequency representations. Wigner Distributions have shown their effectiveness in classification problems in sonar and radar. Resonant features in echoes scattered from targets insonified or illuminated with short pulses are revealed well by WVD. Its effectiveness in ground-penetrating radar to identify buried land mines is shown by Guillermo et al[101].

It is very well known in the signal processing community that the inherent bi-linear characteristics of the Wigner Distribution introduce cross terms. But a modified technique called XCDWR overcomes this problem and performs as good as a matched filter based detector. WD can be decomposed into terms that contribute to the auto-WD and terms that affect cross-WD, via Gabor transform. By deleting the terms that affect cross terms, we get CWDR, which is entirely free of cross-terms. If Gaussian window is chosen, the CDWR is free of negative components also. This technique proposed by Kadambe et al[102], when applied in sonar detection application showed reduced reverberation effects, which is very
much a problem in active sonar operation. The methods also reduced the dimensionality of the problem.

Stochastic modeling methods like AR and ARMA have earlier been applied for classification in radars. However, they suffer from inaccuracies such as showing inexact spectral widths, which is a prime factor to distinguish weather returns from birds, non adaptability to time-varying clutter spectra etc, despite being prominent in resolving spectral peaks. Krishnakumar et al[103] says that WVD can be used as an better alternative for the classification of radar returns, especially from the view point of spectral widths, though faced with the problem of cross-spectral components.

Signal detection techniques based on WVD and XWVD are shown to provide high resolution in the time-frequency plane. The steps involved are computation of AF, WVD, XWVD and finally 2D correlation[104]. A Gabor time-frequency basis element is used to construct a data-adaptive weighting window to suppress cross term artifacts in the complex AF domain. A double Fourier transform of the weighted complex AF will then yield a filtered WVD function without cross terms. This technique proposed by Lawrence Marple[105] can be applied in radars and sonars. Zhang et al[106] have proposed a novel high-resolution TFM for source detection and classification in OTH Radar systems. For target identification and classification in sonars, high range and bearing resolutions are required. However, a conventional beam former using Fourier transform is inadequate for this purpose because it is not sensitive enough for discriminating small objects on the sea bottom. Imail et al[107] have applied WVD for the above application and achieved higher resolution.

A classic problem in sonar and radar is the detection and localization in time of narrow-band deterministic transient signals of unknown Waveform and short duration that are embedded in a noisy background comprised of quasi harmonic and random components. The transients are typically of low energy buried in additive noise comprised of relatively higher energy narrow-band and broadband components. Detection needs to be achieved without a precise knowledge of the transient Waveform or frequency. Conventional methods like STFT, Gabor transform etc have various shortcomings. These shortcomings can be overcome by using Wigner distribution, a joint time-frequency signal representation that is capable of providing concentrated estimates of non-stationary signals. Since the transient signals of
interest are localized in time and frequency, it is natural to consider it for this application. Further, this method can be used to estimate the envelope of the transient waveform, which in turn can be used in the classification of the transient[108]

2.2.5 Ambiguity Function (AF)

The classical definition of narrowband ambiguity function is given by Woodward[109], way back in 1964. Theory of matched filtering, ambiguity function, AF of different types of waveforms, their velocity tolerances etc are dealt with in these classical papers of active sonar[references 110 to 121]

Joao et al[122] have introduced a new definition of AF, which is much broader, capable of handling the various radar/sonar problems. Rahah et al[123] defines a combined NB and WB AF for signal parameter estimation in active sonars. Saini et al[124] discuss the AF analysis signals of a DTV-T and the use of this signal for radar applications. Zhenbiao[125] has studied the wideband ambiguity function of FM signals for radar and sonar systems. Jourdain et al[126] discusses the feasibility of using large bandwidth-duration BPSK signals for target delay and Doppler measurements in sonar/radar. Ning Ma et al[127] in their paper propose two novel methods for DOA estimation of broadband chirp signals via the ambiguity function.

2.2.6 Fourier Mellin Transform (FMT)

A combined Fourier Mellin transform yields a representation of a signal that is independent of delay and scale change i.e. invariant in translation and scale. Given below are some references for applications of FMT in allied areas as well as radars. FMT yields a signal representation that is independent of delay and scale change. Hence it is a useful tool in speech analysis where delay and scale differences degrade the performance of correlation operations or other similar measures. Hence it will lead to a signal representation that is unaffected by scale changes and time shifts. FMT can therefore be used for detection of LPM signals. Such signals are used by echolocation bats and cetaceans[128]. Most of the machine speech analysis and processing is done on warped spectral representation. Douglas Nelson talks of an efficient method for computing warped representation. When Mellin-Wavelet transform is used, the linear convolution of a warped Wavelet basis element and a log-warped
speech signal produces an un-warped Wavelet like signal[129]. Gabriel and Cohen[130] have investigated how the concept of scale transform can be extended for multi-dimensional signals, and in particular images. Scale transform can be applied in image analysis and denoising. The scaling operators permit the analysis of the local frequency contents of an image at different resolutions.

Scaling of signals is not possible by standard means. Ovarlez et al[131] have developed an efficient algorithm using FMT. The algorithm involves only FFT routines. This implementation allows considering the time-frequency representations as practical tools for the study of broadband signals. Cohen[132] has come up with a proper representation of the scale transform. The basic properties of scale are given and it is treated as a physical variable for obtaining a framework for joint representations. Jurgen et al[133] have proposed a correlation method for scale and translation invariance in pattern recognition. Their method overcomes the problems of sampling and border effects. Russel and Duck[134] have studied the relation between the spatiotemporal characteristics of basilar membrane vibration and single fiber response. Eberhard et al[135] have reconsidered the temporal effects in masking experiments. Their studies suggest that the cause for temporal effects leading to the critical band can be attributed to spectral effects. Pickles and Comis[136] have brought out the relation between auditory nerve fiber bandwidths and critical bandwidths in cat. Hari et al[137] have introduced the notion of scale periodic function. Their studies showed that scale limited signals can be exactly reconstructed from exponentially spaced samples. An LTI system can be warped by processing its input signals with a unitary warping transform. John and Sommen[138] have developed an efficient implementation of this warping transform. This warping is based on a non-uniform sampling theorem.

The usual time-frequency representations corresponding to the group of time-frequency translations are shown to give rise to localization anomalies. Instead, Bertrand and Bertrand have used the affine group as the basic group and the affine covariant joint distributions are considered and it gives rise to time localized signal. This technique can be applied to radar applications[139]. Philip et al[140] have developed a modified Mellin transform for digital implementation and applied to range radar profiles of naval vessels. Their modified DMT algorithm overcomes some of the problems in FMT.
McCue[141] have done a detailed study of audio pulse compression in bats and humans. Altes [142,143] has derived a sonar system for generalized target description and found out its similarities to animal echolocation systems. Also, he has examined the pulse compression phenomena in terms of nonlinear phase function in the frequency domain.

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