

RIDDIM: A RHYTHM ANALYSIS AND DECOMPOSITION TOOL BASED
ON INDEPENDENT SUBSPACE ANALYSIS

A Thesis

Submitted to the Faculty

in partial fulfillment of the requirements for the

degree of

Master of Arts

in

ELECTRO-ACOUSTIC MUSIC

by

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DARTMOUTH COLLEGE

Hanover, New Hampshire

May 23, 2001

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Abstract

The goal of this thesis was to implement a tool that, given a digital audio input, can extract and represent rhythm and musical time. The purpose of the tool is to help develop better models of rhythm for real-time computer based performance and composition. This analysis tool, *Riddim*, uses Independent Subspace Analysis (ISA) and a robust onset detection scheme to separate and detect salient rhythmic and timing information from different sonic sources within the input. This information is then represented in a format that can be used by a variety of algorithms that interpret timing information to infer rhythmic and musical structure. A secondary objective of this work is a “proof of concept” as a non-real-time rhythm analysis system based on ISA. This is a necessary step since ultimately it is desirable to incorporate this functionality in a real-time plug-in for live performance and improvisation.

Acknowledgments

I would like to thank Roger Dannenberg for introducing me to this field and writing me a recommendation for Dartmouth. Arigato gozaimasu to my teachers Jon Appleton, Eric Lyon, Charles Dodge and Larry Polanksy for providing the opportunity to attend Dartmouth and the nurturing environment to develop and produce this work.

Ridiculous amounts of thanks to Hany Farid for signing on to be an advisor and tolerating random emails for help throughout the year.

Major muchas gracias to Ray Guillette for moral support, advice and assistance with the design of *Riddim*'s Graphical User Interface. Cheers to Michael Casey for providing MATLAB demos, suggestions and feedback that ultimately enabled this work. Thanks to John Puterbaugh for technical advice and book suggestions like “Conceptual Spaces”.

Vielen Dank to Barbara Tillmann and Dêkuji to Petr Janata for reviewing the final document on such short notice and giving many useful comments.

Tusen takk to the nordic nightmare for offering suggestions, papers to read and numerous insights on rhythm. Mahalo nui loa to Moki for reviewing the kosherness of my diagrams and generally adding value. And grazie mille to Dee for making the world a safer place for everybody!!

This work is dedicated to my parents and to the loving memory of my grandfathers Samuel Moniruvwe Orife and William Kendall Ford.

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1 Introduction

“Jam sessions, jitterbugs and cannibalistic rhythmic orgies are wooing our youth along the primrose path to Hell!” — The Most Reverend Francis J.L. Beckman in an address to the National Council of Catholic Women, Biloxi, Mississippi, October 25, 1938

“Without music, life would be a mistake... I would only believe in a God that knew how to dance.” — Friedrich Nietzsche

1.1 Some Personal Compositional Perspectives on Rhythm

My interest in understanding rhythm comes from my struggles to write music that incorporates rhythm in fresh and exciting new ways. Frequently while listening, I am deluged with musical ideas for a new work or work already in progress. Many times, I find it difficult to pick out the individual patterns or understand, in real time, how different components of a percussion mix interrelate. For the experienced listener and musician this might be a routine task, but I would argue that such transparency is less common in electronic or dance styles that play with the human perceptual limits (Aphex Twin, Squarepusher, etc) especially where the recording is the only source of information about the underlying musical and rhythmic structure.

On one hand, repeated listenings and intensive study would facilitate a better understanding; however, for the compositional novice, a tool that embodied the skills of ear-training and expert knowledge on rhythm perception that could uncover the hidden structure and more complex inter-relationships between various elements of a mix, would be invaluable.

Another compositional use of such a tool would be to aid mapping out new rhythmic configurations in a more orderly fashion. Rather than employing music theoretic heuristics with ad-hoc trial and error experiments, new patterns could be systematically found by traversing *Rhythmic Spaces*, (defined below). The problem with trial and error is that it is time consuming. Managing the complexity of the interrelationships between patterns and where they reside in the rhythmic space becomes unwieldy as the said patterns accumulate. With a computational tool, such an effort would be markedly simpler especially if rhythmic features are modeled parametrically. Several features that might be interesting to explore are onset timing intervals, accentuation of onsets, overall tempo and timing and grouping hierarchies between elements of a mix. A “rhythmic space traversal” would take a rhythmic pattern with certain of the above features and hold all but one constant while creating new patterns that varied the last feature in some way.

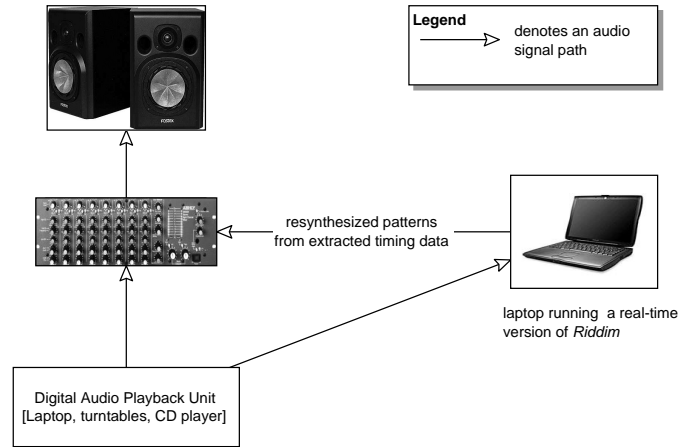


Figure 1: Live performance with Riddim

1.2 Towards a General Perceptual, Transcriptive Computational Tool

A perceptual *and* transcriptive tool is most useful to meet the compositional objectives described above. It needs to be transcriptive because resynthesis can only take place from an abstract representation. It also needs to be perceptual because no score or advanced knowledge about the structure of the music may be available. In this case, any structure the listener perceives is inferred via direct perception. This is especially true of many styles, especially ones practiced solely in oral traditions. Is everything that is needed for a perceptual transcriptive representation of rhythm present in the recorded audio? The information *must* be contained in the structure of the audio waveform because it is this same audio that is processed and understood by the brain.

In the past, researchers have tended to focus on a very specific area of rhythm, be that tempo, beat-induction, metrical interpretation or rhythmic pattern recognition. Only rarely did they ever integrate their work in a framework concerned with the greater experience of rhythm perception. What is attempted here is the development of a tool that explores a more integrative approach. Integrative is used in the sense that the streams of timing sequences extracted from an audio file can be incorporated in existing interpretive models of rhythm, that together can approach a computational correlate of a “rhythmic percept”.

1.3 Thesis Structure

This thesis is structured in the following way. First, I will review the most salient literature on rhythm, by going through its various dimensions, discussing the history and varying opinions. Next I shall discuss my approach to developing

Riddim, a rhythm analysis tool designed to attempt to meet some of the compositional needs described above. This will include some high level algorithmic descriptions of my approach, a brief description of the development platform and how the various components operate. Next, I will explain the workings of each of the algorithms and illustrate why they are important and how they are used, providing detailed diagrams of how data flows through each module.

Finally, I shall show my results which are in-depth rhythmic analyses of a variety of musical recordings using *Riddim*. *Riddim* will be distributed as a stand alone software application from which commands to perform a variety of rhythmic analyses can be executed. Results can then be viewed and saved as MIDI files or rendered as digital audio for use in subsequent creative settings.

2 Rhythm

“Compositional activity proves meaningless in the absence of a corresponding bodily gestalt.” — Ellington’s Law by Warren Burt

“Jazz without the beat, most musicians know, is a telephone yanked from the wall, it just can’t communicate.” — Leonard Feather, British author and jazz enthusiast.

2.1 Understanding Rhythm Through Modeling

A fundamental assumption in this work is that representation and modeling are ways to understand complex real-world processes, in particular, the activity of human perception of rhythm. Theories can then be formulated and verified using empirical data and models, which in turn can be updated to reflect new knowledge. Thus, modeling the cognitive processes associated with rhythm and music in general, can yield better understanding of the functions underlying specific behavioral responses to music and rhythm. This knowledge is potentially useful to composers, performers, designers of codec¹ algorithms, interactive music and media systems [1]. These models embody functional relationships between physical properties of sound and resulting human behavior and sensations. Thus it is important to make distinctions between the physical properties of sound, called *phenomenal*, that can be measured and their perceptual correlates.

¹The term codec stands for “compression/decompression”. A codec is an algorithm, or specialized computer program that does exactly that. Popular audio codecs include MPEG 1 Layer 3 (mp3) and Sony minidisc.

Physical properties of sound	Perceptual correlates
Frequency	Pitch
Signal Intensity	Loudness
Differences in sounds with identical loudness and pitch	Timbre

Table 1: Properties of sound and their perceptual correlates

2.2 Time

2.2.1 Temporal Representations in Music

Music is time-based art form that always occurs in some cultural context. Accordingly, the representation of musical time is culturally dependent. Different cultures have time represented differently in their cognitive structures, for example, in some contexts, time is seen as a circular structure, rather than a continuum [2]. Musical enculturation is thus a combination of perception and the organisation of sounds based on internalised cultural templates [3, p. 5]. Different areas of research also have different perspectives and interests in representing time. For example, while musicologists might be interested in representations that facilitate notation and transcription, composers and performers might be more interested in representations that are designed to work in process-oriented real-time systems [4].

There are several different views of temporal representation in music. These include *tacit*, *implicit* and *explicit* time. Tacit time frameworks do not represent temporal evolution; there is only an idea of “now”. Implicit time structures employ no definitive time relations since time is represented as an absolute, e.g. note lists, while explicit time structures represent time as relations that can be compounded to form higher level notions of time. Other questions that come up in the time modeling process deal with absolute versus relative time

Low Level: Expression	<p><i>Perceptually represented</i> as departures from canonical proportional values; poorly quantified, and experienced as expressive rather than durational effects.</p> <p><i>In Performance</i>, represented as programmed variations in the rate of the clock controlling beat durations (see level 2), and as modifications of the procedures specifying individual notes</p>
Middle Level: Rhythm and meter	<p><i>Perceptually represented</i> as collection of grouped durational equalities and inequalities organized around a metrical framework (when the music has a meter)</p> <p><i>In Performance</i>, represented as a collection of untimed procedures, organized metrical markers which are directly timed by a programmable clock.</p>
High Level: Form	<p><i>Perceptually represented</i> as a structure of hierarchical relations, constructed by means of memory processes and perception, and distinguished from level 2 structures by exceeding the length of the perceptual present</p> <p><i>In Performance</i>, represented as a hierarchical memory structure that forms the highest levels of a motor program</p>

Table 2: A summary of structural levels in musical time based on meter and their cognitive/perceptual properties [5, p. 233].

bases, discrete versus continuous representations and point-based versus intervallic time primitives. Point-based implies that events are duration-less and occur serially while intervallic refers to differences between events that form the basis for meaningful relations (intervals that occur before, after, during) [4].

2.2.2 Causal versus Non-Causal Analysis

A further point regarding time arises when an interpretive process must examine some data structured in time. Some theories propose digesting whole

0 to 2-5 msecs apart	Beats are perceived as simultaneous, indistinguishable (even with different loudness, but same duration) as a single event.
2-5 to 40 msecs apart	Beats are distinguishable, but no order relation can be indicated.
30 to 50 msecs apart	Beats above this can produce an order relation.

Table 3: A table summarising Leigh Smith’s review of Pöppel’s taxonomy of elementary time experiences [3, p. 14]

chunks of data as a unit while others choose to process the data sequentially in time or causally. Opponents of the former process, claim that “theories that behave symmetric with respect to time have to be wrong on that basis alone. ... [those] theories tend to model the perceiver more like a musicologist studying the score, instead of a first time listener” [1]. On the other hand, proponents claim that the non-causality of such models embodies a certain amount of enculturated knowledge that is essential to any interpretation, citing that performers have spent time with their music and have definite intentions before a show [3, p. 55].

Analytically, causal methods can be seen as a subset of the larger non-causal methods, even though the tools may not be related. In any case, because humans infer musical structure based on both universal “hard-wired” low level processing of sensory data (bottom-up processing) *and* culture specific knowledge, expectations and predictions (top-down processing) [6, p. 27], robust models of rhythm perception must account for both bottom-up and top-down processing and cross-cultural variances. This makes room for multiple interpretations in which both causal and non-causal methods are useful.

2.2.3 Specious Present and Auditory Stores

Another aspect of time deals with the interval in which events are processed – the *specious* present. Later called the psychological, perceptual or subjective present, it is related to the duration within which one experiences a sequence of events as simultaneously present in consciousness [7, p. 97][8]. Parncutt defines this perceptual present as a “continuous time interval comprising all real-time percepts and sensations simultaneously available to attention, perception and cognitive processing” [9, p. 451][3, p. 12].

Enabling this present is *echoic* memory, a kind of auditory sensory memory possessing a high degree of structure and organization [9, p. 451]. Echoic memory which is acoustic should be differentiated from *iconic* memory which is visual. There have been many different estimates in the literature on the size of echoic memory and the “integrating buffers”, most of them limiting the echoic memory to less than 500 msec and the subjective present to 4 seconds, with experimental results ranging between 2 and 10 seconds [3, p. 12-3] [9, p. 451]. Parncutt stresses that because the “present is so highly adaptive, no fixed parameter values can be expected to describe it adequately” [9, p. 451-2]. Furthermore, the present also depends on the rate events occur and their complexity and structure [9, p. 452].

2.2.4 Synchronisation

Synchronisation is another aspect of rhythm that emerges from a notion of time. Mari Riess Jones in an article entitled “Time, Our Lost Dimension: Toward a New Theory of Perception, Attention and Memory” states that the “human system in general and the perceptual system in particular depend upon the properties of endogenous rhythmic processes in the nervous system” [10]. Additionally, many studies have suggested the presence of internal clocks and

have noted the strong connection between physical actions and perception [3, p. 17]. In reaction to a sound stream, humans are able to forecast and coordinate motor action to coincide with future events from the sound stream. Smith claims that regularity is less important to the act of synchronisation than the listener’s expectations and ability to predict, since accelerating and decelerating rhythms can be timed [3]. Jones suggests that parts of the synchronisation process are based on temporal expectancy. Thus relations between certain time differences (time deltas) will hold for subsequent ones. This allows the listener to extrapolate time trajectories that predict when events will occur [10].

2.3 Definitions

How should we define rhythm? Smith reports a variety of responses from the literature [3, p. 9-10]. Eric Clarke used the term broadly to apply to “regular, periodic features of the temporal structure of music and to aperiodic features” [11]. Dowling states that rhythm is a temporally extended pattern of durational and accentual relationships, while Parncutt claims that musical rhythm is an acoustic sequence evoking a sensation of pulse [9, p. 451]. Scheirer cites Handel’s claims that “the experience of rhythm involves movement, regularity, grouping, and yet accentuation and differentiation” [12].

According to psychological studies done by Gabrielsson on human “subjects”, rhythm like timbre is a multi-dimensional quality. From the empirical studies, Gabrielsson points out the existence of at least fifteen dimensions which “lend themselves to grouping into three categories” related to structural, motion and emotional aspects of rhythm. Refer to Table 4.

There are problems in modeling the “dimensions” mentioned above because the interpretive ambiguity of the dimensions and characters do not lend easily

Structural Aspects	Meter, position and strength of accents, type and prominence of basic pattern, number of different kinds of subdivisions within beats, uniformity versus variation and simplicity versus complexity
Motional Aspects	Tempo and overall rapidity, however motional characters exist such as walking, dancing, jumping, rocking, swinging, graceful and driving forward
Emotional Aspects	Vitality versus dullness, excitement versus calmness, rigidity versus flexibility and solemnity versus playfulness

Table 4: Gabrielsson’s dimensional categories for rhythm [7, p. 107]

to symbolic or quantitative representation.

2.4 Pulse

Parncutt’s definition of a musical rhythm is an acoustic sequence evoking a sensation of pulse. How should we define pulse? If a car drives by loudly broadcasting music, in the several seconds within which the music is in earshot, the pulse would be the subjective evaluation of the feel or impression of movement inferred from the music. If that music happened to be tabla drumming, Olatunji, Faithless or Subotnick, each would leave a different but firm impression in the mind of the listener. The percept of pulse is confined to the time interval called the subjective present [9, p. 451]. While *rhythm* deals with grouping and time hierarchies and *tempo* involves the rate at which rhythmically significant events occur, pulse corresponds to a “sense of equally spaced phenomenal impulses” [12]. For many kinds of music, the pulse is the “foot-tapping” beat.

Pulse has also gone by other names like *tactus*, defined by Lerdahl and

Jackendoff to describe the most salient hierarchical level at which the listener will tap their foot in accompaniment to a rhythm [3, p. 31]. Parncutt defines it as a trivial sense of expectancy. Once a pulse is established, subsequent events are “expected” with some time trajectory. This is of course related to our discussion above of synchronisation, where motor actions move along time trajectories inferred from this expectation [9, p. 453].

2.4.1 Pulse Outside A Metrical Context

In non-metric musics, the pulse may not necessarily correspond to an isochronous period. Researchers agree that the idea of a basic pulse in such music is “usually misplaced” [13, p. 190] Magill reports that

“West African drum ensemble music takes a multilayered approach to rhythm, with all the parts being related to a fundamental *time line*, often played on a bell... Bell patterns can nearly always be subdivided into a number of regular pulses (usually 8, 12, or 16). Time lines or bell patterns betray asymmetric construction and sound synopated to Western ears” [13, p. 190].

For example, in the *Anlo* drumming style from southern Ghana, Pantaleoni points out that it is “the bell that regulates the play, and not the steady pulse some player or observer might feel, because the bell can put each pattern into its proper places, while a simple pulse can only regulate the speed with which the pattern is played” [14, p. 60]. Magill states that in many West African drumming styles while some instruments reinforce an underlying fast pulse, others might highlight portions of a bell pattern while others provide a complex overlay of rhythms and meters [13, p. 190]. In any case, while regular pulses are quite common, the traditional cultural view is that they are of peripheral importance

compared to the asymmetric time line. The definition of pulse needs to be revised to be a periodic structuring or synchronization framework, since in the kinds of music discussed above, asymmetric time lines and isochronous pulses are both constructs within which higher levels of rhythmic time are structured.

The Fastest Pulse

Attempts to understand non-metric music have also emerged in the form of concepts like the “fastest pulse”. The fastest pulse is defined as the “beat with the shortest duration in the music considered” [3, p. 33] Smith cites Koetting’s warning that the fastest pulse does not seem to describe how African timing is perceived [3, p. 33]. Pantaleoni comments

“I have not found it to be a concept familiar to those African drummer with whom I have worked, nor does it seem to be an easy concept for them to use once it has been explained. Perhaps it is simply difficult for a performer to think as an analyst, but in the absence of some kind of positive support from the musicians themselves it cannot be assumed that a convenient analytical tool corresponds to the basic pulse or principle of timing which actually functions in the playing of the music” [14, p. 58].

While clearly limited from a compositional and improvisational point of view, the idea of the fastest pulse might still be useful in analysis. As will be seen in the next chapter, analyses that determine the fastest pulse, per extracted stream, are a useful interpretive tool.

2.4.2 Beat Induction

Beat induction is a synchronization process with a phenomenal pulse. There have been many successful attempts to measure the pulse computationally in

Western classical and popular music, using that information to drive a foot-tapping process. Eric Scheirer used filterbanks and parallel comb filters to extract amplitude envelopes and infer the beat from arbitrarily complex musical signals [12]. Povel and Essens introduced the idea of “internal clocks” which are selected based on a set of inter-onset interval inputs. Desain and Honing have several computational beat-tracking methods to their name. Given an onset stream, their models return the beat of the stream. Different from the previous methods, Desain and Honing use a causal process model which takes the onset inputs sequentially and updates an expectancy measure. The beat is then inferred from regions of high expectancy. Large and Kolen’s beat-tracking model is based on non-linear oscillators. From a stream of onsets, a “gradient-descent method” is employed to continually update the period and phase of an oscillator, which represents the pulse [15][12]. Goto presented a system which “combines both low-level “bottom-up” signal processing and high-level pattern matching and “agent-based” representation to beat-track and do simple rhythmic grouping for popular music” [12][16]. Leigh Smith has also explored a multiscale representation of musical rhythm. Onset streams or pulse trains are first decomposed using a Morlet wavelet basis. The “foot-tapping” pulse of a section of music is then extracted by correlating frequency modulation ridges extracted from the wavelet decomposition using stationary phase, modulus maxima, dilation scale derivatives and local phase congruency. Refer to [3] for definitions and implementation details.

2.4.3 Tempo

Tempo is defined here as the rate at which rhythmically significant events occur. In music with an isochronous pulse, tempo is the rate of the tactus. In much Western music and popular music, this is the “foot tapping” rate.

In non-metrical settings, there may not be any explicit overall tempo. In this case, there may be multiple rhythmic layers each with its own individual rate. Tempo is however best used in a metrical setting. Commercial software systems express tempo in terms of beats per minute (BPM), while numerous scores have subjective tempo indications like *Larghissimo* or *Prestissimo* or more concrete markings like quarter note equals sixty.

Within the above framework how do variations in tempo affect the interpretation of rhythmic structure? Clarke reports that slow tempi aid the cognitive process of segmentation at structural boundary points while at higher tempi, music tends to be grouped into fewer units. This is claimed to occur because of the limits of the subjective present. In other words, at faster tempi, more elements are packed into the subjective present; thus given fixed maximum neural processing rates, music is subdivided into larger groups in accordance with its structural properties [3, p. 36].

2.5 Grouping

Experiencing rhythm occurs as a whole sequence or pattern, not as individual events. Within the subjective present, how a sequence is formed by segregating a stream of events is called grouping. This can take the form of phrases or motives. In many cases, the number of events grouped together depends on the presentation rate, so the faster the rate, the more members are included in the group. This extends up to perceptual limits. Some rhythmic groups are presented so quickly that they're not perceived as individual rhythms or events but are lumped together to form a pitch percept. Within a sequence of events, certain ones with objective accents or differences suggest or confirm the presence of a perceptual group boundary. Quantities responsible for the percept

of a group boundary include perceptual limits on memory and attention and cognitive representations.

Smith points out two principal grouping principles: The *run* principle and the *gap* principle. The run principle proposes that the longest run of similar events will begin a rhythmic pattern. Listeners tend to group sounds of the same intensity together. Runs of different intensity will be organized so that the longest runs are placed at the beginning or end of the pattern, never in the middle. The gap principle refers to boundaries that are formed between dissimilar elements. Rests partition elements while elements close in time tend to be grouped [3, p. 28]. These principles are based on Gestalt principles of perceptual organization and Bregman’s seminal work on auditory scene analysis [17]. Additionally, higher level grouping structures are bounded by repetitions. A single change in an established pattern can affect the entire grouping percept [3, p. 28].

Parncutt defines two independent kinds of temporal grouping, *serial* and *periodic*. Serial grouping (related to the gap principle) is dependent on the serial proximity of adjacent events in time, pitch and timbre. “According to Lerdahl and Jackendoff (1983), serial grouping in music includes “motives, themes, phrases, periods, theme-groups, sections, and the piece itself” [18]. Periodic grouping on other hand, depends on the relative times and perceptual properties of *nonadjacent* events [9, p. 412].

2.5.1 Meter

Meter can be seen as a form of periodic grouping. Parncutt states that it groups events into equivalence classes, such as all n th beats of a bar [9, p. 412]. He further classifies periodic grouping as being an aggregate of two stages, *pulse sensation* and *perceived meter*. Interestingly enough, Halsband conversely states

that grouping “plays a major role in the perception of all metrically organized music” [19, p. 266].

While grouping is a psychological term, meter is a product of music theory and associated forms of notation. Repetition in meter, as in other groupings, plays an integral role in the determination of a metrical percept. However, the perceived meter in a performance may not necessarily correspond to the notated meter. In many of these cases, some initial metrical orientation influences the interpretation of subsequent sequences.

There are rarely explicit rules to outline an interpretation; in improvisational styles, performances “unbounded by a strict metric frame are not free in the sense of being unrhythmic. Rather, they are driven by rhythmic goals that are elastic” [20, p. 158].

Handel showed empirically that difficult rhythms with atypical metrical accenting were perceived in terms of elemental grouping schemes, rather than as a meter with a timing interval [3, p. 31]. Thus, a metrical percept is a regular alternation of strong and weak beats. What then qualifies an event as particularly strong or weak?

2.5.2 Accentuation

Accents are differences in adjacent events that separate them and suggest particular grouping configurations. Smith cites Lerhdahl and Jackendoff as distinguishing three kinds of accents, *phenomenal*, *metrical* and *structural* according to their effect on groups [3, p. 19]. Phenomenal accents are said to exist superficially, stressing a single moment and enabling syncopations. Metrical accents occur when the emphasised beat is part of a repeating metrical pattern. Structural accents occur at points hierarchically higher than meter. For example, structural accents could delimit phrases or sections. Furthermore,

Well-Formedness Rule 1	Every attack point must be associated with a beat at the smallest level metrical level present at that point in the piece.
Well-Formedness Rule 2	Every beat at a given level must also be a beat at all smaller levels present at that point in the piece.
Well-Formedness Rule 3	At each metrical level, strong beats are spaced either two or three beats apart.
Well-Formedness Rule 4	The tactus and immediately larger metrical levels must consist of beats equally spaced throughout the piece. At subtactus metrical levels, weak beats must be equally spaced between surrounding strong beats.

Table 5: A summary of Lerdahl and Jackendoff’s Metrical Well-Formedness Rules from their “Generative Theory of Tonal Music” [18].

a “phenomenal accent functions as a perceptual input to metrical accent” [18].

Accents occur when there are changes in duration, inter-onset interval (IOI), pitch, articulation, timbre or combinations thereof. Similarly changes in phrase length, time expectancies, event density (flams, fills, etc) or multiple event synchronisation can cause accentuation [3, p. 19] [9, p. 426].

Articulation is one way to signify an accent. Berliner remarks that in the jazz idiom, ghosting “on-beat pitches in an eight-note sequence creates de facto accents on every off-beat, increasing rhythmic tension. Reversing the procedure relieves rhythmic tension... Similarly the varied application of hard, soft and ghosted attacks can shift accents within a repeated eight-note triplet, changing its perceived rhythmic configuration and its relative syncopated quality” [20, p. 156].

Accentuation is not always applicable to specifying grouping structure and its use varies with the musical context. For instance, using the *Anlo* dance

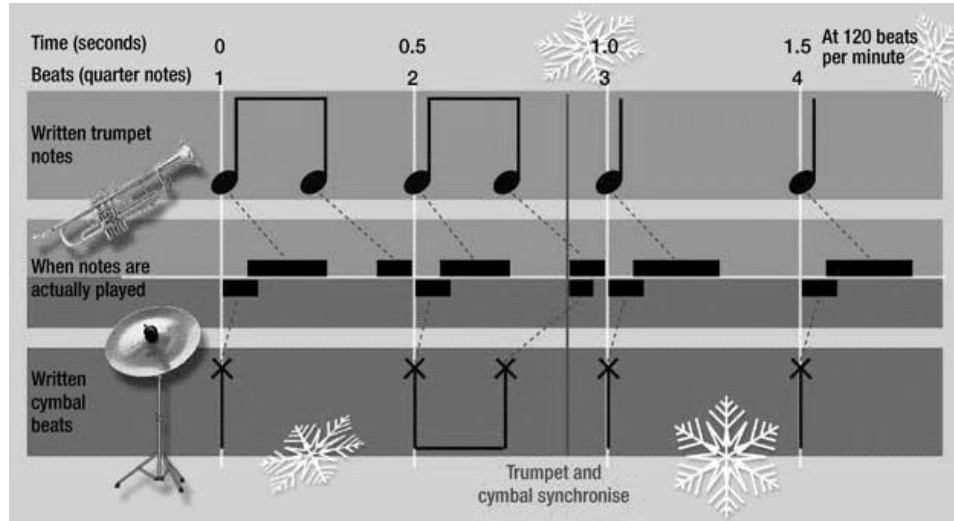


Figure 2: “For a swinging performance the first of each pair of eight notes is played longer than the second. The melody line also hangs behind the cymbal beat, except for the occasional off-beat synchronisation, which keeps the band together” [21]

drumming example, Pantaleoni reports that “variation in loudness (phenomenal accents) is indeed an important part of the character of a rhythmic line in Atsiã, but no evidence indicates that its contribution is more than melodic... Dynamic stress would seem to be individual, decorative, incidental, and often illusory, hardly a likely clue to rhythmic organisation” [14, pp. 54].

2.6 Expressive Timing

Many authors have explored what is known in rhythmic performances as expressive timing. Discrepancies can always be found in the timing dictated by a score and that which occurs in a performance. Barring mistakes, which really are not plausible with highly trained musicians familiar with the music, these deviations are “related to the structural properties of the music and to the ways the performers organize” these properties [11]. In an analysis of Erik Satie’s

“Gnossienne No. 5”, Clarke reports that

“the tempo marking is followed by the instruction “*souple et expressif*”. This is relevant to subsequent analysis, since it suggests that the performance of the piece should avoid metronomic tendencies in the left hand, and may encourage a rhythmic flexibility in the right hand that conventional notation cannot convey” [22, p. 300].

Research has thus focused on determining exactly what deviations from the score contribute to a subjective feeling of expression that is not “metronomic”. First some definitions: expressive timing refers to deviations in notated times that suggest particular structural interpretations or give the temporal development of the music a particular feel or sense of movement. Smith cites certain kinds of expressive timing as having evolved from a performance practice known as *rubato*, literally “robbed time” [3, pp. 37]. In computer music, terms like “micro-tempo” are used to describe this expressive phenomenon of local tempo changing from event to event. In all cases, these expressive timing transformations are used by performers either in response to structural features in the score or in attempt to explore new interpretive configurations or impose a particular structure on structurally indeterminate material [11].

In the jazz tradition, Berliner says that musicians “talk about playing on three different parts of the beat without making any difference in the overall tempo”. On a similar note, Fred Hersch claims that

“there should be ten, fifteen different kinds of time. There’s a kind of time that has an edge on it for a while and lays back for a while. Sometimes it rolls over the bar, and sometimes it sits more on the beats. That’s what makes it interesting. You can set a metronome here and by playing with an edge or playing behind it or right in the

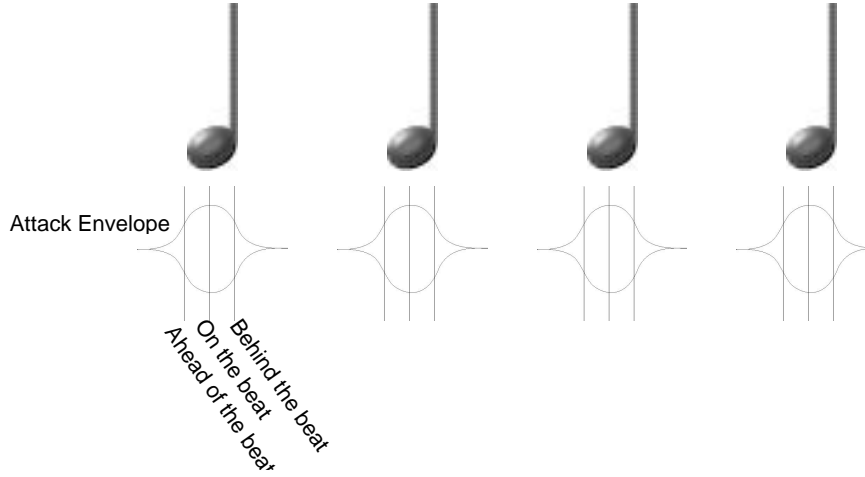


Figure 3: “Imagining the beat as an “elliptical figure”, the drummer or bass player can play either “ahead of the beat” (that is, on the front part of the elliptical figure), “behind the beat”, (that is on the very end of the elliptical figure or in varying degrees toward the center of the figure or “on the beat” (that is, the center of the figure)” [20, p. 151].

center you can get all kinds of different feelings. That’s what makes it come alive. People are human, and rhythmic energy has an ebb and flow” [20, p. 151].

2.6.1 Tempo Curves

Tempo curves are one way that timing deviations are represented in contemporary music software applications for notated music, offering a measure of “beat-by-beat time deviation from a canonical metrical grid” [3, p. 39]. This quasi-instantaneous tempo or *Local tempo* is defined computationally as the event to event ratio of score time interval to performance time interval.

$$Local\ tempo = Global\ tempo \times \frac{Score\ time\ interval}{Performance\ time\ interval}$$

Tempo curves are thus constructed by connecting points with straight line segments between local tempo measures of each performed note in the score [3][23].

2.6.2 Expression and Structure

Desain and Honing warn that while they are useful to study expressive timing, tempo curves can be is a “dangerous notion”. This is because it encourages “its users into the false impression that it has a musical and psychological reality. There is no abstract tempo curve in the music nor is there a mental tempo curve in the head of the performer or listener” [23]. This is because tempo curves by themselves contain no structural information. Thus global transformations, like scaling the overall tempo curve, will change timing relationships between notes that are ornamental and *invariant* with tempo. Furthermore, attempts to impose a tempo curve from the performance of one piece onto the score of another shows that tempo curves convey an erroneous notion that time is independent of the note events that mark it. This clearly ignores the fact that the expression captured by the tempo curve can only function with respect to the original music performance from which it was derived [3][23][24].

For a more in-depth discussion on expressive temporal transformations, refer to Desain and Honing’s paper called “Towards a Calculus for Expressive Timing in Music” [24].

2.7 A West African Concept of Rhythmic Time

2.7.1 Misunderstandings...

The relative lack of serious research on rhythm in sub-saharan African music is frankly baffling, given traditions so rich in rhythmic textures and structures. The *Journal de la Discotheque Internationale de Musique Africaine* and the rare article in *Music Perception* are the only academic journals where such music is even discussed.

On the one hand, this paucity of work is understandable as most of the

research in cognitive sciences is being done in Western universities with funding from local or national organizations; this is compounded by the fact that even the relatively simple rhythmic structure of most occidental music is still largely poorly understood. What is particularly grievous on the other hand is that the choice of music chosen for experiments mirrors the aesthetic imperialism present in the global music industry.

Furthermore, within Africa, the majority of research on rhythm curiously involves Ghanaian dance drumming. This in itself is peculiar, as the Ghanaian styles are hardly representative of West African music. In many respects, any one of the plethora of cultures can hardly represent the varied use of rhythms. Again, I suspect that the popularity of such studies is strongly linked to the patronization of Ghanaian music schools by the Western academic elite and well-known composers like Steve Reich, as well as a much stronger push by Ghanians to export their culture.

2.7.2 Asymmetric Cognitive Clocks

One useful research effort that appeared in a recent issue of *Music Perception* investigated mental models used by expert African drummers in the production of polyrhythmic patterns. The paper fits into the category of work which pursues computational validation of “culturally initiated beliefs” about cognitive models. (That cultural aspects of a music and its production would not even be considered in a computational model is bewildering.)

Experiments were conducted by recording the performance of a master Asante drummer’s spontaneous patterns and responses to a computer generated tone. Quantities such as pulse hand allocation (left or right), pulse stream size (three or four) were varied and measured [13, p. 191]. The goal was to determine if the African cultural description of the cognitive process involved in

producing rhythm is valid by computationally modeling the process and experimentally verifying its validity. The performance data was also run through a pulse-ground(PG) model based on the notion of meter as a means to check the data against another well understood model [13, p. 191].

The model used in the experiments is an asymmetric time-line-ground(TLG) model that represents a computational elaboration of the multilayered traditional West African understanding that all instruments in a percussion ensemble play in relation to a fundamental *time-line*. The asymmetry of the time-line Magill claims doesn't "reduce to one of additive meter" [13, p. 191] because much West African music is founded on the presence of parallel isochronous pulse streams that regularly cut across the additive subgroups. This is possible because the cycle lengths of the asymmetric time-line are composite numbers (6, 8, 12, 24). Furthermore, TLG model "allows for the presence of unequal clock pulses but assumes that where clock intervals are of the same duration, the variance of those two clock pulses will be equal" [13, p. 191].

In performances by the master Asante drummer, Magill reports that the TLG provides a "convincing" fit to seven of the eight tests. The eighth required some modifications to the basic model. Magill comments further that the "asymmetric process" need not be based on the single fastest pulse (additive meter) even though the experimental results can be viewed in this light.

3 Description of Algorithms

“Composers shouldn’t think too much – it interferes with their plagiarism.” — Howard Dietz, American lyricist

3.1 System Architecture

Riddim is a system with two high level functions. The first takes in audio input and extracts sequences of pertinent timing information. The timing information corresponds to temporal gestalt or attack points in the audio. These are points in time where there is a large change in the sound pressure level.

The second part of the system takes as input the time sequences generated by the first and performs rhythmic timing analyses. There are a variety of possible analysis algorithms that try to uncover different aspects the multi-dimensional quality that is rhythm. The present implementation contains one such analysis – the determination per stream of the lowest level pulse.

The first subsystem works in the following way. A single channel audio input is run through a process called Independent Subspace Analysis (ISA). For any kind of input that involves two or more instruments or sources of sound this analysis technique will successfully separate each into a different audio channel. If, for example, we input a recording of a Latin percussion ensemble, we should get at the output a number of different audio channels each containing a different instrument present in the recording.

Each stream is then passed in succession to an onset detection module. In this module, a rectify and smooth algorithm is implemented to prepare the signal for subsequent analyses. Next, a peak detection routine finds the points

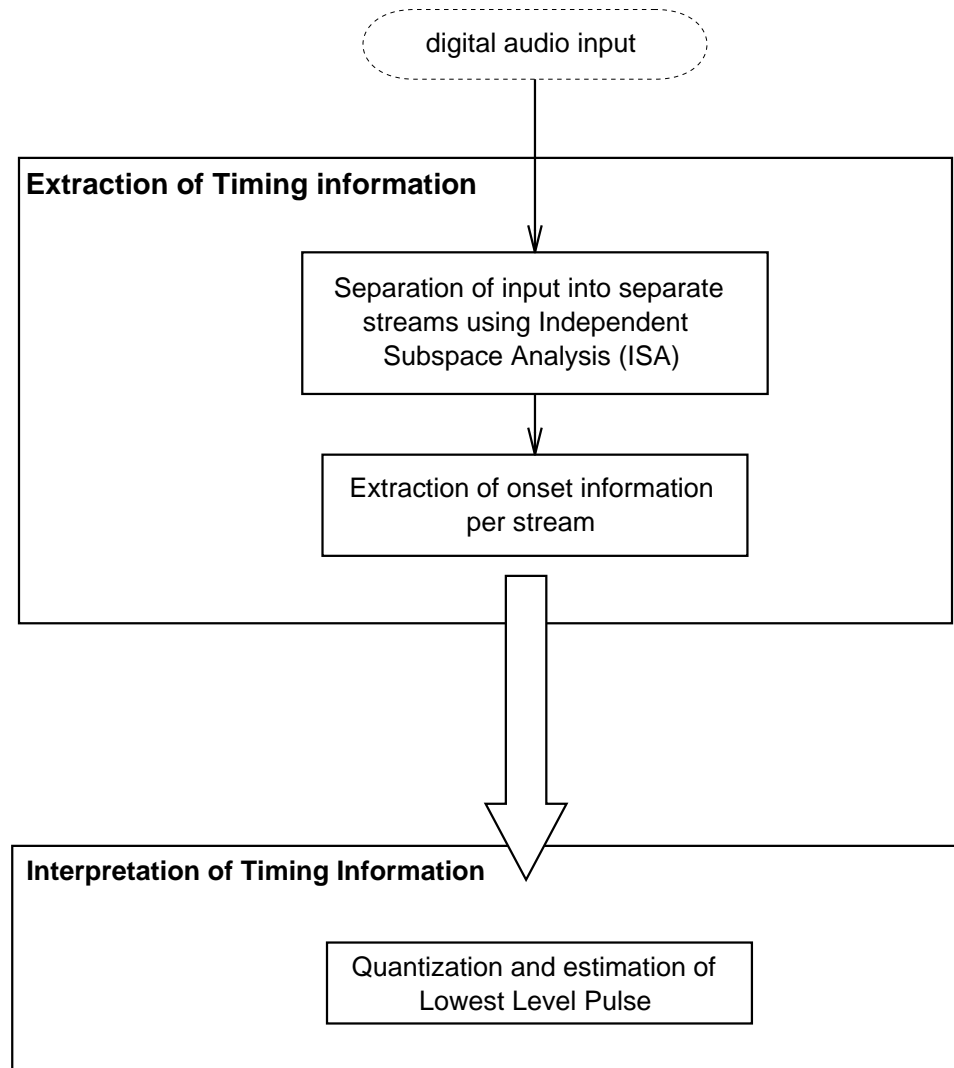


Figure 4: Riddim: The System Architecture

in the stream that correspond to an onset time or an attack point. The onset detection module will return a vector of onset times for each stream that it processes. The second high-level subsystem takes in each vector of times from the onset detection component of the first subsystem, quantizes it and returns the lowest level grid for each stream.

3.1.1 The MATLABTM Platform

I chose the MATLABTM programming language and environment as a development platform for my work for several reasons. First of all it is a common platform for prototyping complex systems, with widespread support among scientists and engineers, in industry and academia. It has numerous highly specialized toolboxes for mathematical analyses and design in a variety of disciplines. This makes it an attractive tool for solving problems quickly, more so than a tool to build a commercial application.

Another motivation for using MATLABTM came from the fact the implementation of Casey’s Independent Subspace Analysis (ISA) was in MATLABTM. Since this was work that formed a central part of my application and was one that I intended to use “as is” with very only minor functional modifications, it made sense to build the rest of my application on the same platform.

3.1.2 Native C subroutines via the MEX interface

However, not all modules were written in MATLABTM. MATLABTM provides an Application Programmer’s Interface (API) called MEX that permits external subroutines written in C or Fortran to be called from MATLABTM functions, script files or the command line. These special C subroutines are compiled into MEX-files which are dynamically linked subroutines that the MATLABTM interpreter can load and execute.

Several parts of this work were implemented in C and called via the MEX interface. These include the bulk of the onset detection module and sections of the grid quantization modules. The main advantages of implementing certain functionality in C is speed. Since MATLABTM is an interpreted language optimized for vector operations, certain constructs like `for` or `while` loops can be very slow for very large arrays. In such cases, to increase the overall performance, such operations can be implemented in C. Another motivation came from the fact that several authors sent me demos of their algorithms as MATLABTM functions. On one hand it would have been easy to simply use their code in my implementation. However, to really internalize their work, I felt that it was important to come up with my own implementation in C.

3.2 Independent Subspace Analysis

3.2.1 Motivation

If the goal was to implement a tool that can extract rhythmic information from digital audio, what would be the best way to decompose the data to facilitate an analysis? How does the brain focus its attention on a singular element in a sound mixture to be able to pick out its rhythmic qualities? Does a model of attention really help? Is any decomposition or data reduction even necessary?

According to Leigh Smith “The perception of musically typical rhythms is achieved by segregation of the received sound complex into separate streams of common sources. It is thereby hypothesized that listeners use timbral, spatial localisation, pitch, tempo and other objective differences between sound sources to distinguish between independent rhythmic patterns” [3, p .85]. A first step in implementing a robust perceptual rhythm analysis tool is to find a scheme that

permits monophonic audio sources to be segregated or “un-mixed” into their source streams. In the past, this problem in the past has been notoriously difficult to solve [25]. This may be due in part to the characteristic representation of the audio signal, the fact that perceptually, sound qualities such as timbre and pitch are still evasively difficult to quantify accurately and the “heuristic nature of psycho-acoustic grouping rules” [26].

First of all, we will discuss some important notions necessary to a complete understanding of ISA.

3.2.2 The Mechanics of Mixing and Unmixing

Let us assume that we have two different recordings of an event involving two instruments. We want to recover the two individual instruments which are mixed in varying degrees in the recordings.

In this simple case, we have two *observed* or *recorded* sounds that we call $y_1(t)$ and $y_2(t)$, each n samples long. We write them in row vector form as,

$$\begin{aligned} y_1(t) &= [y_1(1), \dots, y_1(n)] \\ y_2(t) &= [y_2(1), \dots, y_2(n)] \end{aligned} \tag{3.1}$$

These observed sounds contain a mixture of sounds from different instrument sources that we call $x_1(t)$ and $x_2(t)$ which are also n samples long each and are written as,

$$\begin{aligned} x_1(t) &= [x_1(1), \dots, x_1(n)] \\ x_2(t) &= [x_2(1), \dots, x_2(n)] \end{aligned} \tag{3.2}$$

Because of the relatively high sampling rate of the sounds, we can represent $x_1(t)$ and $x_2(t)$ in a histogram of the signals’ amplitudes. The shape of the histograms are a good approximation of the probability density functions (PDF)

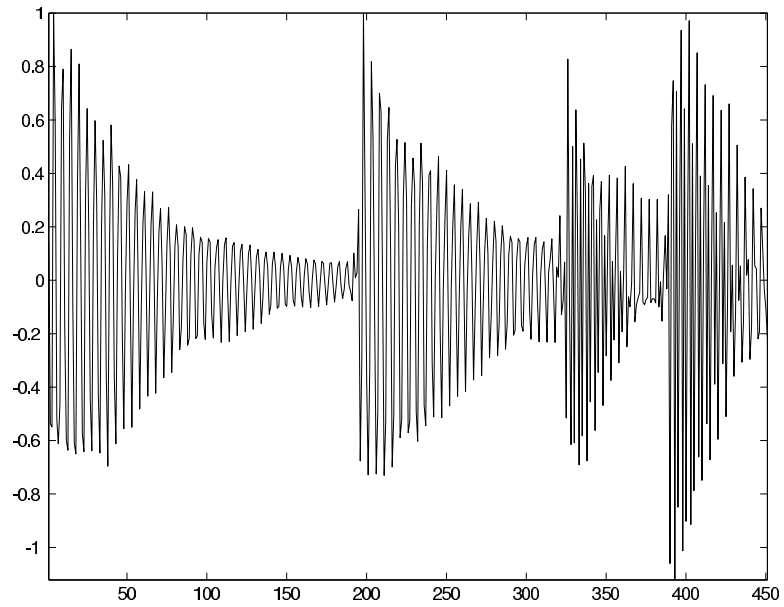


Figure 5: The waveform (time-series) representation of several conga hits 451 samples long.

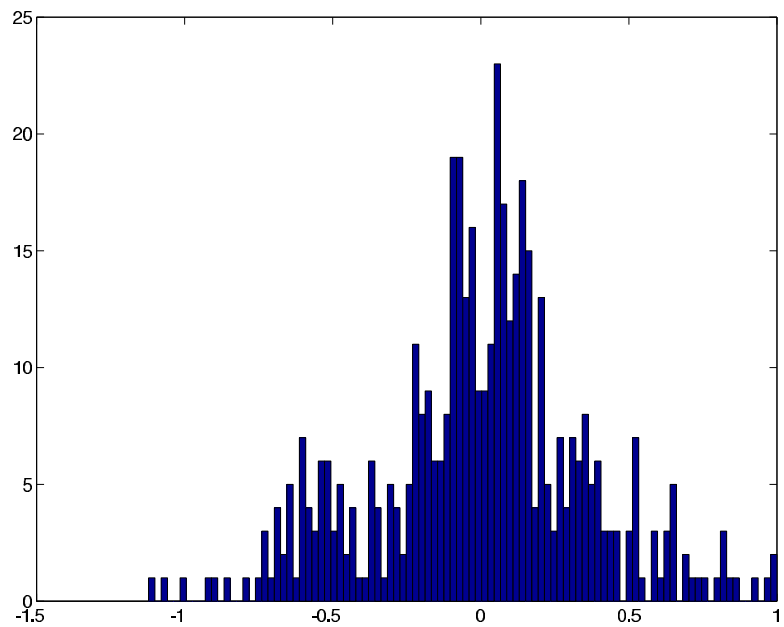


Figure 6: A histogram representation of Figure 5 that approximates a probability density function close to a spiky Gaussian.

$P(X_1)$ and $P(X_2)$ of a pair of random variables X_1 and X_2 . Later we will be making assumptions about the characteristics of these PDFs.

For notational purposes, we put each pair of vectors into $2 \times n$ matrix so

$$X = \begin{bmatrix} x_1(t) \\ x_2(t) \end{bmatrix} \text{ and } Y = \begin{bmatrix} y_1(t) \\ y_2(t) \end{bmatrix}$$

[Note that X is not directly related to X_1, X_2 defined above.] There are two assumptions that we take the liberty of making in discussing how $x_1(t)$ and $x_2(t)$ are combined to yield $y_1(t)$ and $y_2(t)$. The first assumption is that $x_1(t)$ and $x_2(t)$ in X are linearly mixed. This means that Y is the product of X and some full rank mixing matrix M ,

$$Y = MX \tag{3.3}$$

where the M realizes the linear mixing. Thus to recover X given Y reduces to left multiplying Y by the M^{-1} ,

$$X = M^{-1}Y \tag{3.4}$$

In our situation, pulling recordings from CDs or vinyl, we have the mixture Y and we don't know M . Thus to recover the sources X we must try to estimate M^{-1} from the mixture Y . To help this estimation, a second assumption is made. This is that random variables X_1 and X_2 (introduced above), whose PDFs are modeled by the histograms of $x_1(t)$ and $x_2(t)$ respectively, are *independent*. This means that their joint probability distribution given by,

$$P(X_1, X_2) = P(X_1) \cdot P(X_2) \tag{3.5}$$

is separable. Plotting the histogram for two sound sources that are independent with uniform distributions we see that ideally their joint distribution is square-like. On the other hand, plotting the histograms for two highly correlated sounds, we see that their joint distribution tends to run along an axis.

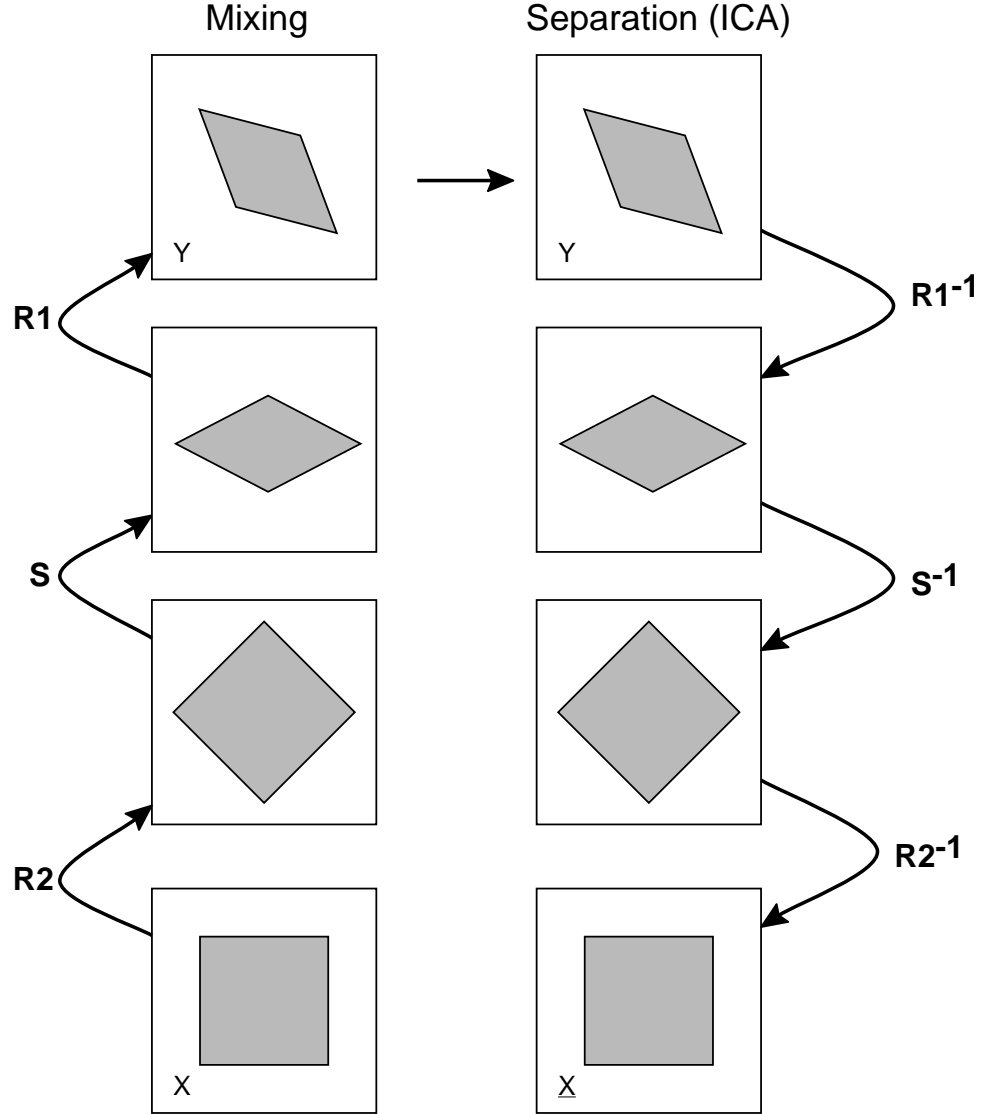


Figure 7: From [27], the bottom left shows an ideal joint probability density function of two independent signals in X . The linear mixing of the signals in equation 3.3 transforms the distribution by rotating it with R_2 , scaling it with S and rotating it with R_1 . ICA tries to estimate M^{-1} by finding rotation matrices R_1^{-1} and R_2^{-1} , and scaling matrix S^{-1} that transforms the joint PDF of the mixed signals back to a square.

Decomposing the Mixing Matrix

From equation 3.3, one can gain some intuition into the mechanics of the mixing procedure by diagonalizing the matrix M . Using the Singular Value Decomposition (SVD), M can be expressed as

$$M = R_1 S R_2 \quad (3.6)$$

where R_1 and R_2 are orthonormal matrices and S a diagonal matrix [27]. Figure 7 shows that M applied to the idealized joint distribution of a pair of independent signals can be seen as matrix R_2 performing a rotation, diagonal matrix S , a scaling, and R_1 a final rotation to yield the joint PDF of the mixed signals.

Thus estimating M^{-1} from equation 3.4 reduces to finding rotation and scaling matrices that undo the mixing operations.

3.2.3 Principle Component Analysis

The matrices, R_1^{-1} and S^{-1} , that perform the first two inverse operations respectively from Figure 7 are obtained via a technique called Principal Components Analysis (PCA). PCA inspects the variance structure of the data looking for components that account for most of the variation in the data. It is about the axis of greatest variance that the joint PDF of the mixed signals (in the upper right corner of Figure 7) is first rotated. This matrix R_1^{-1} is obtained first by calculating the covariance matrix of X , written as,

$$\mathbf{C} = X X^T \quad (3.7)$$

Next, \mathbf{C} is diagonalized using SVD to yield three new matrices [28],

$$\mathbf{C} = U D V^T \quad (3.8)$$

In a zero mean data set, the axis of maximum variance is given by the eigenvector corresponding to the first eigenvalue of the covariance matrix \mathbf{C} . Refer to Figure 8. Thus from equation 3.8, $R_1^{-1} = V^T$.

The next transformation matrix we must estimate as part of the separation process from Figure 7 is the diagonal scaling matrix S^{-1} . In practice it is given by the diagonal matrix D in equation 3.8, so $S^{-1} = D$.

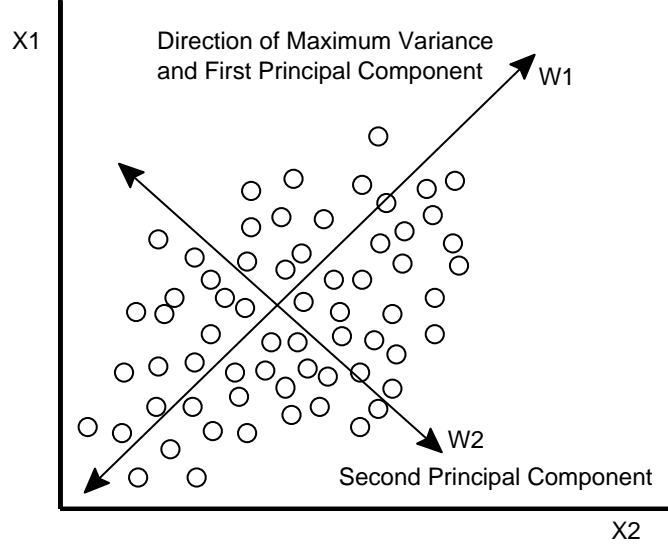


Figure 8: Principal Component Analysis: $Y1$ and $Y2$ show directions of the first two principle components [29].

The final rotation matrix R_2^{-1} from Figure 7 is a bit more difficult to obtain. Typically, the angle that transforms the diamond joint PDF into a square occurs where the kurtosis or fourth-order cumulant of the joint PDF is minimized. So an error function of the orientation angle of the joint PDF is defined and local minima of this function across all orientations angles give a number of possible candidates for the final rotation matrix R_2^{-1} [27].

Statistical Independence and Mutual Information

To narrow down the candidates matrices to one, we must go back to an initial assumption from equation 3.5. Here we claimed that the joint PDF of two random variables is the same as the product of their PDFs when the variables are independent. Generalizing equation 3.5 for N variables instead of

two, statistical independence is given by,

$$P(X_1, \dots, X_N) = \prod_{i=1}^N P(X_i) \quad (3.9)$$

Statistical independence is achieved when the distance between the joint PDF and the product of the marginal PDFs is minimized. To calculate distances between PDFs, a measure like the Kullback-Leibler divergence is used, given for two PDF $P(X), Q(X)$ by,

$$K(P||Q) = \int P(X) \log \left(\frac{P(X)}{Q(X)} \right) dX \quad (3.10)$$

Furthermore, if the probability density function of a mixture factors into the product of the marginal densities (statistical independence) the mutual information between the output joint density and the component densities is said to be zero.

Thus, by applying each of the candidate rotation matrices obtained from the local minima of the error function minimizing the kurtosis of the joint distribution, the mutual information is calculated. The matrix that has the lowest measure of mutual information between the joint PDF and the product of the marginal PDFs obtained from its application to the joint PDF becomes final rotation matrix R_2^{-1} in Figure 7.

In practice, though as was mentioned above, the PDFs are approximated by the histogram representation of the mixed and individual signals.

Summary

The notion of independence is of central importance to this separation scheme. Ideas on independence are heavily influenced by Gestalt grouping rules and Bregman's views on stream segregation in auditory scene analysis [17][26]. Described above are the mechanics of one approach to solving the canonical separation problem using ICA. There are a variety of algorithms in the literature

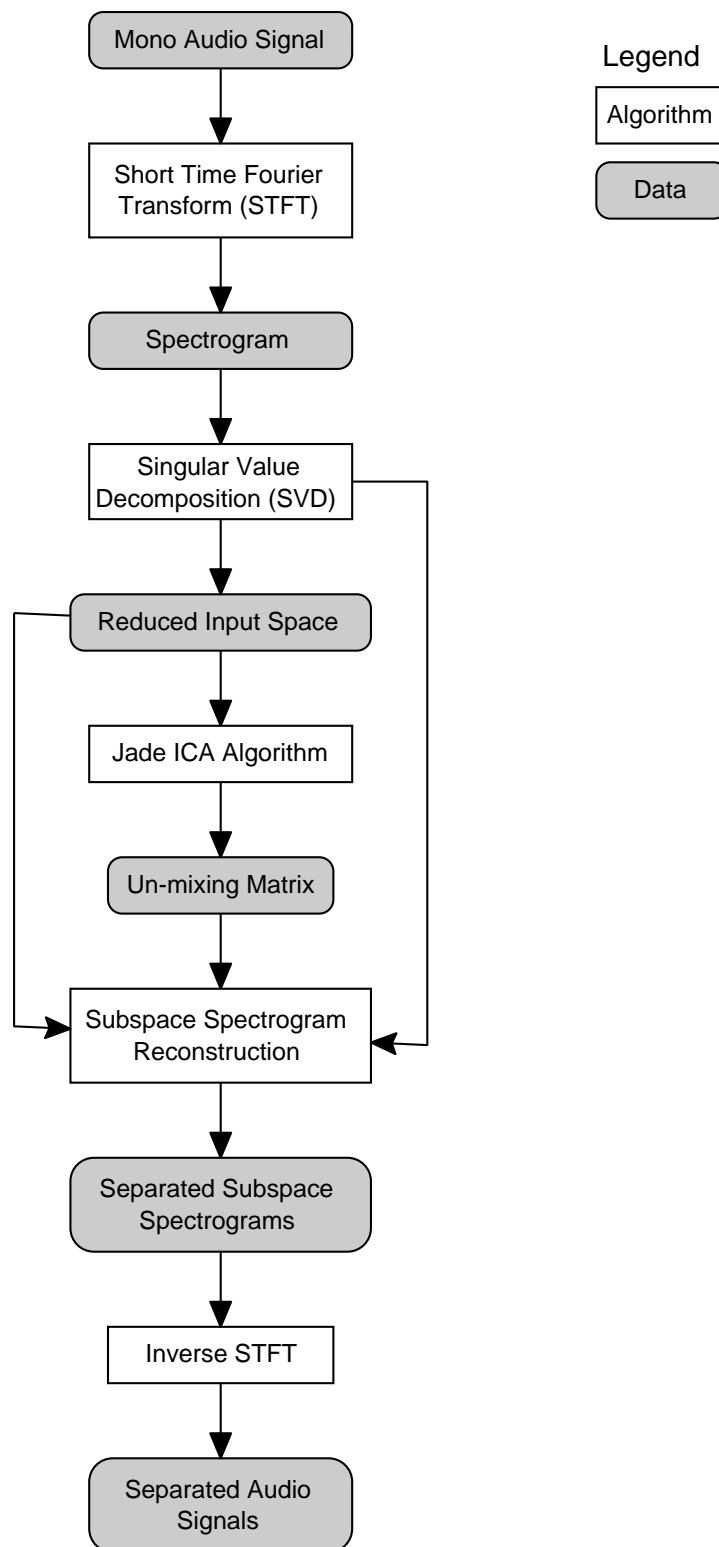


Figure 9: High Level Functional Blocks of the ISA Algorithm

that use “higher-order statistics, minimum mutual information or maximum entropy in their solutions” [30].

For a more in-depth discussion on independence, mutual information, ICA algorithms and their derivations, please refer to [30][31][32][27][26].

3.2.4 Independent Subspace Analysis

Casey’s innovation in ISA was the idea to take a mono signal (that ordinarily cannot be un-mixed directly using ICA) and perform a change of basis operation before employing canonical ICA techniques. A mono signal of size $1 \times N$ is first projected onto a new bases of sines and cosines using a windowed Short Time Fourier Transform (STFT) to yield a spectrogram of size $n \times m$. This new multidimensional manifold is composed of m time slices each containing n frequency bins. The general approach is to do a dimension reduction on this high dimension space to obtain a reduced set of vectors spanning the input space. This is accomplished by performing SVD on the covariance matrix of the input spectrogram. The input spectrogram is then projected onto this new basis to yield a dimension-reduced input space. The input basis vectors are then fed to the Jade¹ algorithm which returns the mixing matrix $A \approx M^{-1}$. The un-mixing matrix is then multiplied against the dimension-reduced basis vectors from the spectrogram projection to yield the independent components oriented in time.

¹Jade is an ICA algorithm by Jean-François Cardoso

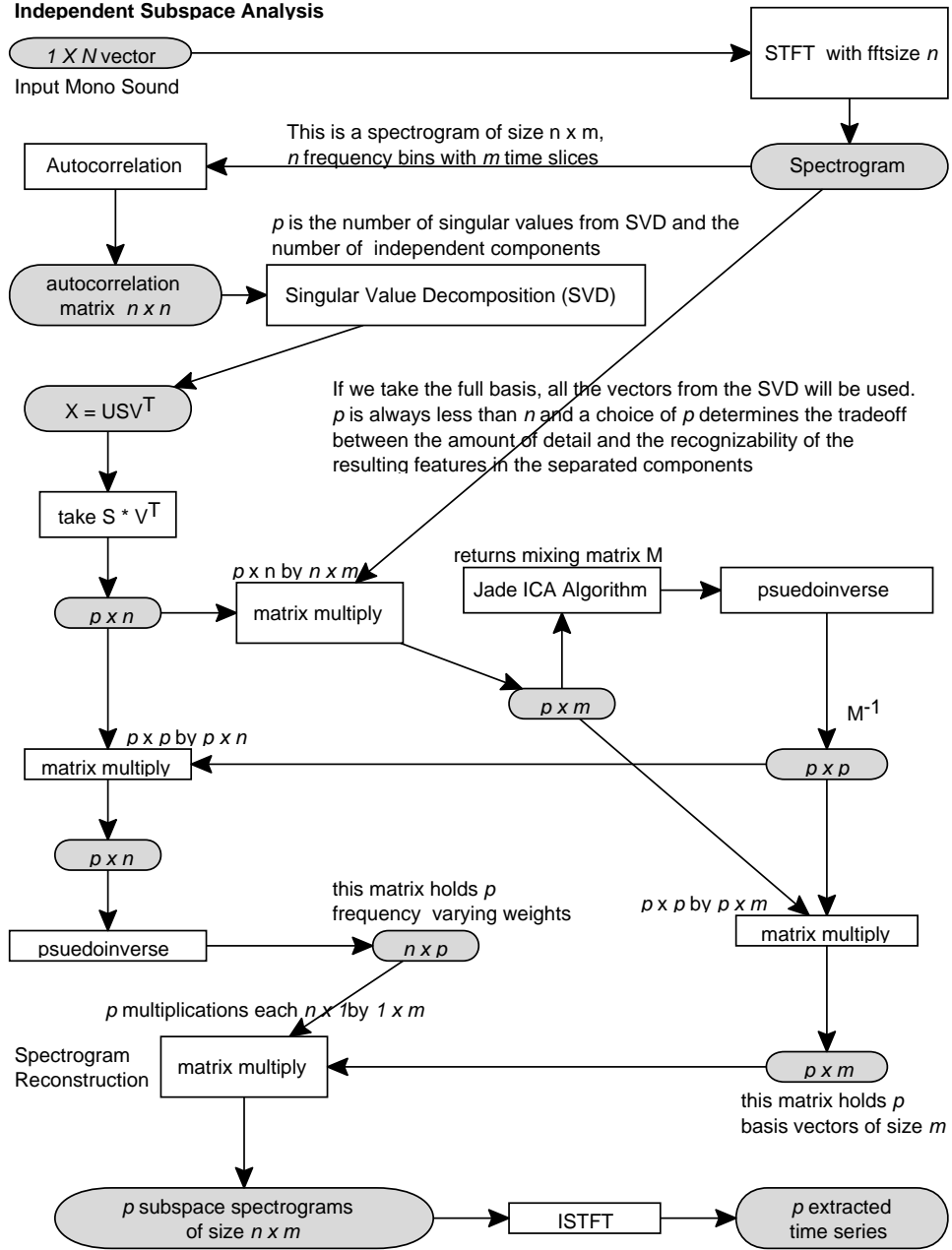


Figure 10: Details of the ISA algorithm

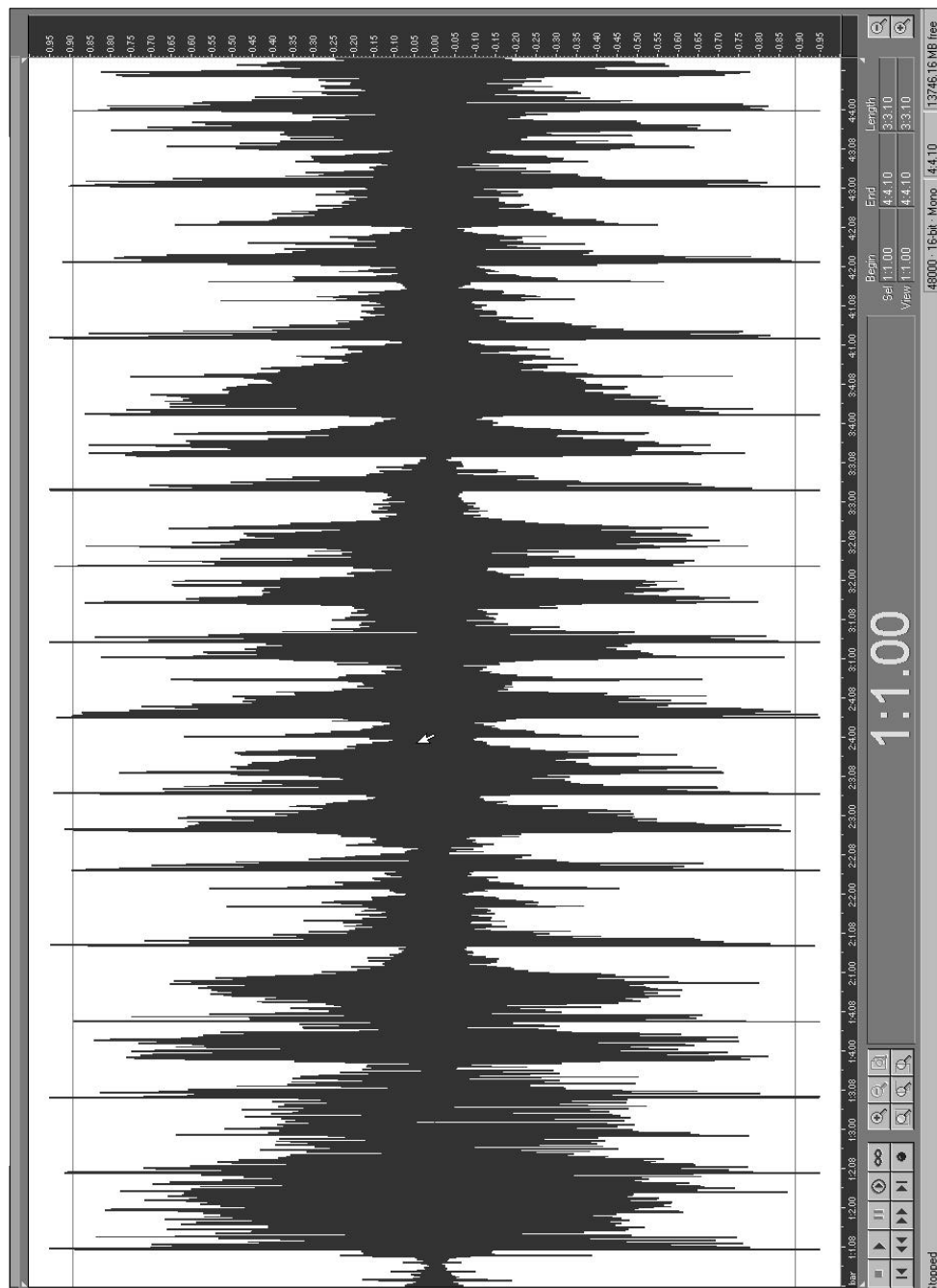


Figure 11: A waveform of an eight second excerpt of Soukous music from the Congo

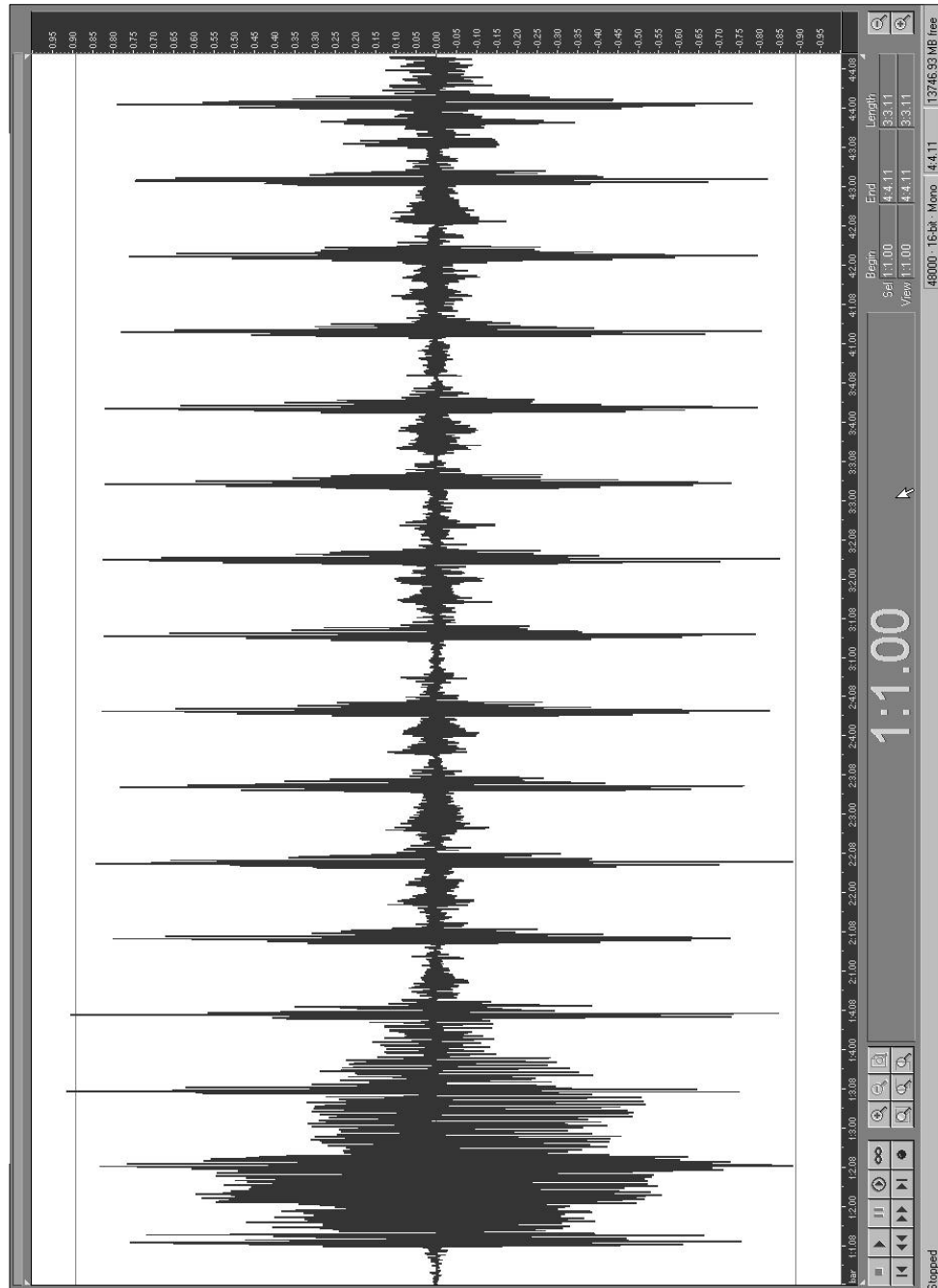


Figure 12: The extracted kick drum from the Soukous music

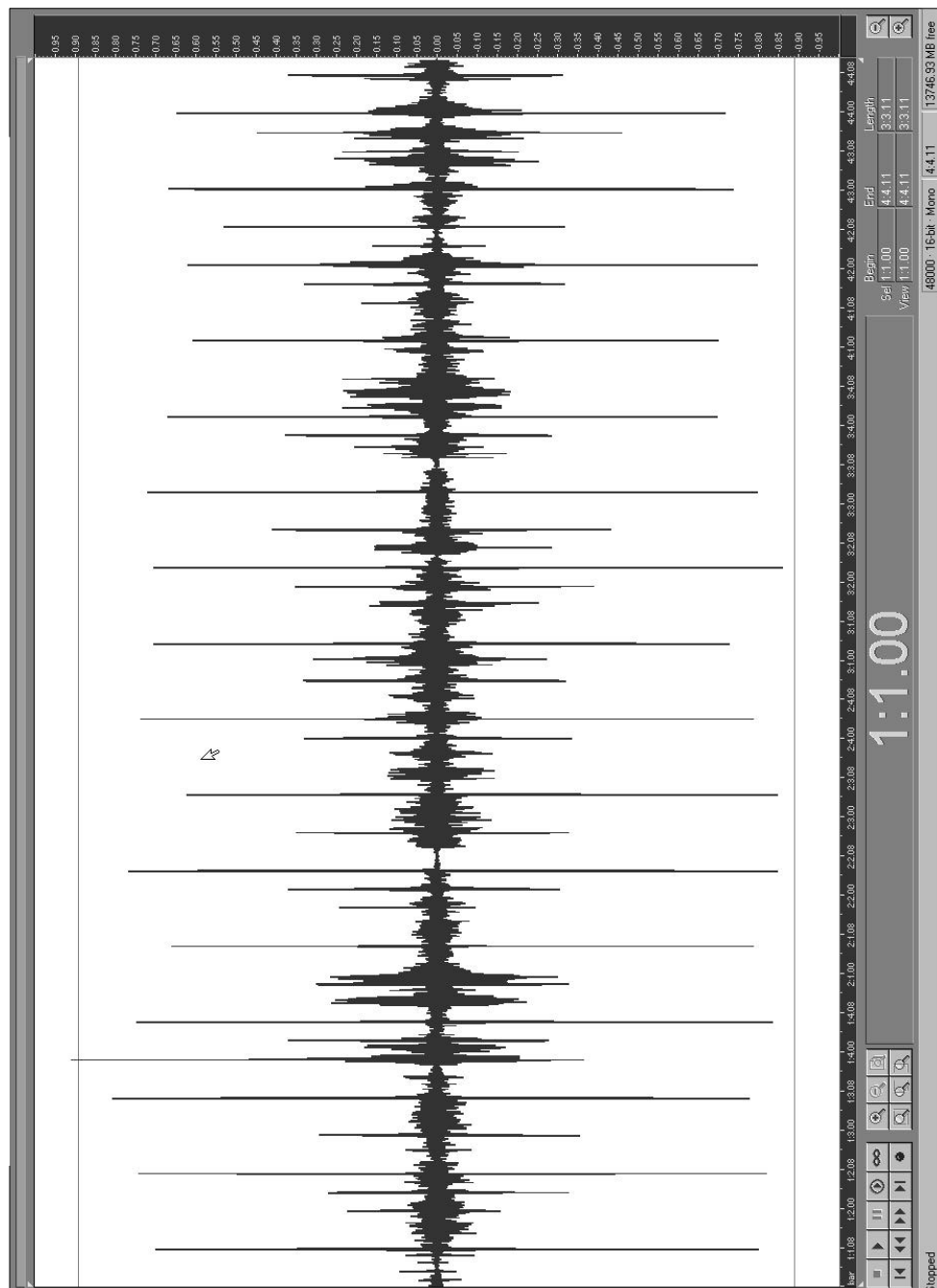


Figure 13: The extracted snare drum from the Soukous music

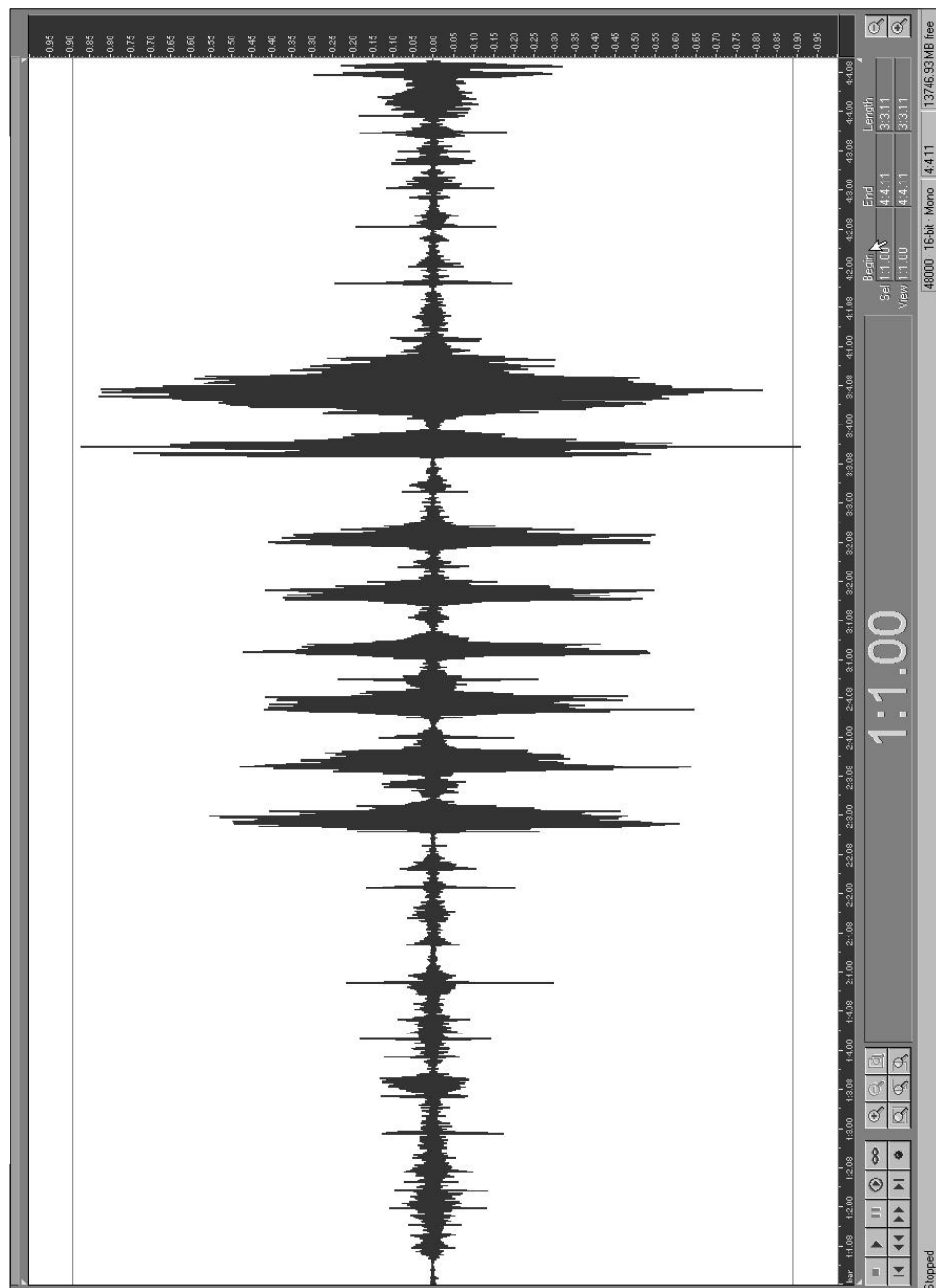


Figure 14: The extracted human vocal chants from the Soukous music

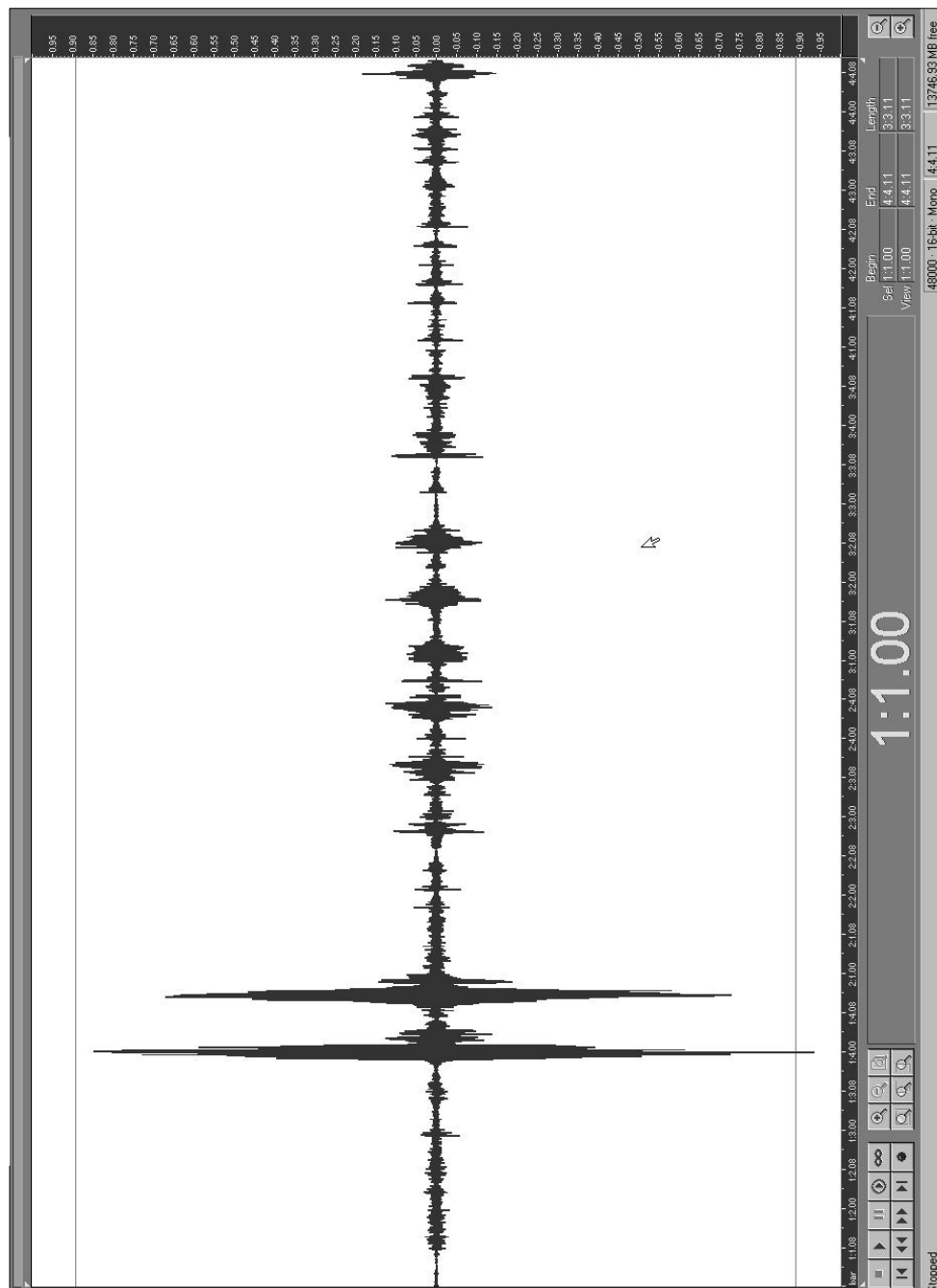


Figure 15: The extracted whistle blow from the Soukous music

Next, frequency varying weights for each independent component are calculated by multiplying the un-mixing matrix against the SVD reduced vectors. Together, the basis vectors and the frequency varying weights are combined to yield individual subspace spectrograms corresponding to independent marginal distributions within the original time-frequency space. These spectrograms are then passed through a windowed inverse STFT (ISTFT) to yield the time domain audio signals corresponding to the extracted streams.

3.3 Detecting Onset Points

The next subsystem in the processing chain extracts onset timing information from each of the audio streams passed to it. The onset detection algorithm used was taken from a paper on sound segmentation by Anssi Klapuri [33] at Tampere University of Technology (TUT) in Finland.

Onset detection is based on the fact that humans are built to detect real-world structure by detecting changes along physical dimensions, representing the changes as relations [10].

Researchers in sound segmentation and onset detection agree that a robust system should imitate the human auditory system by treating frequency bands separately and combining onset information from each band at the end of the analysis. The first detection systems tried to analyse the amplitude envelope of the audio signal as a whole without any band filtering. Since the results were not very accurate, researchers moved to concurrent analyses in different frequency bands combining the various band results at the end [33][12]. In a sense, separately processing frequency bands crudely imitates the human auditory system’s tuning curves. This approximation becomes unnecessary when

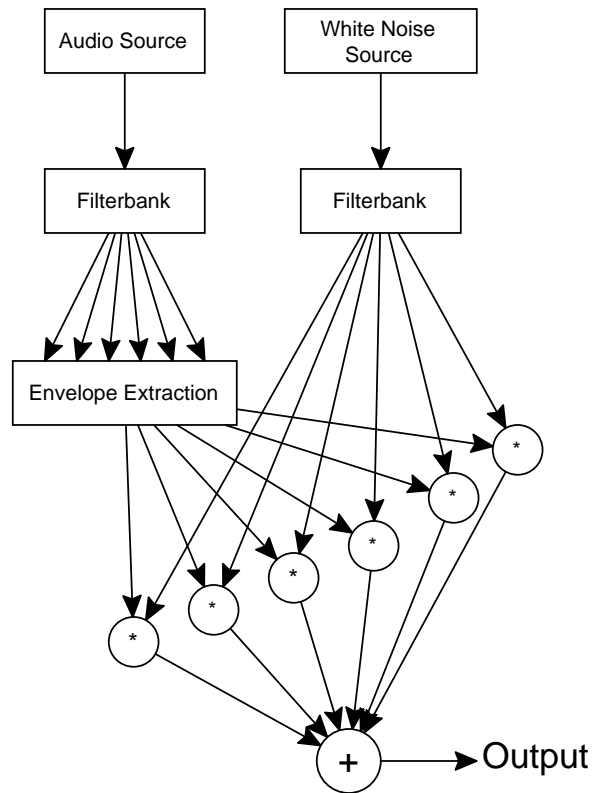


Figure 16: “Amplitude Modulated” white noise from a musical signal has the same rhythmic percept as the signal

individual streams are separated. Thus, in my implementation of Klapuri’s onset tracking algorithm, I do not preprocess the extracted streams through any filterbanks, but analyse each stream as is.

Another psychoacoustic principle that is employed to simplify the onset detection algorithm deals with the perceptual rhythmic proximity of an amplitude modulated noise signal to the original signal. One can construct an amplitude-modulated noise signal by passing a white noise signal and a musical signal through the same set of filter banks. The amplitude envelope of the output of each of the music signal filter banks is used to control the corresponding noise band amplitude. The resulting noise signals are then summed together to form an output signal. Refer to Figure 16.

Scheirer reports that “when an audio signal is divided into at least four frequency bands and the corresponding bands of a noise signal are controlled by the amplitude envelopes of the musical signal, the noise signal will have a rhythmic percept which is significantly the same as that of the original signal” [12]. The importance of this is that since “the only thing preserved in this transformation is the amplitude envelopes of the filter bank outputs, it stands to reason that only this much information is necessary to extract pulse and meter from a musical signal; that is, algorithms for pulse extraction can be created which operate only on this much input data and “notes” are not a necessary component for hearing rhythm” [12].

The importance of this psychoacoustic simplification is that the task of finding rhythmically significant time points in a section of audio reduces to finding the same points on a drastically smaller data set, i.e. the smoothed amplitude envelope of that same section of audio. This smoothed version of the original is attained via a rectification and smoothing algorithm.

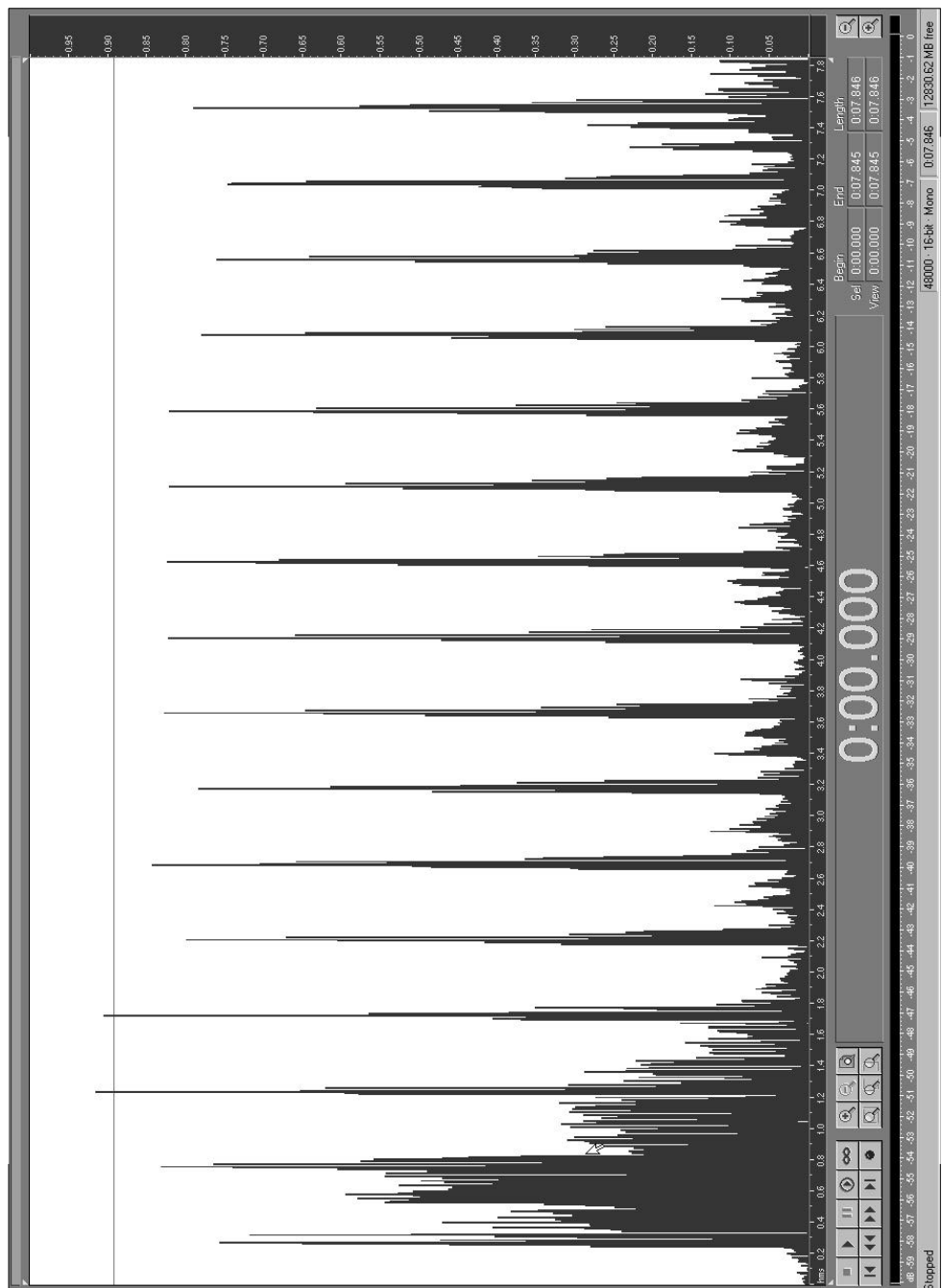


Figure 17: The soukous kick drum after its been half-wave rectified

3.3.1 Rectification and Smoothing

From an audio signal with potentially cumbersome imperfections our task is to prepare the signal for a robust peak searching algorithm by first applying a rectify-and-smooth algorithm proposed by Scheirer [12]. First, the input audio stream is half-wave rectified. This means that all the negative bits of the signal cycle are converted to zero. Refer to Figure 12 and 17 for a graphical representation of what occurs before and after rectification.

At this stage, the rectified signal is decimated. This involves a two step process in which the signal is low pass filtered with a sixth order Chebyshev Type I lowpass filter with a cutoff frequency,

$$w = \frac{.8(\text{ samplerate})}{2R}$$

where R is the decimation factor. The second part of the decimation process is the actual resampling of the signal at $1/R$ its original rate. Decimation is implemented to allow all further operations on the data to be more computationally efficient, since there is minimal data loss for signals with sample rates as high as 44100 *samples/second*.

The decimated signal is then convolved with a 200 msec raised cosine window. The window's discontinuity in time (i.e. the start and end points are not at the same level) means that its Fourier transform has a non-linear phase response. This means that lower frequencies are passed through with a larger delay than higher ones. The window has low pass filter qualities with a -10dB response at 10Hz and *6dB/octave* rolloff afterwards. It performs a kind of energy integration similar to what occurs in the ear by weighting the most recent inputs and masking rapid amplitude modulations in the signal. The net effect of convolving the input data with this window is that all the rough edges, discontinuities, rapid modulations and other "imperfections" in the signal are

smoothed and we are left with just the amplitude envelope of the original signal [12].

3.3.2 Amplitude Envelope Peak Detection

Once the signal has been suitably smoothed, it is now time to search for peaks or sudden changes in signal intensity or representative sound pressure level that would suggest the presence of an attack point or onset. Most algorithms from the literature accomplish this task in one way or another by examining peaks in the first order difference (FOD) function of the original signal, which correspond to the points on the signal with the steepest positive slope. For a time domain signal $S(t)$ the first order difference function FOD can be defined as:

$$FOD(t) = \frac{1}{\Delta t}(S(t) - S(t - 1)) \quad (3.11)$$

Klapuri’s algorithm is different in several ways that directly address two problems when using the peaks of the FOD function. First of all, researchers at TUT noticed that the first derivative measures the signal loudness well, but the maximum value of the signal’s slope does not correspond directly to the start time of the onset. This is due to the fact that if the sound takes some time to rise to its maximum value, the point at which there is a maximum slope might be late in relation to the physical onset of the sound. Note, I refer here to the *physical onset* not the perceptual onset. The distinction is important since the physical onset is the one that would be used in a tool like RecycleTM for splicing and rearranging loops. The perceptual onset, which is where the human would subjectively mark as the beginning of the event, may not necessarily coincide with the physical onset. If the sound rises slowly, its perceptual onset may be significantly later than the physical onset. Additionally, many sounds as they rise from do not monotonically increase. There are usually many local maxima

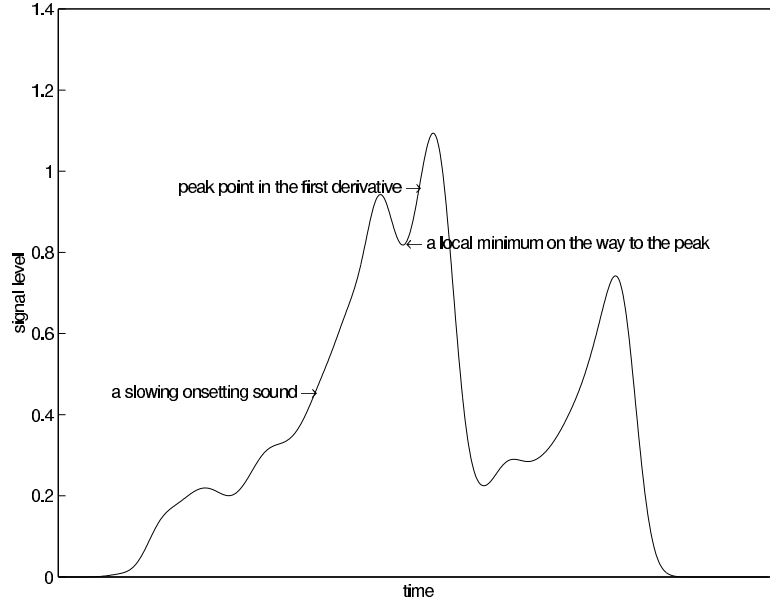


Figure 18: When using peaks in the *first order difference function* to determine onset times, the estimation is sometimes late with slowly onsetting sounds and local maxima may be picked as separate onsets.

and minima throughout the attack point that may be confused as separate attacks [33].

To handle these problems, Klapuri suggests using the *relative difference function* (RDF). The RDF in layman's terms is the ratio of the FOD to the original signal. It measures the amount of change in a signal relative to its level and is equivalent to the FOD of the logarithm of the signal [33].

$$RDF(t) = \frac{1}{\Delta t} (\log(S(t)) - \log(S(t-1))) \quad (3.12)$$

The RDF solves the above mentioned problems since its peaks correspond to the beginning of the physical onset, thereby resolving the problem of late onset estimation with gradually increasing sounds. Moreover, once a signal starts to rise, local minima and maxima along the attack slope are not significant relative to the signal's level.

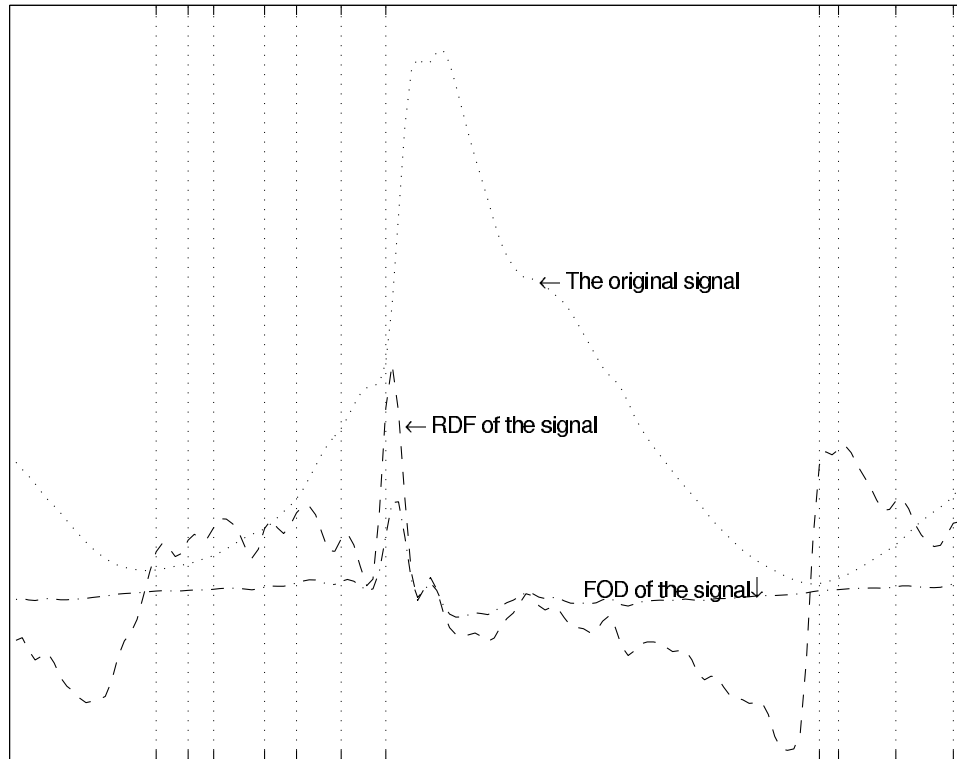


Figure 19: Comparison of the First Order Difference and Relative Difference function of a signal, the grid lines correspond to peaks in the RDF above a particular threshold

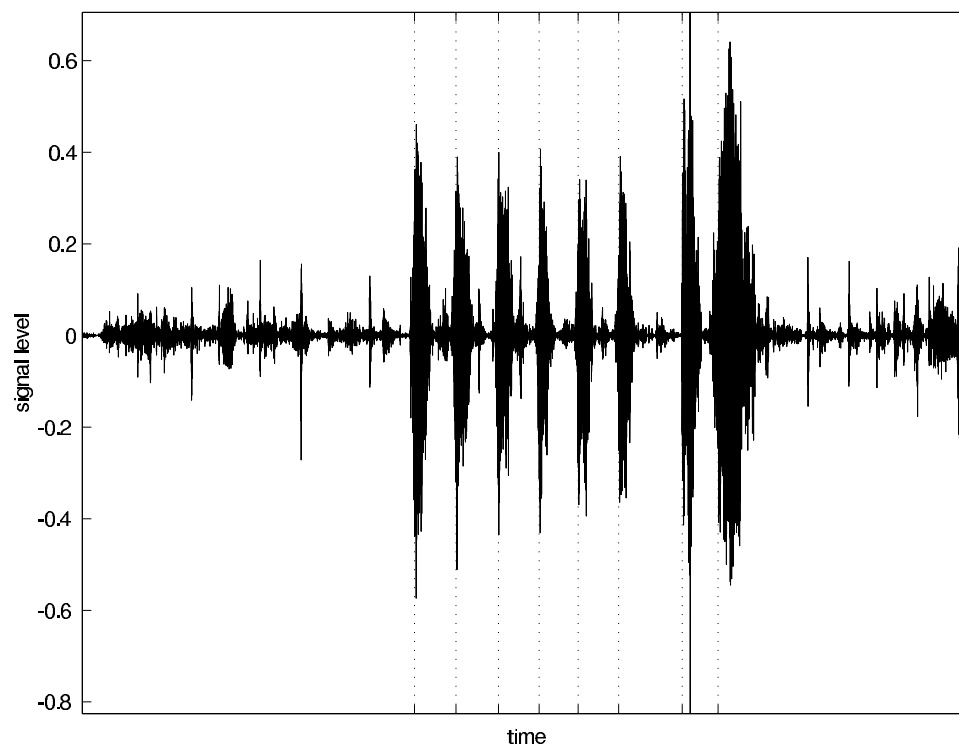


Figure 20: A separated Soukous vocal chant, gridlines indicate time points where onsets are detected. The threshold value is set to .3

So the output of this module returns for each stream it processes, a sequence of time values in the stream when the onsets occur. If for example, an extracted snare drum pattern was passed, it will return the start times of each snare hit. This information is significant for rhythmic analyses since the difference in start times or inter-onset intervals (IOIs) are important in the composition of rhythmic patterns. The snare hit “off-times” are less important from a rhythmic point of view since once the onset occurs and that attack has happened what follows is less important for the determination of a rhythmic percept.

During the determination of the peaks in the RDF, the loudness of each peak is also calculated. The inner or dot product of the FOD of the signal is taken with a Gaussian window equivalent in length to 200 msec. The peak of the FOD coincides with the middle point of the normal Gaussian window. Refer to Figure 21.

After the onsets are obtained as well as their respective sound pressure levels, the timing data is run through a *pruning* routine. Pruning helps eliminate spurious onsets due to imperfections present in the ISA separation. This routine takes the minimum spacing between onsets as an argument set by the user. While the perceptual limit for discriminable sounds tends to be around one sixteenth of a second, percussive flams and grace note flourishes can be much more rapid. Along with the spacing parameter for pruning, users can also set a threshold value. This is useful in dealing with anomalies in the extracted signal that are usually at a much lower level than the salient rhythmic features that characterize a stream. Thus by setting an appropriately high threshold and a minimum onset interval, clean timing and loudness data can be gleaned from each stream.

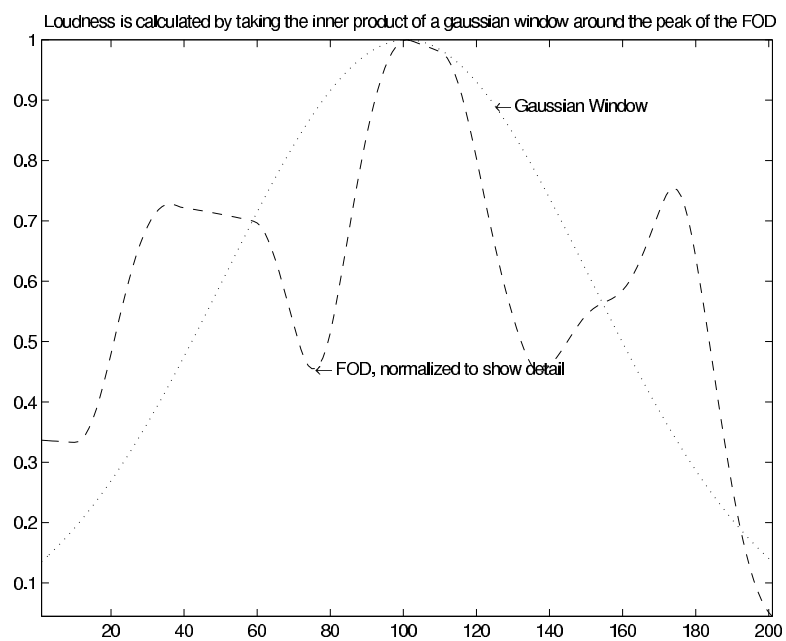


Figure 21: The FOD represents well the loudness of an onsetting sound, the loudness is thus calculated by taking the inner product of a Gaussian window centered about the FOD peak in question

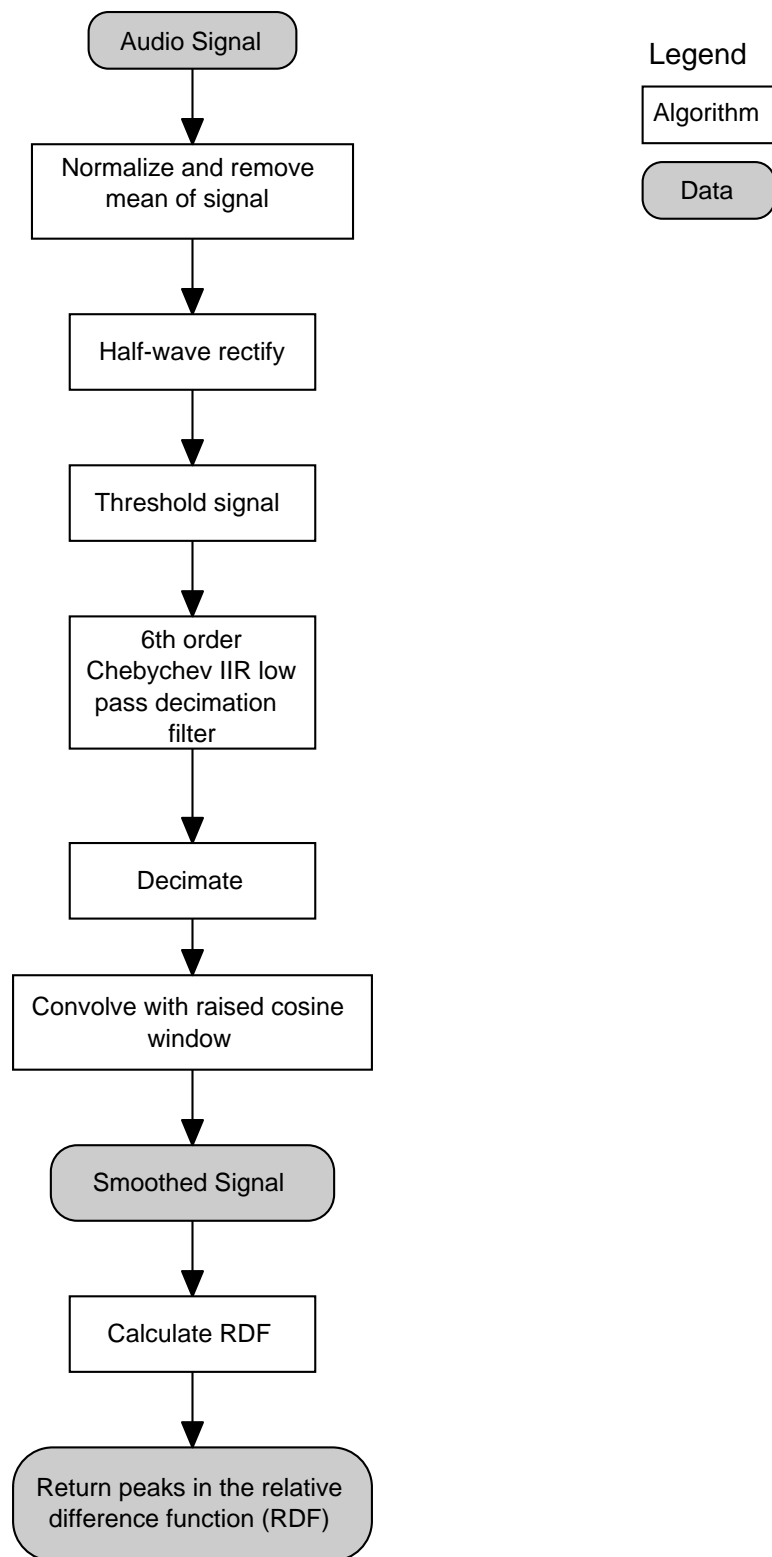


Figure 22: Onset Detection Data Path

3.4 Determination Of A Per-Stream Lowest Level Pulse

Having successfully separated individual streams from a sound mixture, each potentially corresponding to a different instrument or sound source, timing information is then extracted. So far, this is the first of the two high level functions discussed above.

The next step is to make rhythmic sense of the timing information. For example, with the per-stream time vectors we could try to find hierarchical time or grouping structures. We can also try to estimate the overall and per-stream pulse or if applicable estimate quantities like meter and strength of beats. Due to time constraints and the limited scope of this thesis, I implemented only one interpretive module, one that estimates the lowest level pulse per-stream.

This idea of each stream having its own pulse is a deviation from all previous work in the field which has tried, given a piece of music, to estimate the *overall* lowest level pulse [12]. If, however, pulse information is clearly determined for *each* instrument in an ensemble, then it is possible to examine a variety of sophisticated rhythmic inter-relationships between instruments.

3.4.1 The Greatest Common Divisor (GCD) of the IOIs

The algorithm I used to determine the per-stream pulse was incidentally also from Tampere University of Technology (TUT), this time by Jarno Seppänen. It is based on finding the “temporally shortest and perceptually lowest-level pulse” [34]. This idea of a temporal atom is not new and has been suggested by others as tatum, quantum or clock [34][25]. Estimating the tatum of a piece of music is equivalent to approximating the greatest common divisor (GCD) of the all the inter-onset intervals (IOI) present in the music.

First some definitions, the GCD of a set of natural numbers $a_i, i \in I$ is

defined to be the largest integer which divides all a_i with a zero remainder:

$$\gcd(a_1, a_2, \dots, a_n) = \max\{d : \forall i \in I : a_i \bmod d = 0\} \quad (3.13)$$

There is a problem however. The inter-onset intervals that are calculated are not always integral. This is because we are dealing with real performance data, there are never any such guarantees. Seppänen suggests that we try to find a tatum that adequately divides all the IOIs and then define an error function whose local minima represent the best candidates for the GCD. The error function proposed is defined as such

$$e(q) = \sum_i [(o_i + q/2) \bmod q - q/2]^2 \quad (3.14)$$

It is loosely based on the error function of the form $\sum_i (o_i \bmod q)^2$. So if a GCD exists then the remainder function will be zero at the GCD, otherwise the best approximation of the GCD will be the the greatest value q where $e(q)$ has a local minimum. An assumption made by the remainder function is that IOI deviations are gaussian in nature. Barring expressive timing modulations, Scheirer has shown experimental evidence that timing deviations tend to be distributed normally [35].

3.4.2 Histogram Representation of Intra Frame IOIs

Seppänen’s algorithm is designed to run in real time, processing data in frames of 500 msec using a leaky integrator that “remembers”, with some decay, data from previous frames. My implementation of this algorithm is similar despite its non-realtime operation, because the net effect of listening to music in realtime to determine a pulse is identical to analyzing a recording causally.

Because 500 msec analysis frames are a bit short to capture rhythmic modulations, a histogram data structure is used to store IOI data from frame to

frame. Each time the histogram data structure is updated with IOI information from the current frame, the newest additions are weighted higher while histogram data from previous frames are weighted lower with some decay value.

As IOI data is accumulated, it is first discretised to be a multiple of the reciprocal of the sample rate f_s . If $f_s = 44100$ there will be 44100 bins in the histogram each $1/44100$ seconds wide. This also means that the upper bound on the lowest pulse is one second, as an IOI larger than one second will “fall off” the histogram. This is a valid constraint as it matches psychological experiments where events that are spaced more than two seconds apart are not able to be understood in a rhythmic context [36].

As we accumulate IOI data in the histogram form, each bin of the histogram with a value greater than zero can be seen as an element of a set for which we must find the GCD. The error function defined above now must be expressed in terms of the histogram bin values. Seppänen defines such a function as

$$e(q) = \frac{\sum_{k=0}^{M-1} h[k] [(h_x + q/2) \bmod q - q/2]^2}{\sum_{k=0}^{M-1} h[k]}$$

where $h[k]$ is a histogram bin value and $e(q)$ is normalized by the instantaneous histogram mass $\sum_{k=0}^{M-1} h[k]$. Once the error function is calculated, the largest indice containing a local minimum is the candidate for GCD. Picking this candidate involves a traversal of the error function, calculating first and second order difference functions and finding indicies in the histogram that correspond to peaks in the FOD that have positive SOD values (i.e. concave up)

As the signal’s IOI values are successfully accumulated in the histogram, the arrival of each new frame triggers an update of the error function and the estimation of the histogram bin containing the GCD. Accordingly, over the course of a ten second piece of music as many as 20 estimations will be made for the lowest level pulse, thus capturing rhythmic feature modulations. This

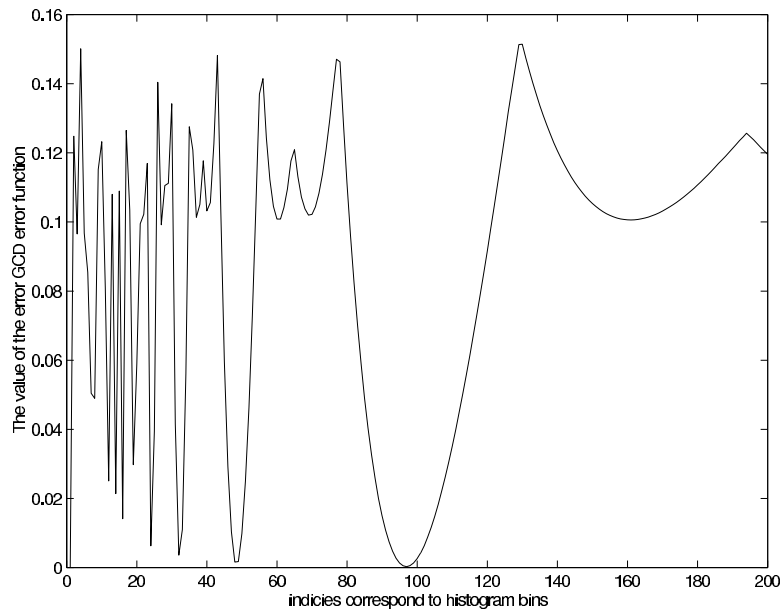


Figure 23: The error function from one 500 msec frame of onset times extracted from the soukous kick drum. The largest indice with the lowest local minima is 96. Thus the lowest level pulse corresponds to the time in seconds represented by histogram bin 96

per-instrument low level pulse trajectory is then plotted over incrementally for the duration of the musical segment in question.

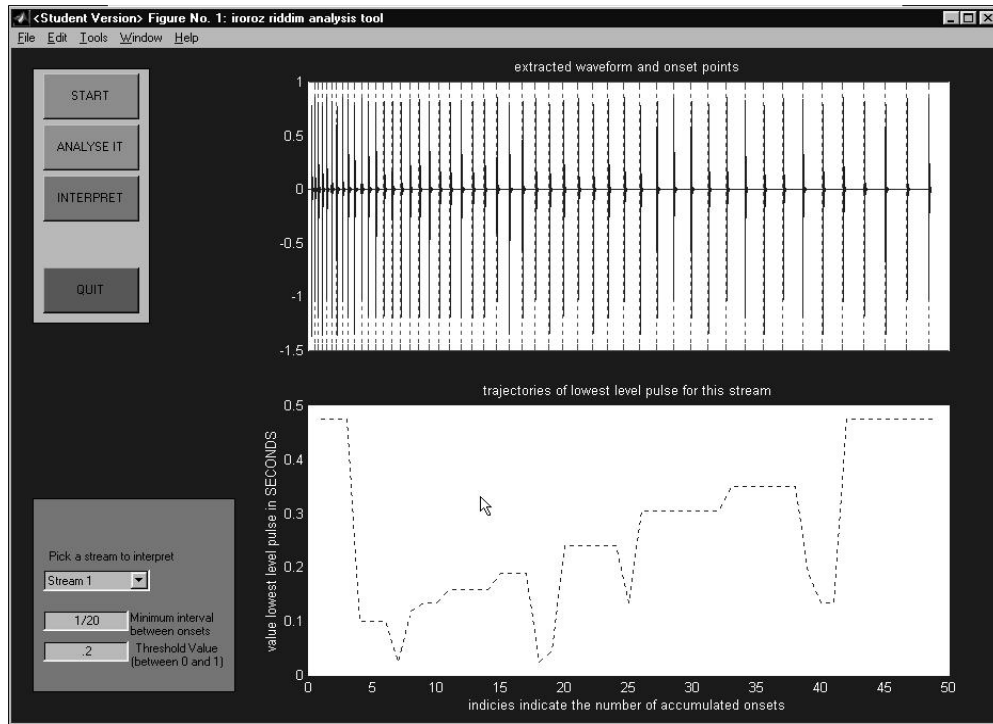


Figure 24: A plot of the trajectory of the lowest level pulse

4 Results

“There’s a basic rule which runs through all kinds of music, kind of an unwritten rule. I don’t know what it is. But I’ve got it.” — Ron Wood, guitarist for the Rolling Stones.

The result of this thesis is a software system that realizes the algorithms described in the previous chapter. The outputs of the software are twofold. First, there is the rhythmic timing data extracted from each stream that is saved as a MIDI file or re-interpreted with an audio sample and saved as digital audio. Second, there is the graphical interpretation of the timing data as the lowest-level pulse in each stream.

The quality of the MIDI rendering and the re-interpreted audio files is very promising. In most cases, a high quality extraction of an instrument virtually guarantees clean results from the onset analysis. Additionally, variable threshold values and minimum onset spacing parameters add to the robustness of the onset detection algorithm. The ISA algorithm works especially well when the number of voices to be separated is low (i.e. less than five). Note that when the number of voices is much higher, a hierarchical ISA technique should be employed. This means that a mono audio file is first run through a ISA extraction with a low number of components, and each component is then run through subsequent ISA extractions to further unmix the latent sources.

The interpretation of the timing data for each stream is also promising. While not having many practical compositional uses, its analytic value is important. For example, an *accelerando* or *ritardando* in an extracted instrument will be seen as an increase or decrease respectively in that instrument’s lowest

level pulse. In most popular dance and folk music, if one of the extracted instruments is a kick drum or some other salient pulse keeper, the lowest level pulse is interpreted as the foot-tapping beat or tactus of the entire work.

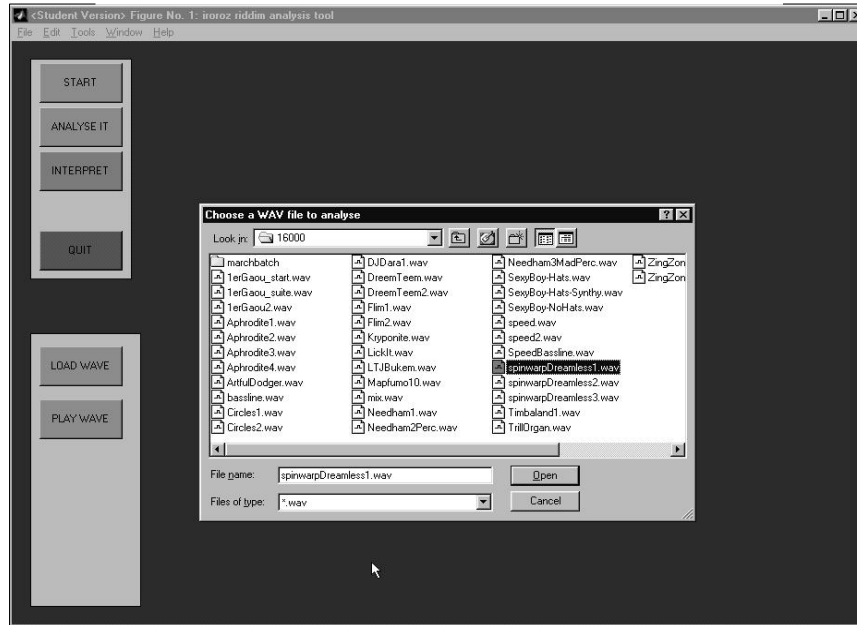


Figure 25: A screen shot of *Riddim* in the Start Mode, with an open dialog box to choose a WAV file to analyse

4.1 *Riddim*

Because *Riddim* is also a proof-of-concept for a real-time rhythm analysis/synthesis engine, the completion of the software goes beyond the scope of this thesis. For that reason, I will not be including the source code in this document. Rather, all the source and examples will be distributed on the web under the GNU General Public License (GPL) agreement, available at

<http://eamusic.dartmouth.edu/~iroro/>

Riddim is organized into three modes, a start, an analysis and an interpretation mode. In the start mode, the user can load and preview audio files. The user then advances to the analysis mode where the loaded audio data is subjected to the algorithms discussed in the previous chapters. The user can enter a set of variables that direct the course of the analysis algorithms. After the user pushes “GO”, the ISA extraction routines proceed and return a menu allowing the user to select a stream to view. By selecting a stream, the onset detection algorithm is run and the user is presented with the extracted waveform with superimposed grid lines indicating onset times. At this stage the user is can enter a spacing parameter and a threshold value to fine tune the quality of the onset detection. The user change also change the parameters used for the ISA extraction by hitting “GO” to re-run the analysis. Once the onset

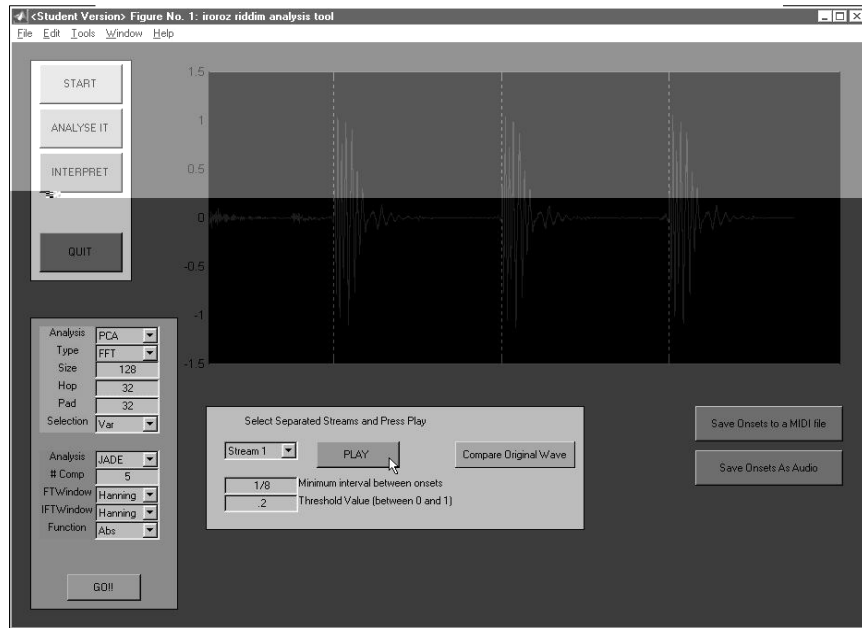


Figure 26: A screen shot of *Riddim* in the Analysis Mode. Shown are an extracted kick drum and the detected onsets indicated by the grid lines.

times per stream are as desired, there are two buttons that allow the timing

information to be saved as a MIDI file or as digital audio. In the latter case, a dialog window will appear asking the user to choose a file with which to render the timing information. At any stage the user can load another audio file or reload the current file.

So far these modes embody the functionality of the first high level subsystem. The next and final mode, the “Interpretation” mode, is the forum for a variety of interpretive analyses of the extracted timing data. As was discussed above, only the determination of the lowest-level pulse was implemented. Accordingly, in this mode, the user is presented with the same menu of extracted streams. This time selecting a stream will return the specified stream, the associated onsets *and* a plot of the movement of the lowest level pulse. Refer to Figure 24.

5 Conclusions and and Future Directions

“I conclude that musical notes and rhythms were first acquired by the male and female progenitors of mankind for the sake of charming the opposite sex.” — Charles Darwin

5.1 Automatic Music Transcription

The most immediate application of the ideas behind *Riddim* is in an automatic music transcription system. Traditionally, transcription involves writing down the notes occurring in a piece of music, converting an acoustic signal to a symbolic representation [37]. Learning to transcribe music involves however, a lifetime of musical training and specific training in the styles to be transcribed. An automatic transcription software tool thus saves time and is useful for musicians and composers for a variety of analytic and pedagogical purposes. For example, in many modern styles where a score is not available, a symbolic representation of what is occurring in the music is very valuable. Similarly, Jazz buffs trying to learn the intricacies of an improvised jazz solo can benefit from such a system as well as can academic composers trying to analyse an electro-acoustic work.

To make *Riddim* a robust music transcription system will entail additional work. First, a fundamental frequency tracking module is essential for a classic “score” transcriber. This module would essentially traverse the extracted streams performing a pitch estimation at every onset point. In this way, the onset times would correspond to the rhythmic part of the score while the estimated fundamental frequency would approximate the pitch. Invariably, as audio

analysis techniques become more sophisticated, automatic music transcription systems like speech recognition systems are going to be increasingly important in the way people learn and interact with music. As Klapuri points out, “Some people would certainly appreciate a radio receiver capable of tracking jazz music simultaneously on all channels” [37].

5.2 *Riddim* in Real-time

One of the central motivations of this thesis was to take the music that I have composed from the studio to the stage, without compromising the complexity. At the same time, moving to a more improvisational performance setting requires a certain mastery of the musical content and the tools to be able to compose on the fly. The current work is of course non-realtime. Working with the algorithms and the code for the past nine months, I have determined that a realtime version is not only feasible, but with minimal additional work, can run as a MAX/MSPTM external DSP module or as a VSTTM plugin.

5.3 Live Improvised Performance and the Meta DJ

Once running in real-time, what are its musical applications? In a live performance setting where listening and playing within a shared musical context is paramount, *Riddim* “listens” to the surrounding music, analyses it and reveals patterns that can be used time a new instrument or part. How these patterns are mixed with the original material is a delicate question of production and aesthetics, as the choice of voicing makes a world of difference.

Furthermore, in analysing and re-interpreting music on the fly, mixing elements occurs in an abstract representation of the music rather than the common

DJ practise of mixing the concrete representation of the music i.e. samples. The idea of a *Meta DJ* involves mixing an abstract representation of “vinyl cuts” or CD tracks by synchronising the realtime outputs of *Riddim* and rendering the patterns with variety of instruments appropriate for the musical setting. The flexibility of not having to mix concrete samples is significant since resampling, time-stretching and vocoder artifacts are common byproducts trying to match two tracks at different tempi. Furthermore, the time patterns extracted in realtime can be used to control light shows, smoke machines or any number of patterned sensory stimuli that accompany the music.

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