



THE CARL PHASE VOCODER

Introduction and Reference Manual by Trevor Wishart

**This program has been made available to CDP by the
Computer Audio Research Laboratory at the
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CDP PVOC Functions (FFT Analysis and Resynthesis)

ANAL

Convert soundfile to spectral file

EXTRACT

Analyse, then resynthesise sound with various options

SYNTH

Convert spectral file to soundfile

Also see:

PVPLAY

Direct playback of PVOC analysis files

PVOC MANUAL (by T.Wishart, with additional material by A. Endrich):

WINDOWS

Introduction and Analysis Windows

CHANNELS

Analysis Channels

ANAL. FILE

Fequency Analysis File

FREQ. BANDS

Points and Frequency Bands

ANAL. RATE

Analysis Rate

TERMS

Summary of Terminology

BANDWIDTH

Analysis Bandwidth

THE OPERATION OF THE PHASE VOCODER (by R.W.Dobson)

A non-mathematical introduction to the Fast Fourier Transform

OLD USAGE

Command Line and Flags before Release 4

PVOC ANAL – Analysis: convert soundfile to spectral file

The flags mentioned refer to Old Usage pre-Release 4. The documentation often refers to the *-N* flag (points) and the *-W* flag (overlap).

Usage

pvoc anal mode *infile outfile* [*-cpoints*] [*-ooverlap*]

Modes

- 1 Standard analysis (was the **-A** flag)
- 2 Output spectral envelope values only (was the **-E** flag)
- 3 Output spectral magnitude values only (was the **-X** flag)

Parameters

infile – the input is a MONO soundfile

Stereo or multi-channel input must first be split into separate channels using **HOUSEKEEP CHANS 2** or **CHANNELX**. The individual mono files are then PVOC-analysed and processed separately, before being re-synthesised (**PVOC SYNTH**) and re-combined using **SUBMIX INTERLEAVE** or **INTERLX**.

outfile – the output is an analysis file in Mode 1

In Modes 2 and 3, the output is thought to be a binary data file. These options have been retained from the original PVOC, but in fact we are unclear as to what they produce and what program could read the data! If any CDP user has made use of these options and could enlighten us further, please do so.

-cpoints – the number of analysis points (2-32768 – powers of 2 only) Default: 1024 (was the **-N** flag)

NB: *Points* is the same as the value entered for the **-N** flag in the previous versions of PVOC. You will sometimes see references to **-N** in the CDP PVOC documentation.

More points give better frequency resolution, but worse time resolution: i.e., detail is lost in rapidly changing spectra.

-ooverlap filter overlap factor (1 – 4) Default: 3 (was the **-W** flag)

End of PVOC ANAL

PVOC EXTRACT – Analyse, then resynthesise sound with various options

The flags mentioned (-W, -C) refer to Old Usage pre-Release 4.

Usage

pvoc extract *infile outfile -cpoints -ooverlap -ddochans -llochan -hhichan*

Parameters

infile – the input is a soundfile

outfile – the output is a soundfile

-cpoints – the number of analysis points (2-32768 (power of 2)); Default: 1024 (was the **-N** flag)

More points give better frequency resolution, but worse time resolution:
i.e., detail is lost in rapidly changing spectra

-ooverlap filter overlap factor (1 – 4) Default: 3 (was the **-W** flag)

-ddochans resynthesise ODD (**1**) or EVEN (**2**) channels only (was the **-C** flag)

-llochan ignore analysis channels below this when resynthesising (Default: 0)

-hhichan ignore analysis channels above this when resynthesising (Default: highest channel)

Lochan and *hichan* refer to channels rather than analysis points. There is 1 channel for every 2 points.

Hichan should therefore not be > analysis points/2

To default to topmost channel, set at 0.

NB: If no flags are set, the output sound will be the same as the input.

See [STRETCH TIME](#) and [STRETCH SPECTRUM](#) for stretching operations.

For the various spectral transposition options, see the [REPITCH](#) group of functions: TRANSPOSE, TRANSPOSEF, COMBINE, COMBINEB.

End of PVOC EXTRACT

PVOC SYNTH – Resynthesis: convert spectral file to soundfile

Synthesis was the **-S** flag in the Old Usage pre-Release 4.

Usage

pvoc synth *infile* *outfile*

Parameters

infile – the input is an analysis file

outfile – the output is a MONO soundfile

End of PVOC SYNTH

INTRODUCTION

The *Phase Vocoder* is a powerful analysis/resynthesis tool with important musical applications. The *Phase Vocoder* was initially ported from the CARL computer music package to the CDP Desktop System through the efforts of Trevor Wishart, Andrew Bentley, and Keith Henderson working with Richard Orton. Later versions were produced by T. Wishart, N. Laviers, and M. Atkins. Martin Atkins has ported it to the Atari Falcon030 and PC. The following is an outline description of its operation from a technical point of view.

Windows

Traditional acoustics based on the analysis of the sounds of pitched musical instruments teaches us that a sound may be analysed into a sum of constituent sine-waves (known as partials), each having a particular frequency and a particular magnitude (amplitude or 'loudness'). This is known as the spectrum of the sound. When these frequencies are integer multiples of the smallest value (e.g. 120, 240, 360, 480 etc.) the resulting sound has one specific pitch and the sine-wave components are known as 'harmonics'. The relative amplitude of the harmonics is partly responsible for differences of timbre between one instrument and another.

When the partials are not related in this simple manner, the resulting sound may have indefinite pitch (like a bass drum) or appear to contain several pitches (like a bell). At the other extreme, a sound whose spectrum changes rapidly and randomly through time produces an unpitched noise spectrum (like a clash cymbal or maracas). It was also understood that the loudness contour (amplitude envelope) of a sound significantly affects our perception of its timbre, particularly such features as the time taken to reach the maximum loudness (attack).

Traditional synthesis methods were modelled on these assumptions. However, more detailed analysis using computer technology has shown us that even the spectrum of pitched instruments varies through time. For an accurate analysis of a sound we must therefore be able to analyse the spectrum of a sound at successive instants of time which we will refer to as 'windows' (to be explained in more detail below). For a good analysis such windows are of the order of milliseconds in duration.

The *Phase Vocoder* is a program which produces such a windowed analysis of a soundfile, and will re-synthesize the sound from this analysis.

In practice one cannot simply chop up a sound into chunks for analysis. The arbitrary editing involved will produce segments of sound which drop to zero suddenly from non-zero values at the edit points (like a synthetic square-wave would do), and on analysis will contain spurious partials as a result.

To circumvent this problem, the sound-segments have a special kind of envelope imposed on them before analysis. There are a number of such window-envelopes. For most musical purposes the Hamming window can be used and this is the default window available in the CDP implementation of the Phase Vocoder (see Fig. 1).

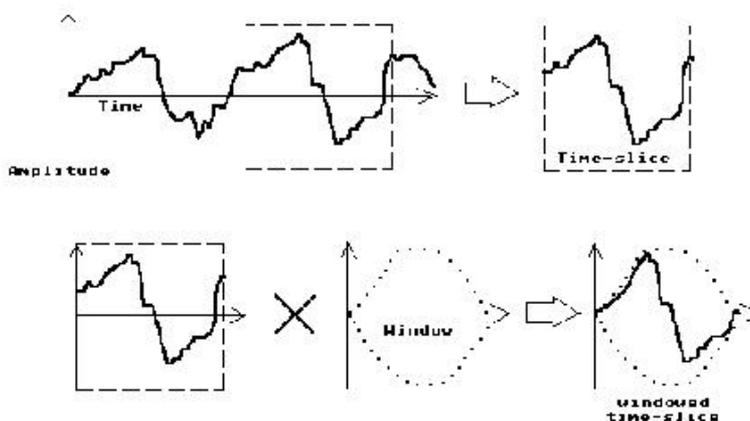


Fig. 1

PVOC, Fig. 1 – window envelopes

Channels

Within each *time*-window the *Phase Vocoder* divides the spectrum into a number of equally spaced bands known as **channels**. It then looks in each channel to see if any component of the sound is present and records its amplitude and frequency. The greater the number of channels, the more detailed the analysis.

For example, a pitched sound at 500Hz will have partials at 500 Hz, 1000 Hz, 1500 Hz (500*1, 500*2, 500*3) and so on. If there were to be 100 analysis channels, the frequency bands will be 110.25Hz wide at a sample rate of 22050 – arrived at by the calculation $SR/2$ divided by the number of analysis channels: $(22050/2)/100 = 110.25$. The various partials comprising the sound will therefore each fall into one or other of these 100 different channels of the *Phase Vocoder* analysis. Note that each channel can handle only one partial.

However a sound at 25 Hz, having partials at 25 Hz, 50 Hz, 75 Hz, 100 Hz etc will clearly have *several* of its lower partials falling within a single channel of such an analysis. In this case the *Phase Vocoder* will fail to distinguish between the several constituent partials found within a single channel and the analysis will be poor (see Fig. 2).

Therefore, the greater the number of analysis channels, the finer the resolution of the analysis. The **general rule** is that the sample rate divided by the value for **-N** should be less than the lowest pitch in the input sound. See below for more about channel width, centres and boundaries.

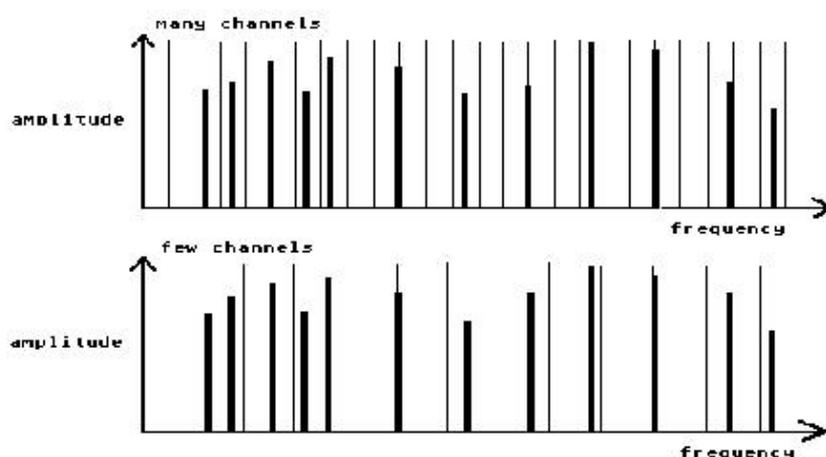


Fig. 2

PVOC, Fig. 2 – illustrating the frequency resolution of the analysis bands (channels)

Structure of the analysis file

The number of channels you wish to use may be entered with the **-N** flag of the *Phase Vocoder* program. This parameter refers to value pairs: the frequency and amplitude for each component. Thus, the number of analysis channels is only half the value you enter. Therefore, always enter twice the number of channels you want. The default value is 256. For most musical applications I would recommend entering a value of at least 1024.

Note that the Fast Fourier Transform (analysis) algorithm, which is at the core of the *Phase Vocoder* is optimised to work extremely efficiently with values which are powers of 2 (e.g. 256, 512, 1024, 2048) and you are advised to use such values unless you have a particular reason not to do so.

The following information should help to clarify some terminology and help you to use the *Phase Vocoder* more effectively, whether for musical or programming purposes.

Study example: when the soundfile sample rate is 44100 and the **-N** value is 512, what happens?

- (1)** The *sample rate* in the header of an analysis file is the analysis sample rate (see below) while the *analysis bandwidth* is equal to the sample rate of the original sound which was analysed divided by the value you entered with the **-N** flag (or, if you did not use the **-N** flag, the default value of 256). E.g., $44100/512 = 86.13$
- (2)** The *Phase Vocoder* produces *two values* (amplitude and frequency) *for every analysis channel*. The convention in using the *Phase Vocoder* is to enter a value of 512, meaning 512 value-pairs per window when you want 256 equally spaced channels (amplitude and frequency value for each) and, in general, a value of **2N** when you want **N** analysis channels. A **-N** value of 1024 is recommended.
- (3)** The analysis actually produces *one extra channel* centering on 0Hz at the start of each window. Thus if you enter 512 you will produce 256 channels plus 1 extra channel at 0Hz, making 257 in all.

(4) Finally, if you need to know exactly where the *channel boundaries* are, you must first calculate the *channel centres*. With a **-N** value of 512, this gives 256 channels. These 256 channels divide up the spectrum between 0Hz and half the sampling-rate (the Nyquist frequency) into equally spaced bands, and so: if the sample rate is 44100, half this is 22050, divided by 256 means that the analysis bands will centre at:-

1/2*SR*Ch_no:	0Hz	$1/2 * 44100 * 1$	$1/2 * 44100 * 2$	$1/2 * 44100 * 3$ etc
Div by Num_chnls:		256	256	256	
= channel centres:	0Hz	86.13 Hz	172.27 Hz	258.41 Hz etc

The *frequency bandwidth*, i.e., the distance between these channel centres is 86.13 Hz. This is quickly calculated as SR/N – but more technically correct expressed as $(1/2*SR)/(N/2)$. You may assume that the *channel boundaries* lie half-way between the channel centres. In the present case (see Fig. 3) this would be at:

Channel boundaries:	43.07 Hz	129.2 Hz	215.34 Hz	301.48 Hz
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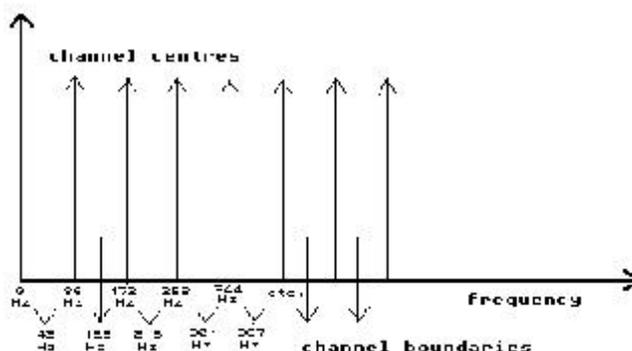


Fig. 3

PVOC, Fig. 3 – channel centres and channel boundaries

If you examine the analysis file you will discover it consists of a list of floating point numbers starting with Window 1, 0Hz channel. If you entered 512 as the **-N** value, giving you (256+1) channels, you will have 257 pairs of floating point numbers before you reach Window 2, and so on. Any number-manipulation program can be applied to this data (bearing in mind restrictions on maximum values for amplitude and frequency) so long as it generates a new analysis file of this same format. (You could generate a file with a different number of channels if you correspondingly altered the information in the analysis file header). This new file may be subsequently run through the synthesis option of the *Phase Vocoder* to generate a new soundfile.

The Phase Vocoder flags – with CDP parameter names (since Release 4) in parentheses

N (= *-cpoints*), **F**, **N**

sets number of channels you want in your analysis windows: $(\mathbf{N}/2) + 1$. Conversely **F** is the analysis sample rate, which is the sample rate of the input sound divided by the **N** value you want to use (frame overlap not taken into account here – see below). You cannot use both **N** and **F**.

b, **e**

As analysis programs always take a long time to run, you will often want to analyse only a short segment of a sound for test purposes; **b** and **e** indicate the beginning (default 0) and ending (default, end of file) samples for the analysis.

i, **j**, **c** (= *-ddochans*)

You may use the *Phase Vocoder* as a powerful filter by excluding certain channels from the re-synthesis. Partials falling in the excluded channels will disappear; **i** indicates the lowest channel to be re-synthesized (default 0) and **j** the uppermost (default, maximum channel value) – revised as *-llochan* and *-hhichan* parameters).

Note that programs can be written to achieve more sophisticated filtering routines including cross-synthesis (imposing the channel amplitudes (which define the spectral envelope) from one sound onto a different sound); **c** is a special option resynthesizing from odd or even channels only.

A (= ANAL Mode 1), **E** (= ANAL Mode 2), **X** (= ANAL Mode 3), **S** (= SYNTH)

If you want to manipulate the sound spectrum with the program described below or with your own programs you will need to select the **-A** (= ANAL Mode 1) option to generate an analysis data file to submit to these programs. On generating a transformed analysis file, use the **-S** (= SYNTH) option to regenerate a sound file. The **(-E)** flag generates data on the spectral envelope of the sound, window by window.

T, **P**

The *Phase Vocoder* already contains a number of options which permit you to transform the spectrum of the input sound. For example, you may transpose the pitch of a sound without changing the duration (the **-P** option) or you may time-stretch or contract the sound without altering its pitch (the **-T** option). **T** and **P** are both multiplying factors and the default value is thus 1.0 in each case. Although these effects are more rapidly achieved with a harmonizer type of program for small degrees of stretching or compression, the *Phase Vocoder* retains much greater fidelity to the original source.

(**NB**: not implemented in Release 4 – use other functions from the REPITCH group.)

Furthermore the *Phase Vocoder* permits much greater degrees of time-stretching or pitch-transposition than any Harmonizer algorithm. For a time-stretching factor greater than ca. 8, I would recommend doing the process in 2 passes - stretch 8 times, re-synthesize the sound, analyse again, stretch and re-synthesize. Beyond a certain limit, even *Phase Vocoder* stretching will produce artifacts of one kind or another. For example, time-stretching the sound of vocal breath by a factor of 64 introduces strange glissing noise-bands (these can be heard in my piece *VOX-5*).

D, **I**

The decimation factor and the interpolation factor express certain mathematical relationships between sound and analysis data.

For further details, see the specialist literature. These two flags should NOT normally be used.

w

Transposing a vocal sound will distort the vowel-image rendering a text distorted or incomprehensible. This is because our perception of vocal vowels (and certain characteristics of other sounds) is related directly to the spectral envelope of the sound. If we examine the spectral envelope in any time-window we will usually find it has a number of peaks, known as formants (see Fig. 4).

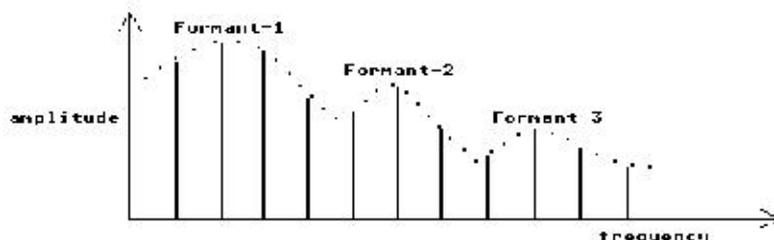


Fig. 4

PVOC, Fig. 4 – formants in the spectral envelope

Our perception of vowel 'type' is related to such formants. We are able to change the pitch of a vocal sound while retaining the vowel by using the **-w** flag. In this process the spectrum as a whole moves, but the spectral peaks remain in place centred at their original frequencies. Note that this implies that individual components of the sound change their relative amplitude to maintain the correct spectral envelope. Unfortunately when we merely transpose a sound in the *Phase Vocoder* we thereby move and stretch the spectral envelope (see Fig. 5).

In order to retain the original formants of the sound we need to 'warp' the spectral envelope back to its original position. The **-w** flag (default value 1.0) should be used to do this. Using just **-w** (i.e. with no value specified) will produce the correct warping for the pitch transposition you are using. Other specified values of **-w** may also be entered. The program SPECF below also attempts to preserve the formants of the originally analysed sound, under transformation.

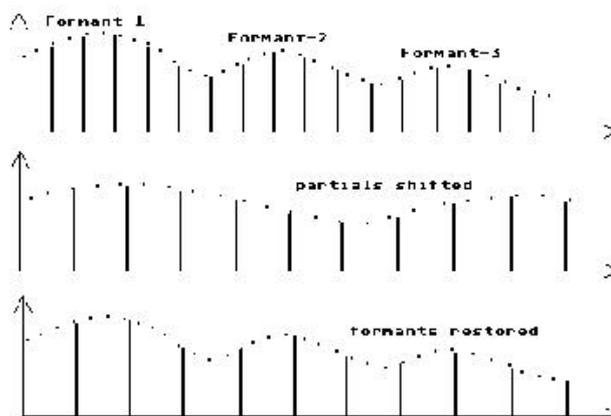


Fig. 5

PVOC, Fig. 5 – using -w to warp, retaining formants

N also sets the analysis channel bandwidth: $SR/N = \text{bandwidth}$. SR is sample rate. For example: $22050/512 = \text{a bandwidth of } 43.07\text{Hz}$.

Let's go through this again a little more slowly

- There is a 1-to-1 match between data in and data out. That is, for every sample of source soundfile, there is *Phase Vocoder* analysis data.
- Two source sound samples are used to create two data of analysis (the frequency/amplitude data pair). This is why it handles up to the Nyquist frequency (ie $SR/2$) and not beyond it.
- The FFT analysis produces a complex number with real and imaginary parts (because it is working in three dimensions). This result is then converted to the amplitude and frequency data (pair) placed in the analysis file.
- The value for **N** input by the user is actually the number of source sound samples used for each analysis frame. In doing so, the user determines the number of analysis channels, which is $(N/2) + 1$ because each pair of source sound samples (time domain) results in a frequency/amplitude pair (spectral domain). The '+1' refers to the extra channel at 0Hz.

Analysis rate: the 'sample rate' of the analysis file:

Each analysis frame uses **N** samples from the source sound. Therefore each second of analysis would use SR/N frames if the frames did not overlap.

Example: $22050/512 = 43.07$ – that is, 512 samples per frame divided into the total number of source samples per second = 43.07 frames per second. This is the 'analysis sample rate' when there are no overlaps.

However, in order to improve the analysis, *frames are made to overlap*, normally by 1/8th the length of a frame (see Fig. 7). Thus the frame position is moved on in steps of $N/8$ samples (e.g., $512/8 = 64$). The analysis 'sample' rate can be described as how many of these steps of 64 samples will fit into one second of sound, or, expressed another way, how many frames will be produced when moving through the file in steps of 64 samples: e.g., $22050/64 = 344$ analysis frames per second; this is the 'analysis sample rate'. More simply, $43.07 \text{ frames} * 8 \text{ steps} = 344.56$ frames per second. Thus the 'sample rate' of the analysis file is really a 'frame rate', and each frame contains $(N/2) + 1$ channels. For convenience, the decimal portion of these calculations is often omitted.

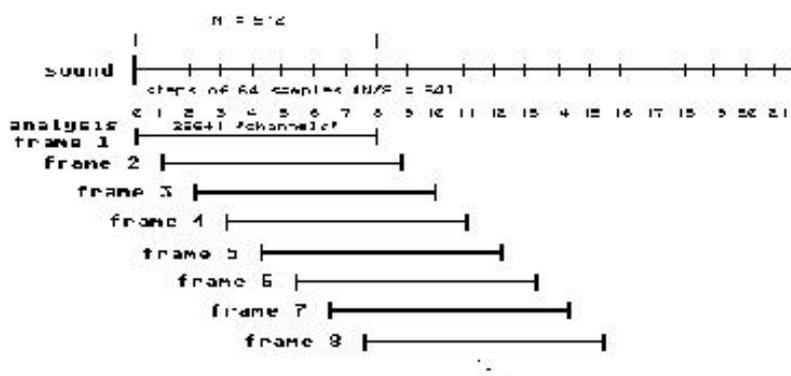


Fig. 7 Frame overlap
(When -D - 8 : 'Decimation flag')

PVOC, Fig. 7 – analysis sample rate = frame rate per sec * number of overlaps per sec

Summary of Terminology:

points (N)

The usage refers to this parameter as the 'number of bandpass filters'. The number **N** is actually the number of samples used per analysis frame from the source sound (time domain), and the number of bandpass filters (or 'channels') is the number of data pairs that result in the frequency or spectral domain after the FFT analysis. This is why the actual number of filters or channels is $(N/2) + 1$.

frame

What, then, is a 'frame'? The frame is the basic unit 'time-slice' of analysis. It contains the data for the $N/2$ bandpass filters or channels. A **'bin'** contains the data for *one of these channels*. The frame is windowed. If **N** = 512, there will be data for 256 channels in each frame: ie 256 data pairs (frequency and amplitude) stored sequentially, plus one extra channel at 0Hz.

Thus the amplitude value of the waveform sample (time domain) has been turned into a frequency/amplitude data pair (spectral domain). Note that each of the 256 bandpass filters or channels has only two values in it: i.e., each can only 'capture' one partial.

step

The number of samples the analysis is moved forward in the source sound before creating another frame. This step is by default $N/8$, but the 8 can be altered by the user. The parameter option to do this is the -D flag, referred to as 'decimation'. The word 'decimation' doesn't seem so odd when one realises that the larger the value for D, the smaller the step will be. The default, 8, is in fact the largest allowed.

window

refers to the fact that the bin or frame has been shaped to avoid glitches. See Fig. 1.

*RECAP: **N** is the number of source sound samples analysed per frame, yielding 256 channels +1 of analysis data in a frame or window. There is by default a frame overlap of $N/8$ samples, meaning that every sample contributes to 8 analysis frames. The amount of overlap can be changed, though this is not normally done. There are therefore $SR/step$ samples per second or $SR/N*8$ frames per second, which is the analysis 'sample' rate.*

The analysis 'bandwidth':

$SR/2$ (the Nyquist frequency) divided by $N/2$ (actual number of bandpass filters) yields the range of frequencies handled by each filter band. Example:

If the sample rate is 22050, half the sample rate is 11025 samples. When **N** = 512, $N/2 = 256$; $11025/256 = 43.07\text{Hz}$ per analysis channel. This can be thought of as the frequency resolution of the analysis. One doesn't divide by 257 here because 257 is the number of channel centres.

Now let's dig deeper still with few more questions.

What does the Nyquist frequency tell us?

First of all, the point of the Nyquist frequency in this context is that it tells us what frequency range is defined by a given sample rate. The Nyquist frequency is simply $SR/2$: half the sample rate. Thus, the Nyquist frequency for a sample rate of 22050 samples per second is 11025 (Hz). This is the highest frequency which can be articulated at this sample rate. Put another way, a sample rate of 22050 defines a frequency range from 0 to 11025 Hz.

When the Phase Vocoder divides up the sound into a number of equally spaced analysis bands or 'channels', it is dividing up the frequency range defined by the sample rate. This is why, when we're calculating the analysis bandwidth, we work with the Nyquist frequency, $SR/2$.

How do we know how large a value to enter for **points (N)** to analyse a certain sound?

This is answered by the section on Channels above. Here's some more ideas on this subject.

The analysis bandwidth is the range of frequencies covered by each bandpass filter or analysis channel. It is calculated by dividing the Nyquist frequency by half the value you provide for **N**: $SR/2$ divided by $N/2$.

Some examples:

SR	Nval	NyqFrq	N/2	Analysis Bandwidth
22050	512	11025	256	43.07Hz
44100	512	22050	256	86.13Hz
48000	1024	24000	512	46.88Hz
22050	1024	11025	512	21.53Hz
22050	2048	11025	1024	10.77Hz

The partials of a 'harmonic' sound are integer multiples of the fundamental, meaning that each harmonic will be equally spaced by the number of Hz of the fundamental itself: $110 * 1 = 110$, $110 * 2 = 220$, $110 * 3 = 330$, $110 * 4 = 440$, etc.

The rule of thumb here therefore, is to make sure that the analysis bandwidth is smaller than the fundamental. Be especially careful when analyzing low sounds that you use a sufficiently large value for **N**. Why? – so that each partial of the source sound will have a separate analysis channel; otherwise, several partials will be lost.

It is important to realise that the above rule of thumb is really valid only for 'harmonic' sounds, sounds in which all or most of the significant partials are integer multiples of the fundamental. Many of the sounds we work with are in fact rather complex, from mildly inharmonic sounds to those containing quite a bit of 'noise' (fairly random combinations of partials). These complex sounds will, of course, also require a fine resolution, such as a **points (N)** value of 2048 (1012 + 1 analysis channels).

It might prove interesting however, to use a low **N** value with a low or complex sound, knowing that you are going to lose data, to create gross approximations, and possibly create some sonic artifacts: sound effects not in the original sound, but resulting from the poor analysis.

In working with the *Phase Vocoder*, it would be useful to have a handy reference chart in the form of a piano keyboard or list of MIDI notes with the appropriate frequency in Hz. It would be easy to work this out using the CDP program INFO UNITS (was MUSUNIT). You'll also find a chart like this on p.21 of *The Science of Musical Sound*, by John R. Pierce.

What about the numbers shown in the soundfile directory display?

The two numbers which may look a bit odd are the one for the number of channels and the one for the sample rate.

The figure shown as the 'number of channels' is easy: it's the value you enter for **N** plus 2 because an extra (real) channel is placed at 0Hz.

Therefore, when you enter 512, the number of channels is given as 514, etc., and the actual number of analysis channels is in this case 257. The use of **N**/2 to calculate bandwidth still seems to be valid, however.

The figure shown as the 'sample rate' for an analysis file has been explained above.

How do we relate bandwidth to channel centres and boundaries?

So far, we have looked into how the analysis frequency bandwidth is affected by the value we give for **N**, why the number of channels and the sample rate for analysis files appear as they do in the soundfile directory display. Try performing these calculations with different values for the sample rate and for **N** and check the results against what actually appears in the soundfile directory display when you actually run the *Phase Vocoder* with these values.

We can now use what we've learned from the above discussion to gain a deeper understanding of two more aspects of the analysis channels themselves: 'centres' and 'boundaries'. Given a sample rate of 44100 and an **N** value of 512, the analysis bandwidth will be 86.13Hz. We're using the same numbers as on p. 3, above. This means that channel centres will occur at every 86.13Hz, starting with 0. The channel boundaries lie half way in between.

Thus the pattern is:

Channel No.	Low boundary	Centre	High boundary	Comments
0	0	0	43	(This is the extra channel, with boundaries from 0 - 43 to 0 + 43)
1	43	86	129	(86 - 43 ... 86 + 43)
2	129	172	215	(172 - 43 ... 172 + 43)
3	215	258	301	(258 - 43 ... 258 + 43)
4	301	344	387	(344 - 43 ... 344 + 43)

As shown in the section on the 'structure of the analysis file' (also see Fig. 3 above).

This shows us what's going on, but of course it's a very tedious way of determining in which analysis a given frequency in the source sound might appear and this is information which we will often need.

(For example, the former programs SPECE, SPECF and SPECSH had a parameter called *fdcno* ('frequency divide channel number'), i.e., the number of the channel in which a certain frequency appears, above or below which we want something to happen – the newer versions have built this calculation into the software.)

This information can be acquired by running the utility program PITCHINFO CHANNEL. The appropriate channel number is output by the program. You can also enter a channel number and find out which frequency it will contain.

But in an effort to take another step in understanding this powerful software, let's review how a particular *fdcno* is calculated.

1. We already know how to calculate the analysis channel bandwidth: the Nyquist frequency (i.e., $1/2$ SR) divided by the number of analysis channels ($N/2$).
2. We then find the channel number we need simply by dividing the frequency we want to use by the analysis channel bandwidth, and rounding up or down depending on whether the decimal part of the answer is above or below .5. This rounding handles the presence of the additional analysis channel at the beginning.

To illustrate with numbers, when $SR = 22050$, $N = 512$, and the frequency in question is 112Hz:

- $11025\text{Hz} / 256\text{channels} = 43.07\text{Hz}$ bandwidth per channel
- $112\text{Hz} (\text{frequency we want to use}) / 43.07\text{Hz}$ bandwidth per channel = channel number 2.6, which rounds to 3 (only integers make sense as channel numbers).
- Thus the frequency 112Hz will be found in the 3rd analysis channel. The following chart confirms this:

Chan No.	Low boundary	Centre	High boundary	Comments
0	0.0	0.0	21.53Hz	(The extra channel)
1	21.53	43.06	64.59Hz	
2	64.59	86.12	107.65Hz	
3	107.65	129.18	150.71Hz	
4	150.71	172.24	193.77Hz	

Conclusion

We have traced what happens to the **source samples** as they are analysed, put into **bins** containing one channel each; then $N/2$ channels are assembled into a **frame**, which is **windowed**, and 8 frames **overlap** if the step by which the analysis moves on is $1/8$ th of a frame. The **analysis sample rate**, which appears in the sample rate part of the soundfile display, is therefore very different from the sample rate of the source sound because it is really a **frame rate**: the number of frames per second -- which must take account of the frame overlap.

The discussion of **channel centres**, **boundaries** and **bandwidth** helps us understand how a given analysis may or may not effectively capture the frequency information of a given soundfile. This information is also relevant to understanding what is meant when one of the spectral manipulation programs does something with channels, such as select the loudest component after comparing equivalent channels in several analysis files. Each channel will contain only one partial, so if there are 5 soundfiles, only one partial will 'go through to the next round': the partials belonging to 4 of the soundfiles will be dropped.

The way the frequency-amplitude information is derived from the amplitude-time information is a complicated process, with many ambiguities and approximations. Many things can be happening in the analysis bandwidth which the software may not deal with as intelligently as it might. This is one of the reasons why experimentation and careful listening are very important aspects of working with the spectral dimension.

Richard Dobson's text [The Operation of the Phase Vocoder](#) takes us deeper into these uncharted but vitally important areas, but the most beneficial of all will be your own exploration of these programs. There is no substitute for entering into the psycho-acoustic experience of the results as a discerning and practiced musician.

OLD USAGE

Retained here for reference: in case any CDP users are still working with previous versions of the program, and to cross-reference any descriptions of pvoc flags which may occur in the CDP documentation. The present CDP version (from Release 4 onwards) is optimised for basic operations, so it is recommended that existing CDP users retain their older version if they have been using the various flag options available there.

pvoc [flags] infile outfile

Examples:

```
pvoc -N1024 -T4.7 in_orig_sound out_stretched_sound
pvoc -N512 -A in_orig_sound out_analysis_file
pvoc -S in_analysis_file out_new_soundfile
```

Flags:

- Rsr** input sample rate (automatically read from stdin)
- Nnum** number of source samples per frame (Default is 256 unless **-F** is specified); values must be powers of 2, e.g., 256, 512, 1024, 2048) – produces **N**/(2+1) chnls
- Ffrq** specify fundamental frequency (**R**/256) DON'T USE **-F** AND **-N**
- Ma_wlen** analysis window length (**N**-1 unless **-W** is specified)
- Wo_factor** filter overlap factor: {**0**, **1**, **2** (Default), **3**} DON'T USE **-W** AND **-M**
- Ls_wlen** synthesis window length (**M**)
- Dd_factor** decimation factor (min (**M**/(8*T)), (**M**/8))
- Ii_factor** interpolation factor (= **T** * **D**)
- Tt_factor** time-scale factor (a multiple: 1.0 = no change) values > 1 stretch, < 1 compress
- Pp_factor** pitch-scale factor (a multiple: 1.0 = no change) DON'T USE **-T** AND **-P**
- Cchannel** re-synthesize odd (**1**) or even (**2**) channels only
- ichannel_no** re-synthesize bandpass filters **i** thru **i** only
- jchannel_no** re-synthesize bandpass filters **i** thru **j** only
- b[s_sample]** starting sample (Default is 0 – beginning of file)
- ee_sample** final sample (end of input)
- w[w_factor]** warp factor for spectral envelope (Default: 1.0) – retains formant shape, e.g., of a voice
- A** analysis only; output will be analysis data
- E** analysis only; output will be spectral envelope
- X** analysis only; output will be magnitude values
- S** synthesis only; input must be analysis data
- K** use Kaiser filter instead of hamming
- V** verbose: summarize analysis data in the file *pvoc.s*

NB – The key flags for the majority of applications include: **-N**, or **-F**, **-A**, **-T** or **-P**, **-S**, **-i**, **-j**, **-b**, **-e**, and **-w**. The others are for specialist applications. A command line with a soundfile input and without **-A** or **-S** will perform both, as in the first example command line above.

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