*	Sampling rate	F_s = Number of samples per second N_{sps}
*	Nyquist frequency	$F_n = F_s/2$; (also called *folding frequency*)
*	Sampling interval	$T_s = 1/F_s$ (in seconds) or $T_{ms} = 1000/F_s$ in ms; (*Keep 5 decimals*.)
	No. of samples per ms No. of samples per sec	$S_{pms} = (=N_{spms}) = F_s/1000$; (*Keep two decimals*) $S_{psec} = (=N_{spsec}) = F_s$
	Aliasing frequencies	Let k = integer part (floor) of f/F_n ; then: $f' = f - (k F_n)$, if k is even (including 0); $f' = (k+1) F_n - f$, if k is odd; where f is the original input frequency and f' is the effective (aliased) frequency in the Nyquist band
	Center frequencies of DFT filters	$f_i = (i-1) \cdot F_s/N_{dft}$, where f_i is the center frequency of the *i-th* filter; N_{DFT} is (integer) length of the sequence to be transformed in sample points.
	Number of useable filters in DFT analysis	$N_{filters}$ = ceiling$((N_{dft}+1)/2)$ {Ceiling means round up to next largest integer so ceiling(1.1)=2This is equal to $(N_{dft}/2)+1$ for even N_{dft} (the usual case) and $(N_{dft}+1)/2$ for odd N_{dft}; }
	Bandwidth of rectangular window in DFT analysis	$bw_{rect} \approx 0.88\, F_s/M_{win}$, where M_{win} is the number of sample points of the nonzero part of window, so $M_{win} = N_{dft} - P$ where P is number of zeros appended to the sequence; or $bw_{rect} \approx 0.88 \cdot (1000/D_{ms})$, where D_{ms} is the duration of the nonzero part of the window in ms = M_{win}
	Bandwidth of Hamming window DFT analysis	$bw_{Ham} \approx 1.3 \cdot F_s/M_{win}$ or $bw_{Ham} \approx 1.3 \cdot (1000/D_{win(ms)})$
*	Conversion of duration in ms to length in samples {-> integer}	$N_{pts} = D_{ms} \cdot S_{pms} = D_{ms}/T_{ms} = D_{ms} \cdot (F_s/1000)$ {-> integer} where N_{pts} is number of sample points, D_{ms} is segment duration in milliseconds (in seconds, the formula is just $N_{pts} = D_s \cdot F_s$)
*	Conversion of length in samples N to duration in ms ; (or seconds)	$D_{ms} = N \cdot T_{ms} = N/Spms = 1000 \cdot N / F_s$; $(D_s = N \cdot T_s = N / F_s)$
	Conversion between advance interval in time units and in sample points	$N_{adv} = D_{adv} \cdot S_{spms} = D_{adv} \cdot F_s/1000$ $D_{adv} = N_{adv} / S_{pms} = N_{adv} \cdot 1000 / F_s$ D_{adv} is frame advance interval in ms {keep two decimal places} N_{adv} is frame advance interval in sample points {-> integer} S_{pms}= number of samples per ms {keep two decimal places}
	Rule of thumb for advance interval for speech	For high resolution analysis of speech signals, $D_{adv(ms)}$ (frame advance interval) should be no more than about ¼ $D_{win\,(ms)}$ (window duration)

	Rule of thumb for filter spacing	For high resolution spectrograms of speech signals, spacing of filters, ΔF, should be no more than about 1/4 of the bandwidth.
	Frequency spacing of filters in a DFT	$\Delta F = F_s / N_{dft}$, where N_{dft} is length of DFT in points and ΔF is frequency spacing in Hz of filters
	Length of DFT required for given frequency spacing	$N_{dft} = F_s / \Delta F$ where ΔF is spacing between centers of adjacent filters (Hz)
	Length of Hamming window required to achieve a particular bandwidth in a DFT analysis	M=1.3 Fs/ bw_{Ham} or $D_{win(ms)}$=1.3 (1000/ $bw_{Ham.}$) where $bw_{Ham.}$ is the desired bandwidth, M is the length of the window in sample points and $D_{win(ms)}$ is the duration of the window in milliseconds.
*	Number of expected formants below a given frequency	$N_{formant} = F_u / (2\ F1_{neut})$ where F_u is the upper frequency limit and $F1_{neut}$ is the expected F1 for a 'neutral vowel' (schwa) for a given speaker. Approx 500 Hz for adult males or 600 Hz for adult females. Can be as high as 800 Hz for small children.
*	Number of LPC coefficients required to model each formant	*2*
*	Rule of thumb for number of LPC coefficients to use to analyze speech	$N_{coef} = 2\ N_{formant} + K_{GR}$; (note: $LPC\ order = N_{coef} + 1$) where K_{GR} is a small integer, usually in the range of 1-4 to accommodate 'glottal source' and 'radiation' characteristics. Number must be adjusted empirically (i.e., by trial and error).
*	Conversion of autocorrelation lags (or cepstral 'quefrency') to frequency estimate	$F_{est} = 1000 / L_{ms}$ *or* $F_{est} = F_s / L_{samp}$ where L_{ms} is lag measured in ms, while L_{samp} is the number of lags counted as sample points.
*	Duration of window for autocorrelation pitch analysis	*2.5 times longest expected period.*
	Search range for pitch estimates in autocorrelation or cepstral function	$t_{max} = 1000 / F0_{min}$; $t_{min} = 1000/F0_{max}$ *where* t_{max} *and* t_{min} *are measured in ms.* $lag_{max} = F_s / F0_{min}$; $lag_{min} = F_s / F0_{max}$ *where* lag_{max} *and* lag_{min} *are measured in sample points.*
	Length of DFT required for given frequency spacing	$N_{dft} = F_s / \Delta F$