```
In [7]:
            import numpy as np
            from numpy.fft import fft
            import matplotlib.pyplot as plt
            import scipy.io.wavfile as spwav
            from IPython.display import Audio
            #from mpldatacursor import datacursor
            import sys
            The entire signal will be voiced.
            The epoch locations in the original signal are precomputed and given as epoch
            a NumPy array containing the indices of each epoch marker.
            TD-PSOLA can be computed over the entire signal, not just individual frames.
            This leaves remapping, windowing, and overlap-adding for you to implement as
            in the Overlap-Add Algorithm section. Your implementation will be expected to
            any synthesis frequency within the nominal human vocalization range [100 Hz,
            A quick implementation note -- earlier in the lab notes, we assumed some fixe
            our entire signal and computed our window as length N = 2 * P 0 + 1. This ass
            signal is unchanging over time, which is clearly not true.
            Instead, compute P_0 as the average of the distance between the nearest two e
            ie P \emptyset = (epoch[i + 1] - epoch[i - 1]) / 2 for your current epoch i.
            plt.style.use('ggplot')
            # Note: this epoch list only holds for "test vector all voiced.wav"
            epoch marks orig = np.load("test vector all voiced epochs.npy")
            F s, audio data = spwav.read("test vector all voiced.wav")
            N = len(audio data)
            #print(audio data)
            Implement TD-PSOLA on the given test file using the starter Python code given
            Try for various frequencies, F_new = 100, 200, 300, 400 etc.
            \#F \text{ new} = 420
            #new epoch spacing = int(F s/F new)
            P0 = np.zeros(len(epoch marks orig)-2, dtype = 'int')
            P0[-1] = epoch marks orig[-1] - epoch marks orig[-2]
            y = [] # store windowed response, len = epoch mark - 2
            for i in range(1, len(epoch marks orig)-1): # extract top and bottom
                start = epoch marks orig[i-1]
                end = epoch marks orig[i+1]
                cur = epoch marks orig[i]
                P0[i-1] = int((end - start)/2) #P0 for each epochs
                # extract the impulse response (rather, an estimate of the impulse respor
                # by windowing ±P0 about each epoch marker.
                w = np.hamming(2*P0[i-1])
                y.append(np.multiply(audio data[ cur - P0[i-1] : cur + P0[i-1] ], w))
                #y.append(audio_data[cur - P0[i-1]: cur + P0[i-1]])
            P0 \text{ avg} = sum(P0)/len(P0)
                                      # fundamental period
            F0 = int(F s/P0 avg)
            Fn = [100, 400, 600]
            ratio = np.zeros(len(Fn), dtype='float64')
```

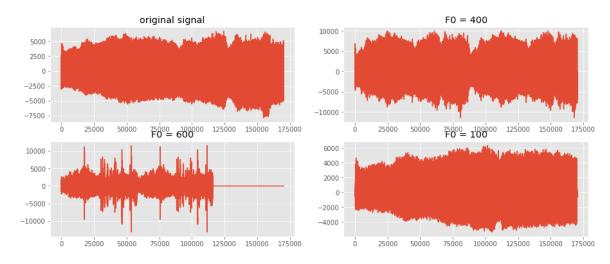
```
Pn avg = np.zeros(len(Fn), dtype = 'int')
                                            # new epoch spacings
Pn = np.zeros((len(Fn), len(P0)), dtype = 'int')
for i in range(len(Fn)):
    ratio[i] = Fn[i] / F0
    Pn_avg[i] = int(P0_avg/ratio[i])
    for n in range(len(P0)):
        Pn[i, :] = P0/ratio[i] # new_epoch_spacing
#print(y)
# Suggested Loop
def get audio(n, N, Pn, P0, y):
    audio_new = np.zeros(N)
    #n = index
    P = 0 #position of the new epoch
    zero start = 0
    for i in range(len(P0)):
        P+=Pn[n,i]
                     # epoch position
        yl = len(y[i]) #length of the windowed frame
        start = P - int(y1/2)
        end = P + yl - int(y1/2)
        if ratio[n] < 1:</pre>
                           #zeropadding
            \#print(P + yl - int(yl/2) - (P - int(yl/2))+1)
            if end > N: #boundary
                audio_new[start: N-1] += y[i][0:N-1-start]
                return(audio_new)
            else:
                audio_new[ start : end ] += y[i]
        else:
                #overlapping
            # boundary
            if (start < 0):
                print(start, end, yl)
                audio new[0:end] += y[i][-start-1:-1]
            else:
                audio new[ start : end ] += y[i]
                zero start = end
    print(audio_new [ zero_start : -1 ])
    length = len(audio_new[zero_start: -1])
    audio new[zero start:-1] = audio new[0:length]
    return(audio new)
out = []
plt.figure(figsize = (15,6))
plt.figure(figsize = (15,6))
for i in range(len(Fn)):
    out.append(get_audio(i,N,Pn,P0,y))
plt.subplot(221)
plt.title('original signal')
plt.plot(audio data)
plt.subplot(224)
plt.title('F0 = 100')
plt.plot(out[0])
plt.subplot(222)
plt.title('F0 = 400')
```

```
plt.plot(out[1])
plt.subplot(223)
plt.title('F0 = 600')
plt.plot(out[2])

plt.show()
-105 327 432
```

-105 327 432 [0. 0. 0. ... 0. 0. 0.] -142 290 432 -69 365 434 [0. 0. 0. ... 0. 0. 0.]

<Figure size 1080x432 with 0 Axes>



In []: ▶