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Optimized Configuration of the Short Time DFT Compander and its Noise Reduction Effect

Short Time DFTコンパンダの最適化構造とその雑音抑圧効果

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ABSTRACT　*Effective utilization of the radio frequency resources is eager to be promised in narrowing spectrum occupancy bandwidth or saving transmission power. Unfortunately, the more efficient in narrowing bandwidth of angular modulation signals the merit of low-noise loses worse in wide-band amplifiers, or the more effective in saving transmission power speech quality is the worse suffered from such noises as thermal and fading. The noise reduction facility of implementing compander or diversity receiving becomes to be inevitable in the narrow band communication systems for preventing frequency resources from exhausting. The short time DFT compander achieves companding on the frequency domain with employing the concept of instantaneous spectrum without detecting the input signal envelopes. Where input signals are analyzed into instantaneous spectrum through the short time DFT, companding is performed via such operations as multiplying or dividing these components of instantaneous spectrum without giving any phase deviation to the vector of the spectrum. The output signal is, therefore, realized with synthesizing from the companded instantaneous spectrums. Noise reduction, which is one of the most important function of the compander, is examined to be sufficient for cellular telephone systems through baseband fading simulation system over 900MHz channel of existing cellular systems.*

1.INTRODUCTION

In vehicular communication system, propagation is generally carried over multipath quasi state field. Figure 1 shows

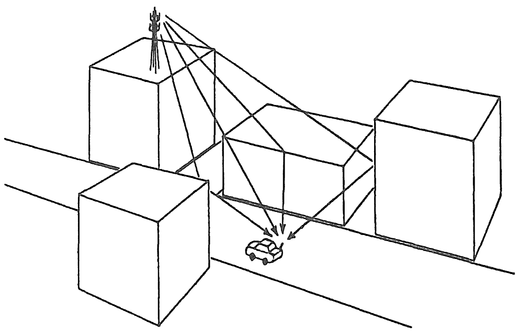

Fig.1 Propagation of vehicular communication.

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the image of occurring multipath fading. The compander or diversity receiving is indispensable to improve speech quality when communication being carried in the multipath quasi state field from passing vehicles. The existing syllabic compander of employing the feed back loop is widely adopted in the vehicular communication

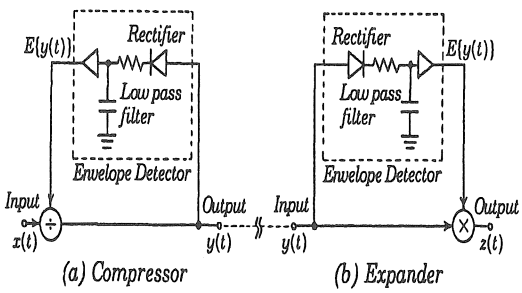


Fig.2 Configuration of FB compressor(a), and expander(b).

systems or cordless telephone sets. The existing syllabic compander takes the theoretical bases on detecting envelopes from input signals. Figure 2 shows the configurations of the existing feed back syllabic companders (ab. in FB companders), and shows which consists of a rectifier and a low pass filter. The circuits, which is originally used as AM demodulator, is able to operate as the envelope detector only if carries is so high frequency as 1,000 times by the baseband signals. In baseband frequencies, it is difficult to directly adopt the AM demodulator to the envelope detector of the compander, because the difference between equivalent carrier of voice and the equivalent baseband signal of voice pitch is at most 10 times. The feed back loop, which is seemed to be essential in companding, is successfully eliminated from short time DFT compander (ab. in ST DFT compander) without any excessive distortions even with improvements of reduction in such distortion as harmonics, transient, intermodulation, and etc.

2.PRINCIPLE AND CONFIGURATIONS OF THE ST DFT COMPANDER

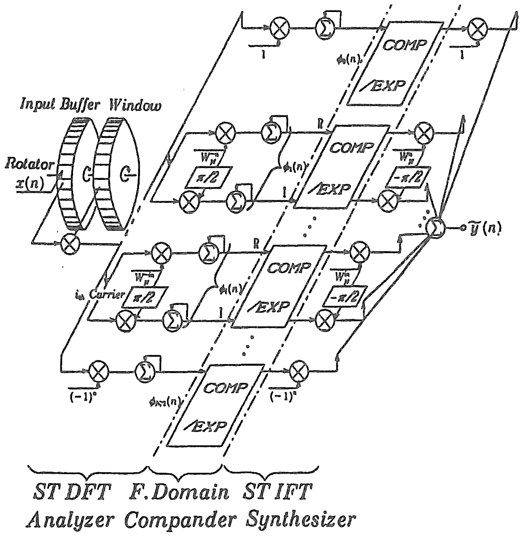


Fig.3 Circuitry configuration of ST DFT compander.

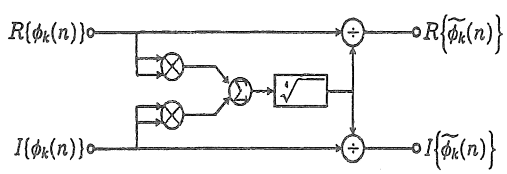
Circuitry configuration of the ST DFT compander is categorized into three major blocks as shown in fig.3. The first block is ST DFT analyzer which consists N/2+1 modules in frequency index k wise. The second block is a significant block in function of compressor/expander on the frequency domain, whose detailed implementation is schemed in figs.4(a) and (b) / figs.4(c) and (d). The last is a ST IFT synthesizer to produce the time domain companding signals from the companded instantaneous spectrum. In similar to the first block, the ST IFT synthesizer plays the role of modulating companded spectrum component $\widetilde{\phi_k}(n)$ with complex carrier W_N^{nk} .

Let the instantaneous spectrum at sampling time n be $\Phi(n)$,

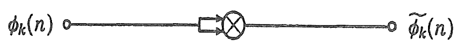
$$\Phi(n) = \{\phi_0(n)\ \phi_1(n) \cdots \phi_{N-1}(n)\}^T. \qquad (1)$$

Here, $\phi_k(n)$ is a spectrum component at frequency index k of $\Phi(n)$. The spec

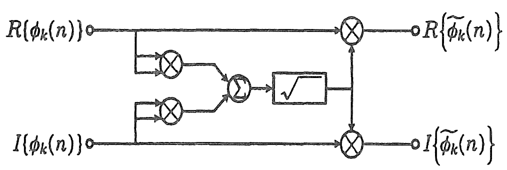
(a) *Sub - module compressor for index k=0 or N/2*



(b) *Sub - module compressor for index 0<k <N/2*



(c) *Sub - module expander for index k=0 or N/2*



(d) *Sub - module expander for index 0<k< N/2*

Fig.4 Detailed schemes of the frequency domain compressor (a)(b), and expander (c)(d).

trum component $\phi_k(n)$ is defined by the ST DFT as follows,

$$\phi_k(n) = \sum_{r=-\infty}^{\infty} x(r)h(n-r)W_N^{-rk} \qquad (2)$$

where, $x(r)$ is an input data at sampling time r, $h(*)$ is a window function,

W_N^{-rk} is the same operator defined in the existing DFT as follows,

$$W_N^{-rk} = \exp\left\{-j\frac{2\pi rk}{N}\right\}, \qquad (3)$$

integer k is $0 \leqq k < N$

$h(*)$ is such apriori window function as truncated Nyquist.

To improve stop-band attenuation and pass-band smoothness, this truncated Nyquist is weighted with Kaiser as follows.

$$h(p) = h_N(p)h_K(p), \qquad -mN \leqq p \leqq mN$$

Here, $h_N(p)$ is truncated Nyquist by 2m frames, $h_k(p)$ is following Kaiser window,

$$h_k(p) = \frac{I_0\left\{\beta\sqrt{1-(p/mN)^2}\right\}}{I_0(\beta)}$$

Where, $I_0(*)$ is the modified 0th ordered Bessel I function, β is arbitrary value to adjust width and concentration of the main lobe energy.

As discussed in aboves, the truncated Nyquist weighted by Kaiser is improved to be almost equal to the ideal filter in characteristics as the proto-type filter without requiring any excessive computing power to the ST DFT.

Compressing $\phi_k(n)$ is interpreted as dividing $\phi_k(n)$ by $|\phi_k(n)|^a$ along to the vector $\phi_k(n)$ on the frequency domain as follows,

$$y_{cmp}(n) = \frac{1}{N}\sum_{k=0}^{N-1}\frac{\phi_k(n)}{|\phi_k(n)|^a}e^{j\frac{2\pi nk}{N}} \qquad (4)$$

When compressing rate is 2 to 1 in decibel meanings, which is same value to the existing companding system, a is chosen to be 0.5. The expanding is also realized as multiplying the instantaneous spectrum component $\phi_k(n)$ by $|\phi_k(n)|^{b-1}$ as given by eq.5.

$$y_{exp}(n) = \frac{1}{N}\sum_{k=0}^{N-1}|\phi_k(n)|^{b-1}\phi_k(n)e^{j\frac{2\pi nk}{N}} \qquad (5)$$

Here, $y_{exp}(n)$ is expanded signal, and

b is expanding rate of 1 to b in dB meanings.

As discussed aboves, both compressed and expanded signals are themselves denoted by similar formula of instantaneous spectrum expansion. If arbitrary compressing and expanding rate are required, values a and b are sufficient to be set to the reciprocal number of required value. It is easy to understand that the ST DFT compander is guaranteed by being free from any distortion in companding according to multiplying or dividing the magnitude of the instantaneous spectrum component with avoiding deviation in phase.

3. FAST PROCESSING FOR THE ST DFT COMPANDERS

3.1 Fast Algorithm based on the FFT Structure

The ST DFT compander is free from any distortion owing to manipulation of instantaneous spectrum. However, it is suffered from great deal of computing in ana lyzing input signal through ST DFT. It becomes an important problem in reducing processing power to put the ST DFT compander on the development stages. Eq.2 is modifies by changing variables $r = n + s$, and $s = lN + m$ as follows.

$$\phi_k(n) = \sum_{m=0}^{N-1} \sum_{l=-\infty}^{\infty} x(r)h(n-r)W_N^{-rk} = W_N^{-nk} \sum_{m=0}^{N-1} \sum_{l=-\infty}^{\infty} x(n+lN+m) h(-lN-m)W_N^{-mk} \quad (6)$$

That is, eq.6 is also modified into following FFT formulation.

$$\phi_k(n) = \sum_{m=0}^{N-1} x_m(n) W_N^{-mk} \quad (7)$$

Here, $x_m(n)$ as follows,

$$x_m(n) = \sum_{l=-\infty}^{\infty} x(n+lN+ [m-n]_N) h(-lN- [m-n]_N),$$

$$[M]_N = M \bmod N$$

Eq.7 reduces the computing amount from $O(N^2)$ to $O(N\log N)$.

3.2 Fast Algorithm based on the Frequency Domain Interpolation

Where only somewhat degradation is allowed to fast processing from practical application of view, $\widetilde{\phi_k}(n)$ may be exactly reproduced from only thinned out $\phi_k(rR)$ at every R sampling clock as follows,

$$\widetilde{\phi_k}(n) = \sum_{r=L^-}^{L^+} f(n-rR)\phi_k(rR) \quad (8)$$

Here, $f(n-rR)$ is, for example, Lagrange interpolation of 2Q frame given by,

$$f(n-rR) = \frac{(-1)^{r+Q} \prod_{i=1}^{Q}(\frac{n}{R}+Q-i)}{(Q-1+r)!\,(Q-r)!\,(\frac{n}{R}-r)} \quad (9)$$

$$L^- = \left[\frac{n}{R}\right] - Q, \quad L^+ = \left[\frac{n}{R}\right] + Q - 1,$$

[∗] *represents the biggest integer contained* ∗.

The output signals $\widetilde{y}(n)$ are consequently synthesized from interpolated frequency given by eq.9 as follows.

$$\widetilde{y}(n) = \frac{1}{N} \sum_{k=0}^{N-1} \widetilde{\phi_k}(n) W_N^{nk} \quad (10)$$

Eq.10 gives a fast processing as shown in fig.5. The processing amount is reduced to $(2mN + 2NlogN)/R$. When R becomes

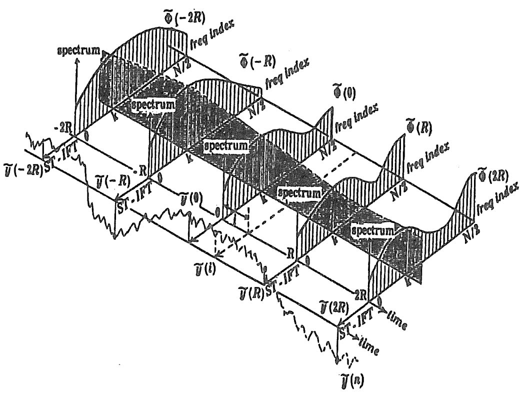


Fig.5 Fast processing diagram in the ST DFT compander based on frequency domain interpolation.

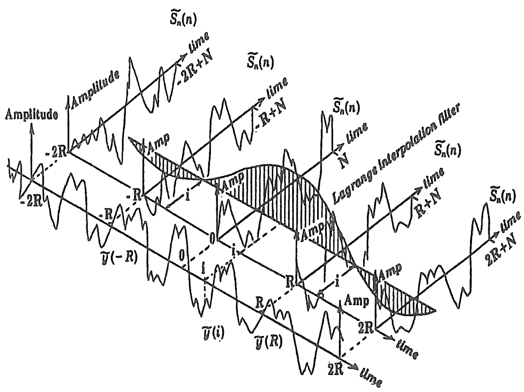


Fig.6 Fast processing diagram in the ST DFT compander based on time domain interpolation.

close to N with troublesome appearance of the Gibbs' phenomenon, the processing is mostly speeded up by N times. R is recommended to be less than N of the frame length.

3.3 Fast Algorithm based on the Time Domain Interpolation

Substituting eq.8 into eq.10, the output signals $\widetilde{y}(n)$ are given as,

$$\widetilde{y}(n) = \frac{1}{N}\sum_{k=0}^{N-1}\sum_{r=L^-}^{L^+} f(n-rR)\phi_k(rR)W_N^{nk} \qquad (11)$$

Since all of the operations in eq.11 are linear on the finite operand, eq.11 holds on exchanging the order of summations for k and r.

$$\widetilde{y}(n) = \sum_{r=L^-}^{L^+} f(n-rR)s_r(n) \qquad (12)$$

Here, every $s_r(n)$ is sub time domain signals as given by followings.

$$s_r(n) = \frac{1}{N}\sum_{k=0}^{N-1}\phi_k(rR)W_N^{nk},\ \ L^-\leqq r\leqq L^+ \qquad (13)$$

Eq.12 clearly shows that output signals $\widetilde{y}(n)$ are reproduced with interpolation from the sub-time domain signals $s_r(n)$ as shown in fig.6.

4.NOISE REDUCTION EFFECT

The ST DFT compander is substantiated to be ideal in companding input signals through computer simulations on CRAY X-MP 14se at AIT to avoid any round-off errors. The noise reduction effect of the ST DFT compander is also examined through computer simulations by using baseband fading simulation when voice being carried over multipath quasi state field. Assuming the frequency being 900

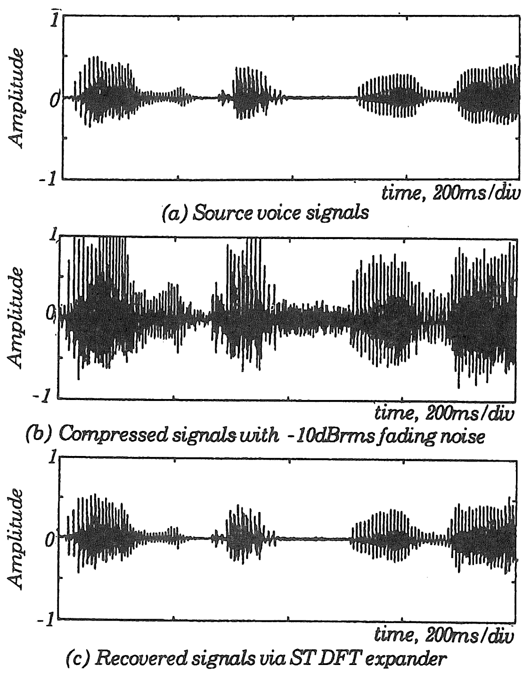


Fig.7 Noise reduction effect observed on the time domain

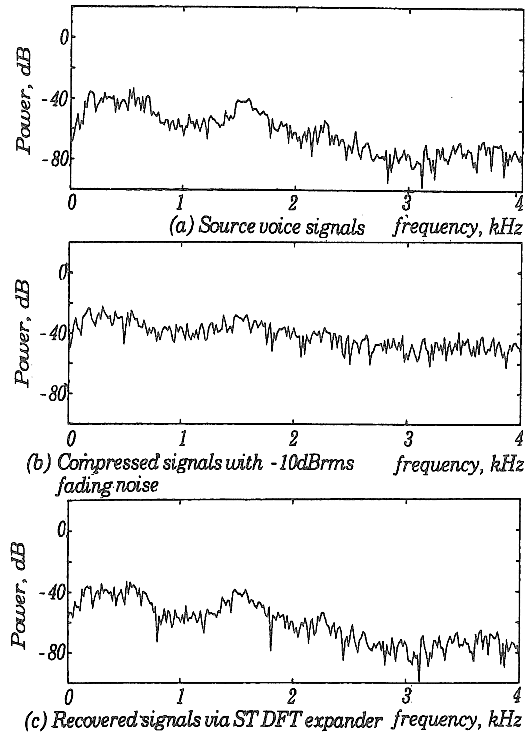


Fig.8 Noise reduction effect observed on the frequency domain

MHz and the noise level being -10dBrms, the speech carried over poor radio channel is suffered from 50Hz pitch fading noise as shown in figs.7 and 8. These figures indicate the noise reduction effect via the ST DFT compander, where window length 2m is 8 and frame length N is 32, interpolation frame number 2Q is 8, and interpolation duration R is 10 both in the ST DFT compressor and expander. These power spectrums are analyzed by the ST DFT under the condition of 2m=8 and N=500 of decimation filter in the ST DFT analyzer.

4.1 Considerations on the Time Domain Responses

Let's the original signal be given as shown in fig.7(a) and be added to the ST DFT compressor. Figure 7(b) shows the compressed signals interfered by -10dBrms noise through fading simulator for poor radio channel. It is clearly shown in fig.7 (b) that fading noise appears in the duration of speech signals power being small. Such damaged signals as shown in fig.7 (b) is almost perfectly recovered through ST DFT expander as shown in fig.7(c). The ST DFT expander is shown to effectively recover damaged signals both in breathed consonants and pronunciation vowels.

4.2 Considerations on the Frequency Responses

In the time domain responses, the ST DFT compander is seemed to perfectly recover the damaged signals as discussed in previous section. To discuss the noise reduction effect in more details, let's consider the frequency responses as shown in fig.8. Figure 8(a), (b), or (c) shows the frequency response corresponding to time domain signals shown in fig.7(a), (b), or (c), respectively. As shown in fig.8(a), the time domain signals features of male pronunciations by possessing 4 formants. The signals is so damaged only with -10 dBrms fading noise after transmission over poor radio channels as detecting no formants as shown in fig.8(b). That is, the communication is carried in very poor quality to put the receiver on the circumstance of confusing in speaker's identifications.

Nevertheless, the ST DFT expander has succeeded as shown in fig.8(c) in almost perfectly recovering the original signals from such damaged signal as shown in fig.8(b). It is ease to recognize that the ST DFT compander is superior in noise reduction more than existing compander.

5.CONCLUSION

A noble compander was discussed with emphasis on the instantaneous spectrum concept. The ST DFT compander, which has implemented by employing the instantaneous spectrum concept, has successfully discussed from circuitry configuration, fast processing algorithm, and noise reduction effect.

Multirate sampling signal processing is also efficiently introduced to reduce amount of processing in the ST DFT analyzer. The ST DFT compander is shown to be almost free from any distortion by avoiding detection of the voice envelopes, and to be effective in sufficiently reducing the fading noise over poor radio channels.

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