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Harmonics Distortion and Intermodulation in the Multi - Channel Compander Based on the Short Time DFT

Short Time DFTを用いたマルチチャネルコンパンダの高調波歪みと混変調歪み特性

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ABSTRACT *Reducing the spectrum occupancy over radio channel is as well known as important to prevent finite radio resources from exhausting. The syllabic compander provides indispensable function both to improve speech quality and to reduce fading noise in the cases of saving transmission power and of narrowing spectrum occupancy. Detailed analysis of previously reported multi-channel compander is shown with harmonic distortion and intermodulations based on the short time DFT*[1, 2]. *This compander is realized with three major functions to avoid envelope detectors, which mainly has caused any distortions in companding. All the input signals are analyzed into instantaneous spectrum via a short time DFT analyzer, then the magnitude of each spectrum component is manipulated by multiplying or dividing to yield companded spectrum, the companded signals are finally given by inverse Fourier transforming from the companded spectrum.*

1. INTRODUCTION

As the syllabic compander provides with indispensable function both to improve speech quality and to reduce fading noise in the case of saving transmission power and of narrowing spectrum occupancy, many investigations are keenly studied on realizing these syllabic companders.

Unfortunately, most of them have been concerned with estimating the envelopes in the approximation of AM demodulation. Where such approximated envelope detector is adopted to get companded signals, both intermodulation error and harmonic distortions are introduced to degrade speech quality and to expand the bandwidth of radio occupancy.

Instead of employing approximated envelop detectors, the short time DFT is adopted to analyze input signals into the instantaneous spectrum in order to generate companded spectrum. The time variant complex real and imaginary parts of each instantaneous spectrum component are interpreted as a kind of time variant coefficient of the Fourier series with exception of possessing common time variant phase function. Where input signals are analyzed into the instantaneous spectra, the envelopes are equal to the absolute of the vector spanned by the complex real and imaginary parts of each instantaneous spectrum component. Dividing or multiplying the absolute value of each frequency component gives

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the compressed or expanded frequency components to avoid inducing any phase deviations.

2. PRINCIPLE AND CONFIGURATIONS OF THE MULTI-CHANNEL COMPANDER

Circuitry configuration of the multi-channel compander is categorized into three major blocks as shown in fig.1 [3, 4]. The first block is ST DFT analyzer which consists of $N/2+1$ modules in frequency index wise. The second block is the significant block in function of compressor or expander on the frequency domain, whose detailed implementation is schemed in figs. 2(a) and (b) or figs. 2 (c) and (d), respectively. The last is the synthesizer to produce the companded signals through inverse ST DFT(ST IFT) from the companded 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 spectrum component $\phi_k(n)$is defined by the ST DFT as follows,

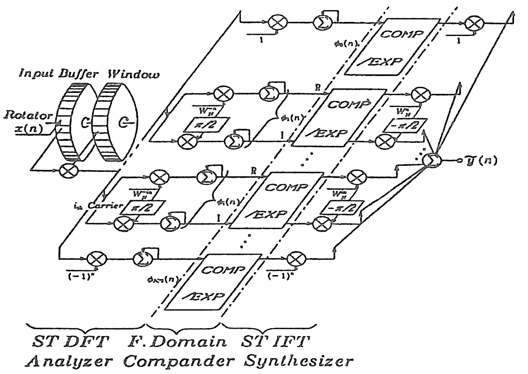

Fig.1 Circuitry configuration of the multi-channel compander.

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

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,

$$W_N^{-rk} = exp\left\{-j\frac{2\pi rk}{N}\right\}, \qquad (3)$$
$$integer\ k\ is\ 0 \le k < N,$$

$h(*)$is such an apriori window function. For example, Kaiser weighted Nyquist is adopted as follows [5],

$$h(p) = n(p)w(p) \qquad (4)$$

Here, $n(p)$ is Nyquist, $w(p)$is a Kaiser window,

$$w(p) = \frac{I_0\left\{\beta\sqrt{1-(p/mN)^2}\right\}}{I_0(\beta)}, \qquad (5)$$
$$-mN \le p \le mN$$

Where, $I_0(*)$ is the modified 0_{th} ordered Bessel function of the first, β is an arbitrary positive real to adjust the width and concentration of the mainlobe energy.

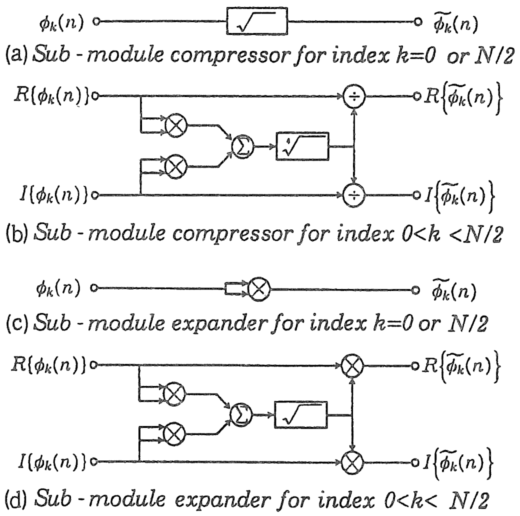

Fig.2 Detailed schemes of the frequency domain compresser(a)(b), and expander(c)(d).

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

Dividing $\phi_k(n)$ along to its vector by $|\phi_k(n)|^{\gamma}$ which is interpreted as frequency domain compressing, it gives the compressed output $y_{cmp}(n)$ synthesized via ST IFT from the compressed spectrum as follows,

$$y_{cmp}(n) = \frac{1}{N}\sum_{k=0}^{N-1}\frac{\phi_k(n)}{|\phi_k(n)|^{\gamma}}W_N^{nk} \qquad (6)$$

When compressing rate is 2 to 1 in decibel meanings, which is same value to the existing companding system, γ is chosen to be 0.5. If arbitrary compressing rate is required, values γ is sufficient to be set to the reciprocal number of required value.

The expanding is also realized as multiplying the instantaneous spectrum component $\phi_k(n)$ by $|\phi_k(n)|^{\delta-1}$ as given by eq.7.

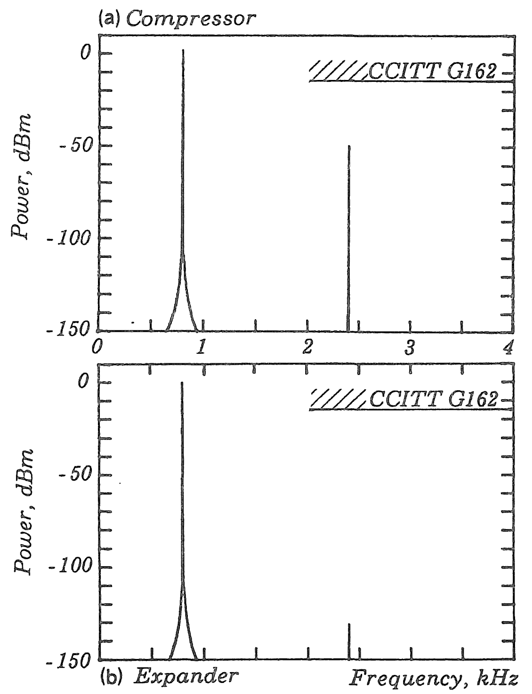

Fig.3 Power spectrum of the multi - channel compresser(a) and of the expander(b) for 800Hz 0dBm tonal signal, 2m=8, N=64, β=6.0.

$$y_{exp}(n) = \frac{1}{N}\sum_{k=0}^{N-1}|\phi_k(n)|^{\delta-1}\phi_k(n)W_N^{nk} \qquad (7)$$

Here, $y_{exp}(n)$ is expanded signal, and δ is expanding rate 1 to δ in dB meanings.

As discussed aboves, both compressed and expanded signals are themselves denoted by similar formula of instantaneous spectrum expansion. It is easy to understand that the multi - channel 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 phase deviation. The more effective computation for the compander was shown in ref.6 based on the multi - rate sampling in order to reduce computing power.

3. EXPERIMENT RESULTS

The multi - channel compander is examined to

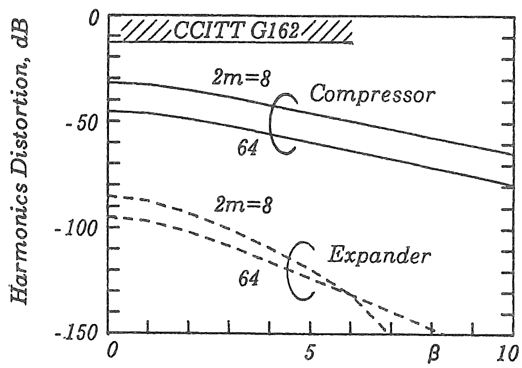

Fig.4 The harmonics distortion vs. β, 2m=8 or 64.

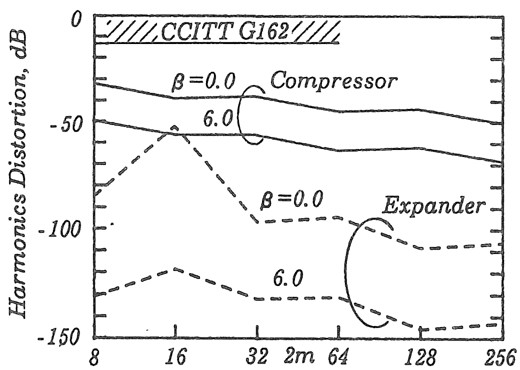

Fig.5 The harmonics distortion vs. the frame number 2m, β=0.0 or 6.0.

be almost ideal in companding through computer simulations. Basically, the multi-channel compander excludes any feed back loops, amplitude of companded signal is precisely compressed or expanded in 2 to 1 or 1 to 2 in dB meanings with almost equivalent level to the granulating error for tonal signals. The transient response are so ideal that both the attack and recovery time are vanished to zero under the CCITT Rec. G162 specifications for the tone-burst if $2m=8$, $N=64$, and $\beta=6.0$ [7].

Harmonics Distortion

Harmonics distortion, measured with 800Hz 0 dBm tonal signal, is recommended to be below 4%(-14dB) as shown in fig.3. Fiqure 3(a) shows clearly that the maximum distortion in the power spectrum appears at 2.4kHz as the third harmonic below -49.4dBm, and the second at 4.0kHz as the fifth harmonic below the -95.3dBm. The harmonics distortion of the multi-channel compressor is observed to be less than -49dB with more than 35dB margin to the -14dB criterion if $2m=8$, $N=64$and $\beta=6.0$. Figure 3(b) shows the spectrum of the multi-channel expander in the lower range than that of the compressor. In this figure, the maximum distortion appears at 2.4kHz as the third harmonic below -131.1dBm. The harmonics distortion of the expander is observed to be less than -130dB with about 120dB margin to the -14dB criterion if $2m=8$, $N=64$ and $\beta=6.0$.

The Harmonic distortion versus β are shown in fig.4, where the frame number of ST DFT window $2m=8$ or 64. Solid curves show the harmonics distortion of the compressor, and dotted curves show those of the expander. The harmonics distortion of the compressor is monotonically improved if β goes large even if the frame number $2m$ being maintained in the same number. The chang-

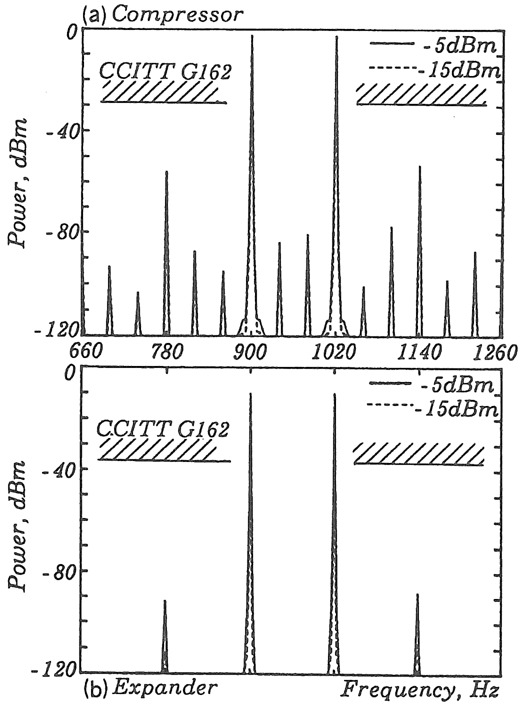

Fig.6 The power specturm of the compresser(a) and of the expander(b), for -5dBm or -15dBm input, 2m=8, N=64, β=6.0.

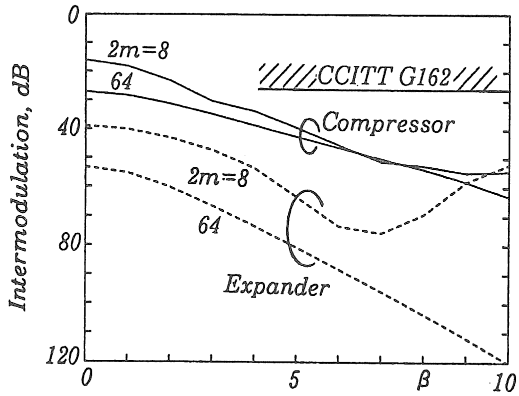

Fig.7 The intermodulation vs. β, 2m=8 or 64.

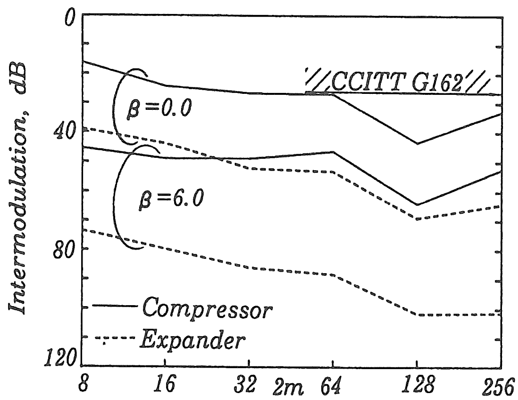

Fig.8 The intermodulation vs. 2m, β=0.0 or 6.0.

ing of β from 0 to 10 improves the harmonics distortion of the compressor by more than 35 dB. More rapid improvements are shown in those of the expander as indicated by dotted curves. Figure 5 shows the harmonics distortions versus the frame number $2m$, where the β of the Kaiser weighting function is set to be 6.0 or 0.0, i.e., β =0.0 means the existing Nyquist truncated by $2m$ frames. The solid curves show the harmonics distortion of the compressor, and the dotted curves show those of the expander. As shown in fig.5, there exist almost no improvement in the harmonics distortion of the compressor while the frame number $2m$ being taken as the parameter.

Intermodulation

The intermodulated signal level, which seems to be adequate for signalling system No.5, is also recommended to be below - 26dB at frequency $2f_1 - f_2 = f_L$ and $2f_2 - f_1 = f_U$ for a compressor and an expander which operate separately. Here, input signals f_1 and f_2 are 900Hz and 1020Hz at a level of - 5dBm or - 15dBm. It is shown in fig.6(a) for the input signals specified in the aboves that the intermodulation of the multi - channel compressor is at a level of - 53.1dB below the - 26dB of CCITT criteria on the lower frequency f_L or at a level of - 50.2dB on the upper frequency f_U for the both - 5dBm and - 15dBm inputs, respectively. The intermodulations of the multi - channel expander are observed at levels of - 81.5dB below the - 26 dB criteria on the lower frequency f_L and at levels of - 77.8dB on the upper frequency f_U for the both - 5dBm and - 15dBm input signals as shown in fig.6(b).

The intermodulations versus the β of the Kaiser weighting function is shown in fig.7, where the frame number of the ST DFT window $2m = 8$ or 64. Solid curves show the intermodulation test results of the multi - channel compressor, and dotted curves show those of the expander. The intermodulations are monotonically improved as β going large. The changing of the β from 0 to 10 also improves the intermodulation of the compressor by more than 35dB. More rapid improvement is shown in the intermodulation of the expander if the frame number $2m$ is set by large number as 64. In the case of $2m = 8$, the intermodulation of the expander is remarkably as β being less than 6.3, however it loses this improvement as being more than 7 as shown by dotted curve in fig.7.

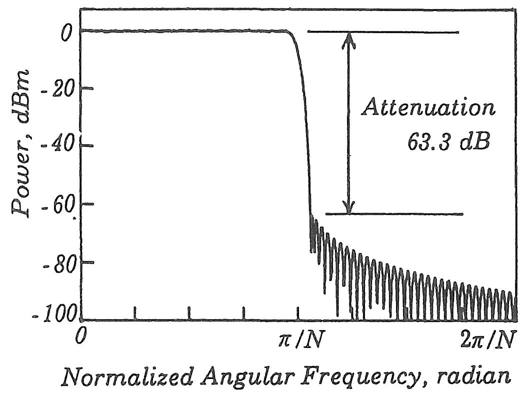

Fig.9 Power frequency response of the Kaiser weighted Nyquist, 2m=8, N=64, β=6.0.

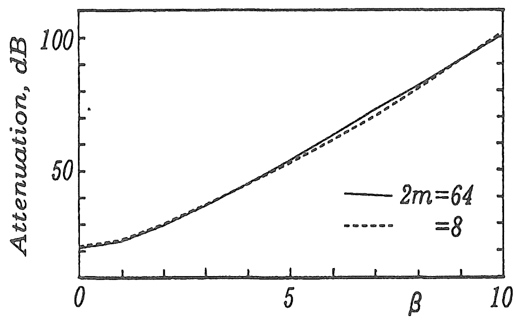

Fig.10 The attenuation vs. β, 2m=8 or 64.

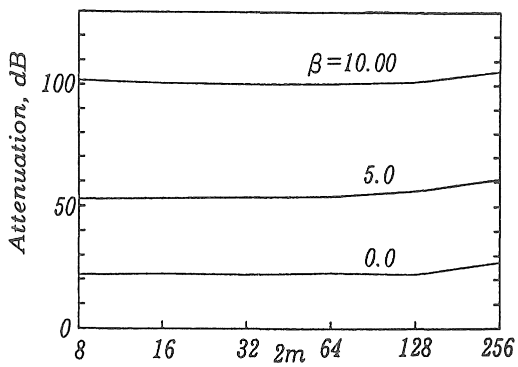

Fig.11 The attenuation vs. 2m, β=0.0, 5.0, or 10.0.

Figure 8 shows the intermodulation versus the frame number $2m$, when β is 0.0 or 6.0. Both intermodulations of the compressor to be slightly improved as $2m$ going large.

Attenuation of the Kaiser Weighted Nyquist

Figure 9 shows the power frequency response of the Kaiser weighted Nyquist function, here $2m = 8$, $N = 64$ and $\beta = 6.0$. The attenuation is measured as shown in fig.9 by the suppress amount between the passband and elimination band. Figure 10 shows that the attenuation of the Kaiser weighted Nyquist is effectively improved by setting the parameter β be large and by about 80dB during β changing from 0 to 10. While the attenuation does not effect by the frame number $2m$ of the Kaiser weighted Nyquist as shown in fig.11.

4. CONCLUSION

A noble compander was discussed with emphasis on the instantaneous spectrum signal processing through transient response, harmonic distortion, and intermodulation. The Kaiser weighted Nyquist function being adopted to the significant decimation filter $h(*)$ of the short time DFT, the multi-channel compander is able to be almost free from any distortions both in steady state of frequency response and transient response.

Farther studies on optimizing the values of the frame number $2m$, the frame length N, and the parameter β will improve the accuracy of the multi-channel companders within the given computing power.

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