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Characteristics of the Feed - Forward Syllabic Compander and its Optimized Configuration

Feed - Forward Syllabic Companderの特性とその最適化

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ABSTRACT *The syllabic compander is inevitable in improving speech quality through suppressing such noise as fading over poor radio channel or thermal noise in the transmission facilities. The feed - back loop is previously seemed to play fundamental role in existing companders of improving speech quality. However according to employing these feed - back loops, the companded signals are become to be unstable and to be degraded when such impulsive fading noise being added.*

The feed - forward syllabic compander (ab. in FF compander) is newly proposed here to successfully discuss elimination of the feed - back loop without any loss of generalities from existing syllabic compander if the root - operator and FIR filter are merely employed in the envelope detector.

The FF compander is improved to quickly converge in transient response by shortening the FIR filter length of its envelope detector. The shorter FIR filter is adopted to FF compander, the severer harmonic distortions are introduced. Optimized structure of the FF compander is shown to solve this contradiction in the meanings of deciding the filter length, and to be examined through computer simulations.

1. INTRODUCTION

The syllabic compander improves speech quality over channels by suppressing dominant noise of fading noise through multiplying or dividing (ab.in companding) signals by the envelope of input signals. Figure 1 shows the level diagram through apriori communication systems.

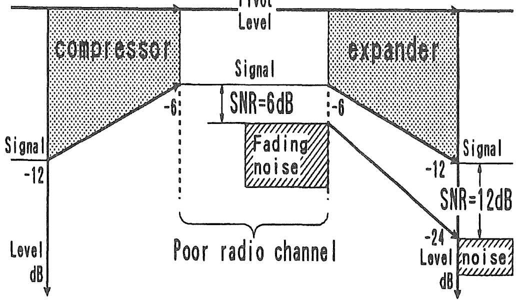

Fig. 1 Level diagram over poor radio channels.

Existing syllabic compander features of employing feed - back loop in its compressor to perform compressing function as shown in fig.2. Hereafter, this existing compander is abbreviated in FB compander. In fig.2, *D* or *M* means divider or

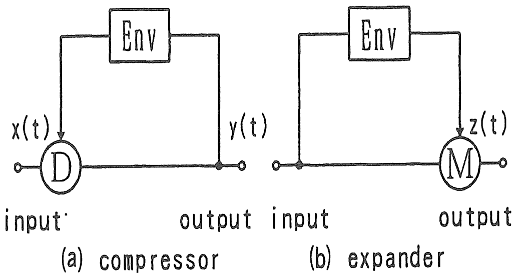

Fig. 2 Configurations of the FB compander.

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multiplier, respectively. Let's consider how the FB compander improves speech quality under following assumption. $x(t)$, $y(t)$ or $z(t)$ devotes such signal as input to the compressor, signals from the compressor, or from the expander, respectively. And the expander is directly cascaded in base band to the compressor.

As shown in fig.2(a), output $y(t)$ is given by dividing input $x(t)$ by envelope signal $E\{y(t)\}$. That is,

$$y(t) = x(t)/E\{y(t)\} \quad (1)$$

The output $z(t)$ of the FB expander is given by the product of input $y(t)$ and envelope signal $E\{y(t)\}$ as follows,

$$z(t) = y(t) \cdot E\{y(t)\} \quad (2)$$

It is easy to understand that relation $E\{E\{A(t)\}\} = E\{A(t)\}$ holds on successive operations, when envelope detector is remembered to be consist of two major parts, rectifier and low pass filter.

Being taken envelope of both sides, eqs.1 and 2 show that syllabic compander exactly compands signals by 1 to 2 in decibel meanings. Envelope of eq.1 gives eq.3,

$$E\{y(t)\} = E\{x(t)\}/E\{y(t)\} \ or$$

$$E\{y(t)\} = E\{x(t)\}^{0.5} \quad (3)$$

Eq.2 is also modified by taking envelopes

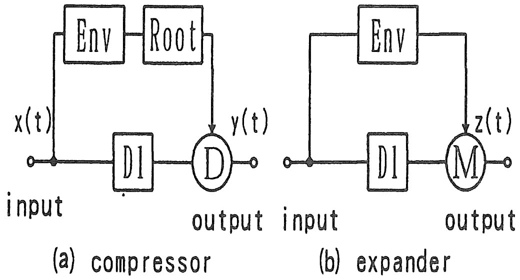


Fig.3 Configurations of the FF compander.

as follows,

$$E\{z(t)\} = E\{y(t)\}^2 \quad (4)$$

FB compander is quantitatively interpreted how to improve speech quality over poor radio channels to double SNR as shown in eqs.3 and 4. However, FB compander is still left in great problem of being unstable for such impulsive noise as fading.

2. PRINCIPLE OF THE FEED-FORWARD SYLLABIC COMPANDER

2.1 Configuration and Principle

The FF compander, which is realized by newly introducing root circuit, perfectly excludes the feed-back loop from circuitry topology as shown in fig.3. In the same figure, ***Root*** or ***Dl*** is newly employed circuit as root or delay, respectively.

Now, we show that this newly proposing FF compander exactly operates in companding facilies. All signals added to the FF compressor of fig.3(a) are simultaneously added to two passes, one is what consists of envelope detector and rooter, the other is delay. The output $y(t)$ is given by dividing $x(t)$ by $\sqrt{E\{x(t)\}}$ with apriori delay,

$$y(t) = x(t-\tau)/E\{x(t-\delta-\varepsilon)\}^{0.5} \quad (5)$$

Here, $\tau = \delta + \varepsilon$, δ or ε is delaying value in envelope detector or square root processing. ε is also negligible small.

A similar structure of the FF expander allows that the output $z(t)$ is given by eq.6 after adjusting delayed amount in envelope detector.

$$z(t) = y(t-\delta)E\{y(t-\delta)\} \quad (6)$$

Now, let's examine the function of the FF compander. Taking envelopes on the both sides of eq.5, it gives

$$E\{y(t)\} = E\{x(t-\tau)\}/E\{x(t-\delta-\varepsilon)\}^{0.5}$$

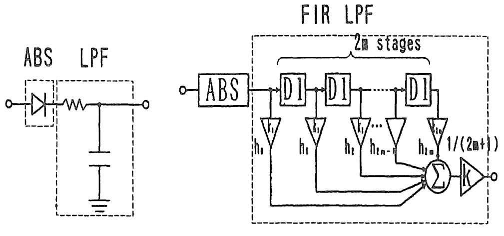
$$\fallingdotseq E\{x(t-\delta)\}^{0.5} \qquad (7)$$

Now, we can get a solution of eliminating feed - back loops from compander by merely employing rootor as discussed aboves[2].

2.2 Envelope Detector

The FIR filter is also important in perfectly excluding feed - back structure from envelope detector, where feed - back structure is eliminated in circuitry topology. For example, transversal filter is employed as the low - pass filter of the envelope detector as shown in fig.4(b). Here, ***LPF***, Σ , or $\boldsymbol{k}$ means the low - pass filter, adder, or multiplying scale, respectively.

According to rapid convergence and linearity in phase characteristics of FIR filters, the envelope detector of the FF compander is also expected in such operating characteristics. Delay is given in the FIR filter by half of filter length multiplied by sampling period τ , here τ is reciprocal number of sampling frequency f_s . While the delay is almost vanished in the rootor according to rootor being consists of a few stage combinatory logics. As discessed in aboves, delay δ in eq.6 or 7 is set to be $m\tau$, $2m$ is FIR filter length.



(a) Conventional envelope detector (b) DSP FIR envelope detector

Fig.4 Configurations of envelope detectors.

3.CHARACTERISTICS

3.1 Transient Responses

Let's consider the transient responses of the FF compander. The output $y(n)$ of the FF compressor is also given by dividing input $x(n)$ by its envelope $E\{x(n)\}$ as follows,

$$y(n) = \left\{\frac{1}{2m+1}\sum_{p=0}^{2m} h(p)a(n-p)\right\}^{-1/2} x(n) \qquad (8)$$

Here, parenthesized term is output of the envelope detector, $a(n)$ is absolute of input signal $x(n)$, and $h(p)$ is coefficients of the FIR filter defined by $h(p)=1,\ 0 \le p \le 2m$.

Under the assumption of input signal being the unit step $u(n)$, the output $y(n)$ of the FF compressor is given in eq.9 by reciprocal number of square root of the hyperbola function.

$$y(n) = \begin{cases} \left[\dfrac{1}{2m+1}\{R(n)+m+1\}\right]^{-1/2}, & if \quad 0 \leqq n < m \\ 1, & if \quad n \geqq m \end{cases} \qquad (9)$$

Here, $R(n)$ is ramp function with unity inclination.

Being given the output $z(n)$ of the FF expander by product of the input signal and its envelope, it gives $z(n)$ as follows,

$$z(n) = \begin{cases} \dfrac{1}{2m+1}\{R(n)+m+1\}, & if \quad 0 \leqq n < m \\ 1, & if \quad n \geqq m \end{cases} \qquad (10)$$

It is clearly shown in eqs.9 and 10 that the transient response of the FF compander converges to unity within $m+1$ steps, i.e. $(m+1)\tau$ clocks.

3.2 Analysis of the Frequency Response

Since the output $y(n)$ of the FF compres sor is given by product of input $x(n)$ and reciprocal number of square root of the envelope $E\{x(n)\}$, the output frequency response $Y(e^{j\omega})$ is defined by the convolution of $X(e^{j\omega})$ and $E^{-\frac{1}{2}}\{X(e^{j\omega})\}$ as follows.

$$Y(e^{j\omega}) = \frac{1}{2\pi}\int_{-\pi}^{\pi} X(e^{j\theta})E^{-\frac{1}{2}}\{X(e^{j(\omega-\theta)})\}d\theta \quad (11)$$

The output $z(n)$ of the FF expander is given by product of input $y(n)$ and its envelope $E\{y(n)\}$ in similar to the FF compressor, the frequency response $Z(e^{j\omega})$ is also defined by similar convolution of $Y(e^{j\omega})$ and $E\{Y(e^{j\omega})\}$ as follows.

$$Z(e^{j\omega}) = \frac{1}{2\pi}\int_{-\pi}^{\pi} Y(e^{j\theta})E\{Y(e^{j(\omega-\theta)})\}d\theta \quad (12)$$

Being remembered that envelope detector consists of low-pass filtering absolute signal of the input, it gives the frequency responses of the envelope signals as follows,

$$E\{X(e^{j\omega})\} = H(e^{j\omega})A(e^{j\omega}) \quad (13)$$

Here, $A(e^{j\omega}) = Abs\{X(e^{j\omega})\}$,
$H(e^{j\omega})$ means low-pass filtering defined by

$$H(e^{j\omega}) = \frac{1}{2m+1}\,\frac{\sin\{(2m+1)\omega/2\}}{\sin(\omega/2)}e^{-j\omega m} \quad (14)$$

The cut-off frequency f_c of $H(e^{j\omega})$ is given by $f_c \fallingdotseq 0.443 f_s/(2m+1)$ [1]. The harmonic distortion is consequently shown in eqs. from 11 to 14. That is, if input signal is $K\sin(\omega n)$, the output signal $y(n)$ or $z(n)$ of the FF compressor or expander is given by Fourier series as follows, respectively.

$$y(n) = \sqrt{\frac{\pi K}{2}}\sum_{i=1}^{\infty} k_i \sin(i\omega n), \quad (15)$$

i is odd integer.

Here, ith coefficient k_i is given as follows.

$$k_1 \fallingdotseq 1 - \frac{350H(2\omega)+35H(2\omega)H(4\omega)}{2100} + \frac{3H(4\omega)H(6\omega)}{2100} + \frac{H^2(2\omega)}{12} - \cdots$$

$$k_3 \fallingdotseq \frac{350H(2\omega)+35H(2\omega)H(4\omega)}{2100} + \frac{3H(2\omega)H(6\omega)}{2100} - \frac{H^2(2\omega)}{24} + \cdots$$

$$k_5 \fallingdotseq \frac{H^2(2\omega)}{24} + \frac{14H(4\omega)+3H(2\omega)H(6\omega)}{420} + \frac{-H(6\omega)+7H(2\omega)H(4\omega)}{420} - \cdots$$

In eq.15, the ith order harmonics of the FF compressor signal when $K\sin(\omega n)$ being added is denoted by ith order coefficients as results of frequency analysis. After similar longsome calculations, the output $z(n)$ of the FF expander is given by the similar Fourier series as theoretical results of analysis as follows.

$$z(n) = \frac{2K^2}{\pi}\sum_{i=1}^{\infty} \widetilde{k_i}\sin(i\omega n), \quad (16)$$

i is odd integer.

Here, ith coefficient $\widetilde{k_i}$ is given as follows.

$$\widetilde{k_1} \fallingdotseq 1 + \frac{H(2\omega)}{3\cdot 1}, \qquad \widetilde{k_3} \fallingdotseq -\frac{H(2\omega)}{3\cdot 1} + \frac{H(4\omega)}{5\cdot 3}$$

$$\widetilde{k_5} \fallingdotseq -\frac{H(4\omega)}{5\cdot 3} + \frac{H(6\omega)}{7\cdot 5}$$

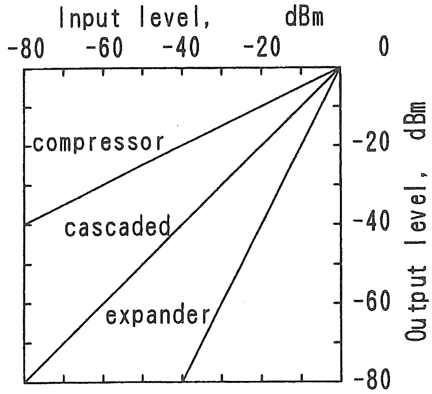


Fig.5 Operating characteristics of the FF compander, input signal is tonal 800Hz.

4. SIMULATIONS FOR THE FF COMPANDERS

Optimum configuration is examined in the meaning of definition for the FIR filter length $2m$ through computer simulations on CRAY X-MP/14se at AIT to avoid round-off errors from algorithmic processing errors.

As shown in fig.5, the FF compander is so precise as observing no displacements from input vs output amplitude operating characteristics if the 800Hz 80dB dynamic range for compressor or 40dB dynamic range for expander input signals being added.

4.1 Consideration from the Transient Responses

Figure 6 shows transient response of the FF compander when the FIR filter length $2m$ is set to be 128. Here, input signal is 2kHz tone-burst with 12 dB steps. According to the CCITT G.162, the attack time t_a is defined by the settling time after sudden rise, and the recovery time t_r is defined by the settling time after sudden descend. These t_a and t_r are recommended to be within 5.0ms and 22.5ms. It is clearly shown in fig.6 that the envelopes both of output $y(n)$ and $z(n)$ monotonically converge into steady-state value after $(m+1)$

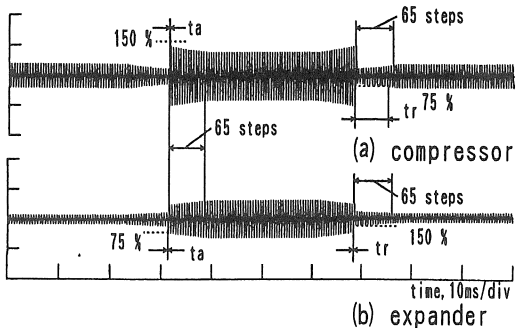


Fig.6 Transient response of the FF compander, 2m=128.

$=65$ steps, i.e. 8.25msec in transient responses as same to the theoretical results as discussed in previous session.

As shown clearly in fig.6, neither overshoot after 12dB rise nor undershoot after 12dB descend dose not exceed more than 150% nor dose not fall less than 75% of steady-state value in transient responses of the FF compressor. Also in transient responses of the FF expander, neither overshoot after 6dB rise nor undershoot after 6 dB descend dose not go out these criteria. These value of t_a and t_r are observed to be almost null of the ideal. The attack time t_a and recovery time t_r show monotonical increase as the FIR filter length $2m$ goes larger as the shown in fig. 7. So long as the filter length $2m$ is less than 752, both of t_a and t_r of the FF com

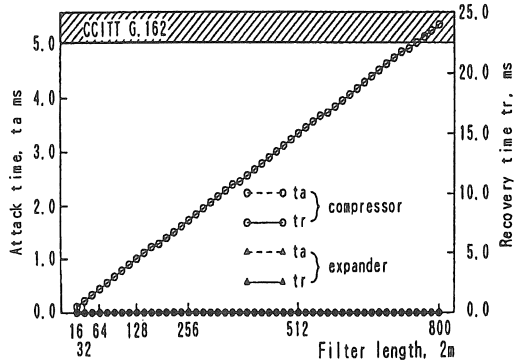


Fig. 7 Characteristics of attack time ta and recovery time tr vs filter length 2m of the envelope detector.

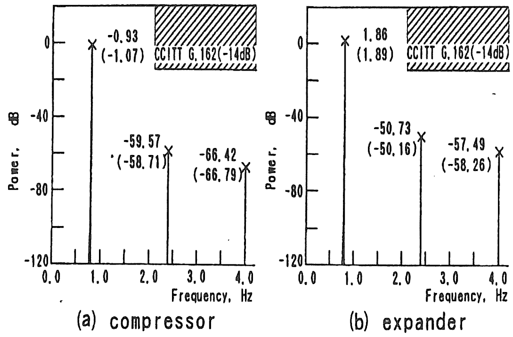


Fig.8 Power spectrum observed at output of the FF compander(2m=128). × or parenthesized value means theoretical values.

pressor is sufficiently quick more than the CCITT G.162 recommendations. In the characteristics of the FF expander, both t_a and t_r have vanished to zero as discussed aboves.

4.2 Consideration from the Harmonic Distortion

Frequency response through the FF compressor or expander, which operate separately with 800Hz 0dBm tonal signals, is shown in fig 8(a) or (b), respectively. Here, the FIR filter length is set to be 128. Harmonic distortion over all frequency is recommend to be below 4%, i.e. -14dB in CCITT G.162.

It is shown in fig.8(a) or (b) by both solid curves and values that the harmonic distortion of the FF compressor or expander is less than -59 dB or -50 dB with margin to CCITT G.162 criteria by more than 45 dB or 36 dB, respectively.

The theoretical results for the FF compressor given by eqs.15 and 16 are also shown to be previously to close to observation values in fig.8 by both mark × and parenthesized values.

The maximum values of the harmonics are plotted in fig.9 as taking the FIR filter length *2m* as a parameter. On the range more than *2m=4*, the maximum harmonic distortion is sufficiently suppressed less than those of CCITT.

4.3 Consideration from the Intermodulation Tests

The level of intermodulation when two frequency f_1 or f_2 -5 dBm tonal signals being added is recommended less than -26dB at f_L and f_U in separately operating compressor or expander. Here, $f_1=900Hz$, $f_2=1020 Hz$, $f_L=2f_1-f_2$, $f_U=2f_2-f_1$.

As shown in fig.10, the maximum two levels of intermodulation signals through the FF compressor or expander in which the FIR filter length *2m* being 128 are observed at f_L and f_U by -51.85dB and -51.56 dB, or, -47.01dB and -46.72dB with margin to CCITT G.162 criteria of -26dB by

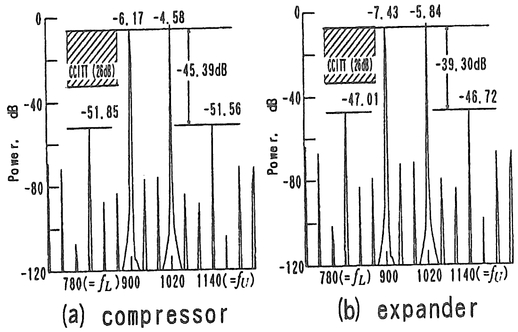


Fig.10 Power spectrum of the FF compander for -5dBm input, 2m=128.

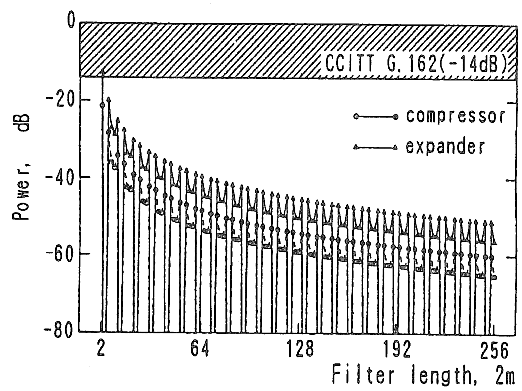


Fig.9 Characteristics of the thrid harmonic distortion vs filter length 2m of the FF compander.

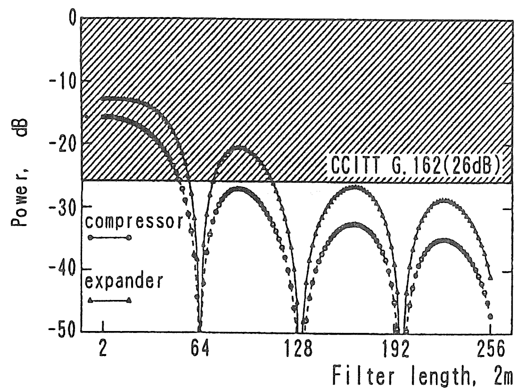


Fig.11 Characteristics of the maximum intermodulation error vs filter 2m of the FF compander.

more than 25dB or 20dB, respectively.

Fig.11 shows the maximum level of intermodulation signals under CCITT specifications as taking filter length *2m* of the envelope detector as a parameter. As shown in fig.11, the length of FIR filter *2m* should be chosen more than 58 in the FF compressor, or *2m* of the FF expander should be chosen more than 114 to clear the CCITT G.162 criteria.

5.CONCLUSION

Novel compander has successfully discussed with emphasis on eliminating feed-back loop and on realizing only feed-forward structure to avoid speech degradation as impulsive fading noises being added.

Optimized structure of the feed-forward compander was also shown by setting the FIR filter length *2m* be 128 through both analysis and experimental examinations. Further studies will be performed on optimizing the FIR filter coefficients.

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