1 # WIDE BAND LF PSEUDORANDOM GAUSSIAN NOISE GENERATION

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4 ## ABSTRACT

A new approach is described for generation of pseudorandom of wide band which could be considered as white Gaussian noise. This is achived by using nonrecursive digital filter with specified transfer function which compensates the spectrum decay of pseudorondom binary sequence (m-sequnce). The optimized filter parameters (i.e. filter length N_f, sampling points M using frequency sample technique) for fluctution ($\mathcal{E}$) $\leqslant$ 0.2 dB in the passband of proposed filter $N_f \geqslant 61$ and $M \geqslant 400$, are considered.
The results of simulation of desinged filter performance gave $N_f = 61$, band width = 0.25 f_c " f_c clock frequency " and the sequence length $L = (2^{N_g} - 1)$, for $N_g \geqslant g$.
The increasing in band width is indicated and is very satisfactory as compared with the existing data (i.e. band width = 0.05 f_c).

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I. INTRODUCTION

The basic random source used in generating pseudorandom noise is normaly a maximum length binary sequence (m-sequence). This sequence can be generated by shift register with proper modulo-2 feedback circuitry. Shift register generates a sequence of length $L = 2^{N_g} - 1$; where N_g is the shifter length. This sequence is also called pseudorandom binary sequence (PRBS), It is periodic sequence with period $T = LT_C$ (T_C is the clock period) [1,2].

The conventional method for generating Gaussian noise is filtering PRBS by low-pass filter. The obtained sequence is approximately Gaussian in amplitude. Filter band width (B.W) is usually 0.05 of the clock frequency f_c within which the fluctuation is maximaly 0.1 dB [3]. This is relatively smaller band. The aim of this paper is to introduce special method to enlarg this band to be more applicable in commuication systems.

II. ENLARGING OF NONRECURSIVE FILTER BAND WIDTH

The power spectral density (p.s.d) of PRBS is shown in Fig.1. This spectrum is a line spectrum whose envelope is proportional to [3] :

$$\mathrm{Sinc}^2\left(\frac{f\,\pi}{f_c}\right)$$

Thus it can be indicated that the spectrum which is centered about zero frequency does not decay monotonically with increasing frequency but has nulls at multiples of clock frequency f_c . An efficient approach, which usually leads to white noise, is to choose the transfer function of proposed filter as the reciprocal of $\mathrm{sinc}^2(\pi f/f_c)$ up to the required cutoff frequency F_c (filter passband) and cutting by chebyshev characteristic in the cutoff band up to $f_c/2$.

The transfer function H(f) of the filter is characterized by two transefer functions as shown in fig. 2.

It can be expressed by [4,5,6] :

$$H(f) = \begin{cases} H_1(f) & 0 < f \leqslant F_c \\ H_2(f) & F_c < f \leqslant f_c/2 \end{cases} \qquad (I)$$

Where

$$H_2(f) = \frac{1}{\sqrt{1 + T^2\,\varepsilon^2\,(f/f_c)}} \qquad (2)$$

where $T^2(X)$ is chebyshev polynomial of order 7

and

$$H(f) = \mathrm{sinc}^{-1}(\pi f/f_c) \qquad (3)$$

For the filter realization the frequency sample technique can be used. Sample points chosen at (M+1) spaced points

over the frequency range $(o - f_c/2)$.

The filter coefficients (γ_ℓ) symmetric filter with length N_f are given by [7];

$$\gamma_\ell = \gamma_{-\ell} = \frac{1}{2M}\sum_{m=-M}^{M} H\left(\frac{mf_c}{2M}\right)\cos\frac{m\ell\pi}{M} \qquad (4)$$

where $1 < \ell < (N_f - 1)/2, N_f$ odd

The general form of filtered sequence (i.e. multilevel sequence) is:

$$S(t) = \sum_{\ell=0}^{N_f} \gamma_\ell \cdot b(t-\ell T_c) \qquad (5)$$

This filter simulation can be carried out using equation (4). A special programme has been desired to fullfill the simulation. The simulation results gave the proposed filter parameters (i.e.N_f, M, F_c) and the amplitude spectrum A(f) of filtered sequence which is illustrated in Fig. 3.

The amplitude spectrum A(f) is computed for different cases as shown in fig.4. The peak A_{min} outside the filter band is due to the repetition property of the transfer function of digital filter. A_{min} as function of the passband is shown in fig.5.

The dependence of the fluctuation (ϵ) within the filter passband is evaluated a function of the filter length (N_f) and its sampling points (M). It is shown in Fig.6 and Fig.7.

III. AMPLITUDE DISTRIBUTION

A measure of the deviation from Gaussian (d) is related to the second and fourth central moments of the sequence, by [9] :

$$d^2 = 2\left[\frac{m_4}{3\,m_2^2} - 1\right] \quad (0 < d^2 < 1) \qquad (6)$$

where m_2 and m_4 are the second and fourth central moments respectively, (for normal Gaussian distribution $d^2 = 0$)

m_2 and m_4 are defined for sequence s_i of length L as :

$$m2 = \left(\frac{1}{L}\sum_{i=0}^{L-1} S_i^2\right) - m_1^2 \qquad (7)$$

$$m4 = \frac{1}{L}\sum_{i=0}^{L-1} (S_i - m_1)^4 \qquad (8)$$

where

$$m_1 = \frac{1}{L}\sum_{i=0}^{L-1} S_i \qquad (9)$$

The dependence of d^2 on the sequence length and the filter bandwidth shown in fig.8 and fig.9.

IV. RESULTS DISCUSSION

The variation of the fluctuation (ε) in the passband of the filter $\varepsilon = f(N_f)$ where (M kept constant) is indicated in Fig .6 and it's variation $\varepsilon = f(M)$ (where N_f kept constant) is illustrated in Fig 7. These two figures indicate that by increasing the number of sample points M (where N_f kept constant), the fluctuation (ε) is decreased. It is also indicated that by increasing N_f (where M kept constant) the fluctuation (ε) similarly decreased. The optimum value for M and N_f for $\varepsilon \leqslant 0.2$ dB can be taken up and is found to be $M \geqslant 400$ and $N_f \geqslant 61$. (simulation results)
Using these optimum values of (M and N_f)fig (4) had been drawn up to represent the output amplitude spectrum in the passband of the proposed filter. From this figure A_{min} as a function of passbond is represented in Fig. 5. It is indicated that A_{min} decreased as the passband increased.
Figure (8) represents $d^2 = f(N_g)$ for different values of N_f . From this figure it is indicated that the minimum sequence length for an accepted deviation (d^2) is $N_g \geqslant 9$. Figure (9) gives $d^2 = f(F_c)$ considering the optimum values of sequence length and the filter parameters. It is Indicated from this fig. that the passband of the proposed filter is developed up to = 0.25 f_c .

V. CONCLUSIONS :

Using the proposed method indicates inlarging the passband of the filter up to 0.25. Testing of the deviation from the Gaussian distribution indicates acceptable deviation which prove the validity of proposed method. Windening the passband gives more application of the proposed filter in communication systems. It gives, also, possibility to decrease the clock frequency compared with the conventional method for the same band.

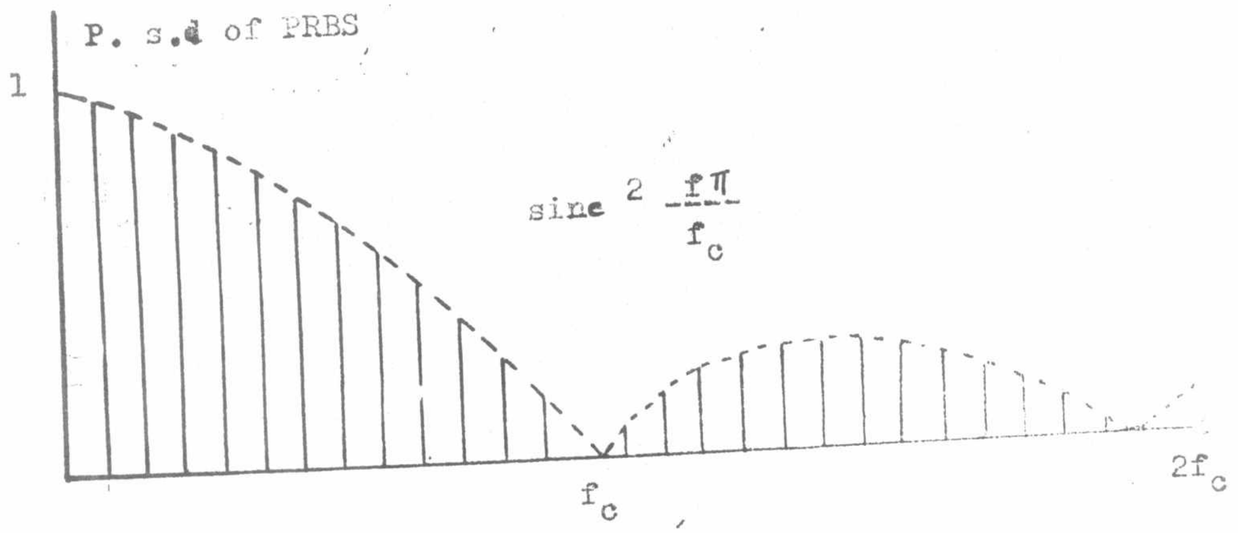

Fig. 1.P. s.d of binary sequence.

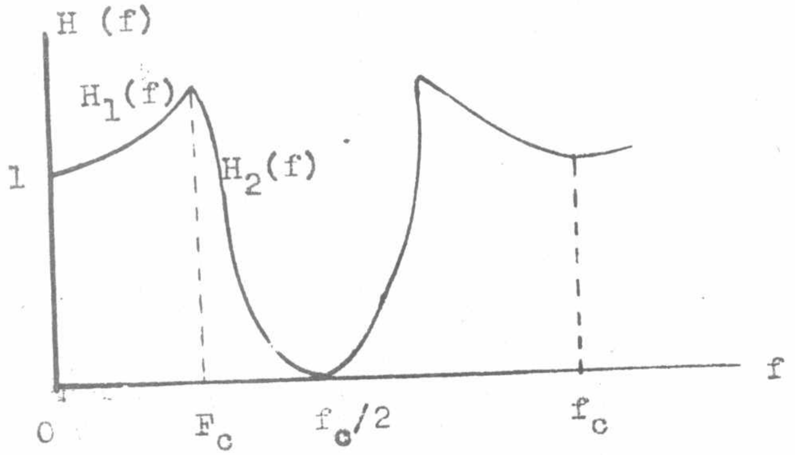

Fig. 2 transfer function of proposed filter.

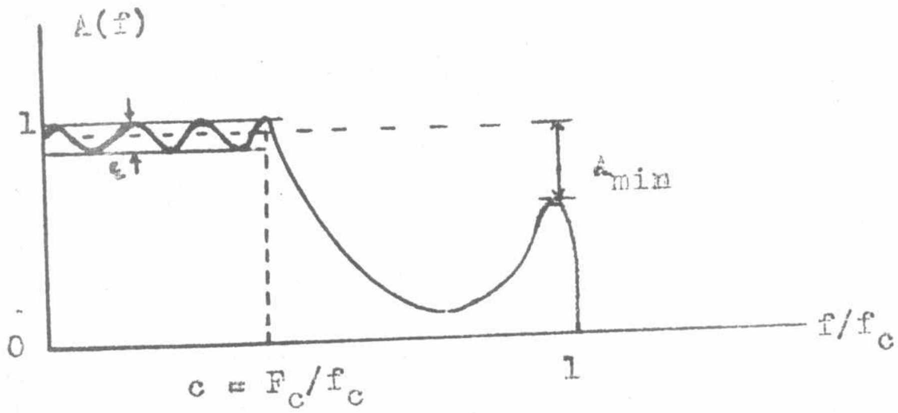

Fig. 3; Amplitude spectrum of miltilevel sequence.

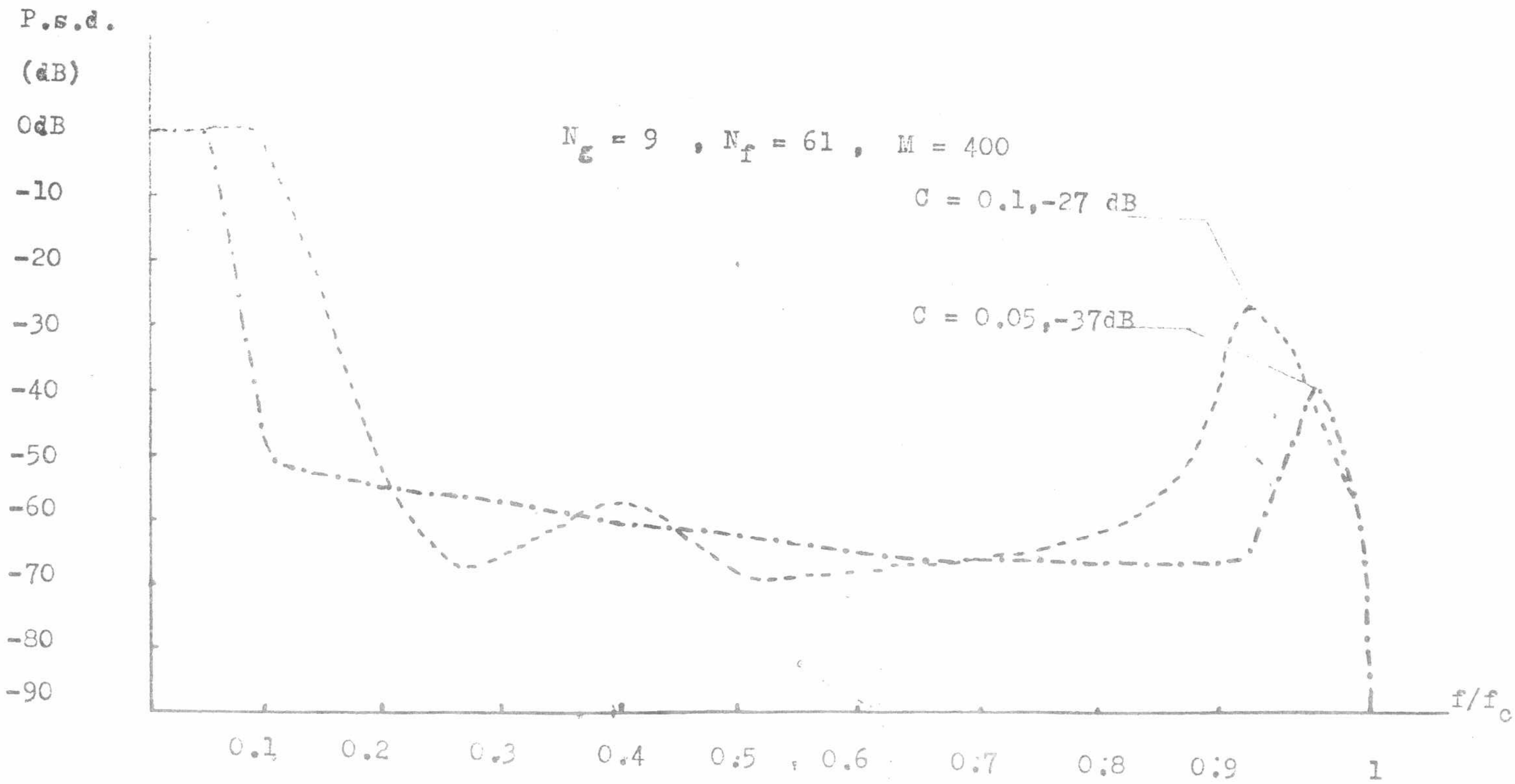

Fig. 4. p.s.d of filtered sequence.

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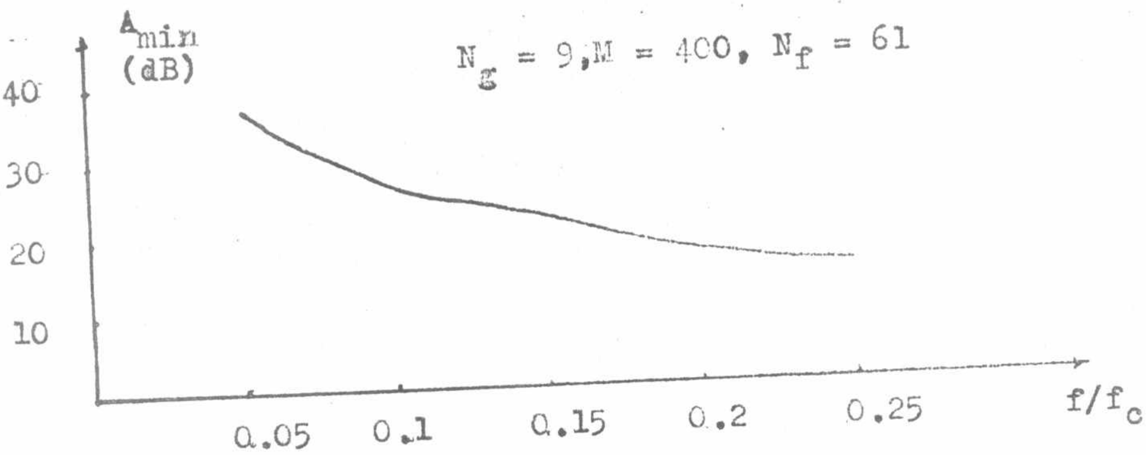


Fig. 5. Dependence of A_{min} on filter bandwith.

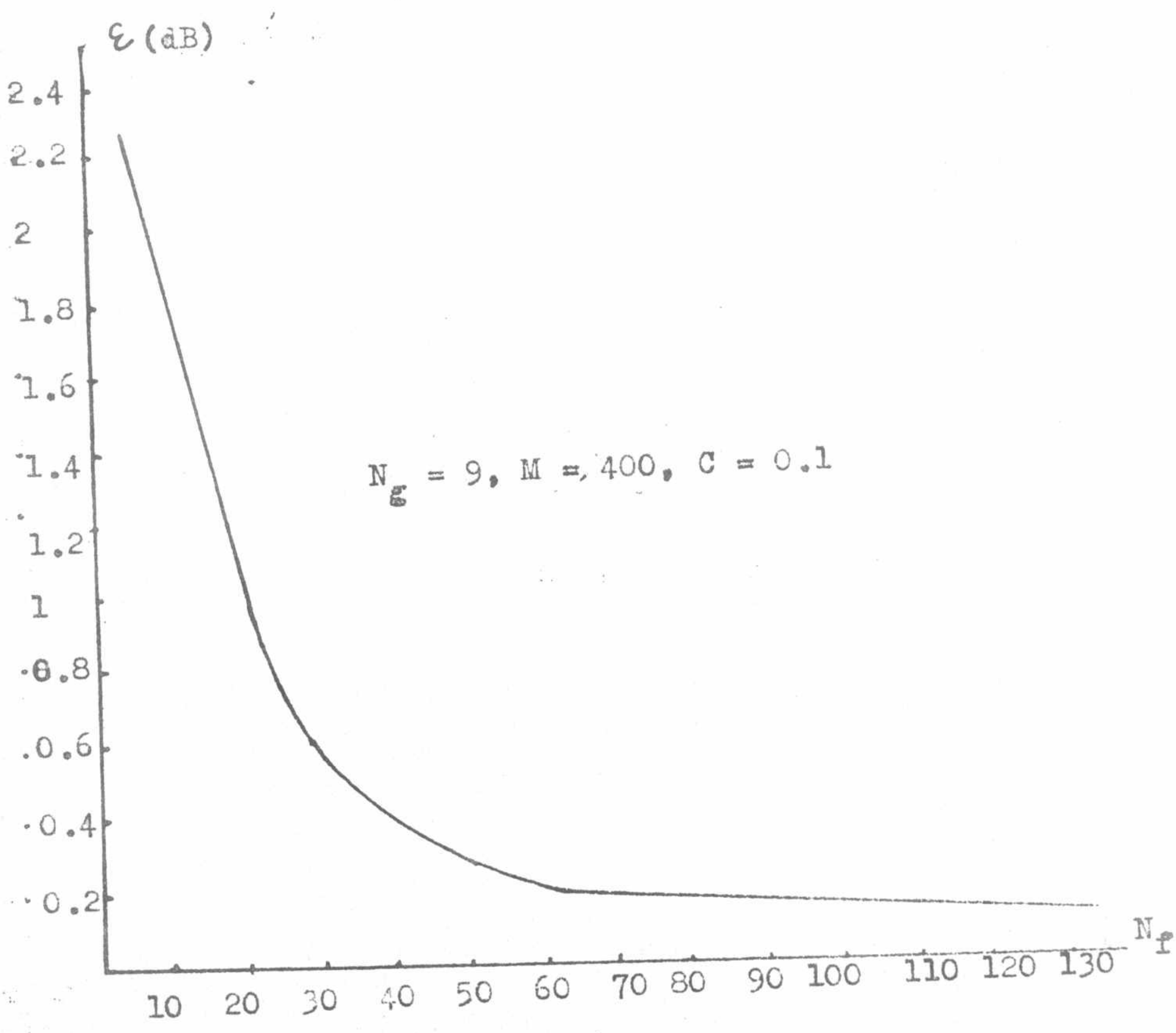


Fig. 6. $\varepsilon = f(N_f)$

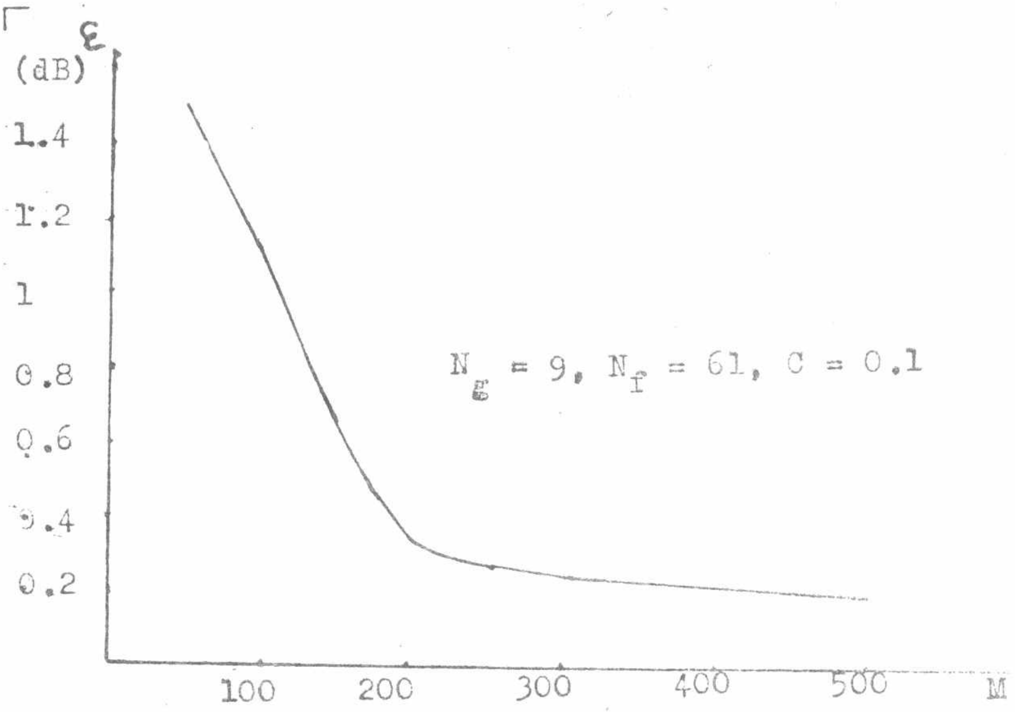


Fig. 7 $\varepsilon = f(M)$

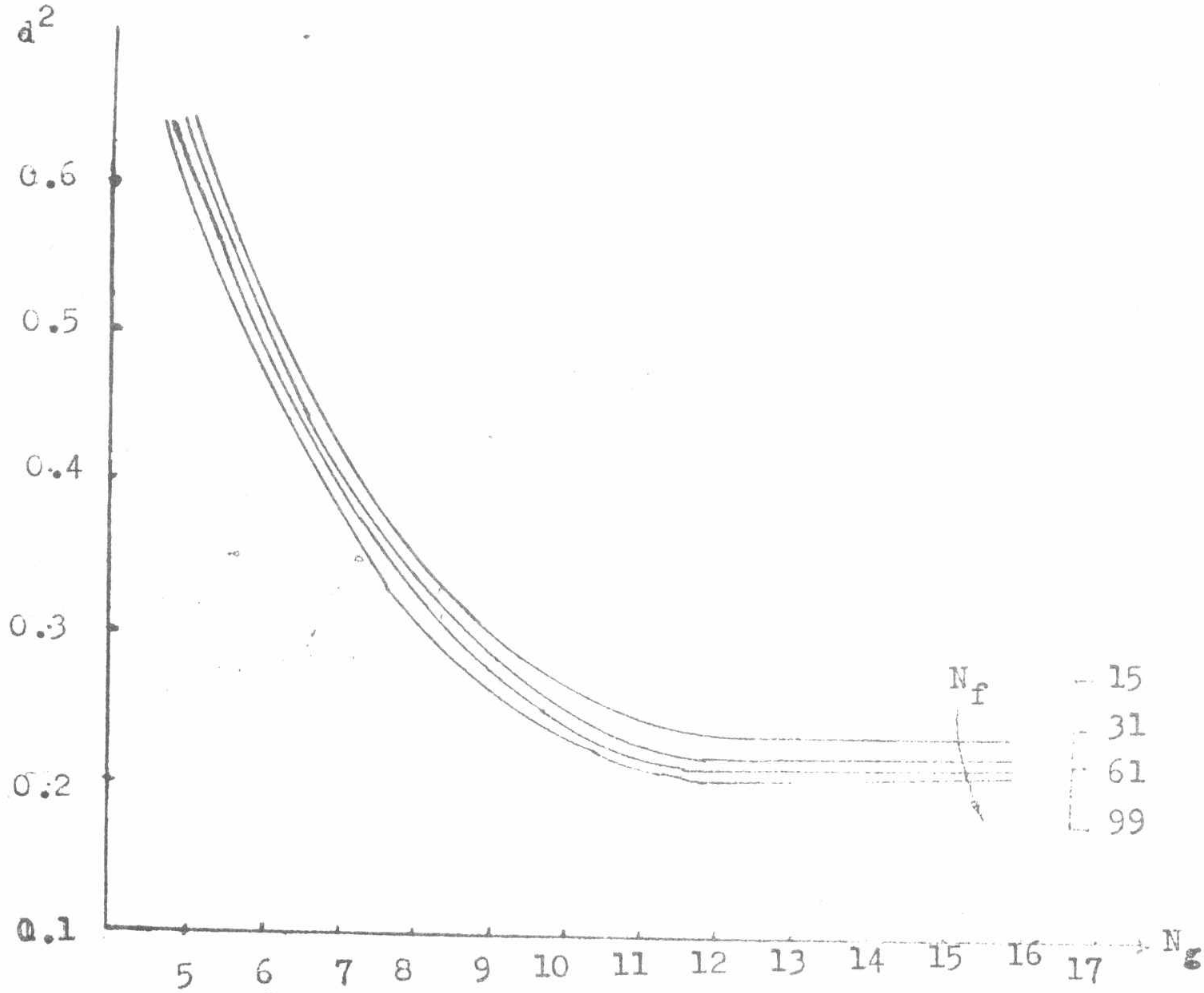
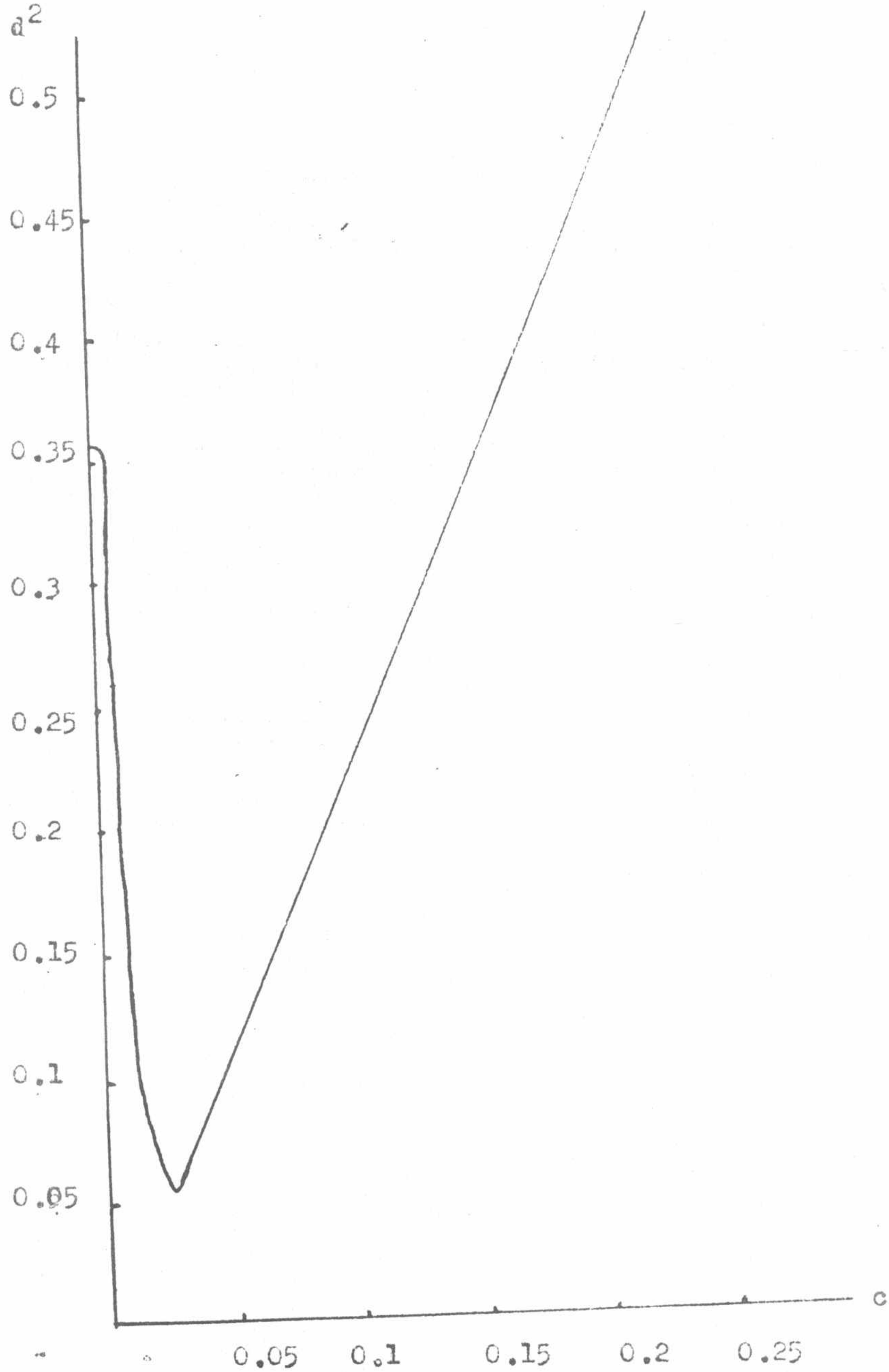


Fig. 8. d^2- measure as function of the generator length.



Fig; 9. d^2 measure as a function of band width.

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