United States Patent [19]

Ostrowski

[11] **4,130,730**

[45] **Dec. 19, 1978**

[54] **VOICE SYNTHESIZER**

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[73] Assignee: **Federal Screw Works**, Detroit, Mich.

[21] Appl. No.: **836,589**

[22] Filed: **Sep. 26, 1977**

[51] **Int. Cl.**[2] **G10L 1/00**

[52] **U.S. Cl.** ..... **179/1 SM**

[58] **Field of Search** ..... 179/1 SA, 1 SG, 1 SM

[56] **References Cited**

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L. Rabiner, "Digital Formant Synthesizer", J. of Ac. Soc. of Am., 43, 822-828, (1968).

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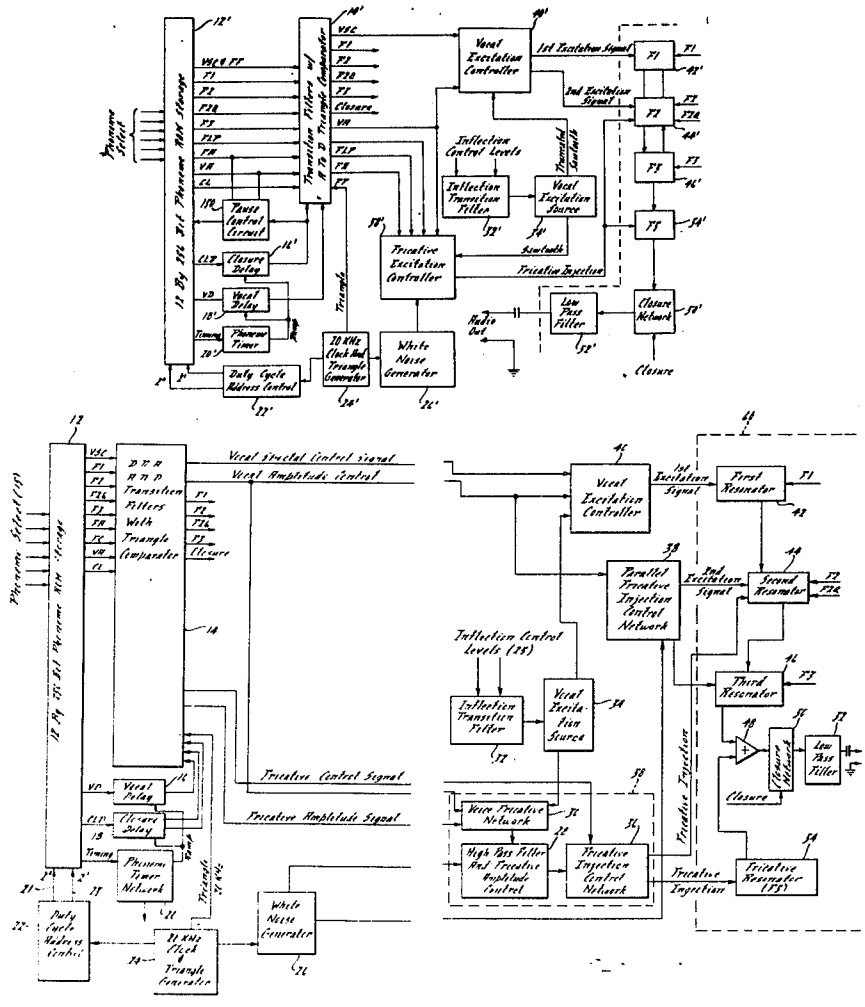
Assistant Examiner—E. S. Kemeny

[57] **ABSTRACT**

A highly simplified speech synthesizer that is capable of producing quality speech. The present speech synthesizer is adapted to be driven by an 8-bit digital input command word. Six of the bits are used for phoneme selection and the remaining two bits for inflection control. In a first embodiment, the system is adapted to generate twelve parameter control signals for each phoneme, with one of the parameters being utilized to control both high and low frequency fricative injection into the vocal tract. This embodiment also provides asynchronous excitation of the vocal tract by including a second fricative excitation control circuit that is adapted to inject white noise in parallel into the second and third resonant filters under the control of the vocal amplitude control signal. In a second embodiment, one of the twelve signal parameters is utilized as two separate control signals thus effectively providing thirteen control signal parameters. The vocal tract in the second embodiment is also driven asynchronously with the glottal waveform being injected in parallel into both the first and second resonant filters. The second embodiment is also adapted to be operated off a portable power supply. A feature of the second embodiment is a phoneme pause control.

18 Claims, 7 Drawing Figures



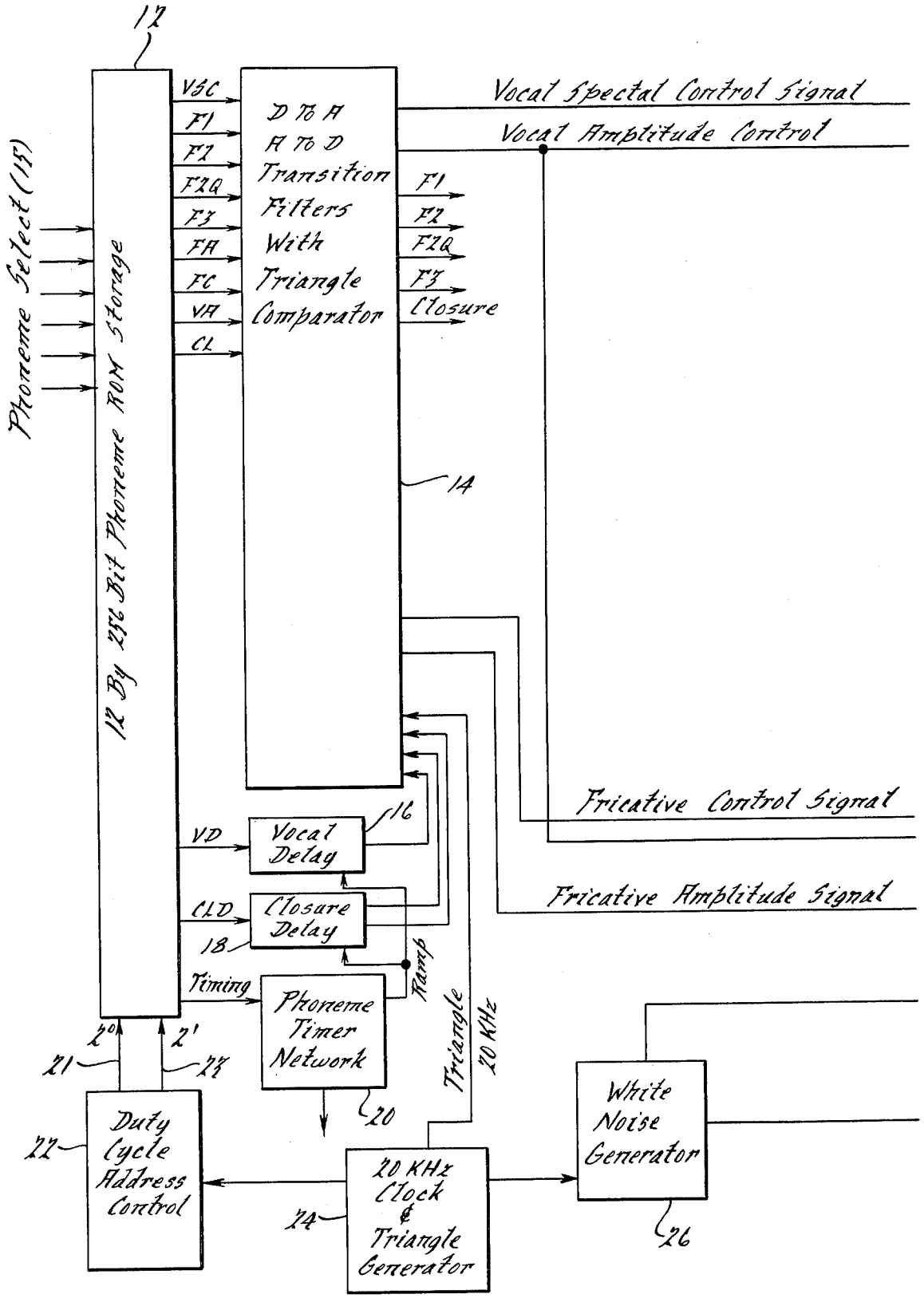

17
Phoneme Select (15)
12 By 256 Bit Phoneme ROM Storage
VSC
F1
F2
F2Q
F3
FA
FC
VA
CL
D To A
A To D
Transition
Filters
With
Triangle
Comparator
F1
F2
F2Q
F3
Closure
14
Vocal Spectal Control Signal
Vocal Amplitude Control
Fricative Control Signal
Fricative Amplitude Signal
VD
Vocal Delay
16
CLD
Closure Delay
18
Timing
Phoneme Timer Network
Ramp
Triangle
20 KHz
20
20'
21
21'
22
Duty Cycle Address Control
20 KHz Clock & Triangle Generator
24
White Noise Generator
26

Fig. 1a.

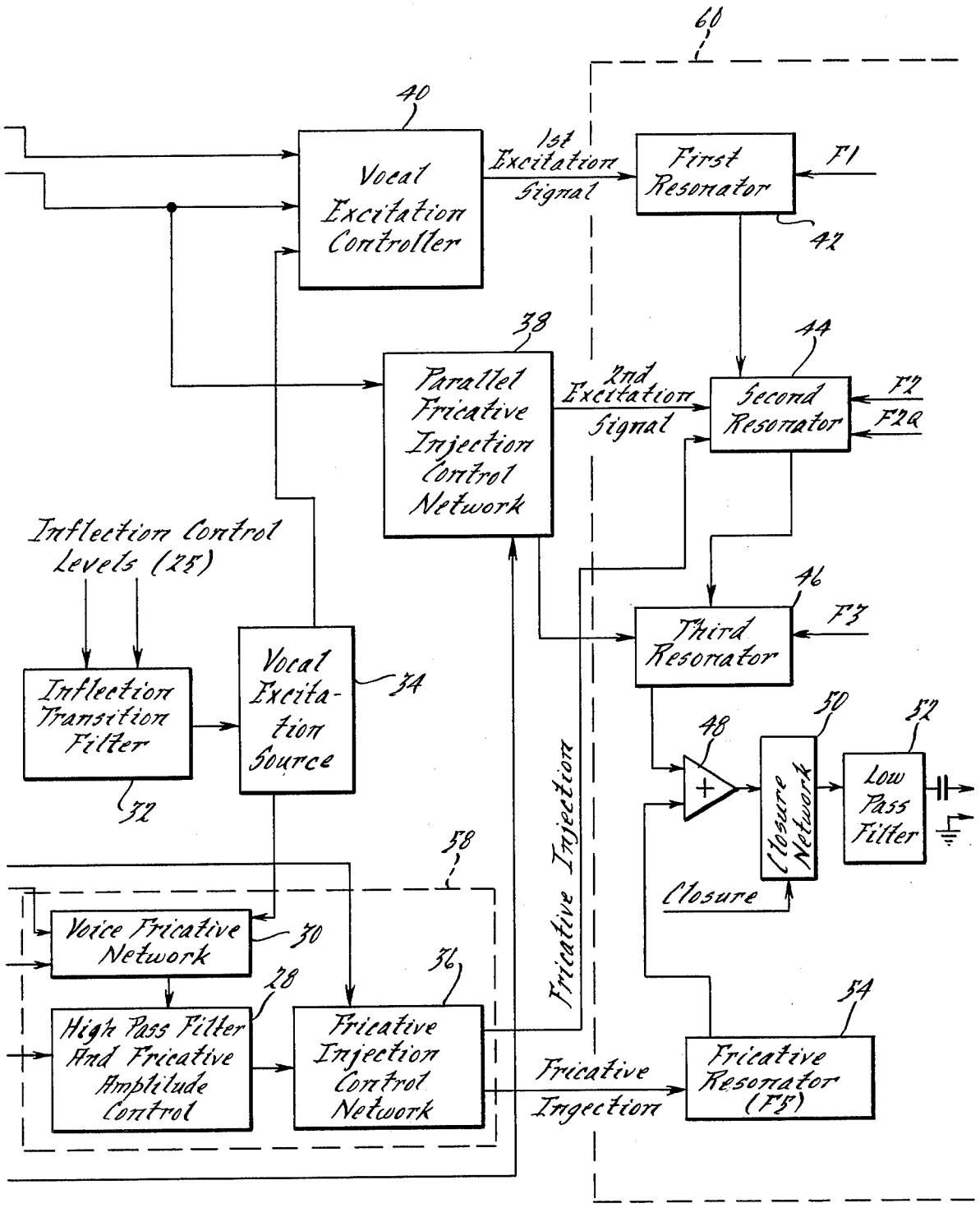

60
40
Vocal Excitation Controller
1st Excitation Signal
First Resonator
F1
42
38
Parallel Fricative Injection Control Network
2nd Excitation Signal
44
Second Resonator
F2
F2a
Inflection Control Levels (25)
46
Third Resonator
F3
Inflection Transition Filter
32
Vocal Excitation Source
34
48
50
Closure Network
52
Low Pass Filter
Closure
58
Voice Fricative Network
30
28
High Pass Filter And Fricative Amplitude Control
36
Fricative Injection Control Network
Fricative Injection
Fricative Injection
54
Fricative Resonator (F5)

FIG. 1b.

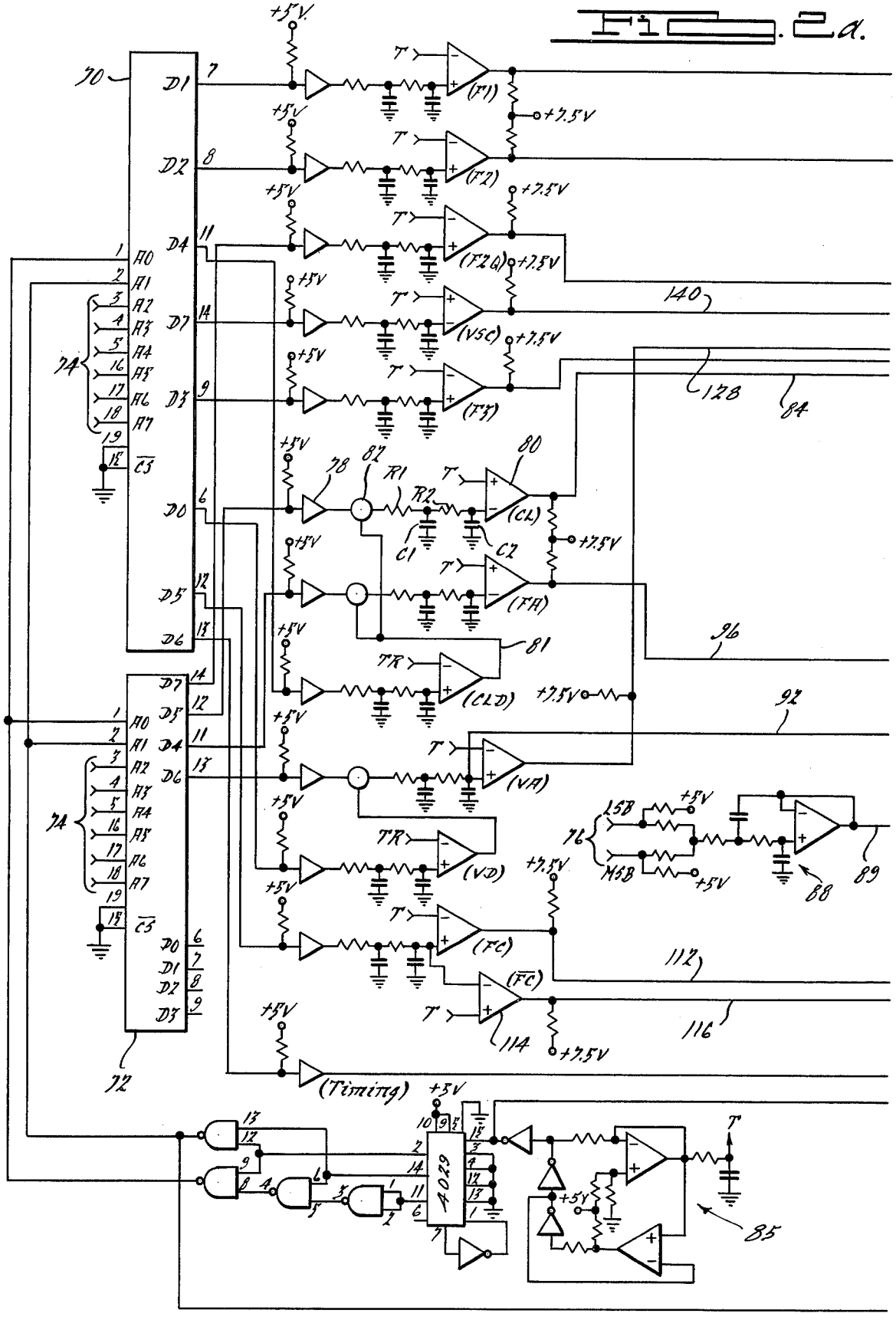


FIG. 2A.
+5V
+7.5V
(F1)
(F2)
(F2Q)
(V5C)
(F3)
(CL)
(FH)
(CLD)
(VA)
(VD)
(FC)
(Timing)
140
128
84
80
78
82
81
96
92
112
116
114
74
72
76
88
89
85
4029

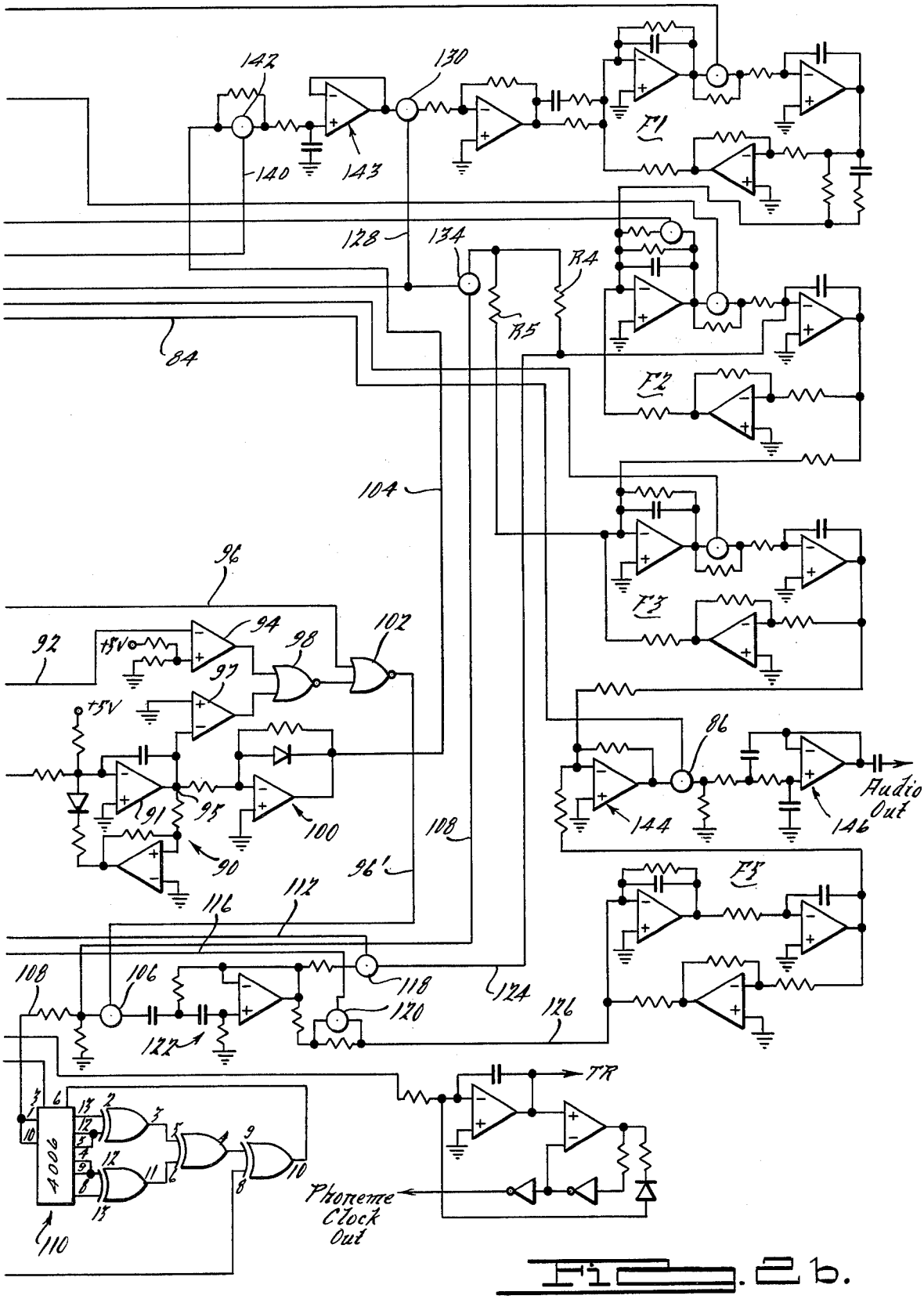

142
130
140
143
F1
128
134
R4
R5
84
F2
104
96
F3
92
+5V
94
98
102
97
91
95
100
90
96'
108
86
144
146
Audio Out
F5
112
116
106
118
120
122
124
126
TR
4006
110
Phoneme Clock Out

Fig. 2b.

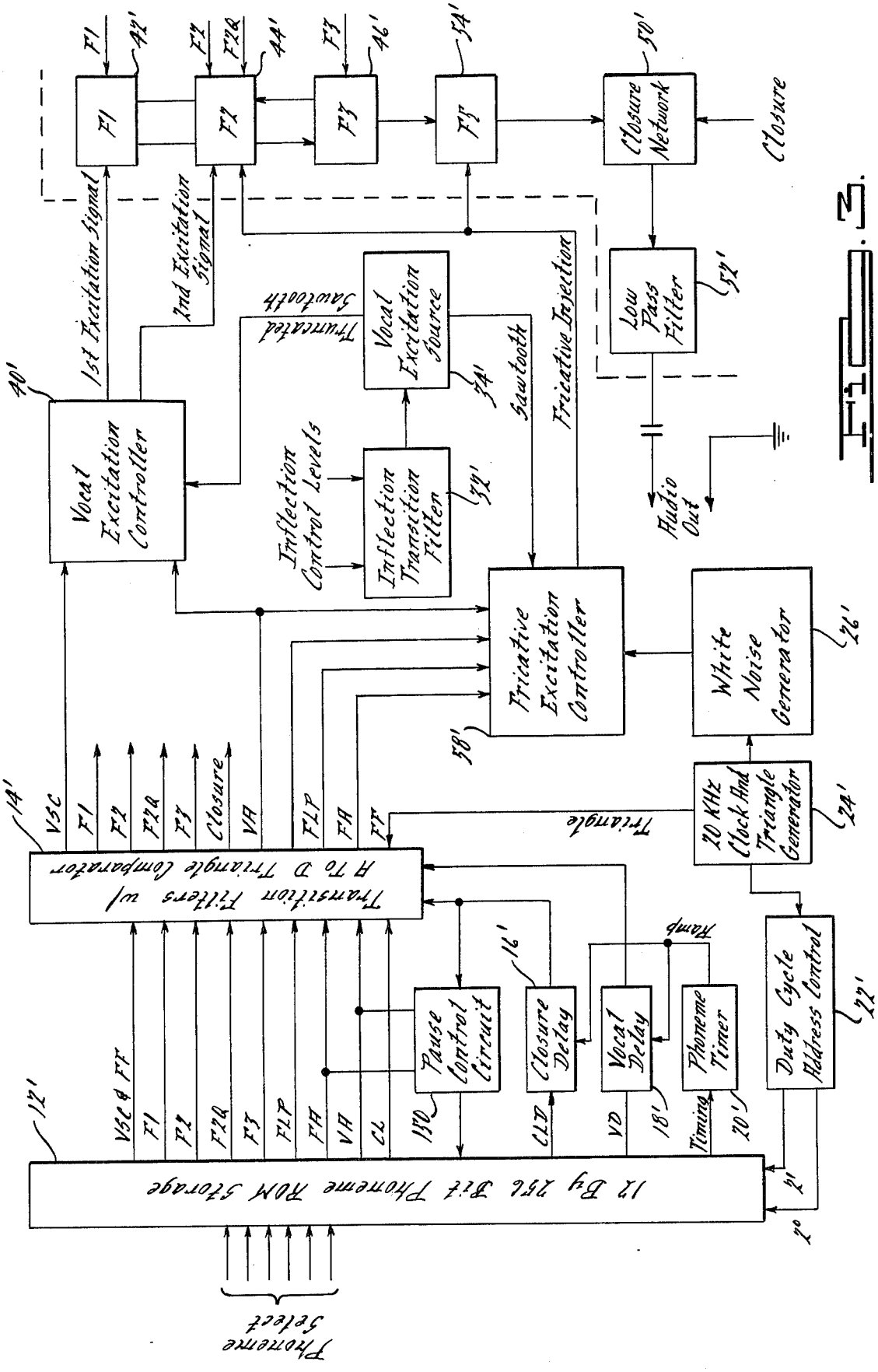
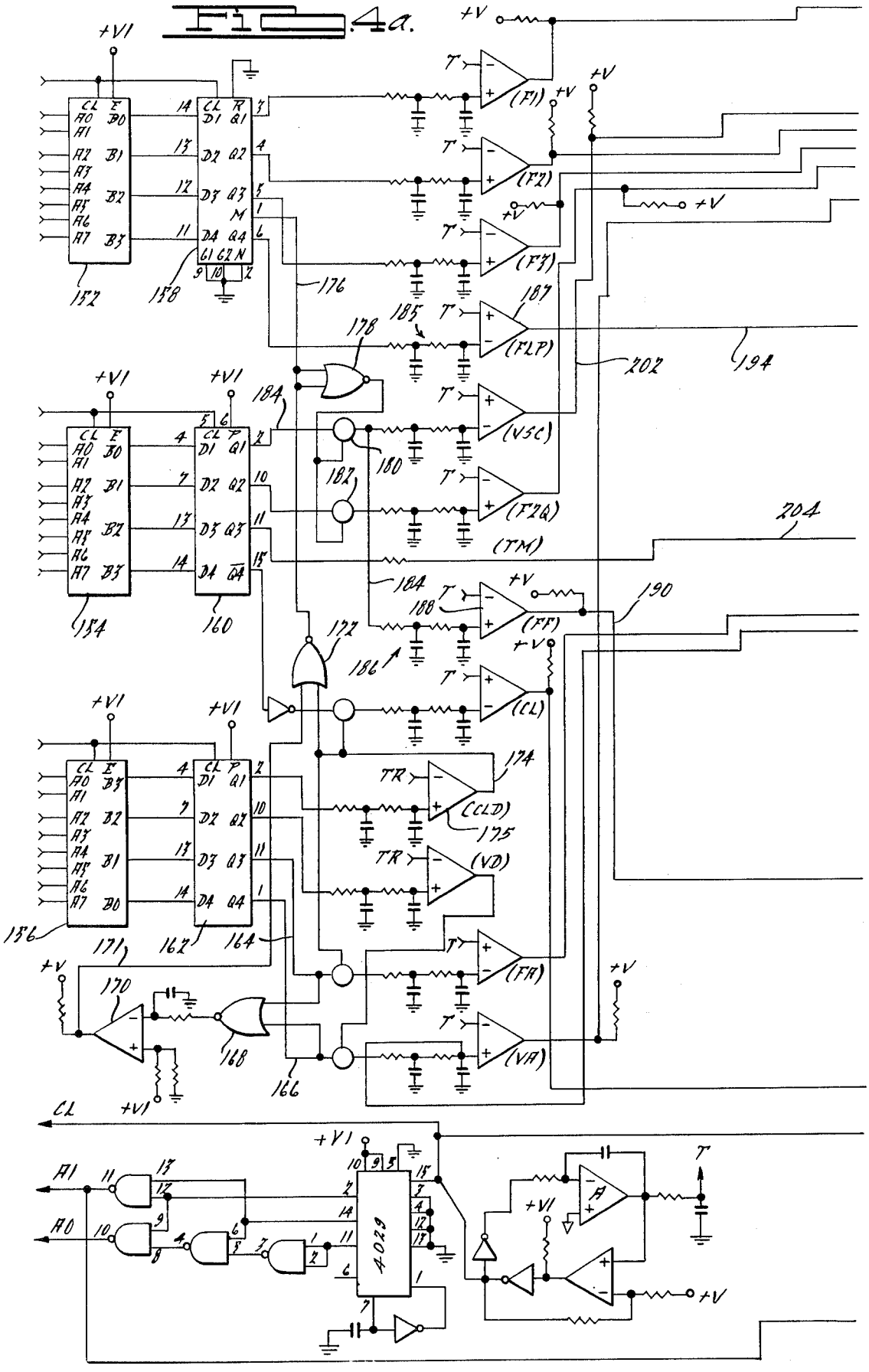

F1
42'
F2
F2A
44'
F3
46'
54'
50'
F1
F2
F3
F5
Closure Network
Closure
1st Excitation Signal
2nd Excitation Signal
Low Pass Filter
52'
Truncated Sawtooth
Vocal Excitation Source
34'
Sawtooth
Frication Injection
40'
Vocal Excitation Controller
Inflection Control Levels
Inflection Transition Filter
32'
Audio Out
Frication Excitation Controller
White Noise Generator
26'
58'
14'
VSC
F1
F2
F2A
F3
Closure
VH
FLP
FH
FF
Transition Filters w/ H To D Triangle Comparator
Triangle
20 KHz Clock And Triangle Generator
24'
Ramp
Duty Cycle Address Control
22'
16'
Pause Control Circuit
Closure Delay
Vocal Delay
Phoneme Timer
12'
VSC & FF
F1
F2
F2A
F3
FLP
FH
VH
T2
150
CLD
VD
18'
Timing
20'
12 By 256 Bit Phoneme ROM Storage
1'
2'
Phoneme Select

FIG. 2


+V1
+V
T
(F1)
(F2)
(F3)
(FLP)
(VSC)
(F2Q)
(TM)
(FF)
(CL)
TR
(CCLD)
(VD)
(FR)
(VH)
4029
CL
A1
A0
152
154
156
158
160
162
164
166
168
170
171
172
174
175
176
178
180
182
184
185
186
187
188
190
194
202
204

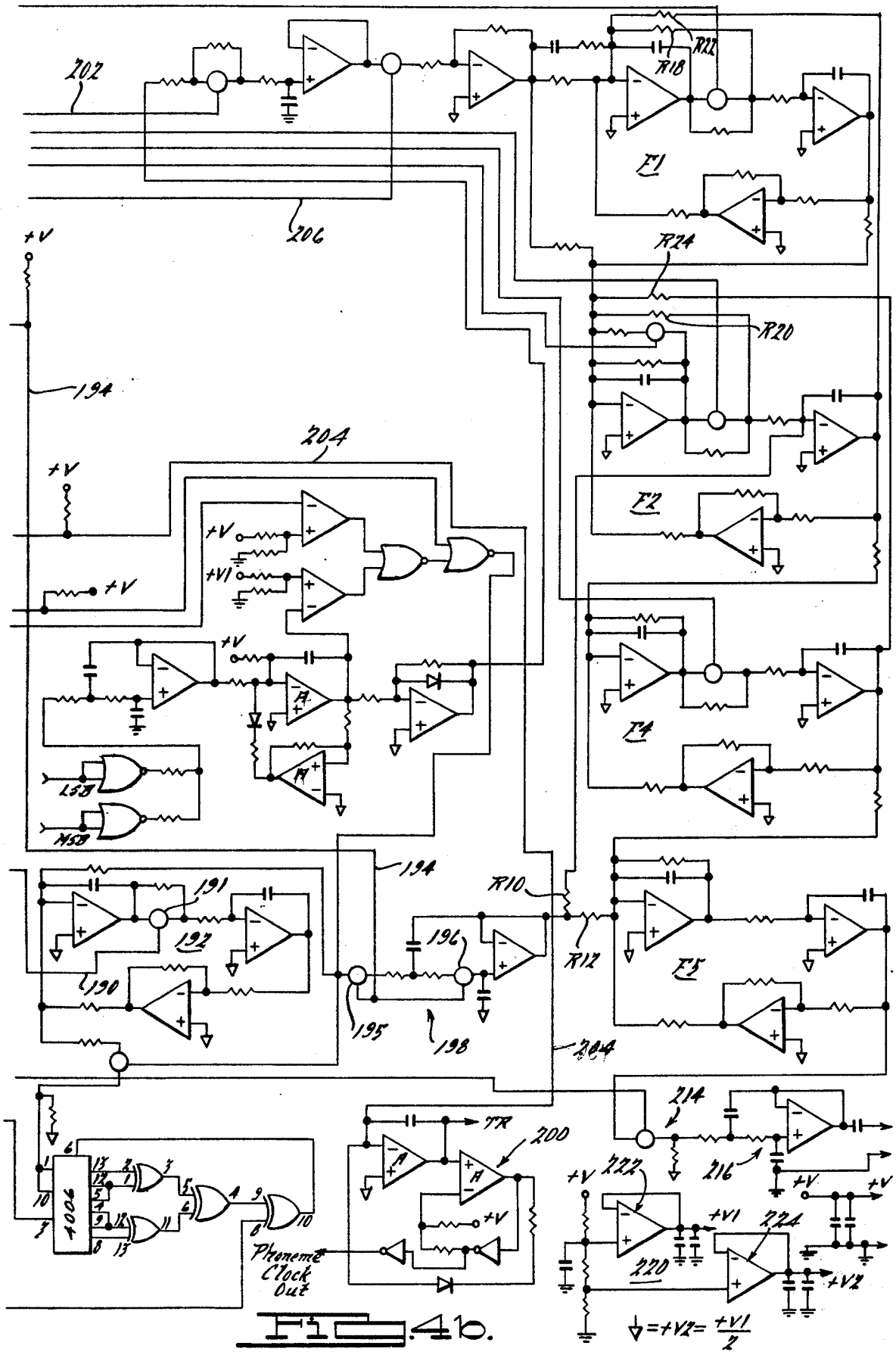

202
206
+V
194
204
R22
R18
F1
R24
R20
F2
F4
LSB
MSB
191
192
190
196
195
198
R10
R12
F5
214
216
200
TR
222
220
224
+V1
+V2
4006
Phoneme Clock Out
=+V2=+V1/2

Fig. 4b.

VOICE SYNTHESIZER

BACKGROUND AND SUMMARY OF THE INVENTION

The present invention relates to voice synthesizers and in particular to a highly simplified voice synthesizer that is capable of producing quality speech.

In general, the present invention comprises a synthesizer of the type disclosed in copending U.S. application Ser. No. 714,495, filed Aug. 16, 1976, entitled "Voice Synthesizer," and assigned to the assignee of the present application. While the synthesizer disclosed in the cited copending application comprises a highly sophisticated synthesizer capable of producing remarkably realistic sounding speech, the present invention is intended to provide a speech synthesizer that is simpler in design, smaller in size, and less expensive, yet nonetheless capable of producing quality speech.

The present speech synthesizer is adapted to be driven by an 8-bit digital input command word. Six of the bits are used for phoneme selection, thus providing 2^6 or 64 possible phonemes, and the remaining two bits are dedicated to inflection control. The system is adapted to generate twelve control parameters for each phoneme. In the first embodiment disclosed herein, one of the control signal parameters, referred to as the fricative control, is utilized to control the injection of both high and low frequency fricative energy into the vocal tract. More particularly, the system utilizes the fricative control signal and the inverse of the fricative control signal to control the parallel injection of fricative energy into the second and fourth (F5) resonant filters in the vocal tract. Thus, as will subsequently be explained in greater detail, for a given phoneme having an unvoiced component, fricative energy is injected directly into the F2 and F5 resonant filters, with the amount of fricative energy that is injected into the F2 resonant filter being inversely related to the amount injected into the F5 resonant filter. Also included in the first embodiment is a second fricative excitation control network that is adapted to control the injection of fricative energy in parallel into the second and third resonant filters in the vocal tract under the control of the vocal amplitude control signal. Consequently, the combination of the glottal waveform which is injected into the F1 resonant filter and the vocal amplitude controlled fricative injection into the F2 and F3 resonant filters, provides asynchronous excitation of the serial vocal tract. The result of using white noise as the primary source of excitation of the F2 and F3 resonant filters provides the synthesizer with a more "breathy" sounding voice.

A second embodiment disclosed herein is adapted to operate off a 12 volt power source and thus is particularly suited for use with a portable power supply. The system is also driven by 8-bit digital command words and is adapted to generate twelve electronic control signal parameters per phoneme. One of the control parameters, however is utilized to produce two separate control signals, thus providing an additional control signal without a lot of additional circuitry.

A unique pause control circuit is included in the second embodiment that is adapted to detect the existence of a pause phoneme, and then maintain the values of certain critical parameters beyond the termination of the phoneme preceding the pause to prevent the characteristics of the vocal tract from changing due to transitional changes in the control signal parameters before

the audio output has completely faded out. Briefly, the pause control circuit functions by producing an output signal whenever the circuit detects a lack of both the vocal amplitude and fricative amplitude control signals. The output signal produced is then utilized to sample and hold the outputs of a tri-state latch which maintains the current values of the affected parameters. The same output signal is also used to simultaneously disable a pair of analog gates to prevent transitional changes of two additional control signal parameters. The output signal is automatically terminated after a predetermined period of time into the pause phoneme less than the entire duration of the pause phoneme.

The serialized vocal tract in the second embodiment is also asynchronously driven as in the first embodiment, however vocal energy is used for the second excitation signal instead of white noise. More particularly, the glottal waveform that is injected into the first resonant filter is also injected in parallel into the second resonant filter. Thus, due to the inherent delay introduced by the F1 resonant filter, the F2 and F3 resonant filters are effectively driven twice; first by the direct parallel injection of vocal energy into the second resonant filter, and secondly by the delayed excitation from the residual vocal energy from the output of the first resonant filter. The result is an improved sounding voice due to the closer simulation of the true action of the human glottis which actually excites the vocal chords twice during each open and close cycle.

Additional objcts and advantages will become apparent from a reading of the detailed description of the preferred embodiments which makes reference to the following set of drawings in which:

BRIEF DESCRIPTION OF THE DRAWINGS

FIGS. **1***a* and **1***b* are a block diagram of one embodiment of a voice synthesizer according to the present invention;

FIG. **2***a* and **2***b* are a circuit diagram of the voice synthesizer shown in FIGS. **1***a* and **1***b*;

FIG. **3** is a block diagram of another embodiment of a voice synthesizer according to the present invention; and

FIG. **4***a* and **4***b* are a circuit diagram of the voice synthesizer shown in FIG. **3**.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Looking to FIGS. **1***a* and **1***b*, a block diagram of one of the preferred embodiments of a voice synthesizer according to the present invention is shown. As previously noted, the present voice synthesizer comprises a simplified and inexpensive version of the more sophisticated type of synthesizer disclosed in the copending U.S. application, Ser. No. 714,495, entitled "Voice Synthesizer," filed Aug. 16, 1976 and assigned to the assignee of the present application. The illustrated system is adapted to be driven by an 8-bit digital command word. Six of the input bits **15** from the digital command word are used for phoneme selection and the remaining two bits **25** for varying the inflection level of the audio output. The six phoneme select bits **15** are provided to a ROM storage unit **12** which has stored therein for each of the 64 (2^6) possible phonemes which can be identified by the six phoneme select bits, 12 different parameters which electronically define ech phoneme. Each parameter stored in ROM memory **12** preferably comprises four bits of resolution for producing the seri-

alized binary-weighted digital control signals described in the aforementioned copending application. Thus, the ROM memory unit **12** utilized in the preferred embodiment must have a storage capacity of at least 4 × 12 × 64 or 3,072 bits. The memory utilized in the preferred embodiment is a 12 × 256 bit read only memory (ROM).

Storage ROM **12** is adapted to be clocked under the control of a duty cycle address circuit **22** which provides the appropriate timing signals on lines **21** and **23** required for the ROM **12** to generate the serialized binary-weighted duty cycle parameter control signals previously mentioned. The duty cycle address control circuit **22** is connected to a clock circuit **24** that is adapted to produce a square wave clock signal at a frequency 20 KHz. The 20 KHz clock signal from clock circuit **24** is segregated by the duty cycle address control circuit **22** into 15 pulse groups, which are then further divided into time segments of 8, 4, 2 and 1 clock pulses. For each group of 15 clock pulses received, the duty cycle address control circuit **22** provides a HI output signal on line **23** or the MSB line during the eight and four time segments, and a HI output on line **21** or the LSB line during the eight and two time segments.

As previously noted, the serialized binary-weighted digital control parameters generated by ROM **12** preferably contain four bits of resolution. In other words, for each phoneme parameter, ROM **12** contains four bits of information, thereby providing 2^4 or 16 possible values per parameter. To provide the four bits with their appropriate binary weight, the first or most significant of the four serialized output bits in the control signal parameter is generated when both the signals on lines **21** and **23** are HI; the second bit when the LSB line is LO and the MSB line is HI; the third bit when the LSB line is HI and the MSB line is LO; and fourth or least significant of the four bits when the MSB and LSB lines are both LO. Thus, it can be seen that the first or most significant bit is produced for a period of eight clock pulses, the second bit is produced for a period of four clock pulses, the third bit is produced for a period of two clock pulses, and the fourth bit is produced for a period of one clock pulse. In this manner, an analog signal can be digitally represented as the average magnitude of a control signal over a 15 clock pulse period.

Although known to the art, the particular control signal parameters generated by ROM **12** will be briefly explained to provide a better understanding of the operation of the present system.

The **F1**, **F2**, and **F3** control signals determine the locations of the resonant frequency poles in the first three variable resonant filters **42**, **44**, and **46** respectively, in the vocal tract **60**. The timing control signal (Timing) is generated for each phoneme and is used to establish the period of production for each phoneme. The vocal amplitude control signal (VA) is generated whenever a phoneme having a voiced component is present. The vocal amplitude control signal controls the intensity of the voiced component in the audio output. The vocal delay control signal (VD) is generated during certin fricative-to-vowel phonetic transitions wherein the amplitude of the fricative constituent is rapidly decaying at the same time the amplitude of the vocal constituent is rapidly increasing. The vocal delay control signal is thus utilized to delay the transmission of the vocal amplitude control signal under such circumstances. The closure signal (CL) is used to simulate the phoneme interaction which occurs, for example, during the production of the phoneme "b" followed by the phoneme "e." In particular, the closure control signal, when provided to the closure network **50**, is adapted to cause an abrupt amplitude modulation in the audio output that simulates the build-up and sudden release of energy that occurs during the pronunciation of such phoneme combinations. The vocal spectral contour control signal (VSC) is used to spectrally shape the energy spectrum of the vocal excitation signal. Specifically, the vocal spectral contour control signal controls a first order low pass filter in circuit block **40** that suppresses the vocal energy injected into the vocal tract, with maximum suppression occurring in the presence of purely unvoiced phonemes. The F2Q control signal varies the "Q" or bandwidth of the second order resonant filter **44** in the vocal tract **60**, and is used primarily in connection with the production of the nasal phonemes "n," "m" and "ng." Nasal phonemes typically exhibit a higher amount of energy at the first formant (**F1**), and substantially lower and broader energy content at the higher formants. Thus, during the presence of nasal phonemes, the F2Q control signal is generated to reduce the Q of the **F2** resonant filter **44** which, due to the cascaded arrangement of the resonant filters in the vocal tract, prevents significant amounts of energy from reaching the higher formants. The fricative amplitude control signal (FA) is generated whenever a phoneme having an unvoiced component is present and is used to control the intensity of the unvoiced component in the audio output. The closure delay control signal (CLD) is generated during certain vowel-to-fricative phonetic transitions wherein it is desirable to delay the transmission of the closure and fricative amplitude control signals in the same manner as that discussed in connection with the vocal delay control signal. Finally, a unique fricative control signal (FC) is provided which replaces two control signals normally provided in synthesizers of this type; i.e., the fricative frequency and fricative low pass control signals. Specifically, it has been determined that, in general, when a fricative phoneme requires low frequency fricative energy in the range of the F2 formant, it does not also require high freguency fricative energy in the range of the F5 formant, and vice versa. Thus, the present invention utilizes a single fricative control (FC) signal, and the inverse of the FC control signal ($\overline{FC}$), to control the injection of both low and high frequency fricative energy into the vocal tract **60**. The specific manner in which this is accomplished will be subsequently explained in greater detail.

The output control signal parameters from ROM **12** are applied to a plurality of relatively slow-acting transition filters **14**. In actuality, the binary-weighted duty cycle control signals are effectively converted to analog signals by the transition filters, and then converted back to duty cycle digital signals by comparator amplifiers provided with a 20 KHz triangle clock signal from clock circuit **24**. The transition filters **14** are purposefully designed to have a relatively long response time in relation to the steady-state duration of a typical phoneme so that the abrupt amplitude variations in the output control signals from ROM **12** will be eliminated. Thus, the transition filters **14** provide gradual changes between the steady-state levels of the control signal parameters to simulate the smooth transitions between phonemes present in human speech. The response time of the transition filters **14** utilized in the preferred embodiment are fixed, thus eliminating the extensive

amount of circuitry necessary to provide variable speech rate capability.

The phoneme timer circuit **20** is adapted to produce a ramp signal that varies from five volts to zero volts in a time period that determines the duration of phoneme production. The slope of the ramp signal produced by the phoneme timer circuit **20** is dependent upon the value of the phoneme timing control signal from ROM **12**. The vocal delay control signal (VD) is provided to a vocal delay network **16** which is adapted to delay the transmission of the vocal amplitude control signal for a predetermined period of time less than the duration of a single phoneme time interval whenever the vocal delay control signal is provided by ROM **12**. The closure delay control signal (CLD) is provided to the closure delay network **18** which functions similar to the vocal delay network **16** and is adapted to delay the transmission of the fricative amplitude and closure control signals whenever the closure delay control signal is provided by ROM **12**.

The two inflection select bits **25** from the 8-bit input command word are provided directly to an inflection transition filter circuit **32** which combines the binary-weighted bits into a single analog inflection control signal, and then supplies the signal to a transition filter which smooths the abrupt amplitude variations in the inflection control signal in the same manner as that previously described with respect to transition filters **14**. The output from the inflection transition filter circuit **32** is provided to the vocal excitation or glottal source **34** which generates the voiced excitation signal or glottal waveform. The output from the inflection transition filter **32** determines the pitch of the voiced component, which corresponds to the fundamental frequency ($F\phi$) of the glottal waveform. In the preferred embodiment of the present invention, the glottal waveform generated by the vocal excitation source **34** comprises a truncated sawtooth type waveform similar to that described in copending U.S. application, Ser. No. 714,495, referred to above.

The glottal waveform from the vocal excitation source **34** is provided to the vocal tract **60** via the vocal excitation controller circuit **40**. The vocal excitation controller **40** is adapted to spectrally shape the energy content of the glottal waveform in accordance with the vocal spectral contour control signal, and modulate the amplitude of the vocal excitation signal in accordance with the vocal amplitude control signal.

The fricative excitation energy or unvoiced phoneme quantity of human speech is supplied by a white noise generator **26**. Injection of the fricative excitation signal into the vocal tract **60** is controlled by the fricative excitation controller circuit **58** and a novel second parallel fricative injection control network **38**. The fricative excitation controller **58** is shown broken down into its three individual circuits **28, 30** and **36** to emphasize the unique manner in which injection of the fricative component into the vocal tract **60** is controlled by this embodiment of the present invention. In particular, a conventional voice fricative network **30** is provided which is adapted to modulate the fricative amplitude control signal in accordance with the glottal waveform whenever a phoneme requiring voiced energy is generated, as determined by the existence of a vocal amplitude control signal. The fricative amplitude control signal is then provided to a high pass filter and fricative amplitude control network **28** which is adapted to filter the fricative excitation signal from the white noise generator **26** and modulate the amplitude of the signal in accordance with the fricative amplitude control signal. The modulated fricative excitation signal is then provided to a novel fricative injection control network **36** which is adapted to control the injection of fricative energy into the vocal tract **60** under the control of a single fricative control signal. The fricative excitation signal from the output of the fricative excitation controller **58** is parallel injected into both the F2 resonant filter **44** and the fricative or F5 resonant filter **54**.

As previously noted, the output from the white noise generator **26** is also provided to a second parallel fricative injection control network **38**. Significantly, it will be noted that the parallel fricative injection control network **38** is adapted to control the injection of fricative energy into the second and third resonant filters **44** and **46** under the control of the vocal amplitude control signal. As will subsequently be explained in greater detail, although the F1, F2 and F3 resonant filters **42, 44** and **46** respectively, are connected in serial form, the vocal excitation signal injected into the F1 resonant filter **42** does not have sufficient energy outside the F1 frequency range to adequately drive the second and third resonant filters, **44** and **46** respectively. Rather, in the embodiment illustrated in FIGS. **1***a* and **1***b*, the second and third resonant filters **44** and **46** are driven substantially with white noise under the control of the vocal amplitude control signal. The result of this arrangement is to provide the present voice synthesizer with a more "breathy" or "hoarse" sounding voice.

The output from the first three serially connected resonant filters **42, 44** and **46** are summed with the output from the fifth or fricative resonant filter **54**, as indicated at **48**, and the combined output is provided through the closure network **50** and a low pass filter **52** to an appropriate audio transducer. The closure network **50** is adapted to abruptly modulate the amplitude of the audio output signal in accordance with the closure control signal as previously described. The low pass filter **52** is adapted to filter the effects of the 20 KHz clock signal from the audio output.

Referring now to FIGS. **2***a* and **2***b*, a circuit diagram of the embodiment of the present voice synthesizer illustrated in FIGS. **1***a* and **1***b* is shown. As previously mentioned in connection with the description of the block diagram, the present voice synthesizer is adapted to be driven by an 8-bit digital input command word. The six input bits utilized for phoneme selection **74** are connected in parallel to a pair of ROM memories **70** and **72**. Two ROM IC chips are utilized to provide the required storage capability previously discussed. As also noted earlier, ROM memories **70** and **72** are adapted to produce binary-weighted duty cycle output control signals comprising the electronic parameters of the synthesized speech. In that the present invention constitutes an improvement in voice synthesizers and much of the circuitry is duplicative for each control signal of the circuitry known to the art, only the circuitry associated with the closure control signal, by example, will be explained in detail.

When a closure control signal is produced at the output of ROM **72**, it is provided through a CMOS buffer **78** to a fixed rate RC transition filter comprising resistors R1 and R2 and capacitors C1 and C2. The transition filter as noted, serves to smooth the abrupt amplitude variations in the binary-weighted digital control signal produced by ROM memory **72**. Additionally, it will be noted that prior to application to the transition

filter, the closure control signal is provided through an analog gate **82**, the control terminal of which is connected to the closure delay control signal on line **81**. As also discussed above, the closure delay control signal serves to momentarily delay the transmission of the closure control signal (as well as the fricative amplitude control signal) during certain vowel-to-fricative phoneme transitions.

Once the closure control signal has been provided through the transition filter and effectively converted thereby to an analog signal, it is converted back to a digital square wave signal having a duty cycle proportional to the amplitude of the analog signal. This is accomplished by connecting the output of the transition filter to the negative input of a comparator amplifier **80**. The positive input of comparator amplifier **80** is supplied with a 20 KHz triangle signal from the output clock circuit **85**. Comparator amplifier **80** effectively pulse width modulates the analog control signal provided to its negative input so that the output signal provided on line **84** comprises a square wave signal whose duty cycle is proportional to the magnitude of the analog signal provided to its negative input. The duty cycle closure control signal on line **84** is then provided to the control terminal of an analog gate **86** which is connected in circuit with the audio output line. The closure control signal on line **84** is adapted to momentarily render nonconductive analog gate **86** so as to cause an abrupt amplitude modulation of the audio output. As previously noted, the closure control signal is generated for certain phoneme interactions such as the phoneme "b" followed by the phoneme "e."

As discussed in connection with the description of the block diagram in FIGS. **1***a* and **1***b*, the remaining two bits **76** in the 8-bit digital input command word are utilized for inflection control. The two binary-weighted bits **76** are combined and provided through a transition filter **88** to smooth the abrupt amplitude variations in the combined signal. The resulting analog signal on line **89** is provided to a sawtooth generator circuit **90** which essentially comprises an integrator amplifier **91** that is adapted to produce a sawtooth waveform at its output at node **95**. The frequency of the sawtooth waveform generated by circuit **90** is dependent upon the magnitude of the signal provided to the negative input of integrator amplifier **91**. Thus, it can be seen that by varying the setting of inflection bits **76**, the fundamental frequency (Fϕ) of the glottal waveform is varied.

The sawtooth waveform at node **95** is provided through an additional waveform shaping circuit **100** that is adapted to effectively truncate the sawtooth waveform by subtracting the lower half of the signal. The resulting output signal on line **104** represents the glottal waveform that is injected into the vocal tract. For a more detailed explanation of the vocal excitation source circuitry, see the aforementioned copending U.S. application, Ser. No. 714,495.

Additionally, it will be noted that the sawtooth waveform at node **95** is also provided through an inverting amplifier **97** to the input of a NOR-gate **98**. NOR-gate **98** is controlled by the output of op amp **94** which is adapted to enable NOR-gate **98** whenever a vocal amplitude control signal is produced on line **92**. When a vocal amplitude control signal is present on line **92**, the output from op amp **94** will go LO, thereby causing NOR-gate **98** to "square-up" the sawtooth waveform from the output of op amp **97**. The square wave signal from the output of NOR-gate **98** is then provided to the input of another NOR-gate **102** which has its other input connected to receive the fricative amplitude control signal on line **96**. Thus, it can be seen that when a vocal amplitude control signal is present on line **92**, thereby enabling NOR-gate **98**, NOR-gate **102** will "chop" the fricative amplitude control signal on line **96** in accordance with the "squared-up" sawtooth waveform from node **95**. When a vocal amplitude control signal is not present on line **92**, NOR-gate **98** is thereby inhibited rendering its output LO, which in turn makes NOR-gate **102** appear like an inverter permitting the fricative amplitude control signal on line **96** to pass unaffected by the square wave signal. It will be noted, that since the frequency of the sawtooth waveform at node **95** is approximately 200 times slower than the duty cycle frequency of the fricative amplitude control signal on line **96** (100 Hz vs. 20 KHz), the "chopping" of the fricative amplitude control signal by the sawtooth waveform is effective to substantially diminish the fricative or unvoiced speech component whenever a phoneme requiring voiced energy, as indicated by the presence of a vocal amplitude control signal, is present.

The fricative amplitude control signal from the output of NOR-gate **102** on line **96'** is provided to the control terminal of an analog gate **106** that is connected in circuit to the output of the white noise generator **110**. The fricative excitation signal on line **108** produced by generator **110** is effectively amplitude modulated by the rapid on/off cycling of analog gate **106** under the control of the fricative amplitude duty signal control signal. The modulated signal is then provided through a 4 KHz high pass filter **122** to an additional pair of analog gates **118** and **120**. Analog gates **118** and **120** are adapted to control the injection of fricative excitation energy into the F2 and F5 resonant filters in the vocal tract. Unlike previous synthesizers, the present invention is adapted to control the injection of fricative energy into the vocal tract with a single control parameter; herein the fricative control (FC) signal. Thus, the circuitry required to generate an additional control parameter is eliminated. Upon examination of the frequency spectrum of fricative phonemes, it was determined that for the most part phonemes requiring substantial amounts of low frequency fricative energy in the range of the F2 formant, do not also require substantial amounts of high frequency fricative energy in the range of the F5 formant, and vice versa. For example, for fricative phonemes used as "f" and "p," fricative energy must be injected primarily into the F2 resonant filter, and for phonemes such as "s" and "t," it is necessary to inject fricative energy primarily into the F5 resonant filter. Consequently, the present system is adapted to generate a single fricative control parameter (FC) on line **112** which is also provided through an inverting comparator amplifier **114** to produce the inverse of the fricative control parameter ($\overline{FC}$) on line **116**. The fricative control parameter on line **112** is connected to the control terminal of analog gate **118** and is adapted to control the injection of low frequency fricative energy on line **124** into the F2 resonant filter, and the inverse of the fricative control signal on line **116** is connected to the control terminal of analog gate **120** and is adapted to control the injection of high frequency fricative energy on line **126** into the fricative or F5 resonant filter. Thus, it will be appreciated that the amount of fricative energy that is injected into the F2 resonant filter is inversely related to the amount of fricative energy that is injected into the F5 resonant filter.

The voiced component or glottal waveform on line **104** from the voiced excitation source is injected into the vocal tract at the F1 resonant filter. Injection of the voiced component into the vocal tract is controlled by the vocal spectral contour control signal on line **140** and the vocal amplitude control signal on line **128**. In particular, the vocal amplitude and vocal spectral contour control signals are connected to the control terminals of analog gates **130** and **142** respectively, which are connected in circuit with the voiced excitation signal on line **104**. As previously noted, the vocal spectral contour control signal is adapted to spectrally shape the energy content of the voiced excitation signal by controlling the cutoff frequency of a first order low pass filter **143**, and the vocal amplitude control signal is adapted to modulate the amplitude of the voiced excitation signal.

Although the F1, F2, and F3 resonant filters are serially connected, the voiced excitation signal in the preferred embodiment herein does not contain enough high frequency energy to adequately drive the F2 and F3 resonant filters. This, of course, is contrary to conventional practice wherein the first three resonant filters in the vocal tract are driven principally by the voiced component of speech. However, in order to provide the present synthesizer with a more "breathy" or "hoarse" voice, the second and third resonant filters herein are driven principally with fricative energy under the control of the vocal amplitude control signal. Specifically the output from the white noise generator **110** on line **108** is injected directly into the F2 resonant filter through resistor R4 and into the F3 resonant filter through resistor R5. Injection of white noise into the F2 and F3 resonant filters is controlled by analog gate **134** which has its control terminal connected to receive the vocal amplitude control signal on line **128**. Thus, it can be seen that the F2 and F3 resonant filters in the present embodiment are driven asynchronously, in parallel, with white noise under the control of the vocal amplitude control signal. The asynchronous drive of the F2 and F3 resonant filters derives from the fact that residual vocal energy from the output of the F1 resonant filter does cause a certain amount of excitation of the F2 and F3 resonant filters. However, due to the inherent delay created by the voice component passing through the F1 resonant filter, the F2 and F3 resonant filters are subject to double excitation; first with fricative energy through resistors R4 and R5 and secondly by the delayed vocal energy from the output of the F1 resonant filter.

Finally, as noted in the block diagram, the output from the F1, F2 and F3 serially connected resonant filters in the vocal tract is combined with the output from the fricative or F5 resonator by summing circuit **144** and provided through a low pass filter circuit **146** to an appropriate audio transducer device.

Looking now to FIG. 3, a block diagram of another embodiment of the present invention is shown. The blocks appearing in FIG. 3 which correspond to blocks shown in the first embodiment illustrated in FIGS. 1*a* and 1*b* are labeled with primed reference numerals. As can be readily seen from the diagram, the embodiment illustrated in FIG. 3 is also driven by an 8-bit digital input command word with six of the input bits utilized for phoneme selection and the remaining two bits used for inflection control. As in the first embodiment, the read-only memory unit **12'** is adapted to generate 12 control signal parameters for each phoneme. However, it will be noted that one of the signal parameters is utilized to produce two separate control signals; i.e., the vocal spectral contour and fricative frequency control signals. The generation of a separate fricative frequency control signal permits the fricative control signal, as it was referred to in the first embodiment, to be used solely as a fricative low pass (FLP) control signal. Thus, a conventional fricative excitation controller network **58'** can be utilized.

The second embodiment also includes a unique pause control circuit **150** which is adapted to "hold" the values of certain critical control parameters from the output of ROM **12'** whenever a pause in the audio output is detected. The purpose of the pause control circuit **150** is to prevent the values of the critical control parameters from changing and thus altering the characteristics of the vocal tract **60** before the audio has completely faded out. The pause control circuit **150** is adapted to detect a pause by continuously monitoring the fricative amplitude and vocal amplitude control signals and providing an output signal whenever both signals are LO. The output signal produced thereby is fed back to the latch circuits at the outputs of ROM **12'** to "hold" the parameters at their current values. The pause control circuit **150** is further adapted to terminate the "hold" signal after a predetermined period into the pause phoneme as determined by the closure delay control signal from closure delay network **16'**.

The remaining differences in the present embodiment are found in the vocal tract **60'** and the manner in which the voiced and unvoiced excitation signals are injected into the vocal tract **60'**. Specifically, the F1, F2, F3 and F5 resonant filters **42'**,**44'**, **46'** and **54'** respectively, in the present embodiment are all serially connected rather than having the F5 resonant filter connected in parallel with the first three serially connected resonant filters as in the first embodiment. Additionally it will be noted that a feedback path has been added between the F2 and F1 resonant filters **44'** and **42'**, between the F3 and F2 resonant filters **46'** and **44'**. These feedback paths are provided to simulate the back pressures which are generated in the human voice system between the tongue, mouth and vocal chords.

Finally, it will be noted that the present embodiment also provides asynchronous parallel excitation of the vocal tract **60'**. However, unlike the first embodiment, the asynchronous parallel excitation herein is supplied solely by the voiced component. In particular, it can be seen that the output from the fricative excitation controller **58'** is only injected in parallel into the F2 and F5 resonant filters **44'** and **54'** in the conventional manner. However, the voiced excitation signal from the output of the vocal excitation controller **40'**, in addition to being injected into the F1 resonant filter **42'**, is also injected in parallel into the F2 resonant filter **44'**. Thus, the F2 resonant filter **44'**, and to a lesser extent the F3 resonant filter **46'**, are driven twice; first by the direct injection of vocal energy into the F2 resonant filter **44'**, and subsequently by the delayed vocal energy from the output of the F1 resonant filter **42'**. The purpose of this arrangement is to more accurately simulate the true action of the human glottis which provides a type of "double" excitation of the vocal chords each time it opens and closes.

Referring not to FIGS. 4*a* and 4*b*, a circuit diagram of the embodiment of the present invention illustrated in FIG. 3 is shown. At the outset, it is to be noted that the voice synthesizer illustrated in FIGS. 4*a* and 4*b* is

adapted to operate off a 12 volt power supply. In actuality, the system will function off a supply that varies anywhere from 6 volts to 15 volts. Thus, this embodiment of the present invention is particularly suited for use in combination with a portable battery power source.

The power requirements of the present system is such that four discrete voltage levels are needed. In addition to the +V (e.g. 12 volts) and ground potentials provided by the battery, the present system includes a power supply circuit **220** that is adapted to generate two additional voltage levels, designated +V1 and +V2, between +V and ground. However, since the voltage output of a battery will vary over its useful life, it is important that the +V1 and +V2 voltage levels vary correspondingly. Thus, the present power supply circuit **220** includes a pair of voltage follower circuits **222** and **224** which are adapted to produce output signals that "follow" variations in the voltage level of the signals provided to their inputs.

Additionally, the change to a variable power source also mandates the use of op amps in certain portions of the circuit that are capable of providing an adequate current sink at their minimum rated voltage. Accordingly, the preferred embodiment utilizes Fairchile 798 op amps for those op amps designated with the letter "A."

The ROM storage requirement is supplied in this embodiment by three individual CMOS ROM memory chips **152, 154,** and **156,** herein No. MC14524. The outputs from ROM memories **152, 154,** and **156** are provided to latch circuits **158, 160** and **162** respectively, which serve the purpose of the CMOS buffers used in the first embodiment to drive the slow-acting transition filters, and also serve to inhibit the CMOS ROM data outputs from going HI during address switching. Latch circuit **158** is a tri-state latch, the third state providing a sample-and-hold function.

As discussed previously, the transitional changes in the values of the more critical control parameters may give rise to a condition most noticeable with the last phoneme before a pause, wherein the value of the control parameters may change prior to complete dissipation of the excitation energy in the vocal tract. The result is that the last phoneme before a pause will begin to take on a different characteristic and therefore a different sound as the audio fades out. To rectify this situation, the fricative amplitude control signal on line **164** and the vocal amplitude control signal on line **166** are provided to a NOR-gate **168** which has its output connected to the negative input of a comparator amplifier **170.** When both the fricative amplitude and vocal amplitude control signals are LO, the output from NOR-gate **168** will go HI, causing the output of comparator amplifier **170** on line **171** to go LO. The LO signal on line **171** in turn causes the output of NOR-gate **172** to go HI, thereby switching tri-state latch **158** to its sample-and-hold state. Additionally, the HI output signal from NOR-gate **172** on line **176** is provided through an inverter **178** to the control terminals of a pair of analog gates **180** and **182.** Analog gates **180** and **182** are connected in circuit with the vocal spectral contour (VSC + FF) and F2Q control signals respectively, appearing at the Q1 and Q2 outputs of latch circuit **160.** When the signal on line **176** goes HI causing the output of inverter **178** to go LO, analog gates **180** and **182** are open circuited, thus isolating the transition filters associated with the VSC + FF and F2Q control signals from further changes in the output state of latch **160.**

Thus, it can be seen that whenever a pause phoneme is detected, as determined by the absence of both the vocal amplitude and fricative amplitude control signals, the **F1, F2, F3,** and **FLP** control signal parameters appearing at the outputs of tri-state latch **158** are held at their current values, and the transiton filters associated with the vocal spectral contour, fricative frequency, and F2Q control signals are isolated from the outputs of latch **160.** Accordingly, it can be seen that the capacitors in the transition filters associated with each of the various critical control signal parameters identified are effectively isolated during the initial part of the pause phoneme from further changes in the ROM outputs to insure that the vocal energy in the vocal tract completely fades out before the existing phoneme parameters are changed.

The HI signal on line **176** at the output of NOR-gate **172** is automatically terminated after a predetermined period of time into the pause phoneme to permit resumption of normal circuit operation. In particular, the other input to NOR-gate **172** is connected to receive the closure delay (CLD) duty cycle control signal on line **174** from the output of comparator amplifier **175.** The output from comparator amplifier **175** is always initially LO at the beginning of a phoneme period due to the triangle ramp signal (TR) provided to its negative input from the phoneme timer circuit **200.** However, after a predetermined period of time less than the duration of an entire phoneme period, the magnitude of the TR signal will drop below the magnitude of the CLD control signal provided to the positive input of comparator amplifier **175,** thus causing its output on line **174** to go HI. The predetermined period of time is, of course, dependent upon the slope of the TR signal which is in turn controlled by the phoneme timing control signal on line **204.** When the closure delay duty cycle control signal on line **174** goes HI, the output of NOR-gate **172** goes LO, thus removing the sample-and-hold signal from tri-state latch **158** and rendering analog gates **180** and **182** conductive.

Additionally, it will be noted that the same control signal parameter from the Q1 output of latch circuit **160** on line **184** is provided to two separate transition filter circuits **185** and **186.** The output from transition filter **185** is provided through an analog-to-digital converter **187** to provide the vocal spectral contour duty cycle control signal on line **202,** and the output from transition filter **186** is provided through an analog-to-digital converter **188** to provide the fricative frequency duty cycle control signal on line **190.** Thus, it can be seen that a single control signal parameter on line **184** is utilized to provide both the vocal spectral contour control signal on line **202** and the fricative frequency control signal on line **190.**

As noted in the discussion of the block diagram of FIG. **3,** the generation of a separate fricative frequency control signal permits the ue of a conventional controller network comprising separately controlled bandpass and low pass filter circuits, **192** and **198** respectively. In particular, the fricative frequency control signal on line **190** is provided to the control terminal of an analog gate **191** which is adapted to control the bandpass of the bandpass filter **192.** The remaining fricative control signal, referred to simply as the FC control signal in the first embodiment, is utilized solely as a low pass control signal. Accordingly, the fricative low pass (FLP) con-

trol signal on line **194** is provided to the control terminals of a pair of analog gates **195** and **196** which are adapted to control the cut-off frequency of the low pass filter **198** in the fricative excitation controller network. The fricative excitation signal from the controller network is injected into the vocal tract at the F2 resonant filter through resistor **R10** and at the F5 resonant filter through resistor R12. Since the value of resistor **R10** is substantially greater than the value of resistor **R12**, the major portion of the fricative excitation energy is injected into the F5 resonant filter.

The vocal excitation signal or glottal waveform on line **200** is spectrally shaped and amplitude modulated under the control of the vocal spectral contour control signal on line **202** and the vocal amplitude control signal on line **206**, respectively. The glottal waveform is then injected into the vocal tract at the F1 resonant filter through resistor **R14** and at the F2 resonant filter through resistor **R16**. Thus, as in the first embodiment, the vocal tract is driven asynchronously due to the fact that the glottal waveform is effectively delayed — i.e., shifted approximately 180° — as it passes through the F1 resonant filter. Accordingly, the F2 and F3 resonant filters are effectively driven twice; first by the direct injection of the voiced excitation signal through resistor **R16**, and subsequently by the delayed injection of vocal energy from the output of the F1 resonant filter.

By driving the vocal tract asynchronously as described, the present speech synthesizer more closely simulates the true action of the human glottis. Specifically, the glottis does not provide a single excitation of the vocal chords by opening and closing smoothly. Rather, it has been found that the glottis initially closes on one side and then subsequently closes completely with a rapid motion. Accordingly, the vocal tract is effectively excited twice with each complete opening and closing of the glottis. The asynchronous drive of the present system thus simulates this action by providing double vocal excitation of the vocal tract.

Moreover, it has been found that, particularly in view of the fact that an F4 resonant filter is not used, the audio output sounds better if the glottal waveform does not have a substantial amount of high frequency energy when injected into the F1 resonant filter. However, with the high frequency energy of the glottal waveform reduced when injected into the F1 resonant filter, there is insufficient energy remaining in the glottal waveform at the output of the F1 resonant filter to adequately drive the F2 and F3 resonant filters. Accordingly, the parallel injection of the voiced excitation signal into the F2 resonant filter also serves to provide adequate high frequency vocal energy to the F2 and F3 resonant filters.

Additionally, it will be noted that a feedback resistor **R22** is provided between the output of the F2 resonant filter and the input of the F1 resonant filter, and another feedback resistor **R24** is provided between the output of the F3 resonant filter and the input to the F2 resonant filter. These feedback resistors simulate the normal back pressures which are present in the human vocal system. Specifically, when the mouth closes, the back pressure created affects the vibration of the vocal chords. Similarly, the movement of the tongue also creates back pressures which affect the vibration of the vocal chords. Thus, the inter-resonant feedback provided by resistors **R22** and **R24** serve to more closely model the present vocal tract to the human voice system. Also it will be noted that a pair of resistors **R18** and **R20** are provided across the bandpass sections of the F1 and F2 resonant filters, respectively. It has been found that "Q" or bandpass of the F1 and F2 resonant filters varies inversely with changes in the resonant frequencies of the filters, although to a lesser extent. Thus, resistors **R18** and **R20** are provided to implement this feature.

Finally, as noted in the block diagram in FIG. **3**, the present embodiment utilizes a completely serially connected vocal tract. In particular, the **F1**, **F2**, **F3** and **F5** resonant filters are all connected in cascaded form, with the output from the F5 resonant filter provided through the closure network **214** and a 20 KHz low pass filter **216** to an appropriate audio transducer device.

While the above description constitutes the preferred embodiments of the invention, it will be appreciated that the invention is susceptible to modification, variation and change without departing from the proper scope or fair meaning of the accompanying claims.

What is claimed is:

1. In an electronic device for phonetically synthesizing human speech including

input means responsive to input data identifying a desired sequence of phonemes for producing a plurality of control signals that electronically define each phoneme in said desired sequence of phonemes, including a first control signal for controlling the amplitude of the voiced component of speech and a second control signal for controlling the amplitude of the unvoiced component of speech;

vocal source means for producing a voiced excitation signal;

fricative source means for producing an unvoiced excitation signal; and

vocal tract means responsive to said voiced and unvoiced excitation signals and certain of said plurality of control signals for substantially producing the frequency spectrums of each of said desired sequence of phonemes, including a first resonant filter tunable under the control of a third of said control signals for producing the first formant in said frequency spectrums and a second resonant filter serially connected to said first resonant filter and tunable under the control of a fourth of said control signals for producing the second formant in said frequency spectrums;

the improvement comprising controller means for injecting said voiced and unvoiced excitation signals into said vocal tract means including first controller means for injecting excitation energy in parallel into said first and second resonant filters under the control of said first control signal and second controller means for injecting excitation energy into said vocal tract means under the control of said second control signal.

2. The speech synthesizer of claim **1** wherein said first controller means is adapted to inject said voiced excitation signal in parallel into said first and second resonant filters.

3. The speech synthesizer of claim **1** wherein said first controller means is adapted to inject said voiced excitation signal into said first resonant filter and said unvoiced excitation signal into said second resonant filter.

4. The speech synthesizer of claim **3** wherein said vocal tract means further includes a third resonant filter serially connected to said second resonant filter and tunable under the control of a fifth of said control signals for producing the third formant in said frequency

spectrums, and said first controller means is further adapted to inject said unvoiced excitation signal into said third resonant filter under the control of said first control signal.

5. The speech synthesizer of claim 4 wherein said second controller means is adapted to inject said unvoiced excitation signal into said vocal tract means.

6. The speech synthesizer of claim 5 wherein said vocal tract means further includes a fourth resonant filter for producing the fifth formant in said frequency spectrums, and said second controller means is adapted to inject said unvoiced excitation signal in parallel into said second and fourth resonant filters.

7. The speech synthesizer of claim 6 wherein said fourth resonant filter is connected in parallel with said serially connected first, second, and third resonant filters.

8. The speech synthesizer of claim 2 wherein said second controller means is adapted to inject said unvoiced excitation signal into said vocal tract means.

9. The speech synthesizer of claim 8 wherein said vocal tract means further includes a third resonant filter serially connected to said second resonant filter and tunable under the control of a fifth of said control signals for producing the third formant in said frequency spectrums and a fourth resonant filter for producing the fifth resonant formant in said frequency spectrums, and said second controller means is adapted to inject said unvoiced excitation signal in parallel into said second and fourth resonant filters.

10. The speech synthesizer of claim 9 wherein said fourth resonant filter is serially connected to said third resonant filter.

11. The speech synthesizer of claim 1 further including pause control means connected to said input means for producing an output signal that is effective to cause said input means to maintain the current values of certain of said control signals beyond the normal phoneme period whenever both said first and second control signals are absent.

12. The speech synthesizer of claim 11 wherein said pause control means is further adapted to terminate production of said output signal after a predetermined time period less than the duration of an entire phoneme period in accordance with one of said control signals.

13. The speech synthesizer of claim 12 wherein said one control signal is a closure delay control signal.

14. The speech synthesizer of claim 1 wherein said vocal tract means further includes a third resonant filter for producing the third formant in said frequency spectrums and a fourth resonant filter for producing the fifth formant in said frequency spectrums, and said second controller means is further adapted to inject said unvoiced excitation signal into said second resonant filter under the additional control of another of said control signals and also inject said unvoiced excitation signal into said fourth resonant filter under the additional control of the inverse of said another control signal.

15. The speech synthesizer of claim 14 wherein said third resonant filter is serially connected to said second resonant filter and said fourth resonant filter is connected in parallel with said first, second, and third resonant filters.

16. In an electronic device for phonetically synthesizing human speech including

vocal source means for producing a voiced excitation signal;

fricative source means for producing an unvoiced excitation signal;

input means responsive to input data identifying a desired sequence of phonemes for producing a plurality of control signals that electronically define each phoneme in said desired sequence of phonemes, including a first control signal for controlling the amplitude of said voiced excitation signal and a second control signal for controlling the amplitude of said unvoiced excitation signal; and

vocal tract means responsive to said voiced and unvoiced excitation signals and certain of said plurality of control signals for substantially producing the frequency spectrums of each of said desired sequence of phonemes;

the improvement comprising pause control means connected to said input means for producing an output signal that is effective to cause said input means to maintain the current values of certain of said control signals beyond the normal phoneme period whenever both said first and second control signals are absent.

17. The speech synthesizer of claim 16 wherein said pause control means is further adapted to terminate production of said output signal after a predetermined time period less than the duration of an entire phoneme period in accordance with one of said control signals that is produced at the beginning of each phoneme.

18. The speech synthesizer of claim 17 wherein said one control signal is a closure delay control signal.

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