SPEECH PROCESSOR

SP-1

DESCRIPTION

The SP-1 Speech Processor can enhance the intelligibility of speech for listeners in a noisy environment as much as six times the peak audio power. Adapted from the flight deck announcing system on US Navy aircraft carriers, it is a cost-effective way to improve coverage of Grand Prix car races, factory floors, or any high noise area. The SP-1 applies a combination of frequency shaping and dynamic range compression by single sideband clipping. The processing can be switched in from the front panel or remotely for background music/paging applications. A 7 hertz frequency shift can be selected for control of acoustic feedback. Input is a 600 ohm line or a low impedance microphone; output is at 600 ohm line level.

SPECIFICATIONS

Type: Speech processor using adjustable peak clipping of a lower sideband signal at a suppressed carrier frequency of 25 kHz.
Frequency response: Optimized for intelligibility with SSB clipping: 6 dB/octave pre-emphasis up to a broad peak from 1500 to 3100 Hz, 18dB/octave rolloff above 3400 Hz. Flat response when processing is not selected.
Output: Continuously adjustable from 0 to 10 volts peak into 600 ohms (19 dBm).
Line input: Input impedance 10 kilohms for bridging across 600 ohm line. 600 mV peak (-5 dBm re 600 ohms) with line level control at maximum will give correct calibration of CLIPPING control.
Microphone input: 0.6 mV peak with microphone level control at maximum will give correct calibration of CLIPPING control. Input impedance greater than 2 kilohms above 50 Hz. 20dB pad selectable on rear panel for high-sensitivity microphones.
Frequency shift: When selected, all frequencies are shifted upward by 7 Hz for control of acoustic feedback.
Remote selection of processing: Contact closure to ground, or floating 24 Vdc 10 mA source.
Front panel controls:
- **Line level:** Continuously variable potentiometer for 600 ohm line input.
- **Mic level:** Continuously variable potentiometer for low impedance microphone input.
- **Clipping (dB):** Continuously variable over 30 dB range. With CLIPPING control set at mid scale (12dB), MICROPHONE LEVEL or LINE LEVEL is adjusted for 12 dB clipping as read on LED bar graph. CLIPPING control can then be used to select the amount of clipping desired. Peak output level will stay approximately the same when processing is switched in or out.
- **Output:** Continuously variable from 0 to 10 volts peak.
- **Processing IN or OUT/REMOTE switch.**
- **7-Hz frequency shift IN or OUT switch.**
- **Display brightness:** Screwdriver adjustment on front panel for CLIPPING and OUTPUT LED bar graphs.
- **Power ON/OFF switch.**

Rear panel controls: Mic Level pad: switchable in or out. Line input: pin 1 ground, switchable on or off. Line output: pin 1 ground, switchable on or off.
Displays: Two 10-segment LED bar graphs, one for CLIPPING (-3 dB to over +24 dB), one for peak OUTPUT (-9 dBm to over +18 dBm). Each display has 5 green, 3 yellow, and 2 red segments.
Connectors: Microphone input: 3 pin XLR female, screw terminal. Line input: 3 pin XLR female, screw terminal. Line output: 3 pin XLR male, screw terminal. Processor in/out remote control: screw terminal.
Power required: 117 Vac, 50/60 Hz, 11 watts
Mounting: 31/2" x 19" rack mount
Dimensions: 31/2" high, 19" wide, 83/8" deep
Weight: 7.5lbs.

COMMUNICATIONS COMPANY inc.

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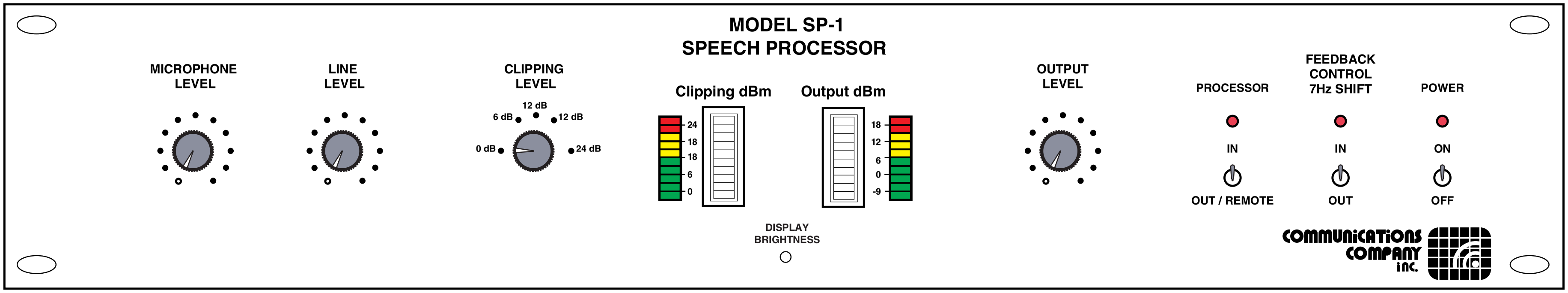
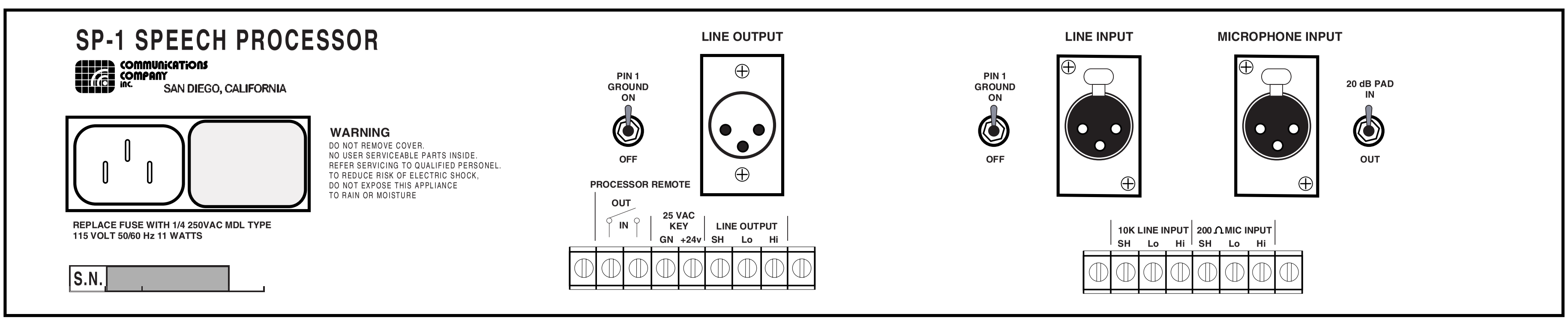

MODEL SP-1
SPEECH PROCESSOR
MICROPHONE LEVEL
LINE LEVEL
CLIPPING LEVEL
0 dB
6 dB
12 dB
12 dB
24 dB
Clipping dBm
24
18
18
6
0
Output dBm
18
12
6
0
-9
DISPLAY BRIGHTNESS
OUTPUT LEVEL
PROCESSOR
IN
OUT / REMOTE
FEEDBACK CONTROL 7Hz SHIFT
IN
OUT
POWER
ON
OFF
COMMUNICATIONS COMPANY INC.


SP-1 SPEECH PROCESSOR
COMMUNICATIONS COMPANY INC.
SAN DIEGO, CALIFORNIA
WARNING
DO NOT REMOVE COVER.
NO USER SERVICEABLE PARTS INSIDE.
REFER SERVICING TO QUALIFIED PERSONEL.
TO REDUCE RISK OF ELECTRIC SHOCK,
DO NOT EXPOSE THIS APPLIANCE
TO RAIN OR MOISTURE
REPLACE FUSE WITH 1/4 250VAC MDL TYPE
115 VOLT 50/60 Hz 11 WATTS
S.N.
PIN 1 GROUND ON
OFF
LINE OUTPUT
PROCESSOR REMOTE
OUT
IN
25 VAC KEY
GN +24v
LINE OUTPUT
SH Lo Hi
PIN 1 GROUND ON
OFF
LINE INPUT
MICROPHONE INPUT
20 dB PAD IN
OUT
10K LINE INPUT
SH Lo Hi
200 Ω MIC INPUT
SH Lo Hi

SP-1 INSTRUCTIONS

Adjusting the SP-1: To set up the SP-1 apply a voice signal at the expected level to either the 600 ohm line input or the low impedance microphone input. Set the CLIPPING control to mid-scale (12 dB) then adjust the LINE LEVEL or MICROPHONE LEVEL so the CLIPPING display indicates 12 dB of clipping: all five green segments lit with the first yellow segment lighting occasionally. If a high sensitivity microphone is close-talked the MICROPHONE LEVEL control may be set near minimum. In this case switch in the 20 dB pad and readjust MICROPHONE LEVEL to avoid overdriving the microphone input amplifier. The input level controls should then be left alone and the CLIPPING control used to set the desired amount of clipping. The peak output will now stay approximately the same as the processor is switched in or out.

When the processor is switched out the processing circuitry is still driven and the CLIPPING display indicates, but the CLIPPING control does not affect the output and all the processing is bypassed. In addition, the LINE LEVEL, MICROPHONE LEVEL, and OUTPUT controls operate, and the frequency response is flat so the unit can be used for background music.

The OUTPUT display indicates the peak output on either processed or unprocessed signals; it is not a VU meter. When the processor is switched out the display may indicate slightly lower on the unprocessed signal, and a VU meter in a following amplifier will indicate roughly 8 dB lower because of the drop in average power.

Setting system levels: In order to realize improved intelligibility from the SP-1, levels must be set correctly throughout the system. It is customary with unprocessed speech to allow about 8 dB of headroom in the power amplifier. That is, the level is set so that the amplifier VU meter on speech reads 8 dB below the power amplifier clipping level to allow room for peaks that extend above the meter reading. With a signal processed by SP-1, however, <u>no headroom is needed</u>. The envelope of the processed signal is nearly constant during speech, with no peaks. Thus the VU meter should be set just below the amplifier's peak power capability to make full use of the power available. It is also important not to overdrive the power amplifier since distortion increases rapidly when the constant envelope processed signal exceeds the amplifier's clipping level.

A limiter or compression amplifier should not be used with the SP-1. The processing already gives inherent control of level, and compression will only increase background noise and make any acoustic feedback worse.

Setting the CLIPPING control: In most cases, about 20 dB of clipping is optimum. To set this level, adjust the CLIPPING control so the five green and three yellow segments light on the CLIPPING display and the first red segment flickers occasionally. Tests show that the intelligibility in noise increases rapidly as clipping is increased from 0 to 10 dB and more slowly from 10 to 20 dB. At 20 dB there is only a slight degradation in voice quality, but above 20 dB the quality gets rapidly worse and intelligibility does not improve much. If the microphone is in the sound field of the loudspeaker system acoustic feedback may limit the amount of clipping that can used without oscillation. Background noise picked up by the microphone can be bothersome at high clipping levels. It can be reduced by a close-talking noise-cancelling microphone.

Acoustic feedback: If feedback occurs, switch in the 7 hertz frequency shift. It has no audible effect on voice quality to a listener, but if the talker can hear both his own voice and the shifted signal from the loudspeakers he may notice a 7 hertz beat or flutter that is produced within his own ears. The flutter is not present on the signal itself. The frequency shift is most effective in a non-reverberant space such as an auditorium, where it may permit as much as 10 or 12 dB additional stable gain. The shift does not help much if the microphone is in the direct sound field of a loudspeaker. A directive microphone, particularly one designed for close talking, will allow more clipping before feedback occurs. Oddly enough a noise-cancelling microphone does not help with feedback, although it doesn't hurt either. The microphone's discrimination against a distant source falls off with increasing frequency and is usually 0 at 3 kHz, which is the frequency at which feedback is greatest because of the frequency response of the processor. As gain is increased with the shift switched in, the onset of feedback is sudden, with a distinctive 7-hertz chirping sound. Only a little less gain will give stable operation.

Reverberant spaces: The SP-1 is highly effective against noise, but tests have shown that it neither improves nor degrades intelligibility where the main problem is reverberation.

How does speech processing work? The dynamic range of the phonemes, or basic sounds, in speech is about 30 dB. Most of the information in the signal, at least in non-tonal languages, is carried by the consonants, but the consonants are weaker on average than the vowels. Thus if the noise level is high enough to mask all but the louder phonemes the voice will be unintelligible. Compressing the dynamic range so all parts of the signal make nearly full use of the power amplifier's peak output capability will make a dramatic improvement in intelligibility and apparent loudness without requiring a larger amplifier.

The simplest way to compress dynamic range would be to clip positive and negative peaks of the audio signal drastically and then increase gain so the same peak output was restored. Unfortunately, heavy clipping generates a great deal of distortion, and the distortion products fall (in the frequency domain) within the speech bandwidth, and thus mask other parts of the signal that are important to intelligibility. In the SP-1 the spectrum of the speech signal is translated upwards to the neighborhood of 25 kHz by single-sideband (SSB) modulation, and the SSB signal is heavily clipped. Most of the resulting distortion products fall outside the bandwidth of the SSB signal and are easily removed by a lowpass filter. The signal is then demodulated back to the audio range. (When the 7-hertz frequency shift is selected the modulator and demodulator carriers are offset by 7 hertz. All frequency components of the output are thus shifted upward by 7 hertz.) The frequency response of the stages preceding the SSB modulator is important. A flat response would give a bassy quality with noticeable distortion. The response in the SP-1 is carefully tailored to give maximum benefit from the clipping. The resulting quality is crisp and distinct, although there is some loss of naturalness, of course.

Intelligibility test results: During the development of the military predecessors to the SP-1 extensive tests were done in the laboratory. Effectiveness was measured by a quantity called the Intelligibility Threshold Improvement (ITI). A subject seated in a semi-reverberant chamber was subjected to 90dBA of pink noise. A tape recording of the phonetically balanced test sentence "Joe took father's shoebench out; she was waiting at my lawn." was played over and over while the subject adjusted the voice level so each word except the two weakest words ("she was") could be understood. This condition is defined to be the threshold of intelligibility. The peak level of the voice was measured by observing two repetitions of the sentence on a storage oscilloscope. "Peak level" here means that value that is exceeded only occasionally, rather than the single highest peak.

When this measurement had been made for unprocessed speech using first 20, then 15, 10, 5, and 0 dB of clipping. In each case the subject switched back and forth between the processed and unprocessed signals until satisfied that the intelligibility was the same.

The ITI for each amount of clipping is defined as the difference between the peak levels of the processed and unprocessed signals, with both at the threshold of intelligibility. It is the number of decibels by which processing allows the peak audio power to be reduced without loss of intelligibility.

The graph (Figure 2, right) shows averaged data from nine subjects, male and female, with normal hearing. Note that 20 dB of clipping, which degrades voice quality very little, is equivalent to using 6 1/2 times the audio power (8.1 dB).

Theory of operation: Refer to Figure 1. In the preamplifier microphone and line inputs are amplified and combined with a frequency response tailored to give best intelligibility with clipping. The phasing networks then split the signal into two components with a phase difference of 90° ± 1° from 200 to 5000 hertz. The SSB modulator switches between 0° and 90° audio inputs and switches the gain of an op amp between +1 and -1. Its output is the phase sequence 0°, 270°, 180°, 90°, going through this cycle 25,000 times per second. The result is a lower sideband signal with a suppressed carrier frequency of 25 kHz. Since a switching modulator also has output components near odd harmonics of the carrier, these are removed by a 25 kHz lowpass filter.

Positive and negative peaks of the SSB signal are removed by the clipper, and distortion products above 25 kHz are rejected by another 25 kHz lowpass filter. The signal is shifted back to baseband in a switching demodulator driven at either 25000 or 25007 hertz. A 3400-hertz lowpass filter removes distortion products more than 3400 hertz from the carrier frequency. The carrier generator is an 800 kHz ceramic resonator followed by a counter chain with outputs at 50 kHz and 25 kHz to drive the modulator and demodulator. The 25007 Hz demodulator carrier is produced by a 7 hertz quadrature oscillator and an SSB modulator similar to the audio modulator.

Each display is a full wave peak detector into a special display-driver chip with 10 outputs spaced by 3 dB, driving a 10 segment LED bar graph. The CLIPPING display is driven by the SSB signal just before the clipper; the OUTPUT display is driven by the output audio signal.

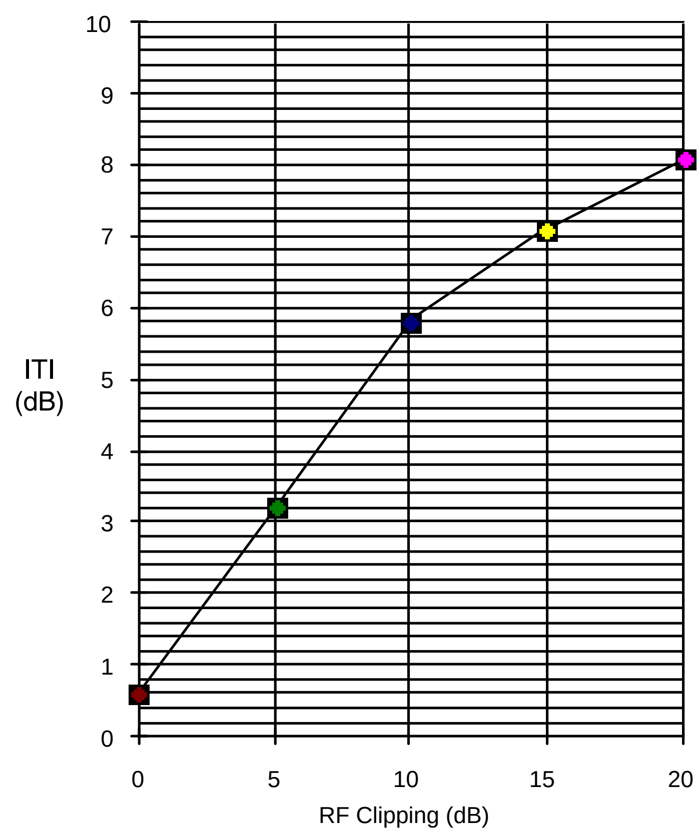
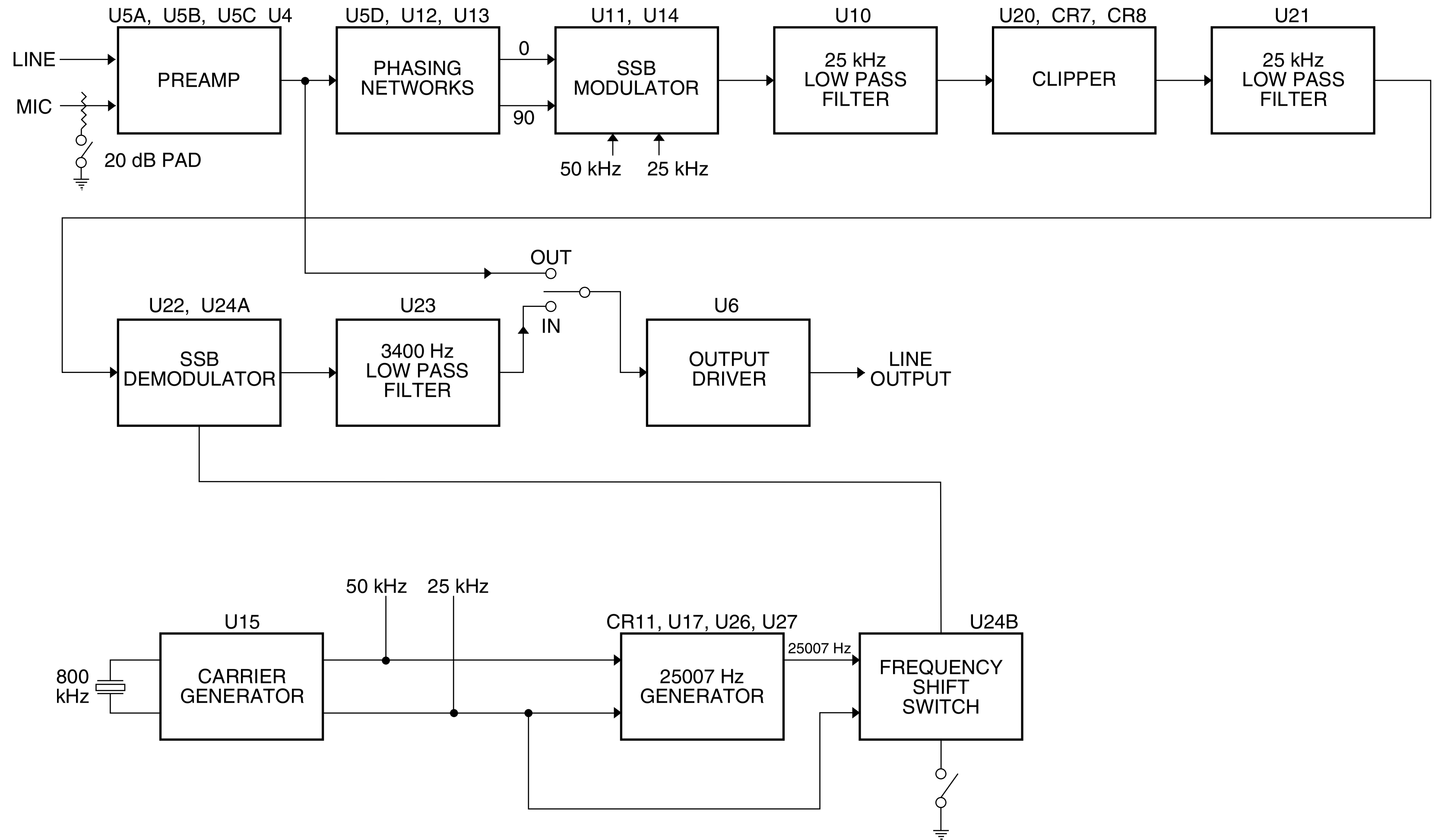


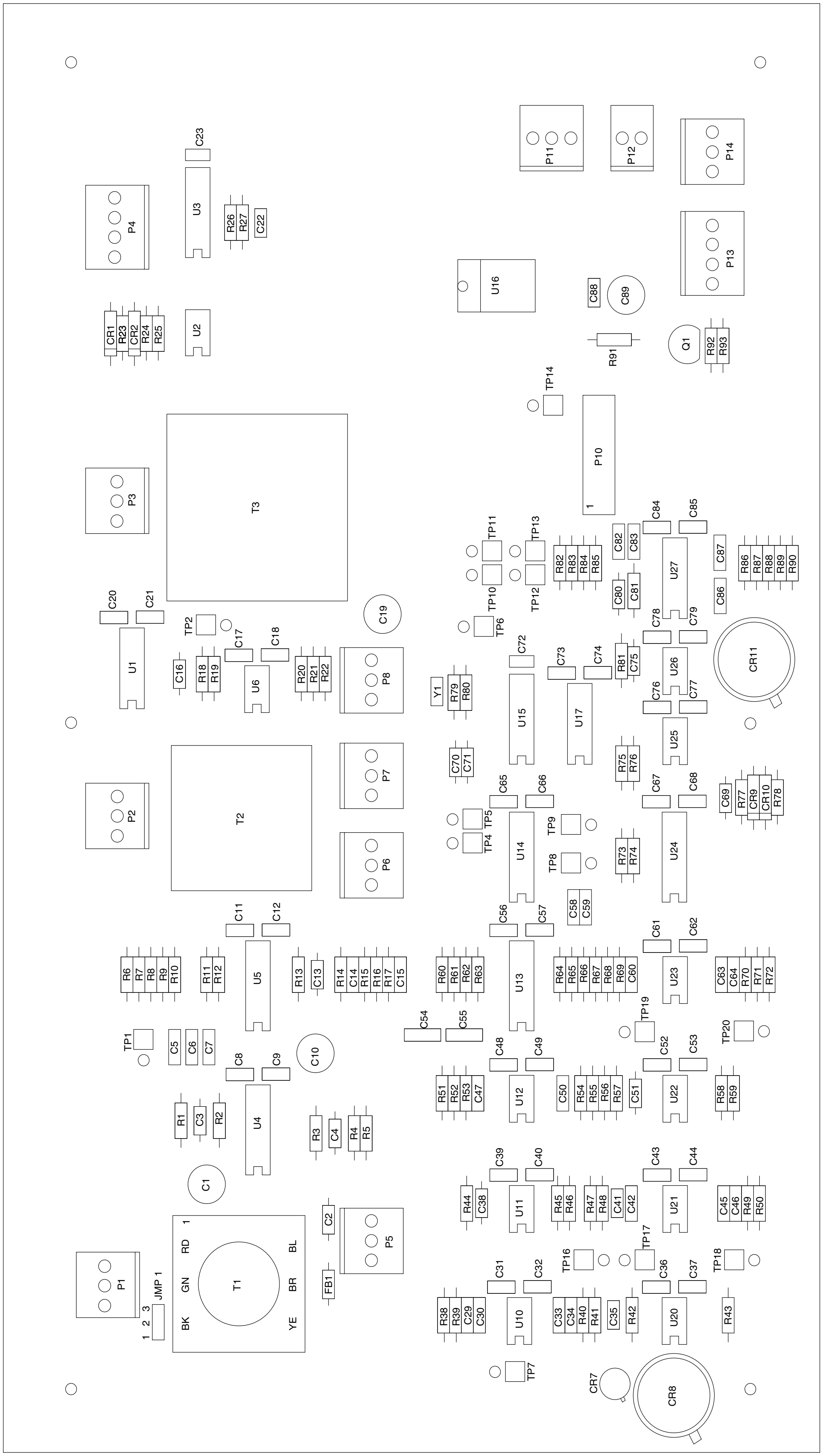
Figure 2



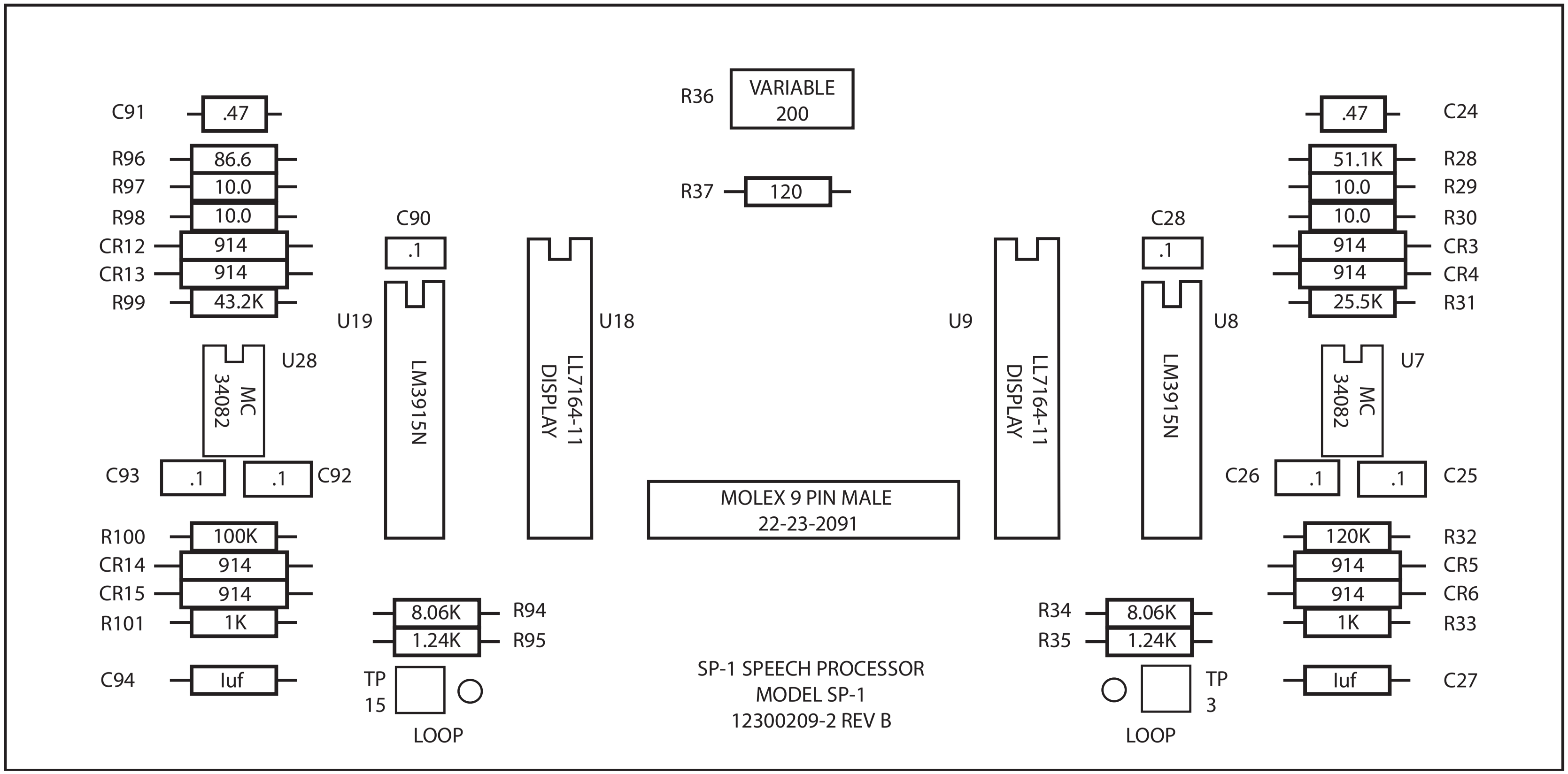
SP-1 SPEECH PROCESSOR
BLOCK DIAGRAM

Figure 1

SP-1 Block Diagram



SP-1 Main Processor



SP-1 DISPLAY BOARD

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