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Speech Processor kit

Introduction

This general purpose kit is intended to increase the 'effectiveness' of microphone audio under difficult conditions - particularly if the TX is QRP. It should be compatible with any make or type of phone transmitter. It has two main sections, the first comprising two adjustable audio amplifiers, the second providing the clipping and filtering to improve the average to peak level of the microphone's signal. The clipping is done after modulation to a higher RF frequency so that the harmonics resulting from the clipping can be easily removed prior to demodulation back to audio. It requires a nominal 12 volt supply of about 40 mAmps. The circuits are on a 50 x 80 mm single sided PCB. The PCB can be mounted at any convenient location. Read all through these notes before building the kit.

Technical Description

The block diagram is shown in Fig. 1 with the full circuit in Fig. 3. (Note that the sequence of numbered test points starts from number four.)

Supply aspects The circuits do not have reverse polarity protection. The nominal 12 volt input is best taken from after any protection circuits of the associated rig. The supply can be from 8 to 15 volts.

Audio amplifiers The microphone input is applied to IC1B which has a fixed voltage gain of 100 and bandwidth of 3 KHz. Its input impedance is 1K which suits most modern mikes. The positive input of both op-amps' is fed from a filtered supply divider so that their outputs are normally at half the supply voltage. The principle mike gain adjusting preset RT1 follows IC1B with the output feeding the unity gain buffer amplifier IC1A. (The gain of this stage can be increased for very low output microphones by reducing R6 down to 22K.) The output of IC1A is applied to a second preset with a much restricted attenuation control range (normally divide by 3 max.) which is actually the Compression control. The correct setting of the first mike gain preset RT1 is when clipping is *just about* to commence on speech peaks with the compression preset RT2 fully anti-clockwise. Thus, if this second *compression* preset RT2 is advanced giving a higher output level, clipping of the speech peaks will occur, resulting a higher average signal level so achieving the desired compression effect. RT2 can be changed to a front panel pot if required. Fig. 4 (top line) shows a typical audio waveform, something like 'Aahh' but somewhat simplified! The lower line of Fig. 4 shows what happens after clipping illustrating that the low level signals are amplified but the high level peaks remain unchanged.

Clipping This cannot be done at audio because the inevitable harmonics caused by clipping low frequency signals (say at 500 Hz) would be within the audio band, so removing them would also remove the rest of the wanted input signal leading to poor speech quality. Instead, the audio signal modulates a 5 MHz carrier so that all the harmonics (at 10 MHz upwards) caused by clipping can be easily removed with a simple 5 MHz low pass filter. The mixer chip IC2 (NE602 or 612) uses its own oscillator section to generate the 5 MHz signal and produces a Double Sideband Suppressed Carrier (DSBSC) signal at its output which carries the basic audio modulation. Fig. 5 shows the waveforms for a single audio input tone (to make it easier to understand) - the lower left diagram showing that tone modulating the 5 MHz carrier, the lower right diagram showing the same signal but with a higher input level so that the peaks of the envelope of the 5 MHz carrier are clipped off. Once the base level of audio amplification is correctly adjusted with RT1, further increases of input signal level occurring with a higher setting of RT2, will cause the excessive peaks to be removed by the diodes D1/2 while not altering the lower levels - so that the average to peak signal ratio is increased achieving compression and improved 'punchiness'. The use of 5 MHz avoids any RX interference problems because all harmonics are outside the HF amateur bands. A buffer stage TR1 is required before feeding the clipped 5 MHz signal to the 330 Ohm low pass filter formed by C16/L1/C17 which removes the harmonics at 10, 15, 20 MHz etc.. Demodulation back to audio is done in the mixer IC3 using the same 5 MHz LO signal from IC2. Although in the RF path, the preset RT3 sets a convenient output audio level for the following rig. The output audio bandwidth is limited by C22 to about 3 KHz to prevent any of the 5 MHz getting into the following rig. The final stage TR2 provides the low impedance output to suit the rig's normal mike input. Either a high or (normal) low level can be selected by the alternative positions for C23. If you wish to be able to switch the circuit out of action, the easiest way is to use a DPCO switch to bypass this circuit completely - because the various presets are normally adjusted for unity gain through this kit no change of the rig's controls will be required.

Building the kit

As you are quite likely to already have some other Walford Electronics kit, the section on general construction advice for that kit also applies here. This kit uses a single sided PCB so there are no top-side solder points. The PCB maybe mounted by drilling holes for bolts in the corners or by soldering the ground track around the outside to any convenient ground plane on the rig. Ideally, the PCB should be located to minimise the length of the mike input lead but this will not be too important if screened lead is used. Normally RT1 should be left as a preset but a pot can be substituted on the front panel for RT2 for a 'processing' control. RT3 must be left as a preset. If you wish to have a processor in-out control add a double pole toggle switch. You should load the PCB before mounting it finally. It will be connected by wires soldered direct to the PCB's larger pads. The parts layout is in Fig. 2

Start construction with the parts whose location is unambiguous and then the supply aspects:-

RT1	1K preset		R3	10K	
RT2	1K preset		C2	10 uF 35v electro	
RT3	1K preset		C5	10 uF 35v electro	
R2	10K				

Connect up your nominal 12 volt supply to the point V+ and any convenient 0 volts/E point. Check the polarity! Switch on and measure the DC voltage on pin 5 position of IC1 with respect to 0 volts - it should be half the actual supply voltage. Switch off and fit the two audio amp stages:-

R1	1K		C1	1 uF 35v electro	
R4	100K		C3	470 pF disc	marked 471
R5	100R		C4	1 uF 35v electro	
R6	100K		C6	33 nF Polyester	
R7	100K		C7	470 pF disc	
R8	1K		C8	1 uF 35v electro	
R9	1K		IC1	TL072	

Arrange to evaluate the audio output on the slider of RT2 by some means, like a high impedance AC voltmeter, scope or the input to a general purpose audio amplifier or Hi-Fi unit. Switch on and measure the DC voltages on pin 1 and 7 of IC1 - both should be half the supply voltage. Fully adance RT1 and RT2. If using a general purpose audio amp, turn its gain down and gingerly do the 'screwdriver/finger hum test' by applying them to first pin 2 of the op-amp, and then more carefully to the circuit input at point I. Both should produce some rough sounds from the AC supply mains pick-up despite the bandwidth being attenuated below 300 Hz! Switch off and fit the two mixer chips:-

R10	1K		R15	1K	
X1	5 MHz crystal		C20	10 nF disc	
C13	10 nF disc	marked 103	C21	10 uF 35v electro	
C11	82 pF cer plate		IC2	NE602/612	
C12	82 pF cer plate		IC3	NE602/612	

Arrange to asses the 5 MHz signal on points 6 & 8. Both should have a DC voltage of roughly 6 to 8 volts on them. If you have a counter or scope, connect either to point 8 - preferably through a divide by 10 probe. Alternatively, drape the antenna lead of a general coverage RX over this kit and tune around at 5 MHz. There should be over 500 millivolts p-p on points 6 and 8. Switch off and fit the other modulator parts:-

C9	1 µF 35v electro		D1	1N4148	
C10	1 µF 35v electro		D2	1N4148	
C14	10 nF disc		R11	330R	
C16	82 pF cer plate		TR1	BS170	

Testing this part is a little harder if you don't have a scope! You need to asses the signal on point 7. The DC voltage here should be about 4 to 6 volts with negligible AC signals under no audio input conditions. Set RT1 midway and RT2 right back. With audio input there should be 5 MHz amplitude modulated signals without any steady carrier - for a single tone audio input it should be like the lower part of Fig. 5, if you have a scope. For a 10 mV p-p single tone input, as you advance RT1 there should be up to about 1.5 volts p-p of modulated signals at 5 MHz which can be observed on the scope or heard on a general coverage RX (with its antenna over this kit) tuned to 5 MHz. Alternatively connect your mike to the input point I. While talking into the mike, gradually advance RT1 till its just below the point where the peak of the signals cease to increase any more. This will be near the optimum setting when in use finally. Switch off and fit the remaining filter and demodulator parts:-

R12	330R		C17	82 pF cer plate	
R13	330R		C18	10 nF disc	
R14	1K		C19	10 nF disc	
R16	100R		C22	33 nF polyester	
R17	1K		C23A	10 µF 35v electro	
R18	100R		L1	10 µH moulded	marked 100J
C15	10 nF disc		TR2	BS170	

Arrange to observe the expected audio output at the point O. Set RT2 fully anti-clockwise and RT3 fully clockwise. Use your general purpose audio amplifier, scope or AC voltmeter. (If this cannot measure down to a few tens of millivolts, change C23 to the B position for an eleven times increase in output.) Switch on and measure the DC voltage on point 9 - it should be 4 to 6 volts with negligible AC without any audio input. Apply your audio source, mike or signal generator to the input point I - initially at a low level and observe the output. It should be a replica of the audio input. Gingerly increase RT1 until the output level ceases to increase. On speech it is quite likely to sound distorted because clipping occurs well before you can sense aurally that the level is not increasing. Carefully adjust RT1 to just below the point where clipping becomes obvious - the audio should not sound distorted at all at this level. Now when you increase the compression preset RT2, the audio will become more 'punchy' as the average levels increase without an increase in the peaks. If you have a scope, comparing the audio input at point 5 to that on the output should show this effect - on point 5 the peaks will be much higher compared to the average signal level. (Note these two points may have different actual signal levels but it is the ratio of peak to average level of each which is different.) As a guide, with RT1 half way, RT2 right back and RT3 right up, a 10 mV p-p input produces about 400 mV p-p at point 9. Switch off and prepare to fit the kit in your rig. It is quite likely that input levels to the rig will be too high if C23 is left in the B position so change it back to the A position.

Connection to the rig

Having decided the best location for the PCB it is probably best to make the lead connections to this kit (and maybe the other ends also) before mounting it. They are:-

Input — From the mike socket to point I. (Use screened lead if over 50 mm, earthed at both ends.)
Wire from mike earth pin/socket to any E point on this kit's PCB.

Output — From point O to the rig's mike input point. (Use screened lead if over 100 mm, earthed at both ends.)
Normally no extra earth lead will be required - if in doubt connect any E to rig 0 volt.

Supplies — Positive from a suitably protected nominal 12 volt supply to point V+ on this kit.
Negative, from the main rig 0 volts or ground plane to any convenient E point.

(If you wish to be able to switch the processor in/out, arrange to mount a DPCO toggle switch on your control panel. Route the rig's mike socket to the common of one pole and the rig's mike input to the other common. Connect the processor input to one side contact of the mike input pole and the processor output to the same side of the other switch pole. Put a link across the other side contact pair for the bypass position.)

Now mount the PCB.

Setting up

Assuming your rig has been properly adjusted for normal direct speech input from the same microphone, you will need to now adjust RT1 and 3 for normal operation without compression. Arrange to monitor the rig's RF output into a dummy load preferably using a peak reading RF volt or power meter, or monitor it on another RX. Set RT2 fully anti-clockwise. Ease the setting of RT1 back a little from that found earlier. Put RT3 to about a quarter up as a starting point. Turn on the rig and go to transmit - while speaking normally advance RT1 until RF peak output ceases to increase. This maybe due to the rig output stage limiting maximum output or it maybe due to the speech processor kit. If the rig appears to be producing full normal output then the limit is due to the rig - in which case reduce RT3 so that output is about half maximum. It is important at this stage that the rig does not limit output. Now increase RT1 again till the output just ceases to increase. Check again if the limiting is due to the rig or this kit. If necessary further reduce RT3 and readjust RT1. If you listen on another rig the audio should sound quite normal - if it sounds more 'punchy' then its likely that RT1 is too high - reduce it a little.

Having found where this onset of clipping appears to take effect, listen carefully while now increasing RT2. This should progressively increase the 'punchiness' of the signal but without increasing the maximum RF output. (This is more easily checked if you have the suggested peak reading RF instrument - voltmeter/power meter or a scope.) Speech quality should remain quite acceptable but will not be as good as without the compression. If necessary re-adjust RT1 just a little to find the optimum spot where the processing effect is just about to become noticeable when listening carefully. This will be the normal position of RT1 so that under difficult conditions, when more compression is needed, all you do is increase RT2. All these adjustments thus far should have been done with the rig producing less than full output. You can now increase RT3 to bring the RF output level of the transmitter up to full power. It is important to do this adjustment last otherwise there is a good chance you will overdrive the rig's output stage causing splattering and interference to others.

Finally, do some experiments with a local station who has good signals to and from you. Get his critical comments on sound quality as you adjust RT2. If he says it sounds bad, reduce RT3 to make certain the rig output is not limiting and then readjust RT1 (with RT2 right back) till quality recovers. Note that it is quite difficult to do these evaluations on air so you might need to play around with RT1 for a while. Its easy to see low level clipping on a scope but quite hard to hear - especially on air! It is possible that for your mike/rig combination, the full increase in gain available from RT2 may cause excessive clipping with an unacceptable reduction in speech quality - in this case removing R8 will decrease the effective range of RT2. Note that since this alters the overall gain with RT2 turned right back, you will need to readjust RT1 as above. Having found the best setting for RT1 and RT3, they are best left alone and not adjusted on air because small alterations may make quite big changes. Use RT2 if you want a compression control!

Do be aware that when the audio is compressed it will cause a higher dissipation in the transmitter's output stage because the average RF output is higher. Ensure adequate ventilation!

That's all! I must record my thanks to Andy Howgate G7WHM for suggesting that this kit would be a useful addition to the Walford Electronics range.

Tim Walford G3PCJ Dec. 11th 2000

Parts list for the Speech Processor

Resistors			Capacitors		
3	100R	R5, 16, 18	4	82 pF cer plate	C11, 12, 16, 17
3	330R	R11, 12, 13	2	470 pF disc - 471	C3, 7
7	1K	R1, 8, 9, 10, 14,15, 17	6	10 nF disc - 103	C13, 14, 15, 18, 19, 20
2	10K	R2, 3	2	33 nF polyester	C6, 22
3	100K	R4, 6, 7	5	1 uF 35v electro	C1, 4, 8, 9, 10
			4	10 uF 35v electro	C2, 5, 21, 23
3	1K Preset	RT1, 2, 3			
Semiconductors			Miscellaneous		
2	1N4148	D1, 2	1	10 uH moulded - 100J	L1
2	BS170	TR1, 2	1	5 MHz crystal	X1
1	TL072	IC1	200	mm multi-core cable	
2	NE602/612	IC2, 3	1	Speech Proc SS PCB	Issue 2
			TRN Walford G3PCJ		Dec. 5th 2000

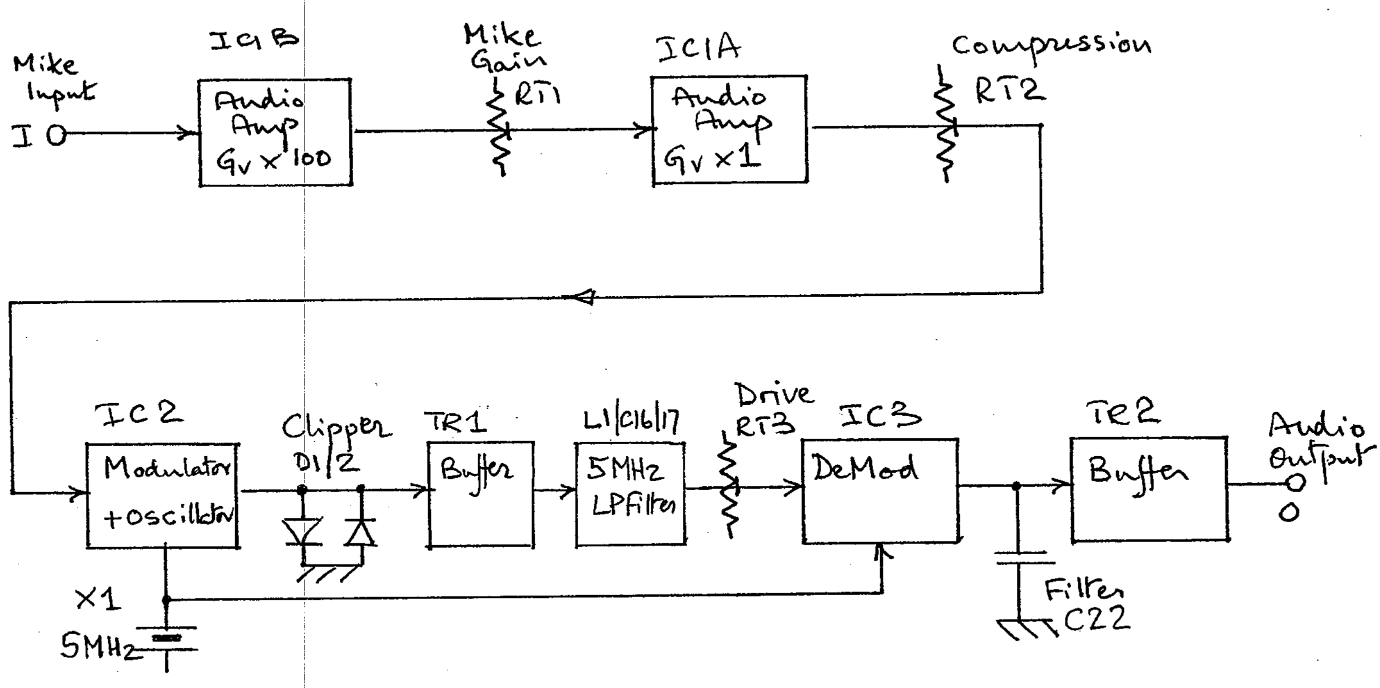

FIG 1 : BLOCK DIAGRAM OF SPEECH PROCESSOR

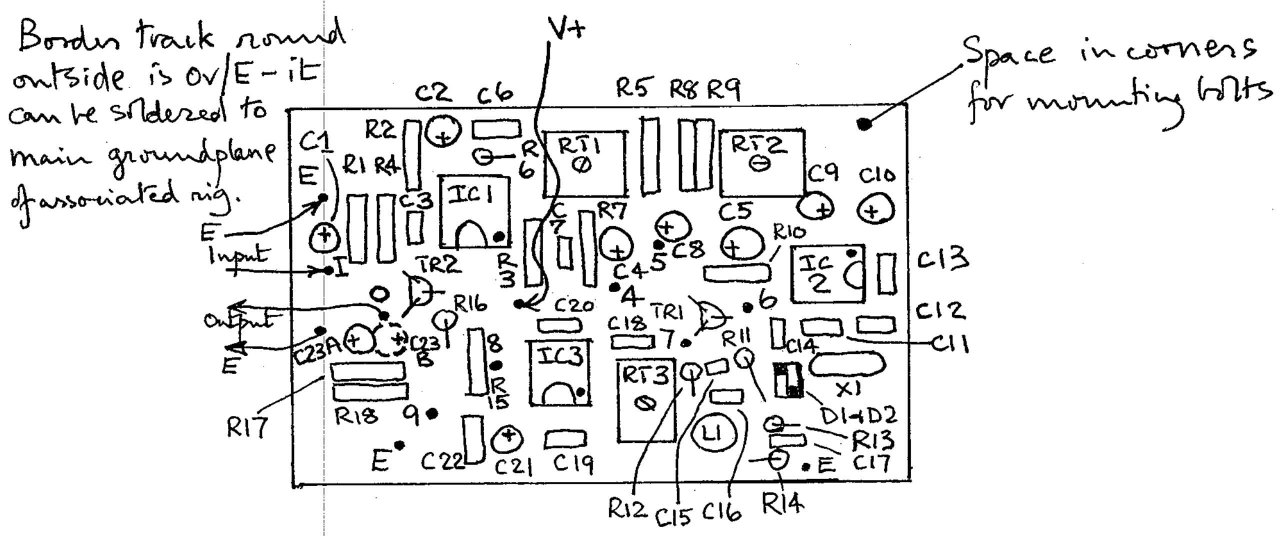

FIG 2 : PARTS LAYOUT OF SPEECH PROCESSOR

Top view of PCB - component side.

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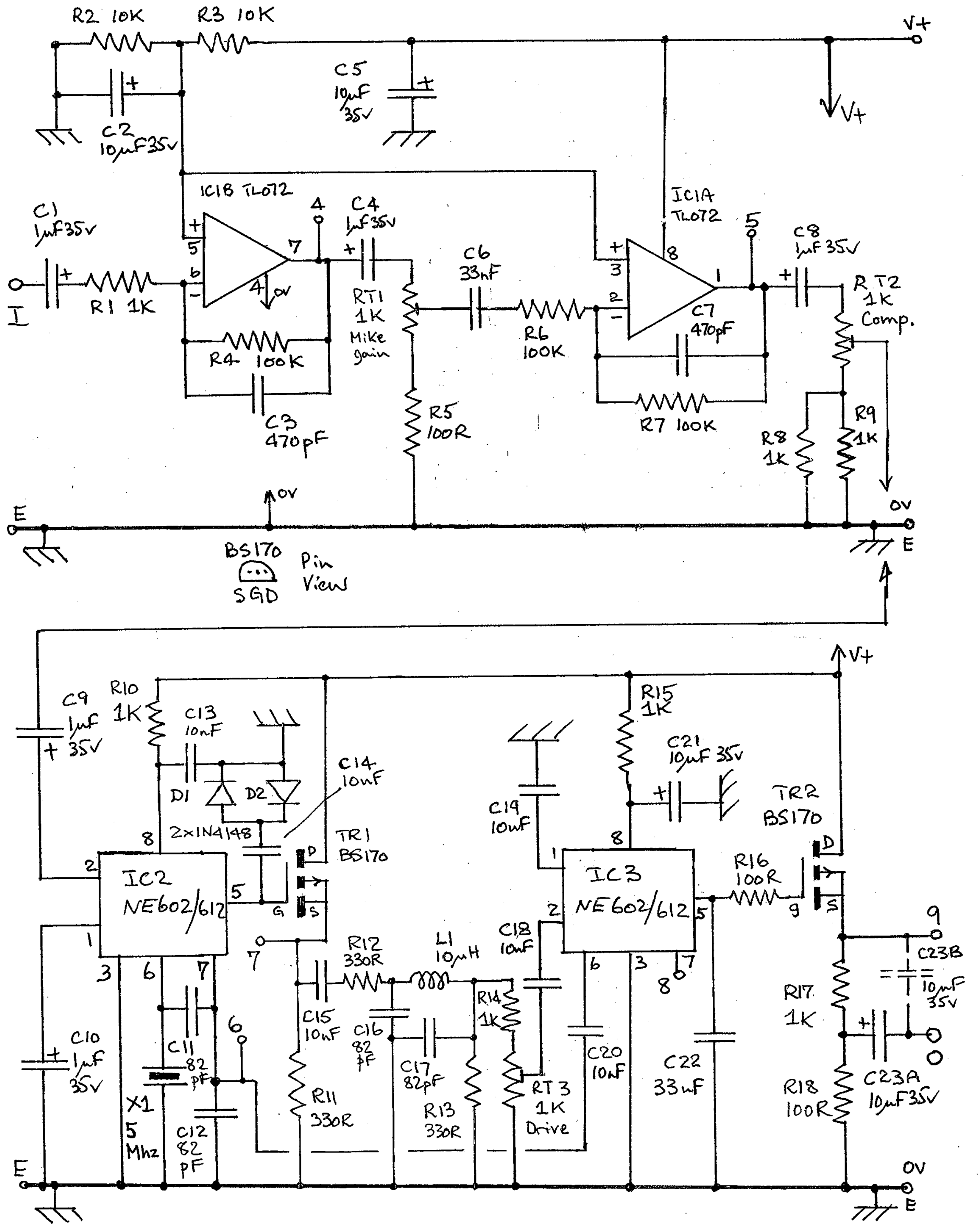

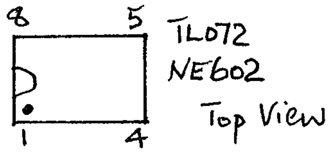

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FIG 3 : SPEECH PROCESSOR CIRCUIT

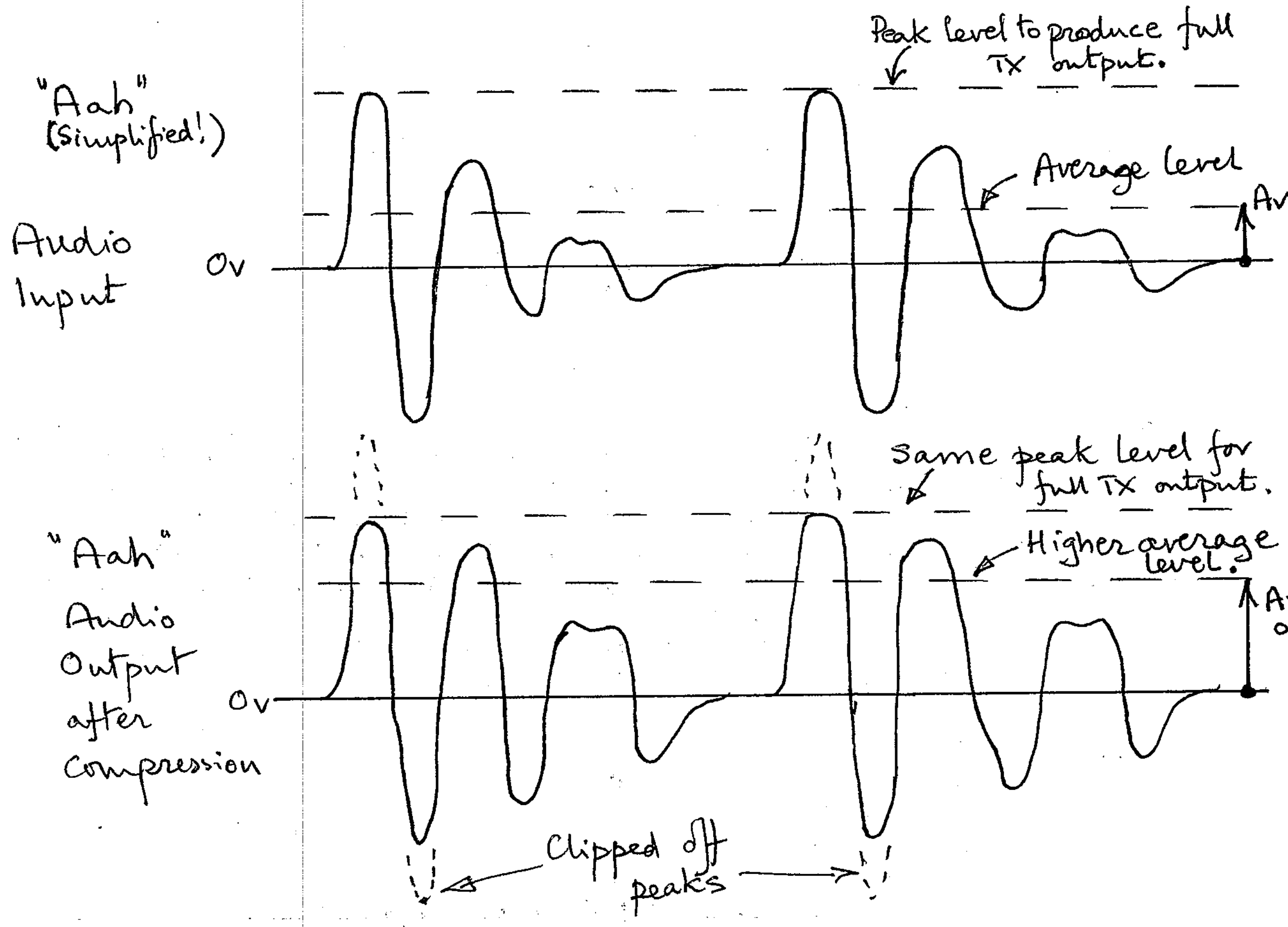

FIG 4 : SPEECH PROCESSING WAVEFORMS

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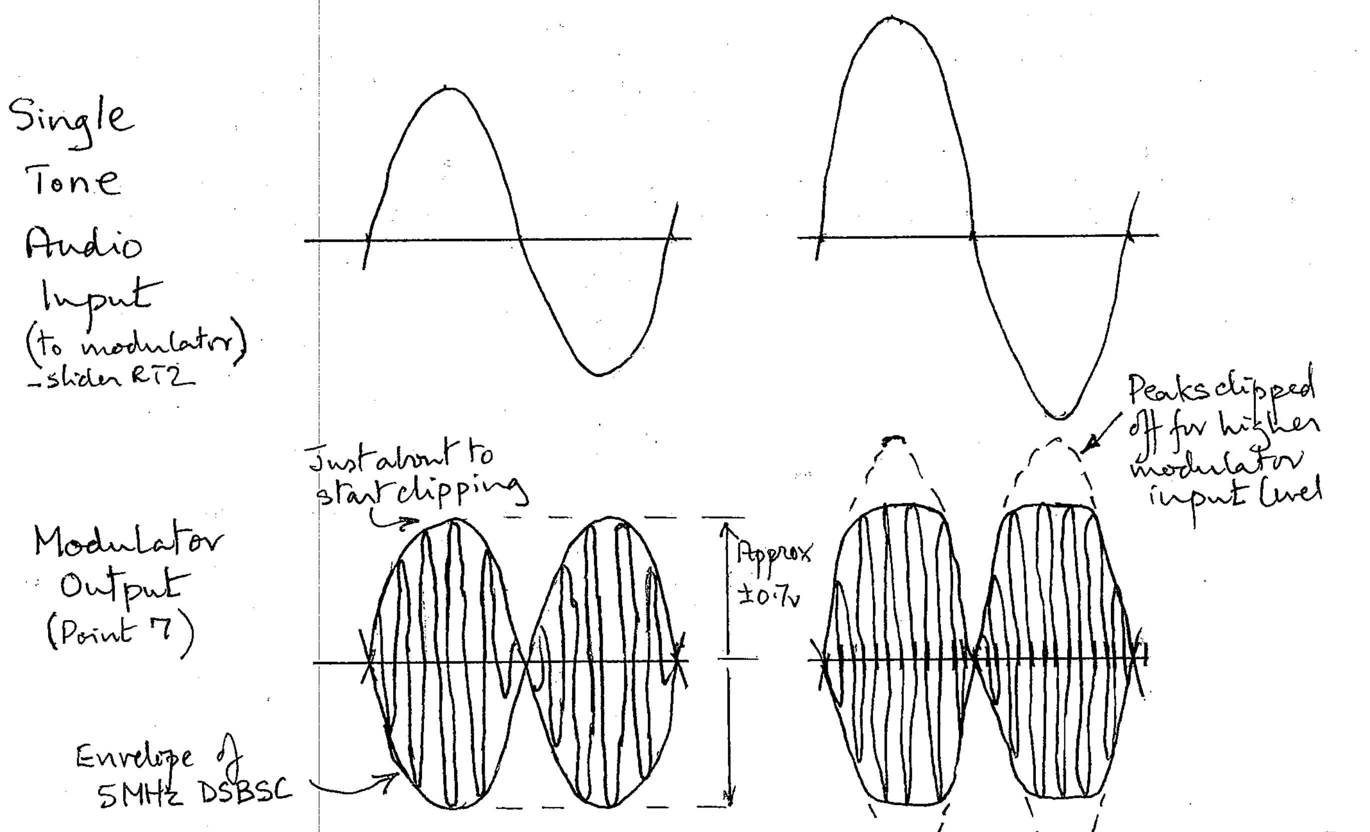

FIG 5: MODULATOR WAVEFORMS