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Noise Reduction for Armature Radio Voice Data and CW Operation using Audio Noise Filters

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ABSTRACT

The aim of our project has the following objectives: To integrate the knowledge and skills acquired from major courses taken so far. To develop allows power FM transmitter to be used in specialized applications for local area entertainment purpose. Provide a reference for further study in a similar streams having ambition to deal with low power FM transmitter design. Random/Tone Noise Reduction, The noise reduction functions of the DSP-9+ operate by examining a characteristic of signals and noise called correlation, and dynamically filtering out the undesired signals and noise. The degree of correlation is relative. Random noise such as white noise or static is uncorrelated. Speech is moderately correlated. Repetitive noise such as a heterodyne is highly correlated. TheDSP-9+ measures correlation and filters out signals and noise that are outside its correlation Thresholds. There is little degradation of the desired speech signal. The amount of noise Reduction varies according to the correlation characteristics of the noise. Typical noise reduction Ranges from 5 dB to 20 dB for random noise and up to 50 dB for heterodynes.

1. INTRODUCTION

Communication system engineers attempt to design communication system that transmits information at a higher rate with a higher performance, using the minimum amount of transmitted power and band width. The purpose of any communication system is to transmit information signals from a source located at one point in space to the user/destination located at another point. The originating in put is frequently referred to as the source, whereas the terminating /end is frequently referred to as the sink. If the message is understandable, then the information has been converted from the source to the destination. Mostly, the message produced by the source is not electrical in nature. But to carry them over an electrical system the message must be converted to an electrical signal in the same manner at receiver. The electrical signal must be reconverted in to an appropriate form. A transducer performs these Functions. Thus, an input transducer used to convert the message generated by the source in to time varying electrical signal called the message signal. Basically, communication consists of three major parts The DSP-9+ is an audio noise filter for amateur radio voice, data and CW operation.

The DSP- 9+ filters and reduces noise and interference to Digital signal processing technology to implement algorithms that perform four basic functions:1) Random noise reduction, 2) Adaptive multi-tone notch filtering (Tone noise reduction), 3) Band pass filtering, and 4) RTTY demodulation. Random/Tone Noise Reduction The noise reduction functions of the DSP-9+ operate by examining a characteristic of signals and noise called *correlation*, and dynamically filtering out the undesired signals and noise. The Degree of correlation is relative. Random noise such as white noise or static is uncorrelated. Speech is moderately correlated. Repetitive noise such as a heterodyne is highly correlated. The DSP-9+ measures correlation and filters out signals and noise that are outside its correlation Thresholds. There is little degradation of the desired speech signal. The amount of noise Reduction varies according to the correlation characteristics of the noise. Typical noise reduction Ranges from 5 dB to 20 dB for random noise and up to 50 dB for heterodynes. Band pass Filters the DSP-9+ has band pass filters that are used in voice, data and CW modes. In a typical Example of a voice mode application, a band pass filter can improve a signal with a poor signal to- Noise ratio. A band pass filter removes the high and low audio frequency components that do not contribute significantly to the speech intelligibility, thus improving signal quality. Another Common voice mode example is the improvement of a SSB signal corrupted by adjacent channel Interference (QRM). The steep skirts of the band pass filters allow the interference to be eliminated with minimal effect on the desired signal. In the voice mode, two front panel push Buttons select one of three voice band pass filter bandwidths from two sets of filters. An internal Jumper behind the back panel selects the filter set, either 1.6, 2.0, or 2.4 kHz. Or 1.8, 2.4, and3.1 kHz. CW signals require band pass filters with steep skirts and linear phase response. Linear phase Response maximizes the usable signaling rate for a given bandwidth and minimizes ringing often heard on extremely sharp filters. The DSP-9+ has 18 different CW filters with skirts so steep That a signal literally falls off the edge of the pass band as you tune through a CW signal. The Bandwidths of the CW filters are 500, 200 or 100 Hz. A front panel push-button selects either of Two CW band pass filter center frequencies chosen from a set of 400, 500, 600 and 800 Hz. Internal jumpers behind the back panel program the two choices. The jumpers also allow the Choice of a special set of filters for the Collins Radio KWM-2. The Collins filters have center Frequencies of 1350 and 1500 Hz. The narrow filters are useful for trying to dig out extremely Weak signals from the noise and QRM. The wider filters allow easy tuning and listening to Multiple CW signals simultaneously. Data signals also require band pass filters with steep skirts and linear phase response. There is an Optimum bandwidth for each signaling rate and modulation type. Any wider bandwidth than Necessary will increase the bit error rate of the data communication link by allowing more noise into the demodulator. The DSP-9+ has four data band pass filters for five popular data types RTTY, AMTOR, PACTOR, G-TOR and HF Packet. (G-TOR is registered trademark of Kantronics Inc.) There is a choice of the center frequency for the data filters, since different Mark-space frequencies are used in different parts of the world. Internal jumpers behind the back panel select one of the four center frequencies.

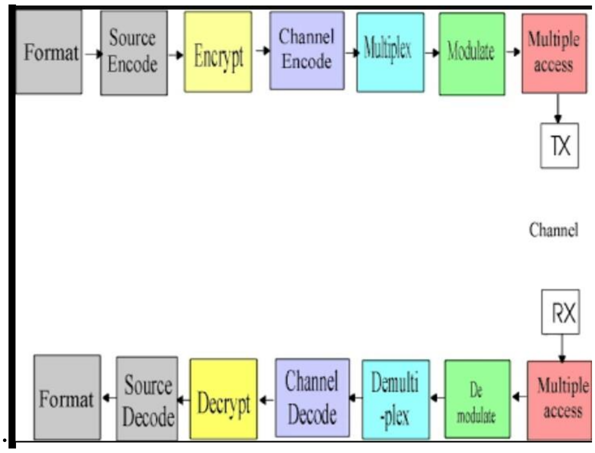

Fig.1 General Block diagram for communication system

Transmitter: The sub-system that takes the information signal and processes it prior transmission. The transmitter modulates the information onto a carrier signal, amplifies the signal and broadcasts it over the channel. That means the main purpose of transmitter is to modify the message signal in to a form suitable for transition over the channel.

It involves modulation and amplification. Channel: The medium which transports the modulated signal to the receiver. Air acts as the channel for broadcasts like radio. Receiver: The sub-system that takes in the transmitted signal from the channel and processes it to retrieve the information signal. The receiver must be able to discriminate the signal from other signals which may use the same channel (called tuning), amplify the signal for processing and demodulate to retrieve the information. It also then processes the information for reception (for example, broadcast on a loudspeaker) in other words the main purpose of the receiver is to reproduce version of transmitted signal after propagation through the channel, this accomplished by using a process of demodulation and amplification. Modulation is employed in order to: More efficiently launch the radiated wave in to space .Permit multiplexing to improve the modulated signal to noise ratio. For efficient launching or reception of an electromagnetic wave or to obtain what is commonly called matching, the radiating or receiving device (antenna) must be a significant portion of wavelength in size. The larger the antenna in the wave length, the greater the antenna's radiation resistance. The antenna resistance can thus closely approximately the driving generator impedance and associated transmission line. The wave length of an electromagnetic wave in free space related to the velocity of light by the following relation = f where: 3 c = the speed of light = 300,000 km/s or 3.0 x 108 m/s= the wavelength of light, usually measured in meter = the frequency at which light waves pass, measured in units of per seconds (1/s).In communication system the information system, maybe transmitted by itself over the medium or may be used to modulate a carrier for transmission over a long distance. The former is a baseband communication, while the latter is band pass (modulated signal).The goal of communication system engineer to design systems that provide high quality service for the maximum number user with the smallest cost and least usage of limited resources. There sources to be conserved include hard ware for generating, transmitting and receiving information signal, the channel band width and the transmitter power.

1.1 BACKGROUND OF FM BROADCASTING STATION

The comparatively low cost of equipment for an FM broadcasting station, resulted in rapid growth in the years following World War II. Within three years after the close of the war, 600licensed FM stations were broad casting in the United States and by the end of the 1980s there were over 4,000. Similar trends have occurred in Britain and other countries. Because of crowding in the AM broad cast band and the inability of standard AM receiver to eliminate noise, the tonal fidelity of standard stations is purposely limited. FM does not have drawbacks and therefore cane used to transmit music, reproducing the original performance with a degree of fidelity that cannot be reached on AM bands. FM stereophonic broad casting has drawn increasing numbers of listeners to popular as well as classical music, so that commercial FM stations draw higher audience ratings than AM stations.

1.2 MODULATION

The information signal can rarely be transmitted as is, it must be processed. In order to use electromagnetic transmission, it must first be converted from audio into an electric signal. The conversion is accomplished by a transducer. After conversion it is used to modulate a carrier signal. A carrier signal is used for two reasons: To reduce the wavelength for efficient transmission and reception (the optimum antenna size is½ or ¼ of a wavelength).To allow simultaneous use of the same channel, called multiplexing. Each unique signal can be assigned a different carrier frequency (like radio stations) and still share the same channel. The phone company actually invented modulation to allow phone conversations to be transmitted over common lines. The process of modulation means to systematically use the information signal (what you want to transmit) to vary some parameter of the carrier signal. The carrier signal is usually just a simple, single-frequency sinusoid (varies in time like a sine wave).The basic sine wave goes like V (t) = Vo sin (2 πf t +ϕ) where the parameters are defined below: V (t) the voltage of the signal as a function of time.Vo the amplitude of the signal (represents the maximum value achieved each cycle) the frequency of oscillation, the number of cycles per second ϕ the phase of the signal, representing the starting point of the cycle. Frequency Modulation Frequency can be defined as the rate of change of phase of a signal. In this type of modulation, information is transferred through a carrier by varying its instantaneous frequency. (t)We have replaced the carrier frequency term, with a time-varying frequency. We have also introduced a new term: Δf, the peak frequency deviation. In this form, you should be able to see that the carrier frequency term. The performance of FM measured by the following parameters as shown below .Bandwidth As we have already shown, the bandwidth of a FM signal may be predicted BW = 2 (β + 1) f m Where β is the modulation index and f m is the maximum modulating frequency used.FM radio has a significantly larger bandwidth than AM radio, but the FM radio band is also larger. The combination keeps the number of available channels about the same. The bandwidth of an FM signal has a more complicated dependency than in the AM case (recall, the bandwidth of AM signals depend only on the maximum modulation frequency). In FM, both the modulation index and the modulating frequency affect the bandwidth. As the information is made stronger, the bandwidth also grows.

1.3 EFFICIENCY

The efficiency of a signal is the power in the side-bands as a fraction of the total. In FM signals, because of the considerable side-bands produced, the efficiency is generally high. Recall that conventional AM is limited to about 33 % efficiency to prevent distortion in the receiver when the modulation index was greater than 1. FM has no analogous problem. The side-band structure is fairly complicated, but it is safe to say that the efficiency is generally improved by making the modulation index larger (as it should be). But if you make the modulation index larger, so make the bandwidth larger (unlike AM) which has its disadvantages. As is typical in engineering, a compromise between efficiency and performance is struck. The modulation index is normally limited to a value between 1 and 5, depending on the application. Noise FM systems are far better at rejecting noise than AM systems. Noise generally is spread uniformly across the spectrum (the so-called white noise, meaning wide spectrum). The amplitude of the noise varies randomly at these frequencies. The change in amplitude can actually modulate the signal and be picked up in the AM system. As a result, AM systems are very sensitive to random noise. An example might be ignition system noise in your car. Special filters need to be installed to keep the interference out of your car radio.FM systems are inherently immune to random noise. In order for the noise to interfere, it would have to modulate the frequency somehow. But the noise is distributed uniformly in frequency and varies mostly in amplitude. As a result, there is virtually no interference picked up in the FM receiver. FM is sometimes called "static free," referring to its superior immunity to random noise.

1.4 FREQUENCY MODULATION ADVANTAGES AND DISADVANTAGES

FM is widely used because of the many advantages of frequency modulation. Although, in the early days of radio communications, these were not exploited because of a lack of understand of how to benefit from FM, once these were understood, its use grew. There are many advantages of FM, but also some disadvantages, and as a result it is suitable for many applications, but other modes may be more suited to other applications. An understanding of the disadvantages and advantages of FM will enable the choice of the best modulation format to be made.

1.5 ADVANTAGES OF FREQUENCY MODULATION

There are many advantages to the use of frequency modulation. These have meant that it has been widely used for many years, and will remain in use for many years.

Resilient to noise:

One of the main advantages of frequency modulation that has been utilized the broadcasting industry is the reduction in noise. As most noise is amplitude based, this canbe removed by running the signal through a limiter so that only frequency variations appear. This's provided that the signal level is sufficiently high to allow the signal to be limited.

Resilient to signal strength variations:

In the same way that amplitude noise can be removed. So too can any signal variations. This means that one of the advantages of frequency modulations that it does not suffer audio amplitude variations as the signal level varies, and it makes FM ideal for use in mobile applications where signal levels constantly vary. This is provided that the signal level is sufficiently high to allow the signal to be limited. Does not require linear amplifiers in the transmitter: As only frequency changes are required to be carried, any amplifiers in the transmitter do not need to be linear. Enables greater efficiency than many other modes: The use of non-linear amplifiers, e.g. class C, etc. means that transmitter efficiency levels will be higher - linear amplifiers are inherently inefficient. RTTY Demodulator the DSP-9+ has a special data function for RTTY only. After passing through the optimized RTTY band pass filter, a precision DSP-based FSK detector in the DSP-9+ demodulates the Noisy incoming RTTY tones and uses the recovered digital data to drive a precision DSP-based AFSK generator. This remodulation process takes place entirely in the DSP-9+. The precise clean tones from the RTTY AFSK remodulator can feed any analog multimode controller or TU via the DSP-9+ audio output. Many analog RTTY demodulators have difficulty with noisy Signals of varying amplitude, but virtually all of them can adequately demodulate the precise DSP AFSK generator output. The RTTY push-button selects either the remodulator or the RTTY filters only. Automatic Gain Control The DSP-9+ has switch-selectable automatic gain control to optimize the signal levels for best filter performance and to enhance listening by minimizing audible signal level variation. Self-Test The DSP-9+ has a self-test mode for digital and analog circuitry, push-button switches, back panel jumpers, LED indicators and connectors. The self-test mode not only verifies the Operation of the DSP-9+, but also aids in verifying the proper installation of the DSP-9+.

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2. SPECIFICATION

AUDIO INPUT

Impedance 2 ohms or 22 Ohms, jumper selectable

AUDIO OUTPUT

Speaker output power 1.6 watts into 8 ohms at 13.8 VDC

2.5 watts into 4 ohms at 13.8 VDC

Line output -6 dB, referenced to input level, into 10K ohms. Not controlled by gain control

Distortion less than 1% at rated output

DATA FILTERS Frequency range Attenuation Type Delay

Band pass - RTTY - 3 dB Bandwidth = 260 Hz 45 dB at 60 Hz Composite 27 msec max

AMTOR - 3 dB Bandwidth = 340 Hz outside the pass band FIR

PACTOR - 3 dB Bandwidth = 440 Hz Linear phase

HF Packet - 3 dB Bandwidth = 540 Hz

All data modes Center frequencies = 2210, 2125, 1700, and 1360 Hz. jumper selectable

Note: RTTY, AMTOR and PACTOR filters have peaks at the mark and space frequencies with a notch at the center frequency.

VOICE FILTERS

Frequency range Attenuation Type Delay

Random Noise Reduction entire freq. range of Up to 20 dB, varies with Adaptive 10 msec max

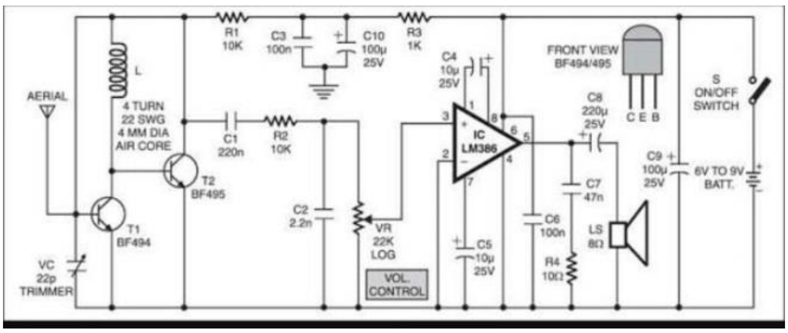
Selected band pass filter noise characteristics

Tone Noise Reduction entire freq. range of Up to 50 dB, varies with Adaptive 10 sec max (multiple automatic notch) selected band pass filter noise characteristics Band pass - Voice 300 Hz - 3.4 kHz, 300 Hz - 2.7 kHz, 60 dB at 180 Hz FIR Linear 10 msec max and 300 Hz - 2.1 kHz, outside the pass band phase or (jumper selectable) 300 Hz - 2.7 kHz, 300 Hz - 2.3 kHz, and 300 Hz - 1.9 kHz

Note: The random noise reduction, tone notch and voice band pass filters can operate simultaneously.

CW FILTERS

Band pass - CW Bandwidth = 100 Hz, 60 dB at 50 Hz FIR Linear 30 msec max 200 Hz and 500 Hz. outside the pass band phase Center freq. = 400, 500, 600, 800 Hz. (select any two - internal jumper), Or Collins KWM-2 mode with center freq. = 1350 and 1500 Hz. Random Noise Reduction entire freq. range of Up to 20 dB, varies with Adaptive 10 msec max selected band pass filter noise characteristics Note: The random noise reduction and CW band pass filters can operate simultaneously.



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AGC

Voice mode 36 dB

CW and Data Modes 18 dB

SIGNAL PROCESSING

A-D/D-A Converter 16 bit linear, sigma-delta conversion Signal Processor 16 bit, 77 ns Analog Devices ADSP-2105

DIMENSIONS

Size 6.0 in. wide x 6.0 in. deep x 1.75 in. high (153 mm wide x 153 mm deep x 45 mm high)

Weight 2 lb. (0.9 Kg.)

POWER 12-16 VDC @ 1A

3. INSTALLATION

To install a DSP-9+ in a station, an operator must provide power to the DSP-9+, make audio Input and output connections to the DSP-9+, and make Push-To-Talk (PTT) connections to the DSP-9+.Receiver/Transceiver Speaker Output Station Power Supply 13.8 Vdc12-16 Vdc Audio Input DSP-9+ Speaker Speaker Output Line Multimode Output Controller PTT Output PTT Input

Power Supply

The DSP-9+ requires a power source of 12 to 16 Volts dc. At 1.0 Amperes. The center pin of The power connector is POSITIVE (+).

Acceptable power sources include:

13.8 volt dc. Regulated external transceiver power supply (recommended power source for the DSP-9+ because it is better regulated than most plug-in wall outlet supplies). Radio Shack 273-1653 12 V.d.c. @ 1 Ampere plug-in wall supply (use the green-tipped adapter Supplied with the Radio Shack unit). Switching power supplies are generally not recommended. Connecting Cables Shielded coaxial cables with RCA phone connectors should be used to minimize the possibility of RF interference to the DSP-9+. Time wave recommends coaxial video cables with metal. The audio input of the DSP-9+ is an RCA phono connector on the rear panel of the DSP-9+. Matching the output level of the radio to the input level of the DSP-9+ is necessary to take maximum advantage of the wide dynamic range of the DSP-9+. The best way to make these levels match is to use an adjustable audio output of the radio (typically the speaker output) as the input to the DSP-9+. After connecting the DSP-9+ to the radio, follow this simple procedure to match the audio levels. First, tune the radio to a strong signal after setting the radio output level Gain control to a convenient midrange position. Then, adjust the output level control on the radio So the Overload indicator LED on the front panel of the DSP-9+ rarely flashes and the Normal indicator LED always flashes with the normal audio input levels. Proper adjustment ensures optimum signal-to-noise ratio and minimum distortion. Adjust the radio output level only to maintain the proper input level to the DSP-9+. Use the Gain control on the DSP-9+ to control the listening volume. The factory default input impedance of the DSP-9+ is 22 ohms. This impedance is appropriate for most radios when driven by the speaker output of the radio. Optionally, configure the DSP- 9+ for a high input impedance by removing the shorting jumper in position 1. Remove the back

Bezel and the back panel of the DSP-9+ to access this jumper. Refer to the Back Panel Jumper Function Table on page 12 for details on the jumper settings.

Audio Output

The DSP-9+ has three audio outputs:

1) On the lower left hand corner of the DSP-9+ front panel is a 3.5 mm headphone jack
Connected for stereo headphones. Use of mono headphones requires a monaural-to-stereo
Adapter (see the appendix on page 20 for details). Direct connection of mono headphones
Will short the DSP-9+ audio power amplifier and may damage the DSP-9+. The DSP-9+
Speaker output is muted when a headphone plug is inserted.

2) The Speaker Output RCA phono jack on the rear panel of the DSP-9+ provides adequate Output to drive a 4 or 8 Ohm speaker. The front panel audio gain control adjusts the audio level From this output. The maximum output power is approximately 2.5 watts into a 4 Ohm speaker, or 1.6 watts into an 8 Ohm speaker. 3) The Line Output RCA phono jack on the rear panel of the DSP-9+ provides adequate output Power to drive a 600 Ohm or greater load. The front panel audio gain control does not adjust the audio level from this output. The output level is 6 dB below the audio input level to the DSP-9+ when driving a 10 kOhm or greater load. When the DSP-9+ power is switched off, the Line Output is attenuated 6 dB in level if it is driven from a low impedance source such as a receiver speaker output. PTTI Input the Push-To-Talk Input electronically bypasses the DSP-9+ in the CW and data modes, and mutes the DSP-9+ in the voice mode. Use the PTTI bypass in the CW mode to hear a fixed frequency side tone which may be different from the frequency of the selected CW band pass filter.

Use the PTTI bypass in the voice mode to prevent unwanted transmit audio from the transceiver From causing audible interference. Many transceivers do not mute their audio outputs completely during transmit. The extra gain from the DSP-9+ with AGC on (up to 36 dB) makes the transmit audio audible and may even cause oscillation from feedback to the microphone. A contact closure operates the PTTI circuit. No external power is required. The return (shield) side of the PTTI jack is connected to the DSP-9+ circuit and chassis ground. Some linear amplifiers have high voltage supplies for their transmit-receive relays. If a transceiver PTT line is used to drive both the DSP-9+ and a linear amplifier, an isolation relay and isolation diode may be required to prevent damage to the DSP-9+ (and any other solid state equipment connected to the PTT line). Internal Jumpers Some operating modes of the DSP-9+ require removal of the back panel to change settings of Internal jumpers. Details of jumper functions are described in the Operation and Troubleshooting sections of this manual. The jumpers are preset for the most common operating Requirements and usually do not need any changes.

Please do not change any of the jumpers without reading the Operation and Troubleshooting sections of the manual. Refer to the Back Panel Jumper Function on the jumper functions. To maintain the integrity of the EMI prevention measures in this unit, it is important to replace all hardware when the unit is reassembled after opening the housing. This includes the star washers around the audio input, line output and PTTI back panel jacks, the ground lugs at the sides of the PC board, and all the panel screws.

4. OPERATION INTRODUCTION

One knob and seven push-button switches on the front panel control the DSP-9+. Eight jumpers located behind the back panel preset options for some of the front panel push buttons. The knob controls power and sets the speaker and headphone audio output level of the DSP-9+. One momentary push-button selects the operating mode of the DSP-9+. The remaining six push Buttons select the operational parameters of the DSP-9+. Power Switch/Gain Adjust Control The gain knob on the front panel of the DSP-9+ is the power switch/gain adjust control. Rotate the gain control clockwise to turn on the DSP-9+ and increase the volume. Rotate the gain control counter-clockwise to turn off the DSP-9+ and decrease the volume.

Selecting the mode The Mode push-button on the DSP-9+ selects the Voice, CW, or Data operating mode. A lighted LED indicates the selected operating mode. In Voice mode, the DSP-9+ filters the audio input using one of six band pass filters, adaptively Reduces random noise, and adaptively eliminates multi-tone noise (heterodynes). These three Functions can operate simultaneously or independently. In the voice mode, two front panel push buttons, marked "Narrow" and "Medium", select the Bandwidth of the voice filter. These buttons select narrow, medium, and wide bandwidths from one of two sets of voice filters. With both front panel bandwidth select buttons out, the filter Bandwidth is the wide. When the "Medium" button is pushed in and the other bandwidth select Button is out, the bandwidth of the voice filter is midway between the wide and narrow filters. If The "Narrow" bandwidth select button is pushed in, it has precedence and the voice filter bandwidth is narrow, regardless of the state of the other button. An internal jumper behind the back panel selects the filter set, either a bandwidth 1.6, 2.0, or 2.4 kHz. Or 1.8, 2.4, and 3.1 kHz. Factory setting is 1.6, 2.0, and 2.4 kHz.

To activate heterodyne/tone elimination and random noise reduction, depress the push-buttons marked "NRt" and "NRr", respectively. Tone elimination, random noise reduction and band pass filtering can operate simultaneously or independently. Just depress the desired combinations of push-buttons. One voice band pass filter is always active in the voice mode.

Voice AGC the AGC (automatic gain control) can add up to 36 dB extra gain for weak signals, and can
Control the variation in levels of stronger signals. Depress the “AGC” push-button to turn on the AGC. Use the AGC whenever it enhances the readability of a signal. Sometimes, the AGC appears to increase the noise level because of the additional gain of the AGC. This is normal when signals are weak, and may enhance readability in spite of the higher audible noise. Voice Bypass Mode Depressing the Bypass push-button places the DSP-9+ into a bypass mode. In this mode, a relay connects the audio input jack of the DSP-9+ directly to the speaker and headphone output jacks. The relay also connects the audio input jack of the DSP-9+ to the line output jack via a 6 dB Attenuator. The Bypass mode has precedence over the voice mode. When the DSP-9+ is in bypass, the settings of the gain control and the parameter select push buttons do not affect the signal. Turning off or removing power

from the DSP-9+ automatically de-energizes the relay and forces the DSP-9+ into the bypass mode.

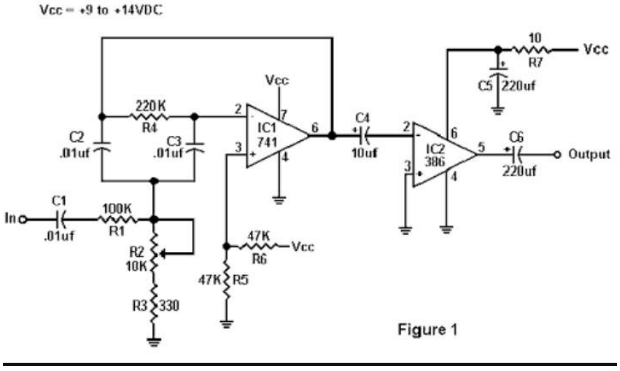

Figure 1

CW Mode

In CW mode, the DSP-9+ filters the audio input using one of eighteen CW band pass filters and Also can reduce random noise. Back panel jumpers preset any combination of two filter center Frequencies from the six available center frequencies. There are three bandwidths for each Center frequency. The front panel "Hi/Lo" push-button selects one of the two preset filter center Frequencies. Depress the button marked "Hi/Lo" to select the highest center frequency. In the out position of the "Hi/Lo" push-button, the center frequency of the CW filter is the lower of the two preset frequencies. Note that depressing a push-button always selects the first of the two parameters. Factory settings for center frequencies are 600 Hz. and 800 Hz.

Back Panel Jumper Function Table on page 12 for details on the Jumper 4, 5, and 6 settings. Two parameter select push buttons, marked "100/500" and "200/500", select the bandwidth of the CW filter. These buttons select a bandwidth of 500, 200 or 100 Hz. When both bandwidth select buttons are out, the bandwidth is 500 Hz. When the "200/500" button is pushed in and the other bandwidth select button is out, the bandwidth of the CW filter is 200 Hz. If the "100/500" bandwidth select button is pushed in, it has precedence and the CW filter bandwidth is 100 Hz, Independent of the state of the other button. No matter what the state of the three CW filter switch settings on the DSP-9+ front panel, one of the six preset CW filters is always active in the CW mode. The CW mode can also operate with random noise reduction. To enable the random noise reduction feature for CW operation, simply press in the button marked "NRr". The AGC (automatic gain control) can add up to 18 dB extra gain for weak signals, and can Control the variation in levels of stronger signals. Depress the “AGC” push-button to turn on the AGC. Use the AGC whenever it enhances the readability of a signal. Sometimes, the AGC appears to increase the noise level because of the additional gain of the AGC. This is normal when signals are weak, and may enhance readability in spite of the higher audible noise. CW Bypass Mode Depressing the Bypass push-button places the DSP-9+ into a bypass mode. In this mode, a relay connects the audio input jack of the DSP-9+ directly to the speaker and headphone output jacks. The relay also connects the audio input jack of the DSP-9+ to the line output jack via a 6 dB Attenuator. The Bypass mode has precedence over the CW modes. When the DSP-9+ is in bypass, the settings of the gain control and the parameter select push buttons do not affect the signal. Turning off or removing power from the DSP-9+ automatically de-energizes the relay and forces the DSP-9+ into the bypass mode. Data Mode In the Data mode, the DSP-9+ filters the audio input using one of sixteen data filters for RTTY, AMTOR, PACTOR, G-TOR or HF Packet. There are four choices of mark-space frequency pairs. Each mark-space frequency pair has four filters, one each for RTTY, AMTOR, and PACTOR and HF PACKET.

Two Back panel jumpers preset one mark-space filter frequency pair from the four available frequency pairs. After presenting the mark-space frequency pair (or using the factory setting), select the desired data filter by pressing the RTTY, AMTOR, PACTOR, or HF Packet push-button (use HF Packet filter for G-TOR). If more than one push-button is depressed at the same time, the narrowest data filter has precedence (RTTY = 260 Hz, AMTOR = 340 Hz, PACTOR =440 Hz, HF Packet = 540 Hz). If no data filter push-buttons are depressed, a wideband filter (100-3700 Hz) is active with a delay equal to the data filters, and AGC is active (if selected). *The mark-space frequencies of the modem, receiver and DSP-9+ must match.* 10 Default mark-space frequencies vary among modem and radio manufacturers, and common mark-space frequencies also vary in different parts of the world. Some modems have default HF Packet mark-space frequencies different from their RTTY, AMTOR, and PACTOR mark-space frequencies. The DSP-9+ mark-space frequencies factory settings are 2125-2295 Hz. for all 4 data modes. *The mark-space frequencies of the modem, receiver and DSP-9+ must match.* Some modems and radios have programmable mark-space frequencies. If your modem or radio defaults to mark-space frequencies other than 2125-2295 Hz. you must change the modem or radio mark-space frequencies to match the DSP- 9+ or change the DSP-9+ mark-space frequencies to match the modem and radio mark space frequencies. The Kantronics KAM+ usually has the HF Packet mark-space pair set to 1600-1800 Hz. See the KAM+ manual for the procedure to change the KAM+ mark-space setting via Software.

See the Back Panel Jumper Function Table on page 12 for DSP-9+ filters and settings. Jumpers 7 and 8 set the mark-space frequencies. RTTY Remodulator-To select the RTTY remodulator, first press in the RTTY push-button to select the RTTY filter. Then, press the RTTY push-button rapidly twice (“double-click”) to enable the remodulator. Leave the button pressed in after you have selected the remodulator. To switch the remodulator off release the RTTY push-button for one second or more. The remodulator mode is easily recognized by a lack of any receiver background noise - only the pure audio RTTY tones are audible when the remodulator is on and a RTTY signal is present. The DSP-9+ mutes the audio output when no FSK RTTY signals are detected. The AGC (automatic gain control) can add up to 18 dB extra gain for weak signals, and can Control the variation in levels of stronger signals. Depress the “AGC” push-button to turn on The AGC. Use the AGC whenever it enhances the readability of a signal. Sometimes, the AGC appears to increase the noise level because of the additional gain of the AGC. This is normal when signals are weak, and may enhance readability in spite of the higher audible noise. Data Bypass Mode Depressing the Bypass push-button places the DSP-9+ into a bypass mode. In the Data mode, the bypass mode routes the signal through an all pass DSP filter which has precisely the same delay as the normal narrow band filter. When switching from data mode to bypass mode, this prevents a time discontinuity which can cause an AMTOR or PACTOR link to lose synchronization. The bypass mode has precedence over the Data mode. When the DSP-9+ is in bypass, the settings of the gain control and the parameter select push buttons do not affect the signal. 11 Turning off or removing power from the DSP-9+ automatically de-energizes the bypass relay and forces the DSP-9+ into the relay bypass mode. In this mode, a relay connects the audio input jack of the DSP-9+ directly to the speaker and headphone output jacks. The relay also connects the audio input jack of the DSP-9+ to the line output jack via a 6 dB attenuator.

5. DISADVANTAGES OF FREQUENCY MODULATION

There are a number of dis-advantages to the use of frequency modulation. Some are can be overcome quite easily, but others may mean that another modulation format is more suitable. Requires more complicated demodulator: One of the minor dis-advantages of frequency modulation is that the demodulator is a little more complicated, and hence slightly more expensive than the very simple diode detectors used for AM. Also requiring a tuned circuit adds cost. There are many advantages to using frequency modulation - it is still widely used for many broadcast and radio communications applications. However with more systems using digital formats, phase and quadrature amplitude modulation formats are on the increase. Nevertheless, the advantages of frequency modulation mean that it is an ideal format for many analogue applications.

6. MOTIVATION

Our motivation to select this project title is that to design and implementation of low power FM transmitter and also to solve the problems of information accuses in our campus and mini-media service. Parallel we will understand the role of communication concept. Secondly we think that.

7. PROBLEM DESCRIPTION

When we do our project there was some problems, these are problems of to get internet access during on time. Due to this reason we start our project late in time than other students, until the program manger adjust the project room and lab class. There were also problems of power during working our project accidentally the power is off, even if we have not saved documents. The other is problem of software; the common software's that was using in the lab class are not easily simulating our project such as, circuit maker and tinna. Finally our advisor gives some comment to use multiuse software for simulation of our project. But multiuse also little difficult to install and immediately license expired. Finally when we come to the implementation part they have been occurred shortage of materials, even if the equivalences of materials are not found in the stores, labs and shops. Due to these challenges our goal of the project is not fully succeed. We hope that the next generation will be finishing this project better than ours.

8. OBJECTIVE

The aim of our project has the following objectives: To integrate the knowledge and skills acquired from major courses taken so far. To develop allows power FM transmitter to be used in specialized applications for local area entertainment purpose. Provide a reference for further study in a similar streams having having ambition to deal with low power FM transmitter design.

9. ORGANIZATION OF THE PROJECT

This section describes the introduction part of our project. It introduces basic concepts FM transmitter which translates information using higher rate with a higher performance using a minimum amount of transmitted power and bandwidth. The second chapter explains the literature review. It includes back ground history of FM transmitter and the documents used to guide during our project. The third chapter discusses the design and analysis/ methodology which describe the body of our project. It explains block diagrams, calculation parts, schematic diagrams and results/out puts. Finally the project also includes summary, conclusion, recommendation, references and appendix.

10. LITERATURE REVIEW

We have performed with high expectation to complete this final year project for the title 'low power FM transmitter. By considering this project can be done for in good result and the output can be performing at FM radio smoothly. This chapter will review some similar project and studies, the solution s, of the project related, over view, on different approaches made by previous researchers and make compares between my final year project and those similar projects**.**

REFERENCES

[1] For more information see Marconi's In 1907, Marconi established the first commercial transatlantic radio communications service, between Clifden, Ireland and Glace Newfoundland. Donald Manson working as an employee of the Marconi Company (England,1906).Julio Cervera Baviera developed radio in Spain around 1902. Cervera Baviera obtained patents in England, Germany, Belgium, and Spain.

[2] June 1899, Cervera had, with the blessing of the Spanish visited Marconi's radiotelegraphic installations on the Channel, and worked to develop his own system. He began collaborating with Marconi on resolving the problem of a wireless communication system, obtaining some patents by the end of 1899. Cervera, who had worked with Marconi and his assistant Georgein 1899, resolved the difficulties of wireless telegraph and obtained his first patents prior to the end of that year. On March 22, 1902, Cervera founded the Spanish Wireless Telegraph and Telephone Corporation and brought to his corporation the patents he had obtained in Spain, Belgium, Germany and England.

[3] http://en.wikipedia.org/wiki/History_of_radioHe established the second and third regular radiotelegraph service in the history of the world in 1901 and 1902 by maintaining regular transmissions between Tarifa and Ceuta for three consecutive months, and between Javea (Cabo de la Nao)andIbiza(Cabo Pelado). This is after Marconi established the radiotelegraphic service between the Isle of Wight and Bournemouth in 1898. In 1906, Domenico Mazzotto wrote: "In Spain the Minister of War has applied the system perfected by the commander of military engineering, Julio Cervera Baviera (English patent No. 20084 (1899))."

[4] http://en.wikipedia.org/wiki/History_of_radioCervera thus achieved some success in this field, but his Radiotelegraphic activities ceased suddenly, the reasons for which are unclear to this day. Using various patents, the company called British Marconi was established in 1897 and began communication between stations and ships

at sea. This company along with its subsidiary Marconi, had a stranglehold on ship to shore communication. It operated much the way Telegraph operated until 1983, owning all of its equipment and refusing to communicate with non-Marconi equipped ships. Many inventions improved the quality of radio, and amateurs experimented with uses of radio, thus the first seeds of broadcasting were planted. The company Telefunken was founded on May 27, 1903 as "Telefunken society for wireless.

[5] http://en.wikipedia.org/wiki/Telefunkentelefon" of Siemens & Halske(S & H) and the Allgemeine Elektrizitäts-Gesellschaft (General Electricity Company)as joint undertakings for radio engineering in Berlin. It continued as a joint venture of AEG and Siemens AG, until Siemens left in 1941. In 1911,Kaiser Wilhelm II sent Telefunken engineers to West Sayville, New York to erect three 600-foot (180-m) radio towers there. Nikola Tesla assisted in the construction. A similar station was erected in Nauen, creating the only wireless communication between North America and Europe. The invention of amplitude-modulated (AM) radio, so that more than one station can send signals(as opposed to spark-gap radio, where one transmitter covers the entire bandwidth of the spectrum)is attributed to Reginald Fessenden and Lee de Forest. On Christmas Eve1906,Reginald Fessenden used an Alexanderson alternator and rotary spark-gap transmitter to make the first radio audio broadcast, from Brant Rock, Massachusetts.

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