**EXPERIMENT-9**

**AIM-** To Study Super Heterodyne AM receiver and measurement of receiver parameters

Viz.sensitivity ,selectivety & fidelity.

**THEORY**

The principle of operation of the superheterodyne receiver depends on the use of

heterodyning or frequency mixing. The signal from the antenna is filtered sufficiently at

least to reject the image frequency (see below) and possibly amplified. A local oscillator

in the receiver produces a sine wave which mixes with that signal, shifting it to a

specific intermediate frequency (IF), usually a lower frequency. The IF signal is itself

filtered and amplified and possibly processed in additional ways. The demodulator uses

the IF signal rather than the original radio frequency to recreate a copy of the original

modulation (such as audio).

Diagram

Description automatically generated

Fig.1. shows the minimum requirements for a single-conversion superheterodyne

receiver design. The following essential elements are common to all superhet circuits: a

receiving antenna, a tuned stage which may optionally contain amplification (RF

amplifier), a variable frequency local oscillator, a frequency mixer, a band pass filter

and intermediate frequency (IF) amplifier, and a demodulator plus additional circuitry

to amplify or process the original audio signal (or other transmitted information). To

receive a radio signal, a suitable antenna is required. This is often built into a receiver;

especially in the case of AM broadcast band radios. The output of the antenna may be

very small, often only a few microvolts. The signal from the antenna is tuned and may

be amplified in a so-called radio frequency (RF) amplifier, although this stage is often

omitted. One or more tuned circuits at this stage block frequencies which are far

removed from the intended reception frequency. In order to tune the receiver to a

particular station, the frequency of the local oscillator is controlled by the tuning knob

(for instance). Tuning of the local oscillator and the RF stage may use a variable

capacitor, or varicap diode. The tuning of one (or more) tuned circuits in the RF stage

must track the tuning of the local oscillator.

Mixer stage: The signal is then fed into a circuit where it is mixed with a sine wave

from a variable frequency oscillator known as the local oscillator (LO). The mixer uses

a non-linear component to produce both sum and difference beat frequencies signals,

each one containing the modulation contained in the desired signal. The output of the

mixer may include the original RF signal at fd, the local oscillator signal at fLO, and the

two new frequencies fd+fLO and fd-fLO. The mixer may inadvertently produce additional

frequencies such as 3rd- and higher-order intermodulation products. The undesired

signals are removed by the IF bandpass filter, leaving only the desired offset IF signal at

fIF which contains the original modulation (transmitted information) as the received

radio signal had at fd.

Intermediate frequency stage:The stages of an intermediate frequency amplifier are

tuned to a particular frequency not dependent on the receiving frequency; this greatly

simplifies optimization of the circuit.[6] The IF amplifier (or IF strip) can be made

highly selective around its center frequency fIF, whereas achieving such a selectivity at a

much higher RF frequency would be much more difficult. By tuning the frequency of

the local oscillator fLO, the resulting difference frequency fLO - fd (or fd-fLO when using socalled low-side injection) will be matched to the IF amplifier's frequency fIF for the

desired reception frequency fd. One section of the tuning capacitor will thus adjust the

local oscillator's frequency fLO to fd + fIF (or. less often, to fd - fIF) while the RF stage is

tuned to fd. Engineering the multi-section tuning capacitor (or varactors) and coils to

fulfill this condition across the tuning range is known as tracking. Other signals

produced by the mixer (such as due to stations at nearby frequencies) can be very well

filtered out in the IF stage, giving the superheterodyne receiver its superior

performance. However, if fLO is set to fd + fIF , then an incoming radio signal at fLO + fIF

will also produce a heterodyne at fIF; this is called the image frequency and must be

rejected by the tuned circuits in the RF stage. The image frequency is 2fIF higher (or

lower) than fd, so employing a higher IF frequency fIF increases the receiver's image

rejection without requiring additional selectivity in the RF stage. Usually the

intermediate frequency is lower than the reception frequency fd, but in some modern

receivers (e.g. scanners and spectrum analyzers) it is more convenient to first convert an

entire band to a much higher intermediate frequency; this eliminates the problem of

image rejection. Then a tunable local oscillator and mixer convert that signal to a

second much lower intermediate frequency where the selectivity of the receiver is

accomplished. In order to avoid interference to receivers, licensing authorities will

avoid assigning common IF frequencies to transmitting stations. Standard intermediate

frequencies used are 455 kHz for medium-wave AM radio, 10.7 MHz for broadcast FM

receivers, 38.9 MHz (Europe) or 45 MHz (US) for television, and 70 MHz for satellite

and terrestrial microwave equipment.

Bandpass filter: The IF stage includes a filter and/or multiple tuned circuits in order to

achieve the desired selectivity. This filtering must therefore have a band pass equal to or

less than the frequency spacing between adjacent broadcast channels. Ideally a filter

would have a high attenuation to adjacent channels, but maintain a flat response across

the desired signal spectrum in order to retain the quality of the received signal. This

may be obtained using one or more dual tuned IF transformers or a multipole ceramic

crystal filter.

Demodulation: The received signal is now processed by the demodulator stage where

the audio signal (or other baseband signal) is recovered and then further amplified. AM

demodulation requires the simple rectification of the RF signal (so-called envelope

detection), and a simple RC low pass filter to remove remnants of the intermediate

frequency. FM signals may be detected using a discriminator, ratio detector, or phaselocked loop. Continuous wave (morse code) and single sideband signals require a

product detector using a so-called beat frequency oscillator, and there are other

techniques used for different types of modulation. The resulting audio signal (for

instance) is then amplified and drives a loudspeaker. When so-called high-side injection

has been used, where the local oscillator is at a higher frequency than the received

signal (as is common), then the frequency spectrum of the original signal will be

reversed. This must be taken into account by the demodulator (and in the IF filtering) in

the case of certain types of modulation such as single sideband.

RECEIVER CHARACTERISTICS:

The important characteristics of receivers are sensitivity, selectivity, & fidelity

described as follows:

Sensitivity:

The sensitivity of radio receiver is that characteristic which determines the minimum

strength of signal input capable of causing a desired value of signal output. Therefore,

expressing in terms of voltage or power, sensitivity can be defined as the minimum

voltage or power at the receiver input for causing a standard output. In case of

amplitude-modulation broadcast receivers, the definition of sensitivity has been

standardized as "amplitude of carrier voltage modulated 30% at 400 cycles, which when

applied to the receiver input terminals through a standard dummy antenna will develop

an output of 0.5 watt in a resistance load of appropriate value substituted for the loud

speaker" .

Selectivity:

The selectivity of a radio receiver is that characteristic which determines the extent to

which it is capable of differentiating between the desired signal and signal of other

frequencies.

Fidelity:

This is defined as the degree with which a system accurately reproduces at its output

the essential characteristics of signals which is impressed upon its input.

Diagram, schematic

Description automatically generated

Figure 2: Setup for Determining Reciever Characteristics

Determination of receiver characteristics:

A laboratory method for the measurement of receiver characteristics is shown in Fig. 2.

We use here an artificial signal to represent the voltage that is induced in the receiving

antenna. This artificial signal is applied through 'dummy' antenna, which in association

antenna with which the receiver is to be used. Substituting the resistance load of proper

value for the loudspeaker and measuring the audio frequency power determine the

receiver output.

Sensitivity:

Sensitivity is a determined by impressing different RF voltages in series with a standard

dummy antenna and adjusting the intensity of input voltage until standard outputs

obtained at resonance for various carrier frequencies. Sensitivity is expressed in

microvolt.

Selectivity: Selectivity is expressed in the form of a curve that give the carrier signal

strength with standard modulation that is required to produce the standard test output

plotted as a function off resonance of the test signal. The receiver is tuned to the desired

frequency and manual volume control is set for maximum value. At standard

modulation, the signal generator is set at the resonant frequency of the receiver. The

carrier output of the signal generator is varied until the standard test output is obtained.

At the same tuning of receiver, the frequency of signal generator is varied above and

below the frequency to which the receiver is tuned. For every frequency, the signal

generator voltage, applied to the receiver input, is adjusted to give the standard test

output from the receiver

Fidelity: Fidelity is the term expressing the behavior of receiver output with modulation

frequency of input voltage. To obtain a fidelity curve, the carrier frequency of the signal

generator adjusted to resonance with the receiver, standard 400 cycles modulation is

applied, the signal generator carrier level is set at a convenient arbitrary level and the

manual volume control of the receiver is adjusted to give the standard test output. The

modulation frequency is then varied over the audio range, keeping degree of modulation

constant.

**RESULT:-**

Superhetrodyne receiver has been studied and plot for receiver parameters viz.

sensitivity, selectivity and fidelity has been studied.

**INNOVATION-1**

**AIM-** to study phase modulation using matlab.

**SOFTWARE USED**- matlab

**THEORY-**

**Phase modulation** (**PM**) is a modulation pattern for conditioning communication signals for transmission. It encodes a message signal as variations in the [instantaneous phase](https://en.wikipedia.org/wiki/Instantaneous_phase) of a [carrier wave](https://en.wikipedia.org/wiki/Carrier_wave). Phase modulation is one of the two principal forms of angle modulation, together with frequency modulation.

In phase modulation, the instantaneous amplitude of the baseband signal modifies the phase of the carrier signal keeping its amplitude and frequency constant

The phase of a carrier signal is modulated to follow the changing signal level (amplitude) of the message signal. The peak amplitude and the frequency of the carrier signal are maintained constant, but as the amplitude of the message signal changes, the phase of the carrier changes correspondingly.

Phase modulation is widely used for transmitting radio waves and is an integral part of many digital transmission coding schemes that underlie a wide range of technologies like [Wi-Fi](https://en.wikipedia.org/wiki/Wi-Fi), [GSM](https://en.wikipedia.org/wiki/GSM) and satellite television.

PM changes the [phase angle](https://en.wikipedia.org/wiki/Phase_(waves)) of the [complex envelope](https://en.wikipedia.org/wiki/Complex_envelope) in direct proportion to the message signal.

If *m(t)* is the message signal to be transmitted and the carrier onto which the signal is modulated is

{\displaystyle c(t)=A\_{c}\sin \left(\omega \_{\mathrm {c} }t+\phi \_{\mathrm {c} }\right).}then the modulated signal is



{\displaystyle y(t)=A\_{c}\sin \left(\omega \_{\mathrm {c} }t+m(t)+\phi \_{\mathrm {c} }\right).}This shows how {\displaystyle m(t)} modulates the phase - the greater m(t) is at a point in time, the greater the phase shift of the modulated signal at that point. It can also be viewed as a change of the frequency of the carrier signal, and phase modulation can thus be considered a special case of FM in which the carrier frequency modulation is given by the time [derivative](https://en.wikipedia.org/wiki/Derivative) of the phase modulation.

The modulation signal could here be 

{\displaystyle m(t)=\cos \left(\omega \_{\mathrm {c} }t+h\omega \_{\mathrm {m} }(t)\right)\ }

The mathematics of the [spectral](https://en.wikipedia.org/wiki/Spectral_density) behavior reveals that there are two regions of particular interest:

* For small [amplitude](https://en.wikipedia.org/wiki/Amplitude) signals, PM is similar to [amplitude modulation](https://en.wikipedia.org/wiki/Amplitude_modulation) (AM) and exhibits its unfortunate doubling of [baseband](https://en.wikipedia.org/wiki/Baseband) [bandwidth](https://en.wikipedia.org/wiki/Bandwidth_(signal_processing)) and poor efficiency.
* For a single large [sinusoidal](https://en.wikipedia.org/wiki/Sinusoidal) signal, PM is similar to FM, and its [bandwidth](https://en.wikipedia.org/wiki/Bandwidth_(signal_processing)) is approximately 

{\displaystyle 2\left(h+1\right)f\_{\mathrm {M} }} where ,

 {\displaystyle f\_{\mathrm {M} }=\omega \_{\mathrm {m} }/2\pi }  {\displaystyle h} is the modulation index defined below. This is also known as Carson's Rule for PM.]

Modulation index-

As with other modulation indices, this quantity indicates by how much the modulated variable varies around its unmodulated level. It relates to the variations in the phase of the carrier signal:

{\displaystyle h\,=\Delta \theta \,}

where {\displaystyle \Delta \theta }  is the peak phase deviation. Compare to the modulation index for [frequency modulation](https://en.wikipedia.org/wiki/Frequency_modulation#Modulation_index).

**SOURCE CODE-**

clc;

close all;

clear all;

t= [0:0.001:1];

fm = 5;

m= cos(2\*pi\*fm\*t);

wm=2\*pi\*fm;

subplot(4,1,1);

plot(t,m);

title('Message Signal');

fc= 100;

c= sin(2\*pi\*fc\*t);

wc=2\*pi\*fc;

subplot(4,1,2);

plot(t,c);

title('Carrier Signal');

kp=input('Enter the value of phase sensitivity');

s=cos(wc\*t + m.\*kp);

subplot(4,1,3);

plot(t,s);

title('Phase Modulated signal');

**OUTPUT-**

**Timeline

Description automatically generated with medium confidence**

**INNOVATION-2**

**AIM-** to study phase demodulation using matlab

**SOFTWARE USED**- matlab

**THEORY-**

in phase modulation, the information is encoded as variations in the phase of the carrier signal. In its generic form, a phase modulated signal expressed as an information-bearing sinusoidal signal modulating another sinusoidal carrier signal is expressed as

x(t) = A cos \left[ 2 \pi f_c t + \beta + \alpha sin \left( 2 \pi f_m t + \theta \right) \right]   \quad \quad \quad (1) 

where, m(t) = \alpha sin \left( 2 \pi f_m t + \theta \right) represents the information-bearing modulating signal, with the following parameters

\alpha – amplitude of the modulating sinusoidal signal  
f_m – frequency of the modulating sinusoidal signal  
\theta – phase offset of the modulating sinusoidal signal

The carrier signal has the following parameters

A – amplitude of the carrier  
f_c – frequency of the carrier and f_c << f_m  
\beta – phase offset of the carrier

## Demodulating a phase modulated signal:

The phase modulated signal shown in equation (1), can be simply expressed as

x(t) = A cos \left[ \phi(t) \right]    \quad\quad\quad (2)  

Here,  \phi(t) is the **instantaneous phase**  that varies according to the information signal m(t).

A phase modulated signal of form x(t) can be demodulated by forming an analytic signal by applying Hilbert transform and then extracting the instantaneous phase. This method is explained here.

We note that the instantaneous phase is \phi(t) = 2 \pi f_c t + \beta + \alpha sin \left( 2 \pi f_m t + \theta \right)  is linear in time, that is proportional to 2 \pi f_c t. This linear offset needs to be subtracted from the instantaneous phase to obtained the information bearing modulated signal. If the carrier frequency is known at the receiver, this can be done easily. If not, the carrier frequency term 2 \pi f_c t needs to be estimated using a linear fit of the unwrapped instantaneous phase.

**CODE-**

clc;

close all;

clear all;

t= [0:0.001:1];

fm = 5;

m= cos(2\*pi\*fm\*t);

wm=2\*pi\*fm;

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title('Message Signal');

fc= 100;

c= sin(2\*pi\*fc\*t);

wc=2\*pi\*fc;

subplot(4,1,2);

plot(t,c);

title('Carrier Signal');

kp=input('Enter the value of phase sensitivity');

s=cos(wc\*t + m.\*kp);

subplot(4,1,3);

plot(t,s);

title('Phase Modulated signal');

z=hilbert(s);

inst\_phase = unwrap(angle(z));

p = polyfit(t,inst\_phase,1);

offsetTerm = polyval(p,t);

d= inst\_phase - offsetTerm;

subplot(4,1,4);

plot(t,d);

title('Phase Demodulated signal');

**OUTPUT-**

Chart

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