Bird Street Media Project
June 6, 2012
Workshop
Preparation Training for Full-Power Broadcasting

Facilitators: Erv Knorzer and Daryl Autry

Original Mission Statement
Bird Street Media Project
January 1, 2000

Proposed by: Erv Knorzer, Tom Opdennaker, Alan Rice. Executive Board

Bird Street Media Project is a non-commercial educational low-power radio station which is: information free from corporate interests; a voice for the voiceless; democracy in action; indigenous/local culture and language; local values, lost dog reports; a forum for in-depth community discussion; exposure to a variety of cultures, music and opinions; cheap; an outlet for genius and disaster, a mirror for the community; an opportunity to participate and learn; real voices, real feelings. It's fun, entertaining, scandalous, not virtual, and a celebration. It helps people find each other.

Allocation of Radio Spectrum

in the United States

	30 km		3 km		300 m		30 m		3 m		30 cm		3 cm		3 mm		
VLF		LF		MF		HF		VHF		UHF		SHF		EHF		Infrared	Visible Light
	10 kHz		100 kHz		1 MHz		10 MHz		100 MHz		1 GHz		10 GHz		100 GHz		

This is the table of contents to a list showing how the radio frequency spectrum is allocated to different users in the United States.

The numbers in brackets "[xx.xx]" refer to a F.C.C. rule section allocating the frequency.

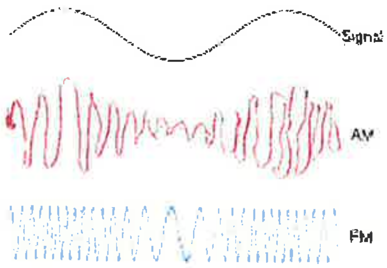
Table of Contents:

30 GHz and above	Extremely High Frequency (EHF)
Other charts of the radio spectrum	
Cable TV channel frequencies	
Letter designations of microwave bands	
Satellite to L-band conversion	
Frequency coordination	
Other communications resources on the net	

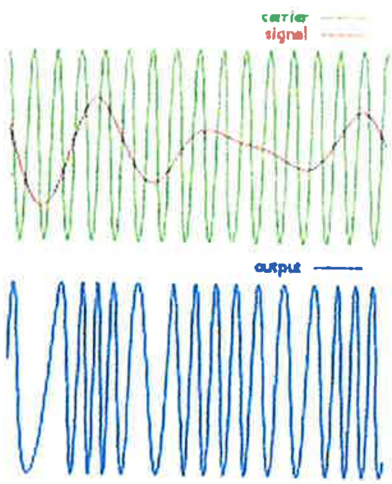
This web page by John Neuhaus, WA2JXE. Please mail your comments and suggestions to <john@jneuhaus.com>

Sun, May 26, 2002

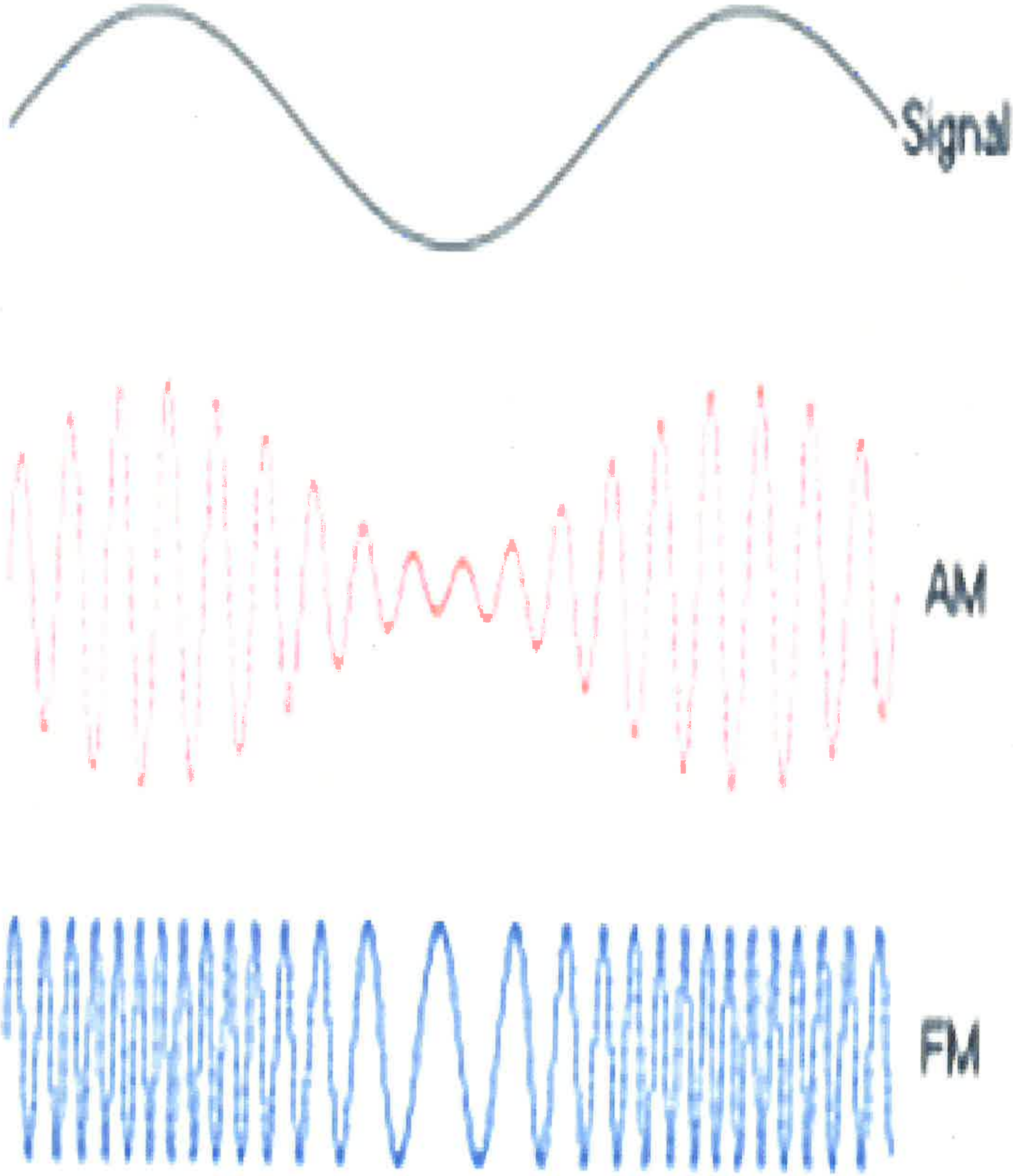
Return to the Table of Contents.



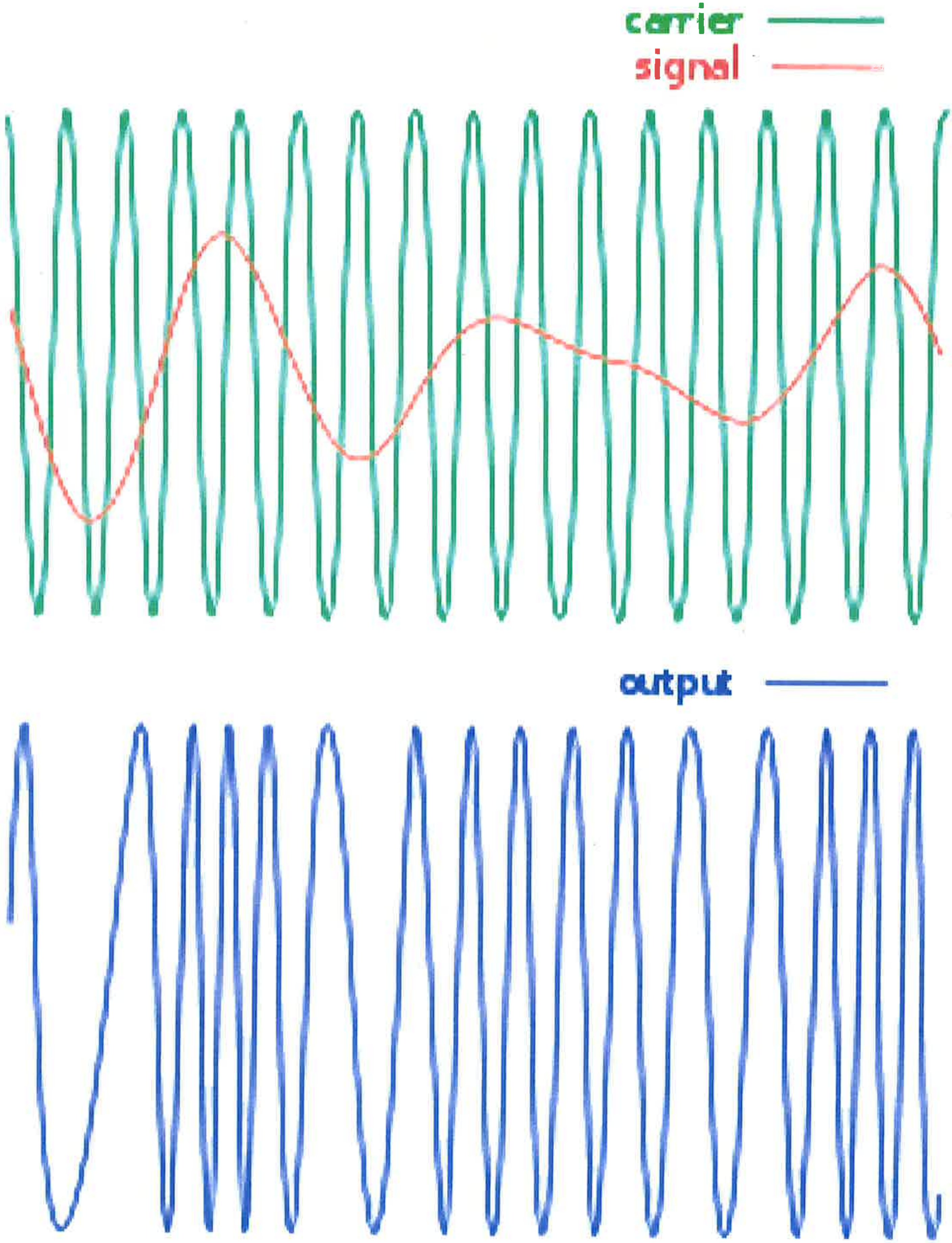
A signal may be carried by an AM or FM radio wave.



A signal modifies the carrier

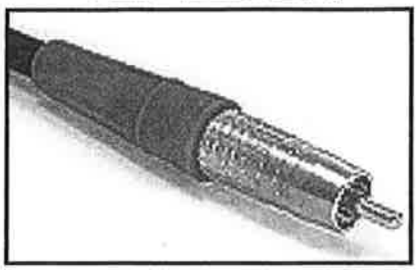


200px-Amfm3-en-de.gif



200px-Frequency_Modulation_svg.png

Balanced vs. Unbalanced Interconnects



In discussing the characteristics and performance of various interconnect systems; two points should be kept in mind.

Balance is defined in terms of the impedance of the two signal conductors with respect to a reference, which is usually ground. If these impedances are equal and non-zero, the system is balanced. If the impedances are unequal the system is unbalanced. A signal conductor with a grounded return conductor is, therefore, an unbalanced (sometimes referred to as a single ended) system.

A small, common-mode, 60 Hz noise, voltage can exist between the chassis of two AC powered devices regardless of whether they are safety grounded (use a three-wire plug) or not.

Unbalanced Interconnections

An unbalanced (single ended) interface uses only two conductors to carry the signal from one device to another, one conductor carries the signal and the other is the grounded return. In consumer audio systems this usually consists of a cable with a center conductor and a shield terminated in an RCA Phono plug.

RCA Phono Plug

The ubiquitous "RCA Phono" plug was developed by RCA over fifty years ago to be used for short interconnections between a turntable and amplifier inside a phonograph, hence its name. This unbalanced interconnect system is simple and inexpensive, but as with many other connector systems has been adopted for uses other than originally intended and has become the de-facto connector for consumer audio/video equipment. Note, however, that the design was original intended only for short cable runs within the same piece of equipment. Being an unbalanced system it is susceptible to common-mode noise voltages.

A problem occurs when there is a ground voltage (common-mode voltage) between the two interconnected devices. Because of this voltage, a small current will flow down the cable shield between the devices (often referred to as common-mode current, or as a ground loop current). If the cable shield were ideal (zero impedance) this current would not cause a problem. However, since the shield has a finite resistance, a small noise voltage will appear across the length of the cable shield. The magnitude of this voltage will equal the common-mode current times the shield resistance. This voltage is in series with the signal voltage and will add directly to it at the receiver. In other words, an unbalanced interconnect system consisting of only two conductors (center conductor plus a shield) has no ability to reject common-mode noise voltages.

This coupling is referred to as **common-impedance coupling** , and is the result of the fact that in an unbalanced two-wire system the shield is performing two functions. It is a shield carrying the common-mode noise current, but it is also one of the signal conductors carrying the return signal current.

For more details on common-impedance coupling see "Noise Reduction Techniques in Electronic Systems," Second Edition, page 54, by Henry W. Ott, published by Wiley-Interscience, 1988.

Example 1: Let's consider a typical case of the interface between two grounded (3-prong AC plug) pieces of audio equipment. Some actual cases will be better than this example and some will be worse. The shield resistance of a fifteen-foot cable might be about 0.25 ohms. If the 60-Hertz shield current is 250 uA, the voltage developed across the shield will be 62.5 uV. For consumer audio products the reference signal level is about 300 mV (-10 dBV). The signal to noise ratio will therefore be 74 dB. For a high quality consumer audio system we would probably like the S/N ratio to be greater than 100 dB. Therefore, we would most likely be able to hear some 60-Hertz hum in quiet passages of the program material.

You might conclude at this point that ungrounded equipment, those using a 2-prong AC plug, might solve this problem by eliminating the ground connections. This often helps, but does not necessarily eliminate the problem. For ungrounded equipment the common-mode ground current can still flow through the inter-winding capacitance of the power transformer. The impedance of the capacitor will normally reduce the magnitude of the current (typically less than 100 uA), and hence the noise voltage, but some noise will still exist. Since the impedance of the inter-winding capacitance is frequency dependent, more current will flow at high frequencies (harmonics of 60 Hz) than at the fundamental frequency (60 Hz). Therefore, the interference will more likely consist of a high frequency buzz instead of a 60 Hz hum.

Despite its shortcomings, this unbalanced system works surprisingly well most of the time. In particular, in cases with short cable runs, and with very little, or no, 60 Hz voltage between the chassis of the interconnected devices.

Unbalanced Interface Cables

From the above discussion, we can conclude that for the case of an unbalanced interface, the only property of the cable that has any significant effect on the common-impedance noise coupling is the shield resistance.

What can we do to minimize the possibility of problems when using this very common unbalanced interconnect system? First, we want to minimize the common-mode voltage difference between the interconnected devices. If possible, plug everything into the same AC power outlet, or power strip. If that is not possible plug the interconnected equipment into power outlets that are on the same branch circuit (same circuit breaker). Another possibility would be to run an additional heavy gauge (low resistance) ground wire between the chassis of the two devices to divert some of the common-mode cable shield current.

Second, we want to minimize the resistance of the interconnecting cable shields. Use cables with a copper braid (or even spiral copper) shield instead of a foil shield. Use cables with the heaviest shield possible, or with double shields in order to minimize cable shield resistance. Do not use cables with aluminum foil shields, since their resistances are much higher. (Note: A foil-braid combination shield is fine, as long as the low resistance copper braid is present). Also keep cables as short as possible, since this will also reduce the total shield resistance.

Thirdly, you can isolate or break the common-mode shield current path. Most people try to do this by removing or lifting grounds on the AC power cord. This can be very dangerous since it can lead to safety problems. The AC power cords and connections should be left alone, exactly as the manufacturer designed them. It is much better, and safer, to do the isolation on the interconnecting signal cables.

This can easily be done by using high quality signal isolation transformers designed specifically for this application, such as the ISO-MAX® line of transformers manufactured by Jensen Transformer Corporation (www.jensentransformers.com). An isolation transformer allows the signal to pass through while at the same time breaking the ground connection and thereby eliminating the common-mode current. These transformers are available for audio signals, video signals as well as RF signals. Although quality isolation transformers are expensive (MSRP of $50 to $100) they work extremely well and their cost is usually negligible compared to the overall cost of a high quality audio system installation. Low quality, inexpensive isolation transformers are also available, however, they will seriously degrade the quality (frequency response) of the audio, video, or RF system.

Troubleshooting Audio Noise Problems

An excellent reference for troubleshooting audio system noise problems is the ISO-MAX® Troubleshooting Guide by Bill Whitlock. Go to Jensen Transformers Website and download the pdf file for the "Jensen Transformer Troubleshooting Guide."

Another approach would have been for the manufacturer to design the product with a balanced interface.

Next Page →
Jump to: Balanced vs. Unbalanced Interconnects - page 2 Go

Description:	Ships:	Price:
The SM7B dynamic microphone has a smooth, flat, wide-range frequency response appropriate for music and speech in all professional audio applications. (SM7B)	24-48 hrs	$349.00

WRITE A REVIEW
Be one of the first to review this item!

Add to Your Basket:
$349.00

SM7B Vocal Microphone

Key Features:

- Flat, wide-range frequency response for exceptionally clean and natural reproduction of both music and speech
- Bass rolloff and mid-range emphasis (presence boost) controls with graphic display of response setting
- Improved rejection of electromagnetic hum, optimized for shielding against broadband interference emitted by computer monitors
- Internal "air suspension" shock isolation virtually eliminates mechanical noise transmission
- Highly effective pop filter eliminates need for any add-on protection against explosive breath sounds, even for close-up vocals or narration
- Now shipping with the A7WS detachable windscreen, designed to reduce plosive sounds and gives a warmer tone for close-talk vocals
- Yoke mounting with captive stand nut for easy mounting and dismounting provides precise control of microphone position
- Classic cardioid polar pattern, uniform with frequency and symmetrical about axis, to provide maximum rejection and minimum coloration of off-axis sound
- Rugged construction and excellent cartridge protection for outstanding reliability

Includes:

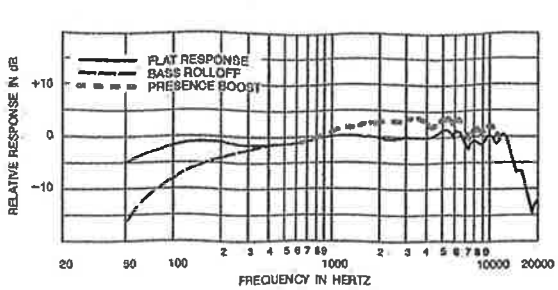
- Foam Winscreen
- Close-talk Windscreen
- Locking Yoke Mount
- Switch Cover Plate

Suggested Applications:

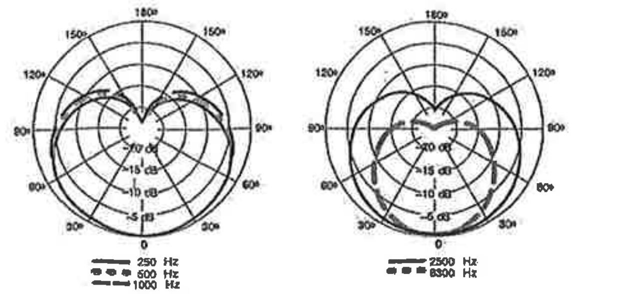
- Lead Vocals, Back-up vocals

Transducer Type	Dynamic
Polar pattern	Cardioid
Frequency response	50 Hz - 20 kHz
Impedance	150 Ohms
Sensitivity	-59 dBV/Pa, (1.12 mV), 1 Pascal=94 dB SPL
Electromagnetic Hum Sensitivity	33 dB equivalent SPL in a 1 millioersted field (60 Hz)
Swivel Assembly	Integrated, captive nut for ease of attachment to stand, fits 5/8 in.-27 thread.
Microphone Connector	3pin professional audio connector (male XLR type)
Weight	321g (11.3 oz.)
Dimensions	189.7mm x 117mm x 63.5mm

The SM7B dynamic microphone has a smooth, flat, wide-range frequency response appropriate for music and speech in all professional audio applications. It features excellent shielding against electromagnetic hum generated by computer monitors, neon lights, and other electrical devices. The SM7B has been updated from earlier models with an improved bracket design that offers greater stability. In addition to it's standard windscreen, it also includes the A7WS windscreen for close-talk applications.



TYPICAL FREQUENCY RESPONSE



TYPICAL POLAR PATTERNS

Warranty: 2 Year Manufacturer Warranty

0 items Checkout / View Cart >>>

Memory	Internal Storage	External Storage	External RAID Storage
MacBook & MacBook Pro	SSD (Solid State Drive)	Mercury On-The-Go Pro	Mercury Elite Pro Dual
Mac Pro	Desktop Drives	Mercury Elite Pro	Mercury Elite Pro Dual mini
iMac	Notebook Drives	Mercury Elite Pro mini	Mercury Elite Pro Qx2
View Memory for all Macs	Mac Pro	OWC Express	Mercury Rack Pro
Wireless	Desktop IDE/ATA	NewerTech Voyager	NewerTech Guardian MAXimus

My OWC | CD / DVD / Blu-ray | Accessories | Laptop Center
Specials | iPod / iPhone | Audio / Video | Batteries for Apple Laptops

Customer Service | Tech Support | Contact Us | Site Map

Other World Computing
Quality Products, Expert Support since 1988

Blog | Twitter | Facebook | YouTube | Newsletter | Catalog
Privacy Statement | About Us | Media/NewsRoom | Site Terms

Load Time: 0.033 seconds, Guest ID: 113651798

Shure SM58 Dynamic Handheld Microphone

Get the legendary live sound of the Shure SM58 microphone. Famous for its warmth, clarity a
SM58 is a favorite for musicians worldwide.

273 People rated this product: Rate and Review This Product

Roll Over to Enlarge

Main

overview **specs** **docs** **more-info**

- Type: Dynamic (moving coil)
- Frequency Response: 50 to 15,000 Hz

- Polar Pattern: Unidirectional (cardioid), rotationally symmetrical about microphone axis, uniform with frequency
Output Level (at 1,000 Hz)
- Open Circuit Voltage: -74.5 dB* (0.19 mV)
- Power Level: -56.0 dB**
- Impedance: Rated impedance is 150 (300 actual) for connection to microphone inputs rated low impedance
- Polarity: Positive pressure on diaphragm produces positive voltage on
pin 2 with respect to pin 3
- Connector: 3-pin professional audio connector (male XLR type)
- Case: Dark gray, enamel-painted, die cast metal; matte-finished,
silver colored, spherical steel mesh grille
- Overall Dimensions: 51 x 162 x 23mm (2 x 6-3/8 x 29/32 in.)
- Swivel Adapter: Positive-action, break-resistant, adjustable through 180 degrees, with standard 5/8 in.-27 thread
- Net Weight: 298 g (10.5 oz.)

*0 dB = 1 v/ bar
**0 dB = 1 mw/

Phantom power

Some microphones such as condenser microphones require power. An alternative to battery power is phantom power, which consists of direct current applied equally through the two signal lines of a balanced audio connector (in modern equipment, usually an XLR connector). The supply voltage is referenced to the ground pin of the connector (pin 1 of an XLR), which normally is connected to the cable shield or a ground wire in the cable or both. When phantom powering was introduced, one of its advantages was that the same type of balanced, shielded microphone cable that studios were already using for dynamic microphones could be used for condenser microphones as well, in contrast to vacuum -tube microphones, which required special, multi-conductor cables of various kinds.

With phantom power, the supply voltage is effectively invisible to balanced microphones that do not use it: e.g., most dynamic microphones. A balanced signal consists only of the differences in voltage between two signal lines; phantom powering places the same DC voltage on both signal lines of a balanced connection. This is in marked contrast to another, slightly earlier method of powering known as *parallel powering* or *T-powering* (from the German term *Tonaderspeisung*), in which DC was overlaid directly onto the signal in differential mode. Connecting a dynamic microphone (especially a ribbon microphone) to an input that had parallel powering enabled could very well damage the microphone severely, but this is not normally so with phantom powering unless the cables are defective or wired incorrectly.

Mute/Solo

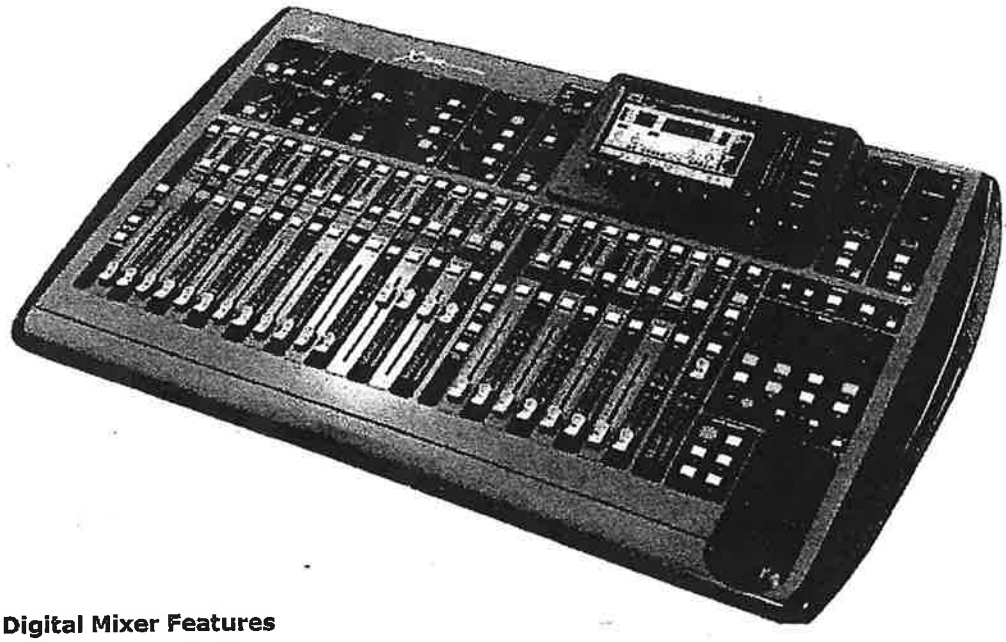
The mute and solo buttons work together to enable or disable a track's audio. This explanation assumes the track is playing and has a non-zero volume slider. Put simply, mute silences the track, while solo makes the track audible, and silences all other tracks that aren't also soloed. Solo "trumps" (overrides) mute, i.e. soloing a track makes it audible regardless of whether it's muted. The following table illustrates Mute/Solo behavior in detail:

Mute	Solo	Other tracks soloed	Track is:
No	No	No	Audible
Yes	No	No	Silenced
No	Yes	No	Audible
Yes	Yes	No	Audible
No	No	Yes	Silenced
Yes	No	Yes	Silenced
No	Yes	Yes	Audible
Yes	Yes	Yes	Audible

While at least one track is soloed, the mute buttons don't affect the audio, and the mixer is said to be in "solo mode". In solo mode, the solo buttons control the mix, but when you exit solo mode, the mute buttons regain control. Solo mode acts like a detour: when you exit solo mode, the pre-solo mix is restored, assuming you didn't change any mute buttons during the solo. If you did change some mute buttons during the solo, the solo becomes a "one-way trip" instead of a "round trip".

Behringer X32 Digital Mixer

Item #: BEH X32 LIST



Behringer X32 Digital Mixer Features

- 32-channel total recall digital mixer
- Fully programmable high-end mic preamps and 16 mix buses, configurable as subgroups
- Main LCR, 6 matrix buses and all 16 mix buses
- Featuring inserts, 6-band parametric EQ's and dynamics processing
- 16 analog XLR outputs
    - 6 additional line in/outputs
    - 2 phones connectors
    - Talkback section with integrated or external mic
- 48-channel digital snake ready via AES50 ports
    - KLARK TEKNIK's SuperMAC networking capability for ultra-low jitter and latency
- iPad app for professional remote mixing available free of charge, no host PC required
- Built-in expansion port for Firewire/USB card or digital networking interfaces
- Virtual FX rack featuring 8 true-stereo FX slots for high-end simulations of famous outboard gear
    - Lexicon 480L and PCM70, EMT250 and Quantec QRS etc. (included)
- Ultra-high power 40-Bit floating-point DSP
    - Unlimited dynamic range
    - No internal overload
    - Near-zero latency between in and outputs
- 6 mute groups and 8 DCA groups on 8 dedicated 100 mm motorized faders
- Super-easy user interface and dedicated channel strip section
    - Direct access controls for intuitive workflow
- High-resolution 7 inch color TFT with associated controls
    - Individual RGB backlit graphic LCD's in every channel
- 25 motorized 100 mm faders on an extensive channel strip
    - Dedicated user-definable control section
- USB type-A connector providing file storage and uncompressed stereo recordings
    - Create show presets and perform system updates
- Connectivity for BEHRINGER's P-16 Personal Monitoring System

Audio Processing and Compressors for Your Station!

Q. Why do I need audio processing for my station? Do I need a compressor? Or a limiter? Or a clipper? Or a Multiband audio processor? Or something with Automatic Gain Control? And does a $2000 audio processor sound 10 times better than a $200 compressor?

One of your most important duties as a low power fm operator is making sure that the sound that you send from your studio into the transmitter is within the limits of volume set by the FCC. Your transmitter will set up a "**carrier**" signal that runs at your frequency. The carrier, running on your frequency, is what makes you distinct from other radio stations. You may have a carrier signal that is 91.3, while your nearest neighbor on the dial may be 91.7. You will add audio on top of that from your mixing board. Then the listener can hear your audio by tuning to your carrier frequency at 91.3 on the dial, and they can listen to your neighbor by tuning to 91.7.

The louder you turn up the volume at your mixing board, the more you modulate the signal, and the louder it gets for the listener. Up to a point, that is good so that you are not dramatically louder than anything else. However, there is only so loud that you can turn up your audio before the kettle boils over onto your neighbors carrier. The FCC has rules about exactly how loud (and how soft) you can be. Radio stations generally have **modulation monitors,** which help you calibrate your equipment and monitor your levels. They also have some form of audio processing, which take the raw audio from out of the mixing board and at least make sure that it does not exceed FCC specifications. Fancier models, that cost more, can create an array of effects that can make your station sound better. The following piece will introduce several concepts: **limiting, compressing, clipping and automatic gain control.** A **compressor** is a piece of equipment which can create a number of these effects on audio.

And we'll look at some of the controls of a compressor, which include **attack, release, ratio and threshold:**

The diagrams below are visual representations of real audio clips, intended to show you what actual waveforms look like. Here are the unprocessed, natural waveforms:

Is there room for community radio in my zip code?
Check!

Latest Headlines

Oakland Local: "Local nonprofits get green light to launch community radio stations"

You're On Air

Radio Matchmaking? Introducing Radio Spark

Nieman Journalism Lab: "New FCC rules offer a "historic opportunity" for low-power community FM radio"

All Gov: "FCC Opens Radio Airways to Small, Nonprofit Local Stations"

STL made simple - with the Barix Reflector

The Barix Reflector service takes the difficulties out of streaming STL audio over the internet.