HANDBOOK OF INTERCOM SYSTEMS ENGINEERING

FIRST EDITION

TELEX®

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The Fine Print

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TABLE OF CONTENTS

LIST OF FIGURES

PREFACE

Welcome to the Telex Communications, Inc. **Handbook of Intercom Systems Engineering**. The idea for this book came, as it does with many books and inventions, over drinks at a bar. A few of us "intercom types" were discussing our varied histories and experiences. We added up the years each of us had in the intercom system industry and between the four of us we hit the 75 year mark. Add the "rest of the gang" at Telex into that estimate and we are well past the century mark, quickly closing in on the two century mark. It was then we decided that we were getting old and had spent too much time dealing with intercoms. Someone commented that it was a shame that "the younger generation" didn't really know what we seasoned pros did and suggested that we should pass down our profound body of knowledge for the good of "intercom-kind."

The idea for the book sort of hibernated for a bit after that (as did we). Weeks later we found ourselves planning for a trade show and discussing the appropriate "swag" for giveaways. After some discussion, we decided a well written, reasonably impartial, complete reference / tutorial on intercom system design would be a great thing – useful, desirable, business related, and maybe something inspirational. We hope those that read this book take advantage of the knowledge they can glean from it and expand the capabilities of their own intercom systems. And, maybe they will use some intercom equipment from Telex. In the process, we may go down in history as the "guys who wrote the book on intercoms."

The book you are starting has a number of goals; it is intended to be a systematic tutorial for the novice user and an encyclopedic reference for the designer in the midst of a project. It is NOT a 100+ page sales brochure for Telex® products. Rather, it is a resource intended to take the reader through the different types of intercom systems and needs, compare them, point out strengths and weaknesses, and provide many "real-life" examples of working systems.

This book will be updated regularly to keep pace with changes in technology. On the enclosed CD you will find a good deal of technical information, systems examples, and some marketing "fluff" such as Telex® product sheets, catalogs, operating manuals, etc. Throughout the book, we have strived to provide real examples with real products. Many of the examples will make use of Telex® products, as that is what we know best – Telex AudioCom®, RTS™ Matrix, RTS™ TW, RadioCom™ Wireless and Telex® Headsets. If we get to a point with an example where the equipment needed or best suited is not one of our products, we will tell you what that product is and how to find it.

I have often joked that intercoms are the "stepchild" of the industry – no one (or VERY, VERY FEW) people decided in high school what they wanted to do with their lives is be Mr. (or Ms.) Intercom. People tend to get dragged kicking and screaming into dealing with the design, installation and support of intercom systems because they were in the wrong place at the wrong time. What they later learn is that they have developed a valuable bit of niche expertise that can be in great demand.

The one goal above all with this book is to provide a solid body of work, in a useful form to all those who have been, and will be, dealing with specification and operation, as well as, design, installation and support of intercom systems. In other words, we hope this book helps you get the absolute most out of your communications systems.

Apart from the story of the bar and the trade show, there is another serious reason why we have written this book. Intercoms (in our opinion) are a neglected, underrated, taken for granted part of the technical world – they are not glamorous nor interesting. I have at times made the comment that intercom systems have a lot in common with toilets (no off color jokes to follow). They are often the last system designed into an environment, they are often cheaply done, they are PRESUMED to be always available and always working, and when they are NOT – it QUICKLY becomes a crisis – and the plumber, all of a sudden is worth ANY AMOUNT OF MONEY to return the toilet to its normal functioning condition, FAST!

Now, let's take the same scenario except in the intercom world. Consider a live television show, a camera fails, or a microphone fails, and the audio operator can't hear the guest, or a tape jams in a VTR. No problem, we'll just TELL the TD to take another camera, and TELL Camera 2 to change its shot, or the audio operator will ASK the stage manager to get a spare microphone to the talent, or the director will TELL the talent to ad-lib until the tape can be salvaged…. "WHAT DO YOU MEAN, NO ONE CAN COMMUNICATE THESE SIMPLE INSTRUCTIONS!?!? Get the PLUMBER (oops… INTERCOM EXPERT) NOW!!!!"

The intercom system, whether in a television station, on the sidelines of a football game, or in a factory is critical, and must be seamless, reliable, and work without fault to allow all needed communications to take place. This book is intended to help make that happen.

We'd love to know if you think we have succeeded, or failed, or fallen short with this effort, so that, as with all things in life, we can learn, grow and improve. Please send your comments to intercoms@Telex.com.

Ralph K. Strader

Vice President & General Manager

Intercom Products

Telex Communications, Inc.

January 2001

About the Authors

This handbook is the work of a number of past and present Telex employees, as well as, some outside experts (such as Stan Hubler).

Among the contributors (in alphabetical order) are: Talal Aly-Youssef, Gene Behrend, Larry Benedict (contributor and editor), Rick Fisher, Stan Hubler, John King, Murray Porteous, Dave Richardson, Ralph Strader, and Tom Turkington. The credits for each chapter reflect the contribution of the primary author for that chapter. Through a group effort such as this, the words may actually be those of a number of individuals in any given chapter.

Many other individuals have directly or indirectly contributed to this book, and not all of them can be recognized here. Many of the illustrations were prepared by John Yerxa, and many of the systems examples came from the work of Shawn Anderson, Chuck Roberts, Gene Behrend, and Geoff Rogers.

CHAPTER

1

INTERCOMS—AN OVERVIEW

RALPH STRADER

Introduction

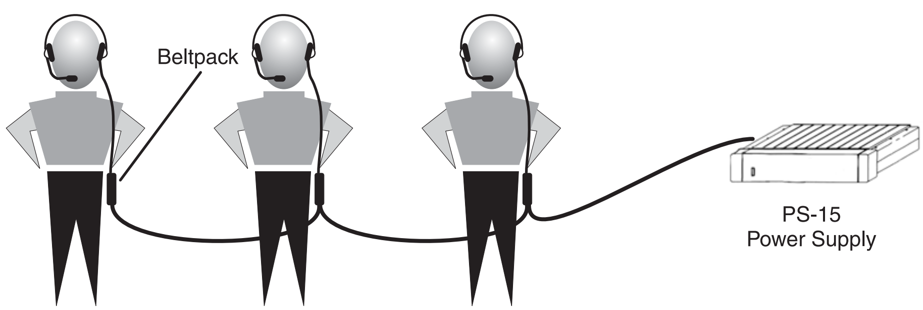
Intercom systems, by definition, may be comprised of many different types of intercoms and subsystems. The basic building blocks can be categorized into four basic types or elements: Party-Line Systems, Matrix Systems, Wireless Systems, and Accessories.

Party-Line Systems

Wired Party-Line systems are systems in which a number of participants are all involved in the same conversation. Think of the telephone extensions in your home, if each person in your family picks up a telephone in your home, you will all be able to hear each other. You can talk to one another simultaneously and the person "on the other end of the line" will be a full participant in one "public" conversation.

Depending on where in the world you are from, (presuming English language), you may also refer to this type of system as "PL" (for "party-line), "TW" or "Two-Wire" from the telephone systems, where on two wires, a full duplex conversation takes place, or "conference" denoting the type of activity taking place in the conversation.

Figure 1.1 Simple Party-Line System



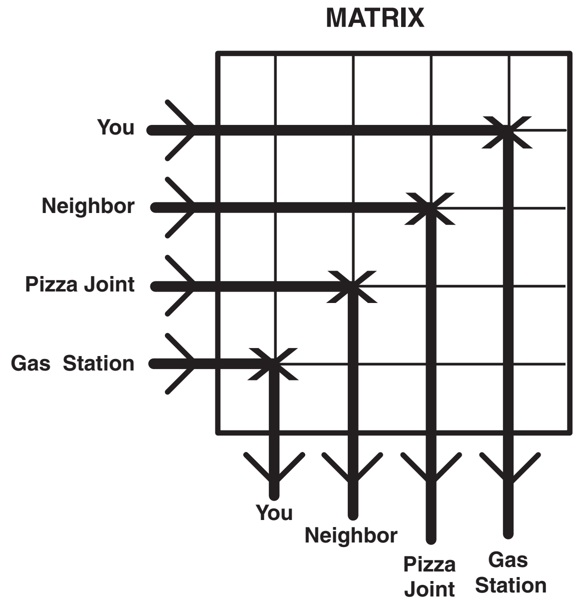
Note, the physical configuration and implementation of that "PL" or "TW" does not necessarily need to be on two physical wires, in most cases it is not. The specific topologies will be addressed in the chapters that follow.

Matrix Systems

Wired Matrix systems are systems in which a large number of individuals have the ability to establish private individual conversations from point A to point B. Again, going back to the telephone system in your neighborhood, you, your next door neighbor, the pizza joint down the street and the local gas station are all connected to the same central office by wires from each location back to the telephone company. At any time, you can be talking to the gas station, while your neighbor is ordering a pizza. The pizza guy does not hear you ask the mechanic about the repairs on your SUV.

Depending on where in the English speaking world you are, you may refer to these types of systems as Matrix systems, crosspoint intercoms, point-to-point systems, private lines (sometimes, confusingly referred to as "PL"), or by some of the brand names used: McCurdy, ADAM™, Zeus™, and others.

Figure 1.2 Simple Matrix System



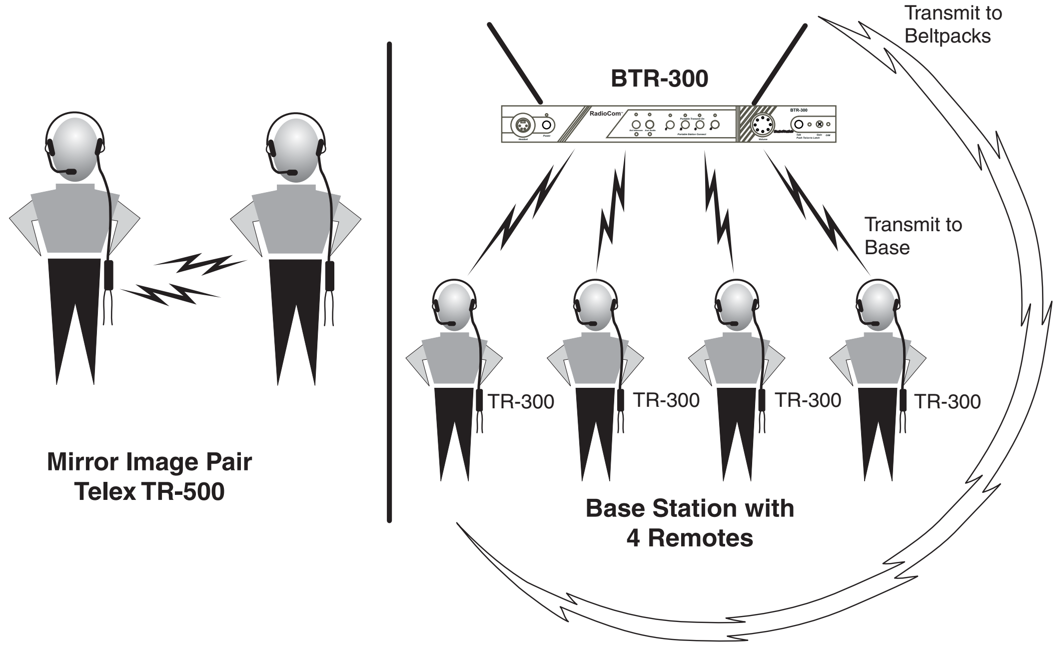
Like the telephone system, matrix systems have other functions and capabilities. Conferences, call waiting, busy signals, and other features are common to many matrix intercoms. They are not limited to simple point-to-point communications. Some systems even allow inter-matrix routing of signals, similar to long distance telephones calls using trunks between central offices. Having a matrix system with a number of conferences configured within it (virtual PLs) is very common.

Wireless Systems

Wireless Intercoms encompass all sorts of systems from the most basic pair of "walkie talkies" to cell phones to dedicated professional full duplex intercom products. The most basic feature of wireless intercoms is that they are not tethered by wires. (Didn't think this was going to be quite that basic, did you?) Seriously, wireless intercom systems are employed where the limitation of wireless systems which can include fidelity, interference, lack of range, lack of security (real or perceived), and battery life limitations are outweighed by the freedom of being cordless. This freedom can be essential in many applications—try dragging a wired intercom cable into the containment vessel of a nuclear reactor.

Wireless intercom systems can be designed, installed, configured and operated in PL or matrix configurations, and may very likely be connected to a hard-wired PL or matrix intercom system at some point. They can range from as simple as a single pair of units talking to one another, to a system in which 24 or more different portable units are dynamically switched between conversations.

Figure 1.3 Wireless Intercom Examples



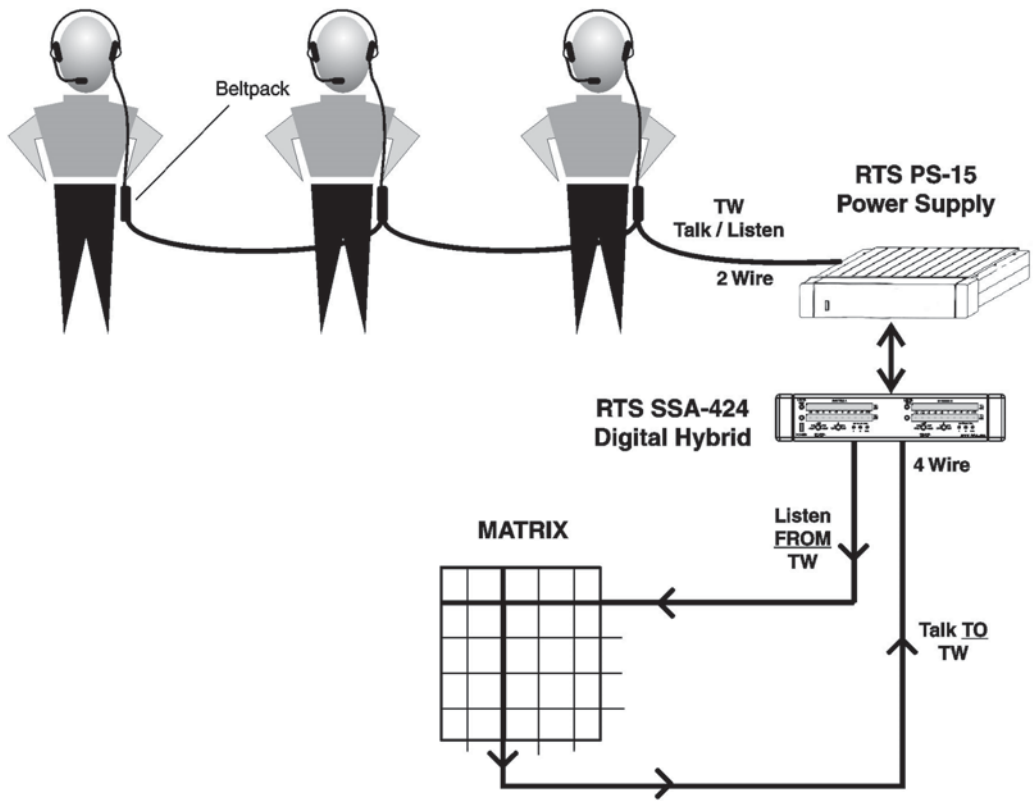
Wireless systems will vary tremendously worldwide, due to varying governmental radio regulations. What is common in America may be illegal in Japan, and may be unsuitable, for other reasons, in Germany. These units may be referred to by any of the types mentioned above, but, again, the unifying feature is the freedom from a wire.

Accessories

The fourth and final category is “accessories”. We are giving accessories its own separate category because of its importance. This book is addressing intercom systems. In all likelihood, many of the systems you encounter will be an amalgam of the three types mentioned above. Without “accessories” you cannot have a system, just a bunch of equipment.

To connect a TW system to a matrix system, a converter is required to change the combined talk and listen signal from the TW to separate talk and listen signals for the matrix – a hybrid provides this conversion.

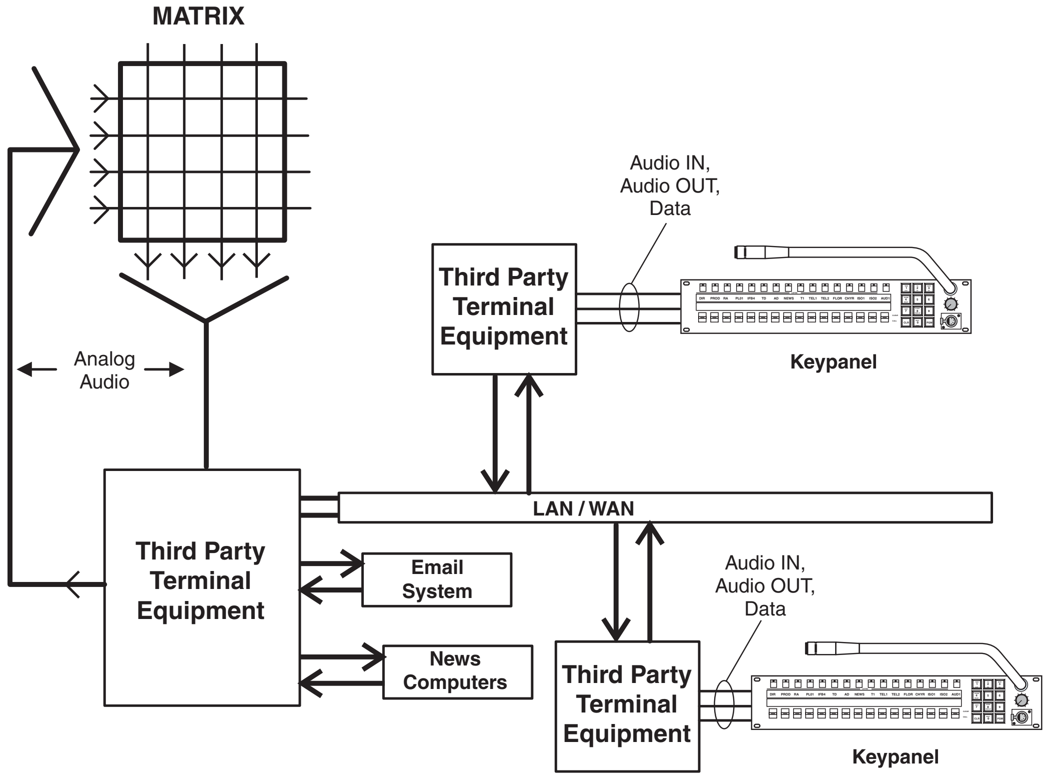
Figure 1.4 Example of Interfacing a TW System to a Matrix System



To connect a matrix intercom system to a Two-way radio system, a contact closure may be required to activate the radio transmitter. A GPI (General Purpose Interface) between the matrix and the base station of the radio can solve this problem easily.

To do intelligent trunking between matrix systems, across campus or across the country, the audio and control signals between the matrices could be transported over fixed pairs of wires. Realistically, however, installing a set of wires between Omaha and Los Angeles may be out of your budget – so an interface allowing the use of dial-up telephone lines may be needed. Other possibilities include muxes and demuxes to allow the audio and data to be carried over an existing corporate Wide Area Network (WAN), or “piggybacked” as subcarriers on an existing satellite feed.

Figure 1.5 Complex Matrix Intercom System



In many cases, connection to “the telephone company” is required to allow a reporter to connect into an intercom from his or her cell phone, or to allow a return program feed to be fed to a remote location. A telephone interface (TIF) unit provides this connectivity.

The most basic accessory in an Intercom system may be the headset. It may provide isolation from ambient noise; it may have a noise-canceling microphone to reduce wind noise, and may have stereo ear pieces to allow program audio and intercom audio to be fed independently to the right and left ears.

Each of these accessories is vital to creating an intercom system that meets the communications needs of the users.

Before We Begin

Throughout this book, you will be subjected to the jargon that permeates the intercom world. In the chapters that follow, you will be presented with definitions specific to the topic being covered. In many cases, there are common terms that will be applicable to all these chapters, and so we will present a few definitions to get us started. We have also provided a comprehensive glossary in the rear of the book.

IFB

Interrupted Fold Back – also referred to as IRF – Interrupted Return Feed. The best way to explain this is to give an example. A news reporter is on the scene of live accident coverage. She needs to not only hear what the anchor back at the studio is saying i.e., “So, Jane, how many chickens were injured when they tried to cross the road during rush hour?” She also needs to hear instructions from the director back in the studio i.e., “Wrap it up, 10 seconds.” The IFB function in an intercom system allows a single audio signal to be sent to Jane, normally containing program audio interrupted by instructions or information from someone not a part of the program audio.

ISO

Camera Isolate – This is not reserved strictly for the domain of cameras anymore. This is truly an isolate function, not unlike the action at a party of grabbing the arm of a fellow guest, dragging them off to a corner for a private conversation, and then returning them to their group. There are instances where it is necessary in an intercom system to establish a momentary private conversation with someone who may be talking and listening to a number of other people. The person who needs to interrupt presses a button or key, which establishes a private two person conversation. Upon releasing the key, the two participants are returned to whatever conversation(s) they were a part of previously. This was called Camera Isolate as it first was used to remove an individual camera from a conference to allow private communications.

Tally

A signal sent for the purpose of indicating status for a particular purpose. The sound of your telephone ringing can be described as a tally. On an intercom panel with multiple channels, it can be a visual signal, such as a blinking light, to indicate which station is calling. It can be used to indicate a particular function is not available due to a conflict – similar to the busy signal you get when calling the radio station trying to be the tenth caller and win a year's supply of cat litter.

The above definitions and many more can be found in the glossary at the back of this book.

The Rest Of The Book

We have organized this book by the above types of systems – two chapters devoted to PL Intercoms, two chapters for matrix systems, two chapters for wireless systems, and one chapter on interfaces, determining systems needs and requirements, technical requirements for installation, and some real world case studies.

Near the end of the book, we have included references for further information, a glossary, and a CD full of information on **Telex®** Products, technical references, and many system drawings.

CHAPTER

2

INTRODUCTION TO PARTY-LINE INTERCOM SYSTEMS

STAN HUBLER

Introduction

Leading off this chapter, **Some Definitions** that may help you understand Party-Line intercoms terms (and buzz-words). Then, a **Short History** of Party-Line intercoms will be presented, leading into a discussion of **Present Day Systems and Manufacturers**. The **System Components and Their Function** will explore the main components of these systems and what they do. Then, **How Each System Works** shows how these system components are put together to make a functioning intercom and some examples of the different systems. **Outstanding Features of Each System** describes application areas and where each system is often marketed. Some important **Limitations of Each System** are described and a **Summary** closes this chapter.

Some Definitions

Party-Line (PL) systems / Conference Line Intercom Systems

A Party-Line system allows a group of people to intercommunicate. For example, one person can talk, while all the others on the bus or channel can hear. When the system is full duplex, anyone can talk and the rest can hear or interrupt the speaker at any time. The Party-Line and distributed matrix systems presently sold today are usually full duplex and are non-blocking, which means that access to the channel is immediate and there is no busy signal. Conversations on Party-Line systems are, in general, non-private. It is important to note that both two wire and four wire type systems support the Party-Line concept.

Two-Wire

A communications system where the path is the same for both talk and listen. In electrical pathways there are, in fact, two wires (one path). Two-wire systems can be two-wire balanced or two-wire unbalanced.

Balanced Line

The balanced line concept reduces noise pickup by outside sources. A balanced two conductor line carries audio that is differentially driven and balanced to ground.

Full Duplex

This is communication that allows simultaneous two-way conversations, that is, one person can interrupt the other. In data communications, full duplex permits confirmation of sent data by the receiving terminal echoing, sending back the same data, or confirming data.

Decibel (dB)

A derived unit of loudness. The human ear perceives a 10 decibel increase as twice as loud, and a 10 decibel decrease as half as loud.

Beltpack

A portable headset user station. This station is designed to be worn on a user's belt, but is also fastened to the underside of consoles, taped to a structure near the user, or mounted on a piece of equipment. The headset plugs into the user station, as does the connection to the rest of the intercom.

Biscuit

Marketing buzz word for a portable speaker station.

Main Station

A multichannel user station. There may be one or more of these stations in a system. Usually the primary station in a system.

Master Station

A user station where a user station and a system power supply are combined into one package

Sidetone

In the truest sense, sidetone is a small amount of microphone signal fed back to the earphone of the individual speaking into the microphone. In a two type user station, the null balance control is sometimes used to adjust the amount of sidetone the user hears. This control is sometimes (erroneously) called the sidetone control. Other equipment has both null balance adjustments and a true sidetone adjustment.

Crosstalk

Unwanted interference caused by audio energy from one line coupling ("leaking") into adjacent or nearby lines.

A Short History

Party-Line intercoms were needed early on by television production crews to coordinate their activities. Some of the activities included on-site sport pickups, entertainment on stage, and videotaping of shows. The crews included camera operators, audio, lighting, stage directors, director, assistant director, production assistant, and others. Originally, these crews shared one intercom channel where the director called the shots. Later, as intercom developed, additional channels were added so each crew could still listen to the director, then could switch to their own channels to coordinate activities without conflict with the director. Party-Line intercom systems were also used by industrial activities to coordinate manufacturing and testing of large systems such as aircraft.

Early intercom systems (1960-1975) were either homemade or accumulations of telephone equipment lashed together. Often, the homemade intercoms worked well enough but lacked the flexibility to expand the system or interface with other systems. The telephone equipment approach had some flexibility, but performance degraded rapidly as the number of stations increased above ten user stations.

In the early 1970s, Clear-Com built Party-Line systems for rock-n-roll concerts, and later for theatrical stage, and eventually for television production. This system was flexible and expandable, but required one three-conductor microphone cable for each channel. In the mid 1970s, another company, RTS Systems, designed a system for television production that had two channels on one three-conductor microphone cable (or one channel on a pair of wires). This system was even more flexible and expandable with a design that allowed up to 50 user stations on a single channel. On the East Coast, a company, Chaos, produced intercoms for the New York and other stages. And, in the Midwest, a company, Telex Communications, produced a balanced Party-Line system. This system was especially useful in noisy electrical environments, because it was immune to induced interference. Other Party-Line systems include systems such as David Clark, which is used for fire trucks and similar public safety and service crews. And, of course, four wire matrix systems can emulate Party-Line intercoms.

As **Clear-Com®** and **RTS™** Systems intercoms became more widely known, compatible systems of both appeared. They included HME and Production Intercoms for Clear-Com, and ROH and Anchor Audio's PortaCom for **RTS™**. Chaos is similar to Clear-Com, except it uses a much higher power supply voltage (46 vs. 24 volts). As the markets expanded, the distinction between theatrical and television production became blurred and Party-Line systems of all types were used wherever they were needed. So a competitive atmosphere developed and continues to the present. ROH and HME are no longer in the wired intercom market.

Present Day Systems and Manufacturers

The three major brands of "two-wire" Party-Line intercoms having the largest worldwide presence are RTS, Clear-Com, and Telex Audiocom. Other brands include Chaos, David Clark, PortaCom, and Production Intercom.

Table 2.1 Intercom brand name vs. manufacturer.

Brand Name	Manufacturer
Audiocom®	Telex Communications, Inc.
Chaos	Goddard Design Company
Clear-Com	Clear-Com Intercom Systems
David Clark	David Clark Company, Inc.
PortaCom	Anchor Audio, Inc.
Production Intercom	Production Intercom, Inc.
RTS™	Telex Communications, Inc.

Note Present day Party-Line intercom systems are mostly distributed amplifier type systems as opposed to a centralized system where all the headset lines plug into one box (Some David Clark Systems are of a centralized type). Oh yes, there is a no-amplifier system called a sound powered system, but we do not discuss it here. Present day Party-Line intercom systems may be wired or wireless or both.

System Components and Their Function

The system components for most Party-Line intercoms consist of power supplies (or master stations), user stations (e.g. belt packs, speaker stations, main stations, etc.), interconnecting cable, headsets, panel microphones, push-to-talk microphones, and a system termination.

The power supply (which is normally centralized) generates the DC power for the entire system (with the exception of self powered user stations). The power supply usually includes system termination for the audio channel, 200 ohms for RTS and Clear-Com, and 300 ohms for Audiocom. This may be as simple as a capacitor and resistor in a series, or, an electronic termination, which is integrated into the power supply voltage regulator.

The user station connects to the power supply and intercom line. The human user connects to the user station via a headset or loudspeaker and microphone or some combination. For a given channel or channels the user stations are connected to each other in parallel.

The interconnecting cable for most intercoms is standard microphone cable with three pin XLR type connectors. The female XLR connects towards the power supply and the male XLR plugs into the user station. This polarity was chosen to prevent putting DC power onto audio microphones which also use this type cable. There are at least two exceptions to the use of microphone cable: the **RTS™ TW** master stations connect audio with unshielded pairs (12 of the 25 pair in a cable). Another exception is where a twisted pair is the only connection between two points. The **RTS™ TW** user stations can connect directly to a twisted pair, while other user stations need adapters of one kind or another, and power may have to be supplied at either end.

The wired systems are of three wiring configurations: 1) separate power, audio, and return conductors (example: Clear-Com), 2)an audio pair which includes phantom power and a common (example: Audiocom), and 3) a conductor that contains one channel and power, a conductor that contains audio with- or without power, and a return (example: RTS™ TWTW intercom system).

Table 2.2 Intercom connector wiring by various manufacturers.

Clear-Com	
Pin #	**Function**
1	Common for Audio, Power, & Shield
2	DC power: 30 volts nominal
3	Unbalanced Audio
Audiocom	
Pin #	**Function**
1	Common for Audio, Power, & Shield
2	Audio + DC Power
3	Audio + DC Power
RTS TW	
Pin #	**Function**
1	Common for Audio, Power & Shield
2	Channel 1 Audio + DC Power
3	Channel 2 Audio

The wireless systems usually include an interface to the wired systems. Principal manufacturers include Telex Communications, Vega (now part of Clear-Com), and HME. We will go into further detail on wireless systems in a later chapter of this manual.

Wired intercoms are mostly of the distributed amplifier kind. The distributed amplifier is built into a User Station. User stations come in various packages and are of three kinds: headset, speaker-microphone, or both. The various packages include a belt pack (worn on the users belt, and of the headset kind), console mount (headset or speaker-microphone), rack mount (headset or speaker-microphone), desk mount (portable speaker station), wall mount (headset or speaker-microphone), and console/rack mount Master Station/Main Station (details later). The distributed amplifier concept allows each user to adjust his/hers own listening level. The user station also includes a microphone amplifier, a line amplifier/buffer, volume control(s), talk switch(es). Some user stations also may have a Call light, status indicators, and a channel selector. The microphone may be in the headset, fastened to or plugged into a speaker station, in a handset, or in a push-to-talk hand held unit.

Belt Pack Headset User Station Functional Description

A typical single channel belt pack headset user station has the following connectors: Intercom Line (XLR-3) and a Headset Connector (XLR-4).
The station has the following controls:
Microphone ON/OFF (sometimes called a TALK switch), and a headset Volume Control. It may also have a Call Lamp and a Call Lamp Send button. Examples of this station are an **RTS**™ BP318 single channel belt pack, or an **Audiocom**® BP1002, or a **Clear-Com**® RS-501.

A typical two channel headset belt pack user station adds a channel selector switch to the above. Examples **RTS**™BP351, Clear-Com® RS-502, **Audiocom**® BP2002

Alternately, newer units have two talk buttons, two volume controls, and two status indicators to tell which talk button is engaged. Examples: **RTS**™ BP325, BP351, **Clear-Com**® RS-522-TW, or **Audiocom**® IC-2B.

Speaker User Station Functional Description

A typical speaker station can function with either a headset or a speaker/microphone. A power amplifier, a speaker, and a speaker on/off switch are added to the electronics of a belt pack. In addition, a nulling adjustment is easily accessible. The nulling adjustment allows for full duplex operation without unwanted feedback. Also added is a connection or jack for either a panel microphone (rack mount stations) or a push to talk microphone (for desk mount or portable speaker stations).

Master Stations

The Master Station allows a user to access multiple channels. This allows different crews to be monitored, cued or updated. If the master station is used for training, again, different crews may be monitored and guided. These master stations have extra features for special tasks such as IFB (Interrupted FeedBack) or SA (Stage Announce), relay closures, "hot" microphones, and microphone kill. Master stations can send and receive call light signals on any channel. Two examples of the Master station are **Clear-Com**® Model 912 (12 channel) and **RTS**™ Model 803 (12 channel). Audiocom's master station is modular and can be as few as 2 channels or as many as 22 channels. Master stations allow simultaneous monitoring of any channel, any combination of channels, or all the channels. They can call or "mic kill" on any given channel. In addition, some master stations can monitor a program source.

Some Technical Notes About The Stations Above

The stations mentioned above generally are designed for the dynamic microphones in the headsets to have an impedance of about 150 to 500 ohms. The speaker station panel electret microphones are designed to have an impedance of 1000 to 2000 ohms and require 1 to 5 volts excitation. And, the push-to-talk microphones have around 500 ohms. This means the actual input impedance of the station microphone preamplifier will range from 470 ohms to 5000 ohms. The low impedance of 470 ohms minimizes the crosstalk in the headset cord. The headphone impedances expected range from 50 ohms to 1000 ohms. The 50 ohm headphones along with suitable headphone amplifiers provide enough SPL (Sound Pressure Level) to overcome the interference from loud concerts and sports events. The headphones also need to have an acoustic isolation of 20dB or more to protect the user. These stations generally have a bridging impedance across the intercom line of 10,000 to 15,000 ohms. A bridging impedance of 10,000 ohms assures that up to 50 stations can be plugged into the systems and the level drop will only be 6dB. The level drop of 6dB corresponds to the level drop when an extension telephone is picked up on an existing conversation-noticeable but the telephone is still usable.

Wiring Notes

1. Clear-Com® and **Audiocom**® two channel stations have 6 pin XLR connectors to connect to the intercom line. Clear-Com also offers the **Clear-Com**® RS-522-TW, which has two channels on a 3 pin XLR.
2. Clear-Com® and **Audiocom**® systems use a female 4 or 5 pin XLR connector on their headsets and a male 4 or 5 pin XLR connector on their user stations. However, RTS uses a male 4 or 5 pin XLR connector on their headsets and a female 4 or 5 pin XLR connector on their user stations.
3. In any system, pin 1 and the shell of the XLR connector should NOT be connected together.

4 The pin out of the headset connectors is as follows:

<u>Four pin XLR</u>
Pin 1 - Microphone common
Pin 2 - Microphone “hot”
Pin 3 - Headphone common
Pin 4 - Headphone “hot”

<u>Five pin XLR</u>
Pin 1 - Microphone common
Pin 2 - Microphone “hot”
Pin 3 - Headphone common
Pin 4 - Left Headphone “hot”
Pin 5 - Right Headphone “hot”

5 Since the power supply has a limited amount of XLR-3 connectors, splitter boxes are used to expand the system. These boxes have all the connectors wired in parallel.

6 Some user stations have “loop-thru” connectors that allow “daisy chaining” stations using a single connection to the power supply.

How Each System Works

Note Drawings at the end of the chapter depict the systems being discussed.

First, please note that although these systems are full duplex and everybody could theoretically talk at once, this is not at all practical or desirable. The usual operation is the director or lead person has their microphone enabled all the time, while all other microphones are switched off. These microphones are switched on only long enough to supply an answer, make a request, or give data. In some cases, especially in noisy environments, all microphones are off and only switched on as required. Because the Party-Line concept has so many signal sources, this operational protocol is the only way the Party-Line can be effective. And this is the reason for the system “mic kill” (microphone turn-off) capability, for the situation where a station is unmanned but has its microphone enabled.

These systems use voltage controlled current sources (or similar electronics) to apply a signal to the intercom line. All the signals applied are summed and converted to a voltage at the single termination resistor or electronic impedance. The current sources (or similar circuits) have output impedances of 10,000 ohms or greater. The loading effect of the station on the intercom, say in a 200 ohm terminated system is, worst case, 10,000 ohms in parallel with 200 ohms. This results in a change of the system termination to 196 ohms, a 2 percent change. This, in turn, causes a voltage change of 2 percent or 0.175dB, an imperceptible change. It takes 20 stations across the line to cause a 3dB change, a perceptible but not significant change. The volume controls in the user stations easily adjust for this change. In the “not so” worst-case situation, these systems can work with up to 75 stations, provided enough DC power is available. The work-around in this case, in the **RTS™ TW** system, is a switch on the power supply which doubles the system impedance. Then, two power supplies can divide the DC load and are coupled together with capacitors to end up with the 200 ohm termination and twice the user stations. In the case of Clear-Com, the system termination is not electronic but a passive resistor. If an adapter is made, the same trick can be done in a **Clear-Com®** system power supply. In the case of Audiocom® intercoms, paralleling two power supplies with capacitors would result in an impedance of 150 ohms which could still be usable in some instances.

System Powering

Systems can be centrally powered with a power supply or they may be individually powered with "local power" modules, also known as built-in power supplies. The systems can also be a mixture of central and local power. In the cases of **Audiocom®** systems and **RTS™ TW** systems, the power and signal share the same wire(s). This means, for those two systems, the power supplies DC source must be ultra low noise/quiet, circa -70dBu or better. Most systems can work using main powers of 120 or 240 volts AC. Some individual stations can be powered with 2 or 3 nine volt batteries in series. Venues such as the Rose Parade may have to use a pair of batteries from the telephone company just to cross the street. Since this may involve a mile of copper wire, there is no central DC source that's going to make it. Out come the nine volt batteries! The **RTS™ TW** power supplies can tolerate only a 5 volt peak-to-peak signal on the powered line. In this system, each station can generate a maximum 2 volt peak to peak signal, so two stations talking simultaneously can add up to 4 volts peak to peak. So, there is just 1 volt of headroom. Clear-Com specifies a range of signal levels of .5 v p-p to a maximum of 4v p-p, but doesn't specify the reference (it is probably dBu or dBv). **Audiocom®** intercoms specify only a nominal level of 1 volt RMS, which is equivalent to 3 v p-p.

Headset User Stations

The microphone preamplifier has a maximum gain in the neighborhood of 53 dB. Many stations have Automatic Gain Control (AGC), which adjusts the gain according to the incoming microphone signal. Some stations also have a limiter that prevents overloading the intercom line. An electronic switch is placed between the microphone preamplifier and the current source (line driver). This substantially reduces noise on the intercom line. A hybrid connection is necessary to sort out the talk and listen signals (a two wire to four wire converter would work best). The listen signal goes from the hybrid to the listen volume control. The listen volume control drives the headphone amplifier that has a gain in the range of 30 to 40 dB. For a 50 ohm headset, the headphone amplifier produces maximum peak sound pressure levels of around 105dB. This is the level needed at concerts and sporting events (along with 20dB acoustic isolation of the headset). In less strenuous situations, a handset instead of a headset may be used with these stations. These stations must have a bridging impedance of 10,000 ohms or higher. The current drains range from 30 to 65 milliamperes. Most systems have signal levels that range from -15dBu to 0dBu. In the case of **Clear-Com®** and **RTS™ TW** systems, the AGC / limiters in the microphone preamplifier tend to keep the level in the -10dB range. This enhances intelligibility and compensates for differences between voices and headset microphones. Usually the headset amplifier has enough gain to make up the differences (by readjusting the volume control).

Speaker User Stations

Most of these stations can operate in both a speaker/microphone mode and a headset mode. The difference between a headset only station and the speaker station is that a speaker amplifier, switching electronics, and a null pot are added. Usually the portable speaker stations use a push-to-talk microphone, whereas the fixed speaker stations use a panel or gooseneck microphone. The stations that have microphone and speaker on the same panel have less available speaker level because of feedback. The push-to-talk microphone has much better isolation. Speaker stations often have "dimming" or "ducking" which attenuate the speaker output when the microphone is keyed. This allows more gain and less feedback. Speaker stations use a very substantial amount of current, about 120 milliamperes. So, fixed speaker stations are ideally operated with local power, to prevent overloading the central power supply. Some RTS™ TW are direct AC powered and do not use central power.

Master Stations

These are multichannel stations. Some Master Stations are balanced (**RTS**™ **TW** Model 802/803) and require an interface (**RTS**™ **TW** Model 862 or 4012) to work with unbalanced channels. Master Stations can be configured to work with their respective systems with a minimum of interfacing. Master Stations have many functions which we go into to detail later.

Cabling

Usually the intercom system's specifications are based on the use of 22 AWG microphone cable. Microphone cable of 22 gage measures 3 ohms per 100 feet or about 30 ohms per 1000 feet (round trip resistance). The wire table says 32 ohms per 1000 feet round trip, but the shield resistance is much lower than the wire resistance. The **Audiocom®** system uses both wires and the shield to transport DC so the calculations will be different for DC voltage drop versus distance.

Outstanding Features of Each System

The **Audiocom®** system is immune to noise and is a lower cost system. It is used in difficult environments, i.e.: churches, concerts, theaters, and sporting events.

The **Clear-Com®** system is robust, relatively lower in cost, and rental systems are readily available. It is often used for concerts, rock-n-roll tours, and in theaters. It is also used in remote trucks, uplink trucks, and low budget venues.

The **RTS**™ system is also very robust, reasonable in cost, and rental systems are readily available in most countries world wide. Because the **RTS**™ intercom has two channels per microphone cable, it is used where many channels are required, such as the Oscar and Emmy award shows. It is also used for events such as the Superbowl. Most larger TV trucks carry both a four-wire system and an **RTS**™ Party-Line system. These systems are interfaced together so the four-wire is used inside the truck and the **RTS**™ system is used outside the truck.

In addition to these features, most systems support extra features such as, "microphone kill" and "call light". The microphone kill feature allows all microphones in a given channel to be switched off. In the case of Audiocom and RTS, the signal is an inaudible 24 kilohertz. In the case of Clear-Com, the power is interrupted for a long enough time to reset the microphones to off.

Call Lights

The Call Light Signal allows user stations to generate and display a visual signal for attention-getting and cueing purposes. The flashing light of the **RTS**™ and **Audiocom®** systems alerts the crew to put their headsets back on. The steady light of the **Clear-Com®** system can also be used for this purpose, however, it has another purpose: when the director holds the call light on, this is a standby signal. When the light goes off, this is the execute signal (raise/lower the scenery, follow spot on, et cetera). Call signals can also be used to key 2-way radios, sound alarms, and activate lighting controls. **Audiocom®** and **RTS**™ systems use an inaudible 20 kilohertz signal for the call signal; **Clear-Com®** systems use a DC voltage added to the audio signal. Telex manufactures a call signal detector / display (Model CIA-1000) which provides both a high visibility light and a relay closure when a call signal is sent. The CIA-1000 works with **RTS**™**TW** and **Audiocom®** systems. Clear-Com and other manufacturers also provide similar products. The company VMA supplies a bright strobe lamp that is triggered by the **RTS**™ system call signal. This strobe is powered from the RTS line but only draws 10 milliamperes. It also supplies a relay closure and a logic signal.

Limitations of Each System

Cable capacitance, resistance, and crosstalk affect all three systems. The longer cables (over 2000 feet) limit the number of belt packs at the end. A system with cumulative cables adding up to 10,000 feet will have a reduction in frequency response due to cable capacitance. Both resistance and capacitance affect crosstalk.

If all you have is a twisted-pair cable, then the **RTS**™ system is most useful. If you have severe coupling with power cables, the **Audiocom**® system will help.

Some of the information in this chapter is repeated in the next chapter, but in a different context.

Summary

(Some Definitions)

1 A Party-Line system allows a group of people to intercommunicate.

2 "Two-wire" means a communications system where the path is the same for talk and listen.

3 A balanced line reduces unwanted noise and crosstalk pickup.

4 A full duplex intercom allows simultaneous two-way conversations.

5 The human ear perceives a 10 decibel increase as twice as loud.

6 A belt pack is a user station designed to be worn on a user's belt.

7 A main station is a multichannel user station.

8 A master station combines a user station and a power supply.

9 Sidetone is a small amount of microphone signal fed back to the user's ear.

10 Crosstalk is unwanted interference.

(A Short History)

1 Television, theatrical, and concert production crews need Party-Line intercoms. Party-Line intercoms are also used for training and for industrial crews.

2 Early intercoms were inflexible and limited to small groups of users and sometimes short distances.

3 In the 1970s, fresh new designs were the beginning of the modern Party-Line intercoms we use today.

(Present Day Systems and Manufacturers)

1 Principal "two-wire" Party-Line brand names today are Audiocom, Clear-Com, and RTS. Other brand names are Chaos, David Clark, PortaCom, and Production Intercom.

2 With the exception of David Clark, present day Party-Line intercoms are the distributed amplifier type.

3 This format allows louder and clearer communication. Party-Line intercoms can be wired or wireless or both.

(System Components and Their Function)

1 The system components for most Party-Line intercoms consist of power supplies (or main stations), user stations, interconnecting cable, headsets, panel microphones, push-to-talk microphones, and a system termination.

2 The power supply (normally centralized) generates DC power for the entire system (exception: self powered user stations).

3 The power supply usually includes the system termination.

4 User stations connect to the power supply and intercom line(s).

5 For a given channel, user stations are connected in parallel.

6 The interconnecting cable for most intercoms is standard microphone cable.

7 For the major three intercoms (and their clones) there are three wiring schemes.

8 Wireless intercoms usually include and interface to the wired systems.

9 Wired intercoms are mostly of the distributed amplifier kind.

10 Distributing the amplifiers allows for better performance and more features.

11 A single channel belt pack has an intercom line connector, a headset connector, volume control, and a talk or microphone on/off switch. A two-channel belt pack adds a channel selector or two talk switches and two volume controls.

12 A speaker station usually can be a headset or a speaker station.

13 Speaker stations add a power amplifier, speaker, and speaker on/off switching to the headset station electronics.

14 Master Stations are multichannel and allow a director or lead person to have separate conversations with various crews in any combination. Master Stations often have many additional functions.

15 Dynamic microphones used with intercom stations usually range from 150 ohms to 500 ohms impedance. Electret microphones range from 1000 to 2000 ohms impedance and require 1 to 5 volts DC excitation voltage.

16 Headphones range in impedance from 50 to 1000 ohms. For concerts and athletic contests, 50 ohm headsets work better. The headphones should also have at least 20dB acoustic isolation for concerts and athletic contests.

17 The bridging impedance of each station should be 10,000 ohms or greater.

18 Since the power supply has a limited number of connectors, splitter boxes are needed to expand the number of user stations in a system.

19 Both male and female XLR-4 connectors are used to connect the headset with the user station. XLR-5 connectors are used for binaural headsets.

20 Some stations have loop through connectors to allow daisy chaining of stations.

(How Each System Works)

1 The stations use a voltage controller current source or similar electronics to apply signal to the intercom line, yet exhibit a bridging impedance of 10,000 ohms or greater.

2 Systems can be powered from a central power supply or local powered modules. Using local power modules allows more stations to be on the system.

3 If a station is too far away to get enough DC power, batteries can be used as a work-around.

4 Headset User Stations have a microphone preamplifier with a maximum gain around 53dB. Many stations have an AGC (Automatic Gain Control) that adjust the gain to the incoming microphone signal level. Some stations also have a limiter to prevent overload to the intercom line.

5 Headset User Stations have a hybrid function to convert the two-wire signal to a four-wire signal. The listen part of that signal is sent to the volume control and then to the headphone amplifier.

6 Portable Speaker User Stations usually have a push-to-talk microphone that gives good speaker to microphone isolation. Fixed stations have a panel microphone.

7 The cabling used in these intercom systems is usually called out as 22 AWG. Use of a smaller diameter wire such as 24 AWG shortens maximum distances and the number of user stations on a cable.

(Outstanding Features of Each System)

1 The **Audiocom®** system is immune to noise and is a lower cost system. It is used in difficult environments, i.e.: churches, concerts, theaters, and athletic contests.

2 The **Clear-Com®** system is robust, relatively low cost, available as a rental. It is often used for rock and roll tours, other concerts, in theaters, and in smaller outside broadcast trucks.

3 The **RTS™TW** system is very robust, reasonable in cost, rental systems are available almost worldwide. Used every place, but especially where many multiple channels are needed such as the Oscar ceremony.

4 Most larger TV trucks carry both a four-wire system and an interfaced **RTS™TW** system.

5 Call Lights and “mic kill” features are in all three major brands. The Call Light signal can be used to operate relays, radio keying, and warning lamps.

(Limitations of Each System)

1 **Audiocom®** and **Clear-Com®** systems require three wires for a single channel.

2 The **RTS™ TW** system may have crosstalk (but this is rarely a complaint).

3 All systems that use microphone cable are subject to distance limitations, as well as the number of stations per cable.

4 **Clear-Com®** systems and **RTS™ TW** system have less immunity to outside interference (practically speaking, rarely a problem).

Figure 2.1 **Audiocom®** intercom concept.

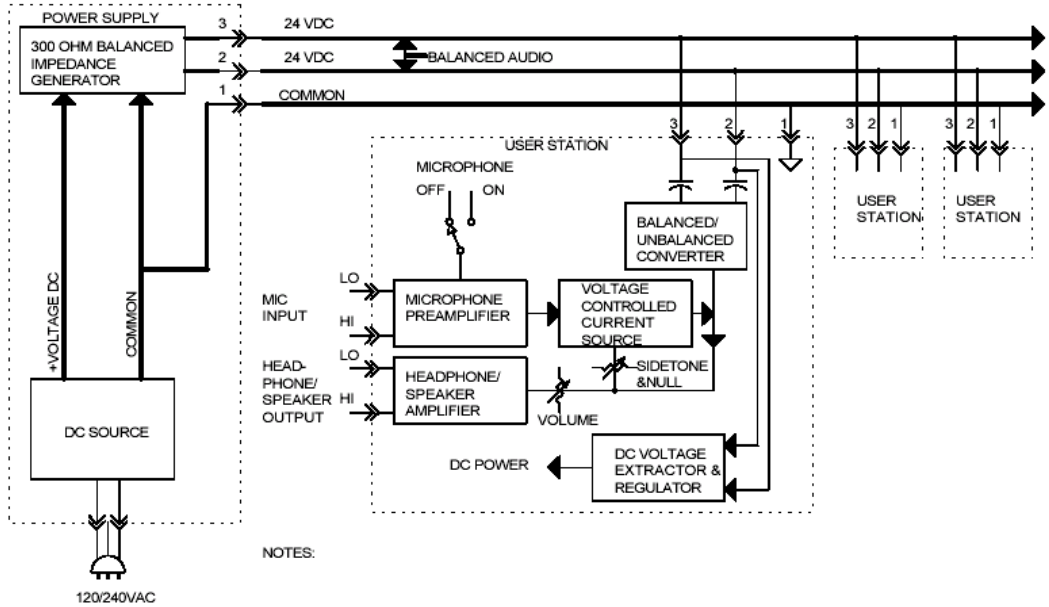


Figure 2.2 **Clear-Com®** intercom concept.

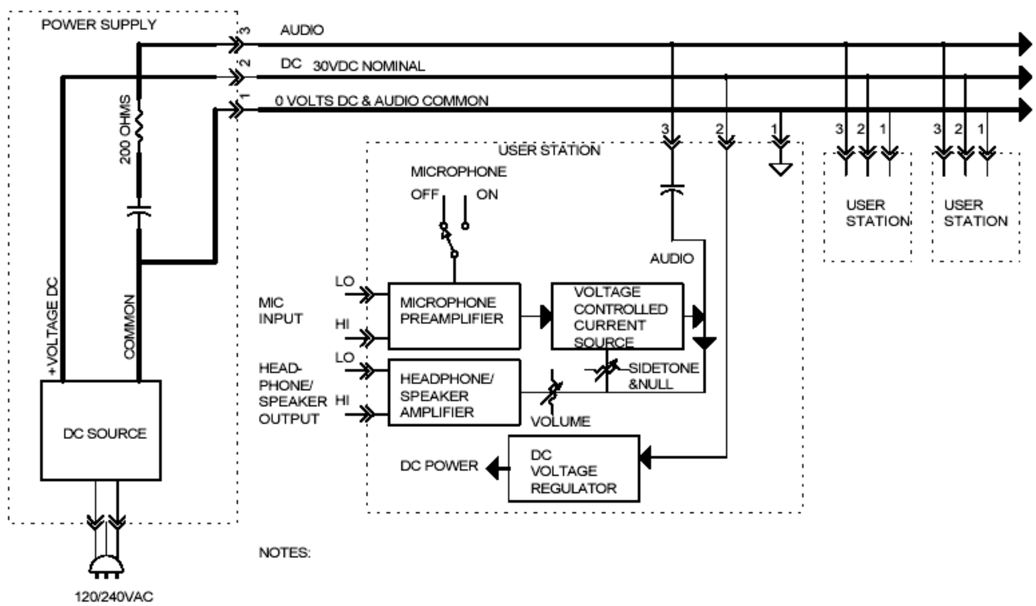
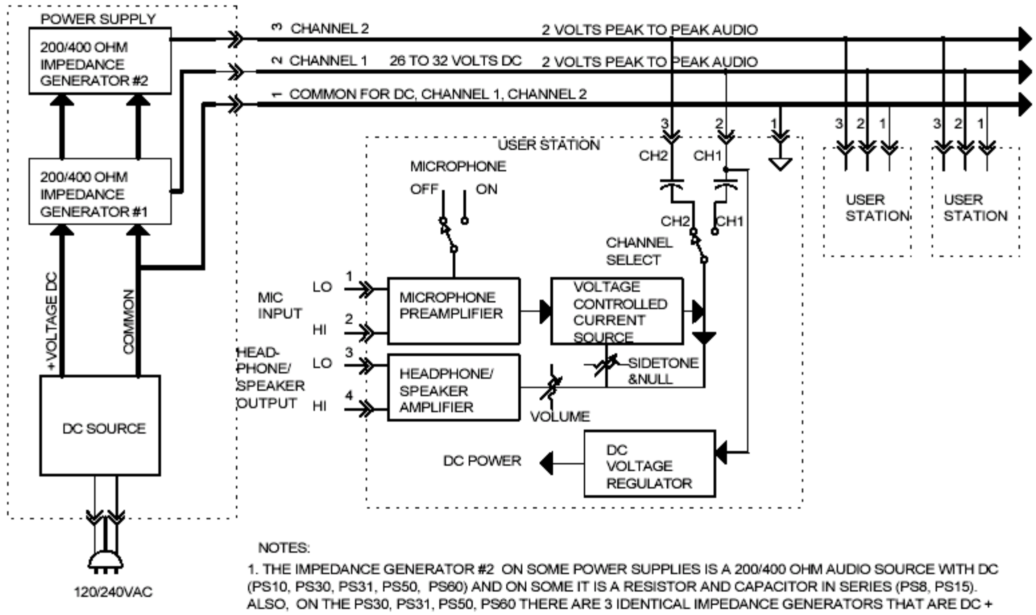


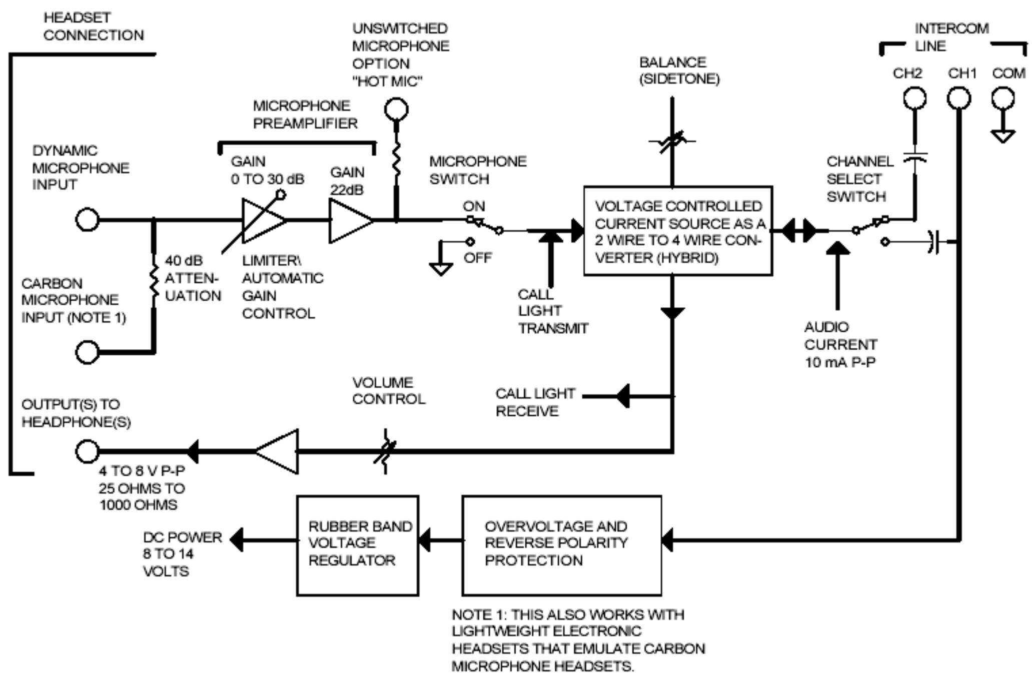
Figure 2.3 **RTS™ TW** intercom concept.



NOTES:

1. THE IMPEDANCE GENERATOR #2 ON SOME POWER SUPPLIES IS A 200/400 OHM AUDIO SOURCE WITH DC (PS10, PS30, PS31, PS50, PS60) AND ON SOME IT IS A RESISTOR AND CAPACITOR IN SERIES (PS8, PS15). ALSO, ON THE PS30, PS31, PS50, PS60 THERE ARE 3 IDENTICAL IMPEDANCE GENERATORS THAT ARE DC + AUDIO IMPEDANCE.
2. ON RTS TW POWER SUPPLIES, THE IMPEDANCE IS SWITCHABLE 200 / 400 OHMS TO ALLOW PARALLELING AUDIO WISE BUT SPLITTING THE LOAD DC WISE.

Figure 2.4 **RTS™ TW** user station block diagram.



CHAPTER

3

DESIGN OF PARTY-LINE INTERCOM SYSTEMS

STAN HUBLER

Overview

In this chapter, designing a system based on your needs is first approached by **Defining And Meeting Your Needs**. This topic is designed to help you choose or at least understand the system. IFB is described in **The IFB System (One Way Communications System)**. Then **Connecting (Interfacing) to Other Communications Systems** discusses real world solutions to interfacing these systems. The **Some Practical Considerations** section discusses real world environments and some work-arounds. A **Summary** closes this chapter.

Defining And Meeting Your Needs

Your needs could include buying, renting, assembling or expanding a system. Application Block Diagrams are a good starting place to define a system. In this section, block diagrams of applications in each of the three leading systems will be shown and discussed. These diagrams will range from relatively simple to complex systems. One of these block diagrams could be close to what you need to know, give or take a station or so. If you make a copy of the diagram and mark it up, this could define your system.

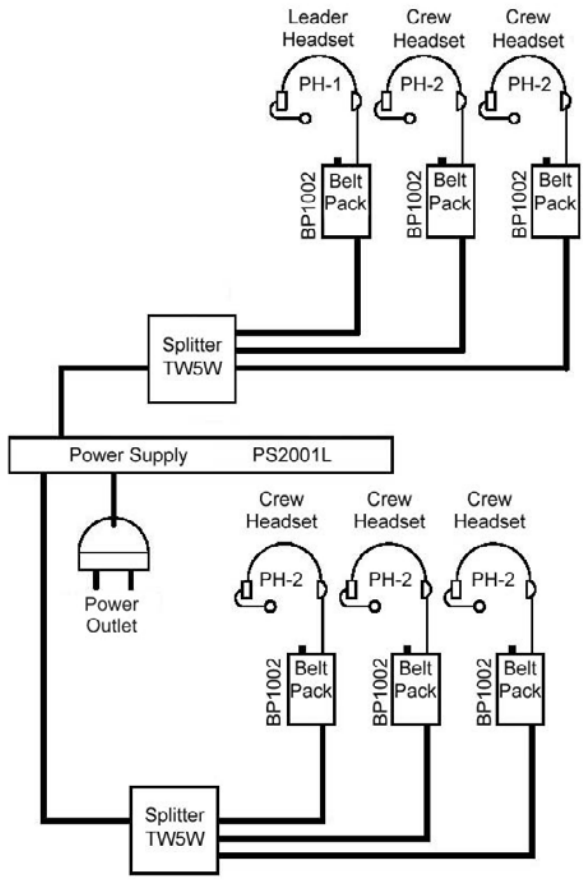
Disclaimer The block diagrams are for instructional purposes, and though every effort has been for accuracy, the manufacturers offerings are often changing. It pays to double check with the manufacturer or rental house to verify the exact system available before buying or renting.

Application 1 Generic Single Channel Systems

The first applications are generic single channel systems, see Fig.1.3. They consist of a power supply, belt packs, headsets, splitter boxes, and microphone cables. These are systems that could be used in a small television studio production, a small outside television field production, or an industrial test of a large system. Depending on the detail of the block diagram, you may be able to compile an equipment list from this diagram.

Audiocom Party-Line Intercom Equipment Listing #1

Figure 3.1 Generic single channel **Audiocom®** system.



Power Supply: PS2001L

Splitters: TW5W

Belt Packs, Single Channel: BP1002

Headsets: Leader Person: Single Muff PH-1; rest of crew: Double Muff PH-2

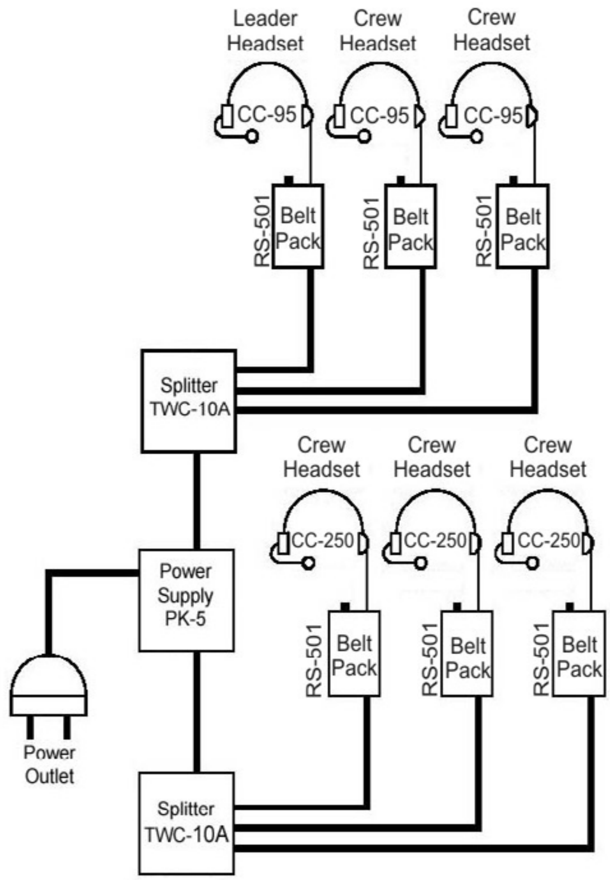
Cables: Standard Microphone Cables with XLR-3 connectors

The first block diagram, Figure 3-1 shows a simple single channel Audiocom intercom system. We start with a 2-channel PS2001L phantom power supply, two TW5W splitter boxes, two strings of three each single channel BP1002 belt packs, one PH-2 single muff headset and five PH-2 double muff headsets.

Note A switch on the PS2001L power supply allows both channels to be combined for one large Party-Line.

Clear-Com Party-Line Intercom Equipment Listing #1

Figure 3.2 Generic single channel **Clear-Com®** system.



Power Supply: PK-5

Splitters: TWC-10A

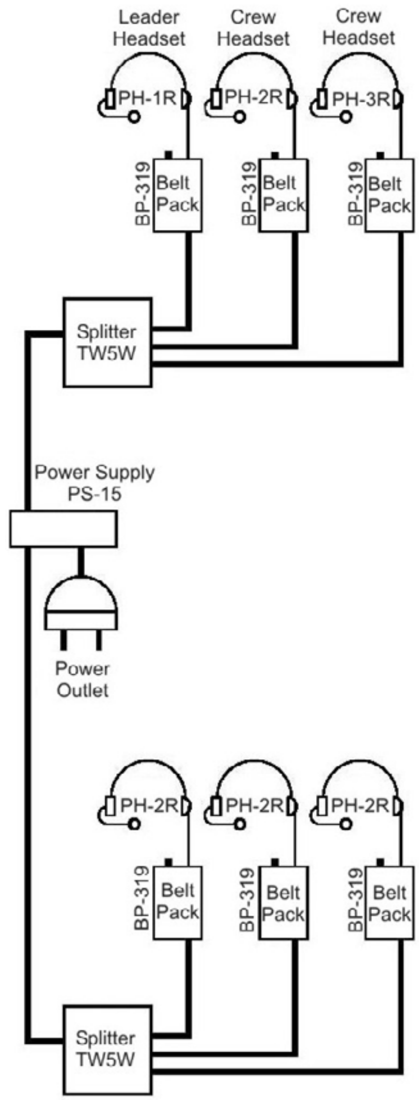
Belt Packs, Single Channel: RS501

Headsets: Leader Person: Single Muff CC-95; rest of crew: Double Muff CC-260

Cables: Standard Microphone Cables with XLR-3 connectors.

RTS TW Party-Line Intercom Equipment Listing #1

Figure 3.3 Generic single channel **RTS™ TW** system.



Power Supply: PS15

Splitters: TW5W

Belt Packs, Single Channel BP319

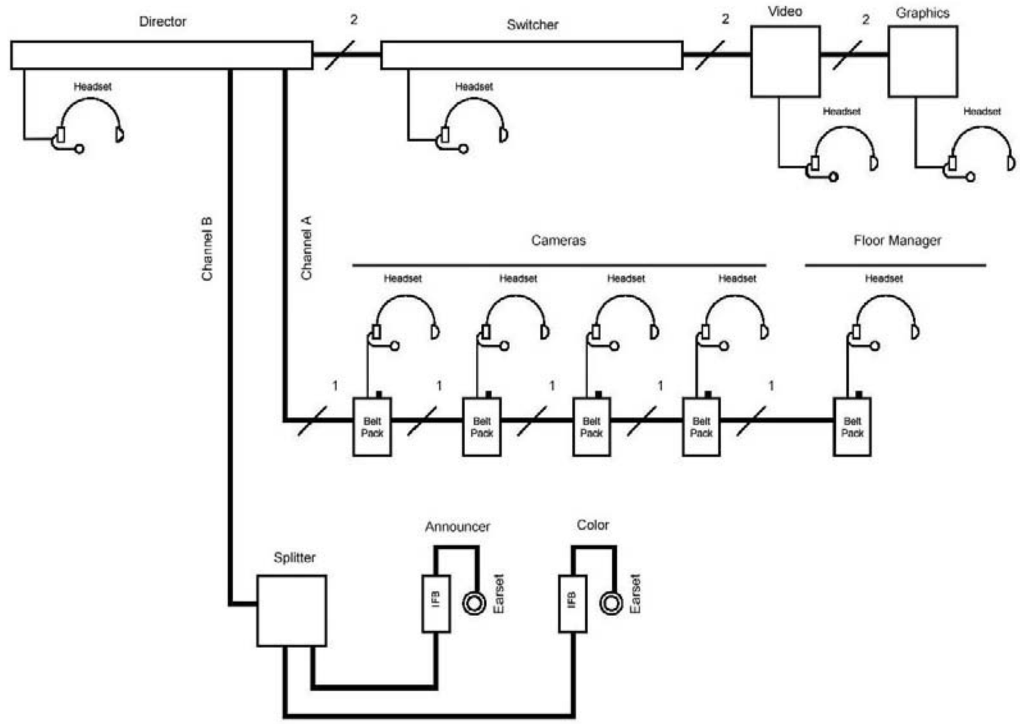
Headsets: Leader Person: Single Muff PH-1R; rest of crew: Double Muff PH-2R

Cables: Standard Microphone Cables with XLR-3 connectors.

Application 2 Two-Channel System: TV, School, Cable

The second application is a two-channel system for a small TV operation (Studio or Truck), school or cable access. The **Audiocom®** and **Clear-Com®** systems will require two 3-conductor microphone cables between director, switcher, video, and graphics. The **RTS™ TW** system only requires a single microphone cable for all hook-ups.

Figure 3.4 Small TV operation.



Audiocom Party-Line Equipment Listing #2

Power Supply: PS2001L (Rack Mount, 1RU)

Director's Station: US2002 (Rack Mount, 1RU)

Video: WM2000 (Wall Mount)

Graphics: WM2000 (Wall Mount)

Cameras and Floor Manager: BP1002 (Belt Packs)

Headsets Director, Switcher, Floor Manager, Video, Graphics: Single Muff PH1

Headsets: Cameras: Double Muff PH2

Splitter: TW5W IFBs: IFB-1000

Earphones (Earsets): CES-1

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per channel.

Note /2 indicates two microphone cables & /1 indicates one microphone cable.

Clear-Com Party-Line Equipment Listing #2

Power Supply: PS22 Rack Mount with RK-101 kit (2RU)

Director's Station: RM220 (Rack Mount, 1RU)

Switcher's Station: RM220 (Rack Mount, 1RU

Video: MR202 Wall Mount (2-gang box)

Graphics: MR202 Wall Mount (2 gang box)

Cameras and Floor Managers: RS-501 (Belt Packs)

Headsets Director, Floor Managers, Video, Graphics: Single Muff CC40

Headsets: Cameras: Double Muff CC60

IFBs: TR-50 (Includes earset)

Splitter: TWC-10A

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per channel.

Note /2 indicates two microphone cables required.

RTS TW Party-Line Equipment Listing #2

Power Supply: PS31 Rack Mount (2RU)

Director's Station: MCE325 (Modular Mount: Rack/Desk/Console (1RU)

Switcher's Station MRT327 (Modular Mount: Rack/Desk/Console (1RU)

Video: WM300L: Wall Mount (2 gang box)

Graphics: WM300 Wall Mount (2 gang box)

Cameras and Floor Managers: BP351 Belt Packs

Headsets Director, Floor Managers, Video, Graphics: Single Muff: PH-1R

Headsets Cameras: PH-2R

IFB's: IFB325 Earsets: CES-1

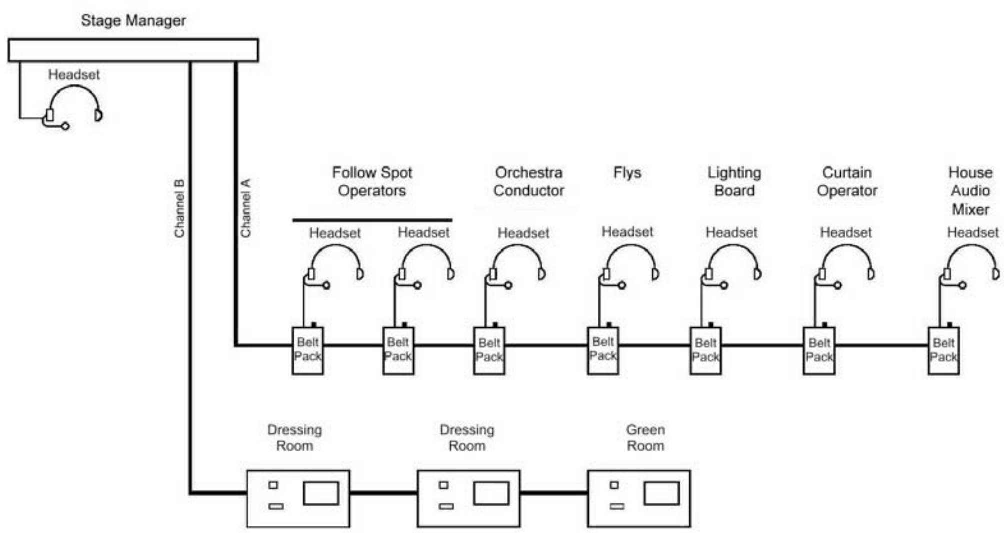
Splitter: TW5W Cables:

Standard Microphone Cables with XLR-3 connectors. One cable per two channels.

Note Ignore /2, both channels are in one microphone cable.

Application 3 Theater System

Figure 3.5 Theater application.



The third application is a theater application, see Figure 3-5. A two-channel system is used in this application. Channel A connects the crew together and channel B is used by the stage manager to cue the actors. This is done using three wall mount or portable speaker stations. For all three systems, only standard microphone cable is required. In the case of the **RTS**™ **TW** system, Channel B is available to the crew, but except for rehearsals or set-up they would stay on Channel A.

Audiocom Party-Line Equipment Listing #3

Power Supply: PS2001L (Rack Mount, 1RU)

Stage Manager's Station: US2002 (Rack Mount, 1RU)

Dressing Rooms and Green Room: SS1002 (Single channel wall mount station; if a portable speaker station is desired, add an S, U, or P box).

Crew: BP1002 (Single Channel Belt Packs)

Headset: Stage Manager Single Muff PH1

Headsets: Crew: PH2

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per channel.

Clear-Com Party-Line Equipment Listing #3

Power Supply: PS22 Rack Mount with RK-101 kit (2RU)

Stage Manager's Station: RM220 (Rack Mount, 1RU)

Dressing Rooms and Green Room: KB-212 (Single channel wall mount speaker station, if a portable speaker station is desired, add a V-Box portable enclosure.)

Crew RS-501 (Single Channel Belt Packs)

Headset: Stage Manager: Single Muff CC40

Headsets: Crew: Double Muff CC60

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per channel.

RTS TW Party-Line Equipment Listing #3

Power Supply: PS31 Rack Mount (2RU).

Floor Manager's Station: MRT327 (Modular Mount: Rack/Desk/Console (1RU).

Dressing Rooms and Green Room: SS1002 (Single channel wall mount speaker station).

Crew: BP319 Belt Packs (Set to work on Channel A).

Headset: Stage Manager: Single Muff: PH-1R.

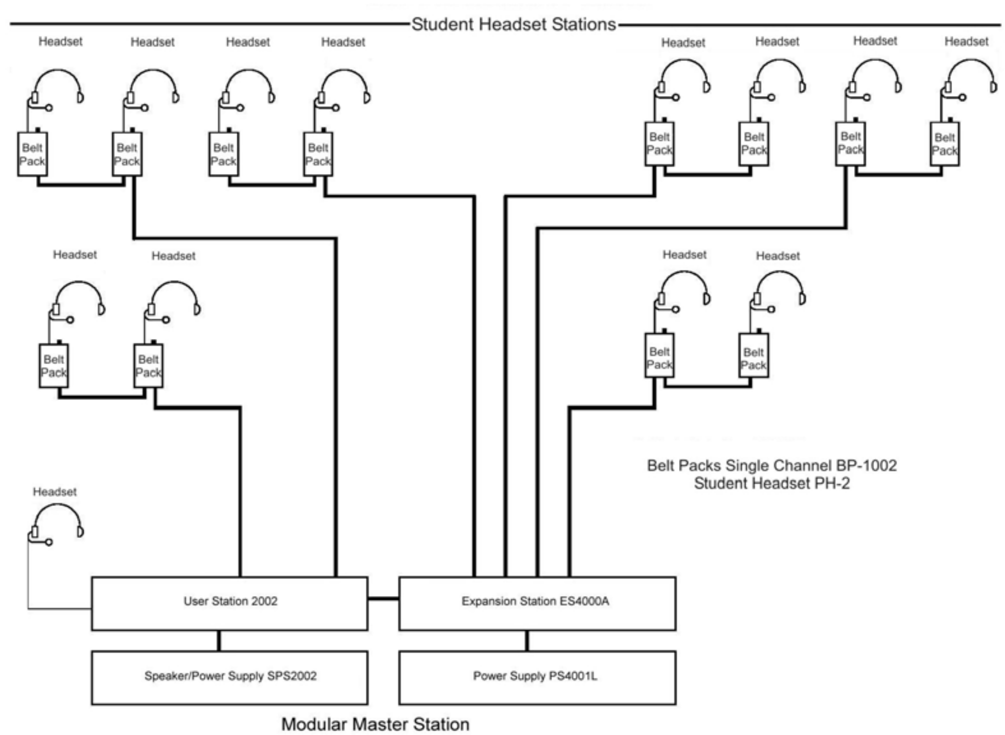
Headsets: Crew: PH-2R.

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per two channels.

Application 4 Training Systems

Audiocom

Figure 3.6 **Audiocom®** based training intercom system.



The training system consists of an instructor and multiple two-student crews.

In the case of Audiocom, each of the six two-student groups are independently addressable by the instructor. When the student groups are not talking to the instructor, each two-student group can have semi-private conversations. The call light tells the instructor which group is paging. The balanced Audiocom system is ideal in hostile electrical noise environments.

Power Supplies: SPS2001 and PS4001.

Instructor's Station: US2002 and Expansion Station.

Students' Stations: BP1002 Single Channel Belt Packs.

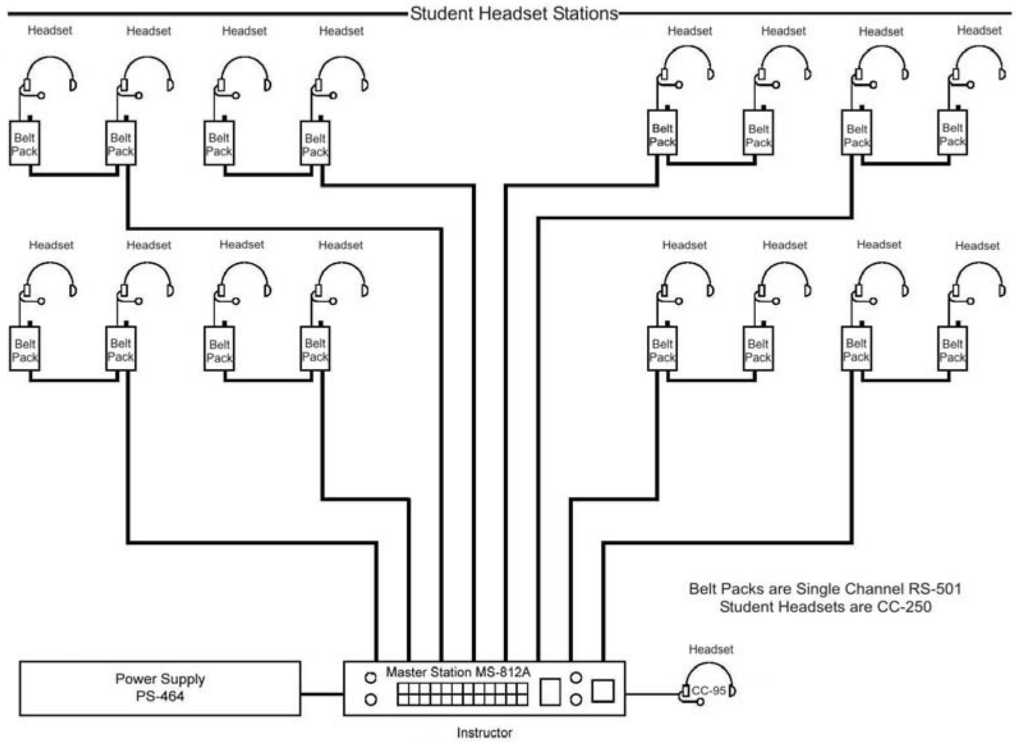
Instructor's Headset: PH-1, Single muff headset.

Students' Headsets: PH-2, Double muff headsets.

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per channel.

Clear-Com

Figure 3.7 **Clear-Com®** based training intercom system.



It just happens that the **Clear-Com®** system is the simplest for this application, since the Master Station, MS-812A has the three pin XLR connectors for 12 channels on the rear panel. The MS-812 has several configurations, and will have to be specified for this application (No IFB, 12 Clear-Com standard PL channels).

Power Supply: PS-464.

Instructor's Station: MS812A.

Students' Stations: Single Channel Belt Packs RS-501.

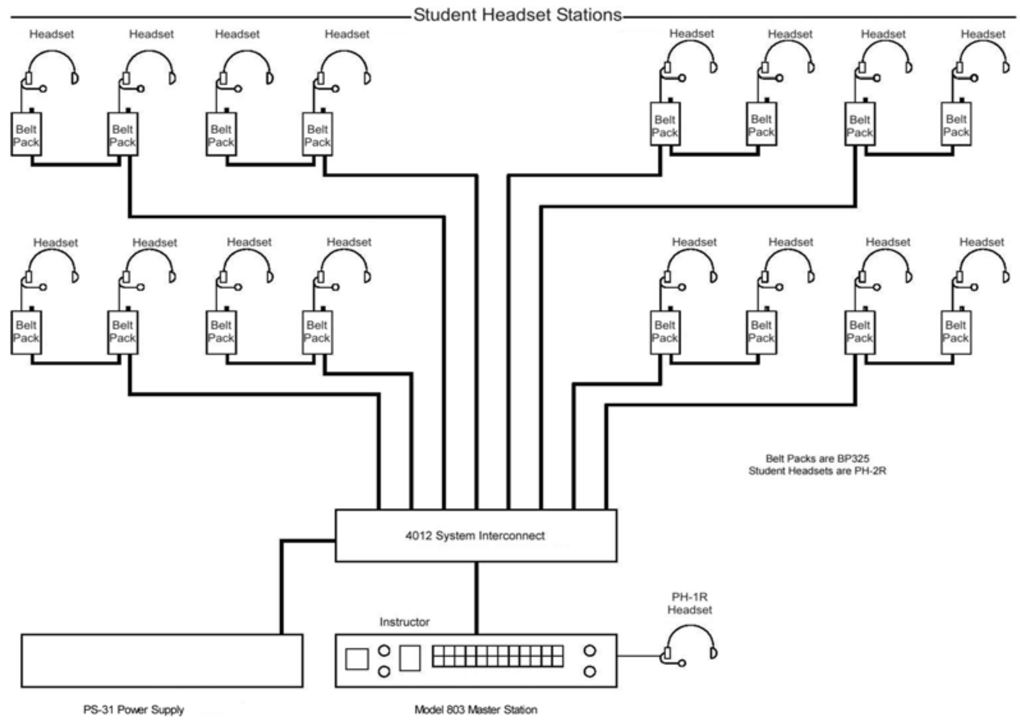
Instructor's Headset: Single Muff CC-95.

Students Headsets: CC-250.

Cables: Standard Microphone Cables with XLR-3 connectors. One cable per channel.

RTS™ TW

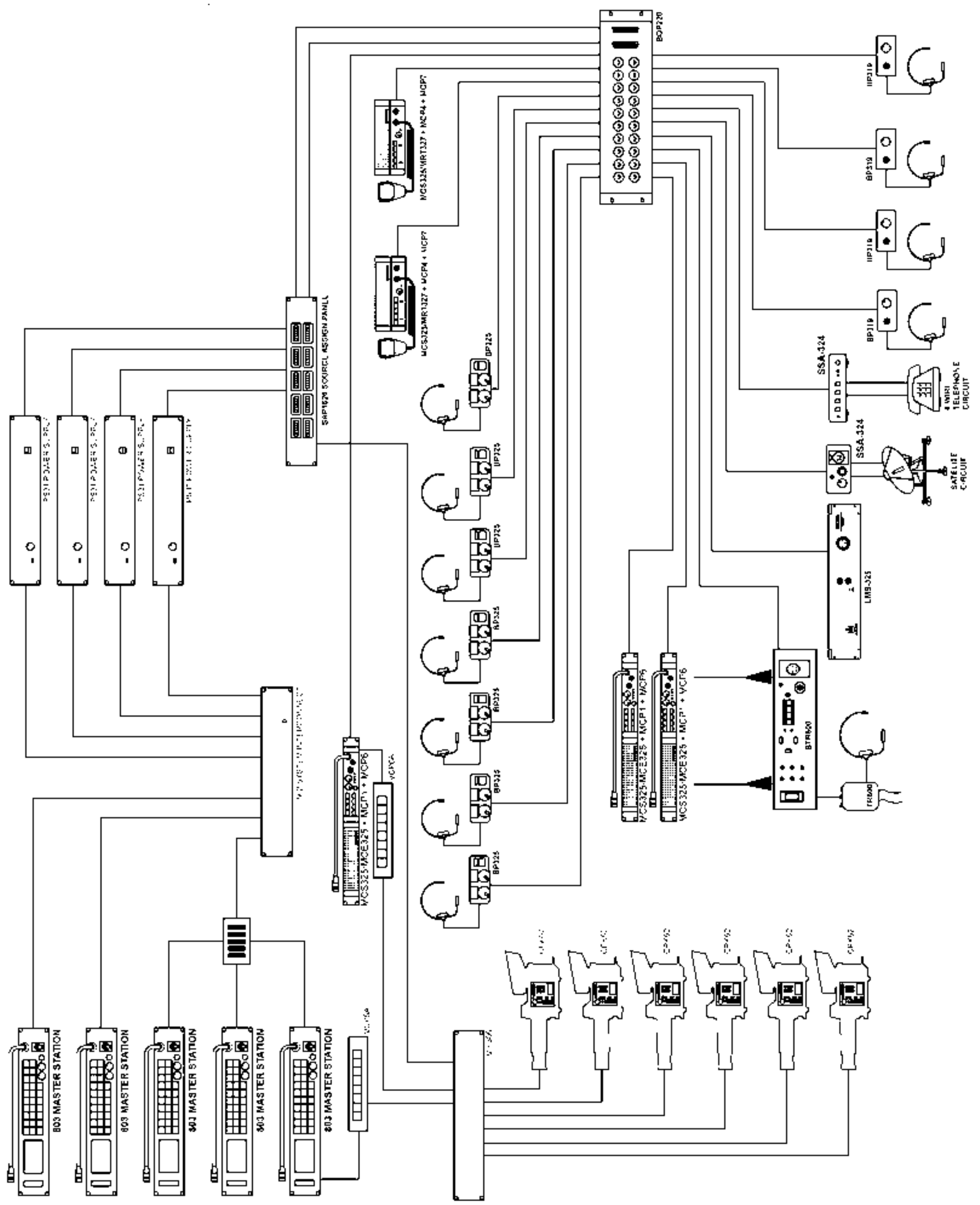
Figure 3.8 **RTS™TW** based training intercom system.



The **RTS™ TW** system for this application is the next simplest, and has added features. The student crews can have completely private conversations, yet are still reachable via the call light paging system. Each BP325 belt pack can be configured to accept an individual program source (but the loop-through is lost and the two students line connection will be through a simple one to two splitter). The program source is often a training audio/video tape, along with a monitoring computer tests the reaction time and correctness of the students reaction.

Application 5 Medium System for Television

Figure 3.9 Medium intercom system for television.



This shows an **RTS**™ **TW** large 12-channel system. This is a system that is in medium trucks that haven't yet switched over to a combination matrix and Party-Line system. This system consists of five Model 803 Master Stations, four PS31 Power Supplies, one SAP1626 Source Assign Panel, a BOP220 Break Out Panel, a VIE Video Isolate Panel, a four belt pack Telex BTR600 Wireless Intercom, and various belt pack and other user station. Also are interfaces to a telephone and a satellite communication link. Many trucks have a similarly configured Clear-Com® system. The Master Stations are usually for: the Director, Assistant Director, Lighting Director, Audio Mixer and Video operator (the one with the VCP6A isolate panel. No IFB (Interrupted FeedBack) is shown in Figure 3-9, but an IFB system is easily married to the Master Stations. A large **RTS**™ IFB add-on is shown in Figure 3-10. Note that Model 4020 is now Model 4030. Similar **IFB** systems are available from Clear-Com and Audiocom. The Control Station connects to the "Hot Mic" output of a Master Station or User Station with a “Hot Mic” output. The IFB electronics receives its program audio from the audio mixer board.

The IFB System (One Way Communications System)

IFB is a television acronym for Interrupted FeedBack, Interrupted FoldBack, Interrupted Return Feed (IRF). An IFB system permits a director or producer to talk to the talent, typically an “on air” announcer, newscaster, or sportscaster. Normally the talent hears the broadcast program audio. When the director or producer activates the IFB, the program audio is replaced by the director’s or producer’s voice. Sometimes the program audio continues in the other ear, sometimes the program audio is reduced instead of completely removed.

How an IFB Works

Those in control positions (the director, producer, or assistant director for example) control the interrupt and or announce functions via control stations. Those in receive positions (on-air talent, floor managers, studio or field crew, audience, talent and crew in remote locations) are on the receiving end of the user station feed or on the actual user stations (talent electronics or talent station) via headphones, headsets, earphones, and / or loudspeakers. In the middle, the central electronics unit provides all the necessary inputs and outputs, processing, switching, and power distribution.

Studio and Some Field Applications

Note Model numbers of the different parts of the IFB are as follows:

Control Panel
Audiocom®: Built into US2002, ES4000A. Clear-Com: MA-4, AX-4. **RTS™ TW**: Models 4001, 4002, 4003

IFB Electronics
Audiocom®: Built into US2002, ES4000A; Clear-Com: PIC4000B; **RTS™ TW**: Model 4010

Talent Receiver
Audiocom®: IFB1000; Clear-Com: TR-50; **RTS™ TW**: Model 4030

Earset
Audiocom: CES-1; Clear-Com: (part of Model TR50); **RTS™ TW**: CES-1

In non-sports activities, the talent normally uses only the interrupt output (mono) of a Talent User Station. The earphone is hidden behind the talent’s back; a plastic tube runs from the earphone to the talent’s ear.

Field Application, Sports

In the sports broadcasting or sports communication field, the talent uses a noise resistant headset. The microphone on the headset is the “air” microphone; the headphone is double muff, stereo. The talent is plugged into the stereo output of (for example) the Model 4030 Talent Receiver User Station. At the IFB Control Station, each talent’s name is marked on a strip of tape pasted adjacent to the push buttons.

In stadium sports, there is usually little problem in getting a microphone cable from the IFB Electronics to the Talent Receiver. In the case of golf, auto racing, and sports venues over an extended area, the distances may be too great. In this case, a four wire circuit can be run to the talent location and adapted to the connector on the Talent Receiver.

In some more extreme cases, only a single pair of wires may be available. In this case, plug the talent's stereo headset into the stereo connection on the talent receiver, then connect the high side of the pair to pins 2 and 3 of the XLR3 connector and the low side to pin 1 (pseudo-stereo mode). This will give a mono feed with each ear individually adjustable and both ears interrupted.

For runs of two miles of number 22 gage twisted pair, at least one talent receiver station should be operable. For a run of one mile, two talent stations should be operable.

Some users have increased the number of talent stations by using higher impedance (300 ohms) headsets. In the case of auto racing and similar loud environment situations, low impedance noise isolating headsets will be necessary to overcome the volume and amount of sound. It may be necessary to use a four wire circuit to connect up each talent station, paralleling the pairs, and running the talent receiver in pseudo-stereo mode, using only the interrupt ("wet") output of the IFB electronics.

Field Application, ENG (Electronic News Gathering)

In this case, the earphone is again hidden as in the studio case above. If the talent has to carry on a conversation with other talent at the studio and other venues, the program feed should be a mix minus feed. The mix minus feed will allow the talent to hear the other talents loud enough without hearing their own self too loud.

Connecting (Interfacing) to Other Communications Systems

What is interfacing? Interfacing is either:

1 The interconnection of two normally separate communications systems into one system.

-OR-

2 The connection of a communications station or device that is not directly compatible within a system.

To accomplish this, voice and data information is adjusted and then transmitted to the other system. The adjustments include level translation, impedance compensation, mode translation, and compensation for parameters of each system.

Some examples are:

1 *System to system*: connection of a four-wire matrix system installed on a large mobile unit to two-wire belt packs outside of the mobile unit.

2 *System to terminal*: connection of a camera with a built-in intercom to an intercom system, or connection of a radio transceiver into an intercom system.

Why is there interfacing, operationally? From an operations point of view:

1 An operation requires a larger collection of personnel and equipment than normal.

2 A mobile unit is used with a permanent installation to conduct an operation.

3 Coordination between personnel / equipment is required at a remote location.

4 A special part of the operation requires communication with an odd system or terminal.

5 A redundant "backup" path is required.

Why is there interfacing, technically? There are system to system, system to terminal or, system to device differences.

Some of these are:

1 *Mode differences.* There are several not directly compatible modes of operation: two wire mode, four wire mode, full duplex mode, half duplex mode, simplex mode. Examples: the **TW** System is two-wire full duplex, the **ADAM**™ matrix is four-wire full duplex, the telephone is two wire full duplex except some long distance calls are half duplex (both people cannot talk at once), a walkie talkie is simplex, AudioCom is two-wire full duplex, Clear-Com is two-wire full duplex, office intercoms are often simplex operation.

2 *Level and Impedance differences.* System voltage levels range from - 40 dBu to + 21 dBu with peaks to +28 dBu (where 0 dBu = 0.7746 volts). See Table 3.1, for typical ranges.

Table 3.1 Typical system impedances and ranges.

Intercom or Audio System	Nominal Impedance (Ohms)	Nominal Level (dBu)	Level Range (dBu)
Telephone	600 to 900	-15	-40 to 0
Old Clear-Com	200, 10k	-30	-45 to -15
New Clear-Com	200, 10k	-14	-14 to +5
Audiocom	300, 10k	0	-8 to +1
RTS™ **TW**	200, 10k	-10	-10 to -1
Recording Studio	600, 10k	+4	-6 to +24

There are different modes of intercom operating modes because each mode offers a different advantage for different needs and situations. For example, two-wire is quick and easy to hook up, while four-wire is easier to interface to other systems.

A Typical Interfacing Problem

A television camera uses a triax cable to connect the camera to the rest of the electronic system because a triax cable allows operation over longer distances with more consistent quality. This is because the triax cable uses radio frequencies to transmit information both ways on the cable. This is, in effect, four-wire (two path) communication. The following implementations often need interfacing:

1 Television camera intercoms to intercom systems.

2 Two-wire systems to four-wire systems.

3 Full duplex systems to simplex systems.

4 When transmission medias change.

Interfacing Issues

There are three tasks to interfacing:

1 Mode Conversion.

2 Level Problems.

3 Signal / Data Conversion.

Modes

The following modes exist in intercom systems:

M2) Two-Wire.
M4) Four-Wire.

The following sub-modes are considered for two-wire and four-wire:

M2F) Two-Wire, Full Duplex.
M2H) Two-Wire, Simplex.
M4F) Four-Wire, Full Duplex.
M4H) Four-Wire, Simplex.

Level Problems

One problem in interfacing from two-wire to two-wire is caused by the 2 wire systems' use of 2 to 4 wire hybrids. Interfacing requires conversion from two-wire to four-wire twice to allow level adjustments to and from systems. The quality of the two-wire to four-wire hybrid limits the amount of make-up gain available to match levels in one system or the other.

Another problem with interfacing is that level adjustment is difficult when interfacing from a limiter controlled system, such as the **TW** Intercom System, to a non-limiter controlled system, such as some two or four wire systems. The reason for the difficulty is that the perceived loudness is greater on the TW System and much less on the non-limiter controlled system. This difference can be improved or eliminated depending on two limiting factors: 1) the headroom of the electronics involved, and 2) the quality of any two-wire to four-wire hybrids in the path. Interfacing from two-wire to two-wire systems is the most difficult. Interfacing from two-wire to four-wire is easier, and interfacing from four-wire to four-wire is the easiest. The problem in two-wire / two-wire interfacing is getting the levels right and preventing oscillations.

The level of the **TW** and 800 Series conference intercom systems ranges from -10 dBu to 0 dBu, with an average value of - 6 dBu, and is limiter controlled.

Some other systems are listed in Table 3-2. The objective is to convert the modes and to adjust the levels.

Signal / Data Conversion

Call Light

Some intercom systems use a "Call Light" signal to illuminate lights in individual stations. This signal may be a 20 kHz tone, a DC level, or a digital logic level. An interfacing device may handle the method conversion to carry the call light signal.

Data

Other systems have data flow via various methods including: contact closure, logic level, RS485 bus, RS422 bus, and RS232 bus. The handling of the RSxxx signals is done best on a case-by-case basis. At this point, system-to-system communications is done via RS232 communications by wire, fiber optic, or telephone lines via modem. Some system-to-system communication is accomplished through user specified hardware imbedded in special products.

Some Master Stations have an RS232/485 connection that allows control of the station over a terminal or another computer.

Interfacing Practice

Interfacing Television Camera Intercom Systems to TW Systems

General Camera Configuration Information for Television Cameras (except ENG units)

Television cameras used in broadcast and industry usually have two parts: a camera head and a camera control unit (CCU). The camera head assembly usually contains the lens equipment, camera electronics, and triax adapter (if used). The CCU contains additional electronics for processing video, the other end of the triax adapter, an interface for microphone audio, and the intercom interface. The intercom interface usually incorporates switches and electronics so that the intercom can be two-wire or four-wire.

The Problems in Interfacing to Cameras

There are two problem areas in television camera intercoms:

1 The electronics in the camera head.

2 The intercom interfacing electronics at the CCU.

Some possible problems with the camera head intercom electronics are as follows:

- Inadequate headphone drive (Not loud enough for athletic contests and studio shows)
- No limiter in the microphone preamplifier (level variations are too much)
- The headphone and the microphone share a common circuit return conductor (headphones oscillate when volume is turned up)
- The Triax Adapter / electronics does not give the camera intercom enough headroom, so there is a trade-off between signal clipping and signal to noise ratio.
- The microphone on/off switch does not disconnect the microphone preamplifier thus adding noise to the system.
- Some possible problems with the CCU intercom interface electronics are:
- An earth ground is applied to the wiring usually in two-wire mode (causes hum loops in the system)
- The four-wire input to the camera is not bridging impedance
- The two wire “RTS0153 Systems compatible” interface loads the line
- No safety capacitors are installed in the CCU, thus causing burnt transformers if connected to the intercom line

Alternatives for Interfacing to Television Cameras

1 Bypass the camera, tape a microphone cable to the camera cable, and plug a TW belt pack in at the end.

2 Use the existing camera intercom, interface it to the TW system with a Model SSA324 or SSA424 interface (if camera intercom is four-wire).

3 In multi-core connected cameras, use the camera wiring to allow a TW belt pack to be plugged into the camera head. This allows the camera operator to use a portable User Station mounted on his belt or attached to the camera body. (

Note: This requires significant modification to the camera head and CCU)

Table 3.2 Intercom comparisons.

Intercom Type	Nominal Impedance (Ohms)	Impedance Range (Ohms)	Output Type	Mode	Estimated Peak TX Power (mW)	TX/RX Levels (dBu)
TW	200	50 to 400	Un-Bal	two-wire	5	0 to -10
TELCO	600	600 to 900	Bal	two-wire	1	0 to -10
Two-Wire	150 to 200	100 to 1k	Un-Bal	two-wire	0.7	-10 to -20
Four-Wire	600	600 to 10k	Bal	four-wire	7	+8
Carbon Mic	150*	4 to 150	Un-Bal	two-wire	2	0 to -30

TELCO = Telephone-lines in two-wire mode

Two-wire = Clear-Com, ROH, HME, R-Columbia, Protech, Theatre Techniques, Telex**, some television cameras

Four Wire = **RTS**™ **ADAM**™ intercom, Philip Drake, Link, McCurdy, Ward Beck, ADM, Farrtronics, PESA, Audix, Datatronics, all triax television cameras, some multi-core television cameras, Radio-telephones, Telephone-Line circuits, Wireless Intercom systems

Carbon Mic Interphone = RCA, Daven, Video Aids, General Electric, Colorado Video, many low-cost television camera intercoms

* Per Station
** Telex(r) Phase 2 = 300 ohm, 5 mW balanced line.

Some Practical Considerations

Headset Cable Lengths

The dynamic (low level) headset cable carries signal levels that differ by as much as 34 dB + 52 dB = 86 dB. Ordinarily, there are three types of unwanted coupling possibilities: resistive (through a common ground), capacitive and inductive. Since separate grounds are carried back to the microphone preamplifier and headphone amplifier, the common ground resistive coupling is, in this design, negligible. The capacitive coupling can be made non-significant by a 100% shield in the cable. The inductive coupling mode dominates in this design, and can be offset in several ways:

- The distance between the microphone and headphone pairs can be increased, while the mutual inductive coupling is decreased by the use of “ribbed” cable (two cables molded together side-by-side).
- Both the microphone cables and the headphone cables can each be tightly twisted.
- Two or four separate cables can be run. A balancing transformer on the microphone circuit may be used. Estimated, Safe Operating Distances are as follows:
  - Single cable, two shielded twisted pair: 10 feet.
  - Dual ribbed cable, two shielded twisted pair: 30 feet.
  - Separate cables, shielded twisted pair in each: 50 feet and more.
  - Balanced microphone input: up to 100 feet depending on cable used.

Headphone Impedances

Low impedance headphones are louder, causing the user station to draw more current from its power source. High impedance headphones are not as loud, drawing less current. Many user stations have a headphone impedance range from 25 - 600 ohms.

Headphones up to 2,000 ohms will function but greatly reduced levels. In a double muff headset such as a Beyer DT-109, there are two 50 ohm headphones connected in parallel resulting in an impedance of 25 ohms.

Wiring Practices/Workmanship Standards

The two most significant wiring practice/workmanship problems are as follows:

1 Unintentional grounding, phase reversals (channel reversing) and power reversal. Cable shields must not touch connector shells or be tied to the connector shell lug. Cables (especially the vinyl insulated type) must not be pulled tight around sharp edges.

2 Line noise due to an intermittent connection:
Poor solder joint.
Corroded connector.
Loose screw terminal.
An non-insulated cable shield touching the metal shell of the connector.

Portable user stations should not arbitrarily be taped or fastened to metal structures. Grounding the case of the user station to an arbitrary structure may introduce large noise voltages due to local ground currents or due to the completion of a "ground loop antenna".

Phase reversals are most common with portable microphone cable that has not been checked with a standard cable tester after fabrication or repair.

DC power reversals are usually not harmful to user stations since there is normally a protective diode in the circuit. The station simply doesn't work. Remember: negative is ground in this system.

Always clear all earth grounds from the RTS™ TW System circuit return ground. The only ground should be the 22,000 ohm resistor in the power supply.

Unbalanced vs. Balanced

Intercom systems such as the TW System, in the standard, unbalanced configuration have been operated at distances of up to two miles with acceptable system noise levels. Routing the intercom cables along the same ductways and pathways as the main power cabling can increase the noise and hum levels in the system.

If intercom cables have to be routed in this manner at distances over 300 meters (1,000 ft.), a balanced conversion should be made.

Alternatively, the entire system can be operated in an optional balanced mode and be powered at each station with the "local power" option. This is sometimes called "dry line, balanced" operation.

Extended Range On Part Or All Of The System

If a station is locally powered, operational range can be extended up to five miles, using two transformers to step up the line impedance to 800 ohms (for lower losses). When the users station has the four wire / 800 ohm option installed, operation is possible up to 20 miles along Telco dry pairs. Operation over longer distances (3000 miles) is possible using dial up or minimum loss dry lines and the TW series of interfaces.

Cable Considerations

Crosstalk

Use shielded cable to interconnect user stations in areas of possible electrical interference, (areas such as those near: digital equipment, high current primary power conductors “power outlets”, transformers, transmitters, and lighting dimmers. Do not run TW Intercom System cables along the same ductways and pathways as these cables.

Standard wire size for the an intercom system interconnection is #22 gauge shielded cable, such as Belden 8761, 8723, 9406.

In permanent installations, to reduce both capacitive and resistive crosstalk and to afford a degree of RF and electrostatic shielding use a cable that has a shielded twisted pair for each channel, such as Belden 8723. Each pair consists of a conductor for the channel, a conductor for circuit ground return and shield around the two conductors. The shield is accessed via a drain conductor. This drain conductor and the shield can augment the circuit grounds and thus lower the ground resistance. Do not tie the shield to chassis, earth, or connector shell ground.

Crosstalk Through A Common Circuit Ground

Since, in the unbalanced version of a TW intercom, all channels share a common circuit ground return, crosstalk due to common ground resistance can occur. This crosstalk is proportional to the ratio of the common ground resistance to the system terminating impedance, 200 ohms. This occurs when a talker on one channel is heard by a listener on another channel due to the common ground resistance (see Figure 8-4). Reduction of this crosstalk can be accomplished by reduction of the circuit ground resistance. Reduction of the ground resistance can occur as a side benefit of using shielded cable, since the shield drains can be tied together and electrically parallel the circuit ground.

Another way of lowering this kind of crosstalk is to “homerun” all interconnecting cables to a central or “home” location. This causes the common circuit ground path to be very short, and other things being equal, makes a low common ground resistance.

Crosstalk Through A Mutual Capacitance Of Two Conductors

Two conductors such as a twisted pair can accumulate a large mutual capacitance over long distances. Using a figure of 100 picofarads per meter and a distance of 1 kilometer, results in a total capacitance of 100 nanofarads or 0.1 microfarad. The reactance of 0.1 microfarad at 800 hertz is 2000 ohms. Referred to the system impedance of 200 ohms, the apparent crosstalk is about 20 log (200/2000) or about -20 dB. Separating the two channel conductors by a shield greatly reduces the capacitive crosstalk, so that the resistive crosstalk discussed above dominates.

A Low Crosstalk Approach To Interconnection

To reduce capacitive and resistive crosstalk and to afford a degree of “RF” and electrostatic shielding, a shielded, twisted pair per channel type cable can be used. Each pair consists of a conductor for the channel, a conductor for circuit ground return and, of course, the shield as a conductor and the shield drain conductors. These drain conductors and the shield can augment the circuit grounds and, thus, lower the ground resistance.

Distances/Conductor Sizes/Distributed vs. Central Connection

Systems that stretch over distances of kilometers are more subject to power losses and crosstalk. These problems can be minimized through the use of large enough wire, shielded cables and central connections.

System Current/System Capacitances/Loading

The system currents are determined by several parameters:

1 The current required to supply standby current for each user station.

2 The current required to supply the dynamic current to generate line signal, headphone signals, speaker signals and call lamp signals.

3 The current required to start up a system (inrush current) by charging up to (50) 4000 microfarad capacitors or 0.2 farad.

4 The current limit imposed by the power supply to protect itself.

5 The secondary current limit imposed by the power supply when a fault is close to the power supply (little or no circuit resistance). This limit, called the foldback current, further protects the power handling electronic devices in the supply and determines the system start-up time.

Currents 1 and 2 can be calculated by multiplying the number of user stations times the user station current data in the Complete User Station Specifications. Current 3 is usually limited by current 5. Currents 4 and 5 are listed in the Power Supply Specifications. Current 5 can be used to calculate the system start-up time: where:

T is the start-up time (approximated) in seconds.
N is the number of stations.
C is the capacitance per station = 4 millifarads
i is the power supply foldback current
dV is a change in voltage across the capacitors, say 10 volts.

For a 20-station system, a 1 ampere foldback current, and a 10 volt change on the capacitors:

The actual system start-up time will be longer since voltages in each user station have to stabilize before audio can be transmitted. This time is on the order of several seconds.

Temperature Range Consideration

All of the elements of the TW Intercom System have been designed to operate over the temperature range of 0 degrees Celsius (32 degrees Fahrenheit) to 50 degrees Celsius (122 degrees Fahrenheit). The high temperature range is extended another 15 degrees Celsius if the units are not operating at full capacity or some other worst-case condition. The low temperature range is extended another estimated 20 degrees Celsius if the full system gain range is not required. The major operating problem at lower temperatures will be the dew point and the resultant condensation. If this is the typical operating environment, then it is recommended that the equipment be opened, cleaned, dried and sprayed with several light coats of plastic spray. This will lessen the noises generated by leakage currents that occur when the moisture and any dirt or film combine. Cleaning can be accomplished a rinse of alcohol, a very mild detergent (saponifier type) wash and 2 or 3 thorough rinses with distilled water. This routine is to first wash off the nonpolar soluble substances, then the polar soluble substances.

Cooling Requirements

In general, only the power supplies require cooling consideration. Normally, leaving 2 inches clearance above and below the rack-mounted supplies is adequate. Portable supplies should not be left in the sun and these supplies should have clearance of 6 inches from five of the six surfaces. All other elements of the TW Intercom System require no special consideration. It is important to note that belt packs and other equipment left in the sun can cause burns to human flesh, due to the large amount of heat transfer possible. The user stations will normally continue to operate if one can only figure out a way to flip the switches and touch only the knobs.

Moisture / Contamination Protection

If, in the field, a soft drink or something like it is spilled into the equipment, the equipment can be dismantled and cleaned gently with clean water. After the equipment is dry it can be returned to service. If this happens fairly often, residues in the water can be deposited on the equipment. It should be noted that a build-up of contaminates and humidity can cause audible noise on the intercom line. If it is likely that the equipment is continually to be exposed to contaminating liquid, suitable plastic covers should be employed. It may also be necessary to add a plastic coating as described above. When using equipment in the rain always protect the equipment with plastic covers - also, make sure all cable connectors are lifted out of the mud or snow and protected with plastic bags. Rain, mud and snow in connectors can cause considerable audible noise in any communications system.

Magnetic Fields: Hum Problems

When the balanced type of intercom equipment is used, it is still possible to induce hum into the system by placing or locating user stations or system interconnects near a hum source, such as, power transformers or electrical switch panels or lamp dimmers. When the microphone switch is turned on and a dynamic microphone headset is used, the dynamic microphone is a sensitive antenna for magnetic fields. Often, operating personnel will go on a break, leave the microphone on and lay the headset on equipment with power transformers or near TV cameras or monitors with vertical deflection yokes. This is the reason for the system microphone turn-off scheme (Mic Kill).

SUMMARY

(Defining and Meeting Your Needs)

1 Application Block Diagrams are a good starting place to define a system.

2 The generic block diagrams show a basic small system and how things plug together.

3 A generic system could used in a small television studio production, an outside television field production (such as ENG and EFP) or an industrial test of a large system (such as an aircraft).

4 A generic system can be created using almost any Party-Line system. Audiocom, Clear-Com, and RTS TW systems block diagrams are shown.

5 A switch on the Audiocom PS2000 two channel power supply can combine channels into one large Party-Line.

6 Equipment available from any one of the three illustrated manufacturers intercom systems can be assembled into a two-channel system.

7 A two-channel system can be used for a small TV operation (Studio or Truck) or cable access. One channel can be used for the director and crew, and the other channel can be used as a public address or stage announce system. The stage announce system can cue talent for the show, or allow the director to talk to the performing crew and talent during rehearsals.

8 All three manufacturers make equipment suitable for theater applications use. Again, one channel of a two-channel system can be used to cue the actors.

9 A training system usually consists of a station for the instructor and multiple, independently addressable student stations.

10 In a large system for television production, additional accessory equipment allow expanding the Party-Line into 12 or more Party-Line channels, isolated camera channels, IFB capable stations, and wireless intercoms.

(The IFB System (One Way Communications System))

1 IFB is an acronym for Interrupted FeedBack, Interrupted FoldBack, or Interrupted Return Feed (also known as IRF).

2 An IFB system allows people running the show, such as director, producer, and mixer to talk to the talent or actors directly. The talent may receive cues, additional information, or hear other talent in other locations to be able to talk with them.

3 The IFB system consists of a 1) a hot mic feed from a director, producer, et cetera, 2) a control panel, 3) connecting cables, 4) talent station, 5) talent headset or earset. Some IFB systems are wireless. This requires some different equipment, and the wireless feature eliminates the connecting cables.

4 IFB systems are often required to operate over large systems, as much as a mile.

(Connecting (Interfacing to Other Communications Systems))

1 Interfacing is connecting two separate communications together or not directly communications to the Party-Line.

2 One modern interface requirement is two connect a two-wire Party-Line system to a four-wire intercom system.

3 Interfaces often can compensate for system to system: a) level differences, b) mode differences, c) impedance differences, and can translate call light and other data signals into suitable formats.

4 Interfacing to various television cameras is often challenging and may require extra equipment and extra efforts.

5 There are three tasks to interfacing: mode conversion, level changing, and signal / data conversion.

(Some Practical Considerations)

1 A too long headset cable may cause feedback or crosstalk problems.

2 Low impedance headphones, in general are louder and cause the user station to draw more current. Higher impedance headphones lower current drain but may not be loud enough for use during concerts or athletic contests.

3 Accidental connection of the shield in a microphone cable to earth grounded objects may cause hum and noise in the intercom system.

4 Taping or fastening metal intercom stations to metal structures may introduce into the Party-Line intercom system.

5 Cabling in poor condition may introduce noise / intermittent operation into a system.

6 It may be necessary to convert the intercom audio to a balanced configuration to cover long distances or to overcome strong interference from adjacent cables.

7 Extending the range of the Party-Line intercom may require using heavier gage cables, or using special schemes of “local powering” the remote user station.

8 Extending the range and using “local powering” may reduce a two-channel system to one channel at the remote station.

9 Crosstalk in a two channel system such as the RTS TW system can be reduced by “home running” the cables to a central point where the splitters and power supply are.

10 Crosstalk can also occur across the ground connection, especially where long cables have built up the ground resistance.

11 System currents are defined by the type of user station, its current drain, and the number of stations on a power supply feed.

12 If the system is operated too close to its maximum current, it may have trouble starting due to the “foldback” current limiting in a power supply. The work around for this is to break the system into several subsystems, then power up each subsystem in sequence.

13 Temperature Range Consideration: Condensation due to low temperatures may cause noise in a system.

14 The power supplies are generally very rugged and withstand a wide range of temperatures. But it is still important to take precautions to prevent overheating of the power supplies.

15 The dynamic microphone in a headset can pick up stray magnetic fields and introduce unwanted hum and noise into a system. Don’t place the headset on or near other equipment that has strong magnetic fields.

16 If equipment gets contaminated with a spilled drink, mud or snow, it may require cleaning with distilled water and gentle drying.

CHAPTER

4

INTRODUCTION TO MATRIX INTERCOM SYSTEMS

RALPH STRADER

Introduction

While there is an extensive glossary in the back of this book, some definitions will be given here to aid in the following chapter.

Definitions

Ports Refers to the number of connections available to external devices from the matrix. In typical usage a logical port consists of an audio input to the matrix, which is used to bring the talk signal from a user station, an audio output used to take listen audio to the same panel, and a bi-directional data signal for control and status information between the matrix and the user station. In the RTS™ ADAM™ intercom system, the inputs and outputs can be assigned to completely separate functions, allowing the port to be “split.” A typical application would use the output portion of a port for a feed to a paging speaker, while using the input portion to provide program audio to be used with IFB feeds.

Matrix The audio router that establishes communications paths from user to user. A matrix must not only provide the routing, it must do so reliably, remembering configuration and status and reporting on them. They must also have some degree of reliability – which, as with all things in the world, is related to needs and budget.

User Station Also referred to as a keypanel. Using the telephone system analogy, the matrix is the central office switch or PBX and the keypanel or user station is the telephone instrument. These devices can range in complexity from a simple microphone with a single push button and a loudspeaker to a fully programmable keypanel with alphanumeric displays, DSP signal processing, user programmable features and volume controls. The RTS™ KP-32 (see figure 4.1) is a good example of the latter.

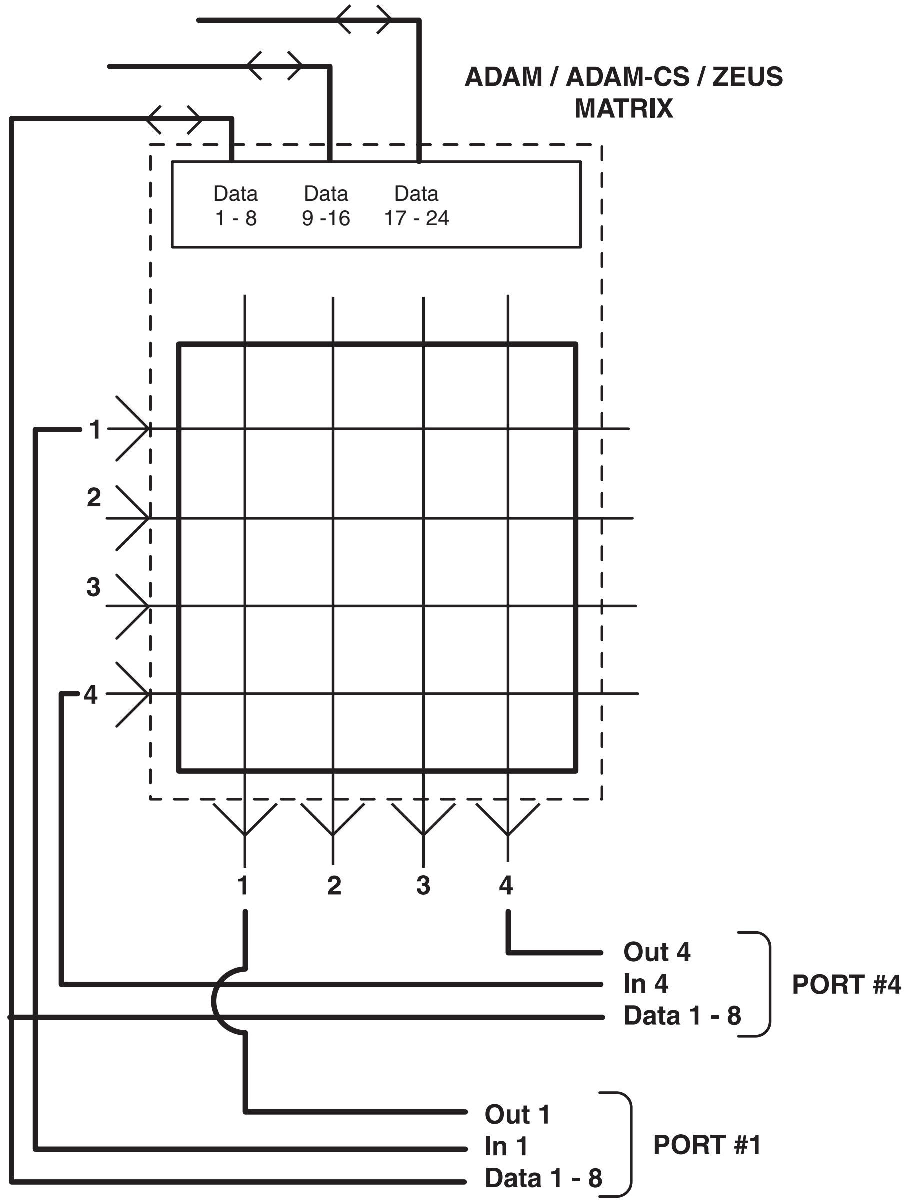
Figure 4.1 The KP-32 is a good example of an advanced user station (keypanel).

GPI (or GPI/O) General Purpose Interface or General Purpose Input/Output. This refers to logical inputs and outputs that can be wired to external devices for various purposes (hence the term "General Purpose"). Typically, these are optically isolated logical inputs and relay outputs. However, other variations exist.

Rack Unit(s) (RU) A standard unit of measure used when dealing with electronic equipment racks. 1 RU = 1.75" (44.45 mm). For example: a particular piece of equipment is described as being 3 RU in height. This means that it is 5.25" (3 x 1.75") in height. Detailed information on the specification of standard electronic equipment racks can be found in EIA RS-310**.

**International Standard from Electronics Industry Alliance. See http:\\www.eia.org.

Figure 4.2 Example of Matrix Ports



History of Matrix Intercoms

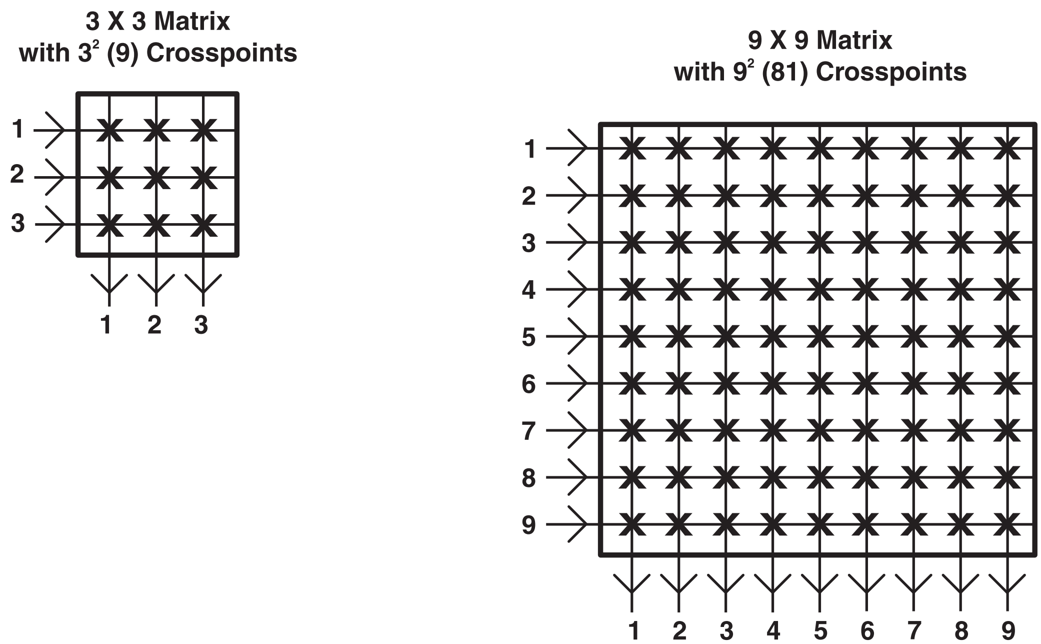
Properly, it can be said that matrix intercom systems go back to the advent of automated central office telephone switching systems in 1892. Matrix intercoms, even today, owe a great deal to the concepts and technologies of those systems.

In the 1950's, McCurdy Radio Industries of Canada introduced the 7000 Series matrix intercom based on wire per crosspoint and reed relay technology. Its basic building block was a crosspoint card containing six crosspoints. It was the first known matrix intercom system developed for the broadcast industry. In the early 1970's, in a project for the CBC, a solid state crosspoint was developed and the resulting matrix intercom system was named, the 9100. This was still "wire per crosspoint" technology, but density increased to allow a 10 X 1 format on a single crosspoint card. A 10 X 10 system could be built in only 7RU. The 9100 gradually kept expanding and graduated to the 9200 series. The largest system built was a 60-port system delivered to CBC Winnipeg.

In the late 1970's, microprocessors became available and the first truly intelligent intercom system, the McCurdy 9400, was delivered. This was the first system that used data sent from the user stations as opposed to one wire per intercom key. As microprocessor technology improved, the 9400 was replaced by the 9500 series. This series was more dense, allowing a 50 X 50 system in 3RU. The technology was modern; a very conventional square array of switches allowing any input(s) to be switched to any output, but the implementation was somewhat limited by what is called the "square law" problem.

Briefly, in traditional matrix technology, in communications, audio, and video routing systems, the size (electrical and physical) of a matrix is related to the number of inputs and outputs, or "ports", in a mathematical "square law" relationship.

Figure 4.3 A Comparison 3x3 vs. 9x9 Matrices



If you examine Figure 4.3, you can see that the 3 x 3 matrix, which is needed to support a three-user intercom system, has nine crosspoints. The 9 x 9 matrix, for nine users has 81 crosspoints, so by tripling the number of users, the size of the matrix has increased from 9 to 81 crosspoints or nine times. As nine is equal to the threefold increase in number of ports squared, the term "square law" has come to represent the problem.

Table 4.1 Number of Users vs. Number of Crosspoints

Number of Users	Number of Crosspoints
10	100
25	625
50	2,500
100	10,000
200	40,000
400	160,000

As you can see in Table 4.1, while a ten-user system "only" requires 100 crosspoints for all possible communications paths, a 100-user system requires 10,000 crosspoints. Now, realize the number of crosspoints has a direct correlation to power consumption, physical size, and cost. It becomes apparent that with a traditional architecture, crosspoint matrices have a pretty small limit on maximum practical size.

When McCurdy Radio Industries introduced their 9500 series matrix intercom product, 50 ports required a rack frame 3 RU in height, and weighed 20 pounds. At the time, the size limitation was understood, but not regarded as a problem because it was thought that no one would ever need more than 50 users in a single intercom matrix. Today, we can look back and put that statement in the same category as IBM's assertion in the 1950's that "the world market for computers is 5 systems – TOPS," or the apocryphal Bill Gates quotation to the effect of, "Who will ever need more than 640K of memory?" In 1985, the market for systems as large as a 50-user intercom was primarily limited to the major television networks.

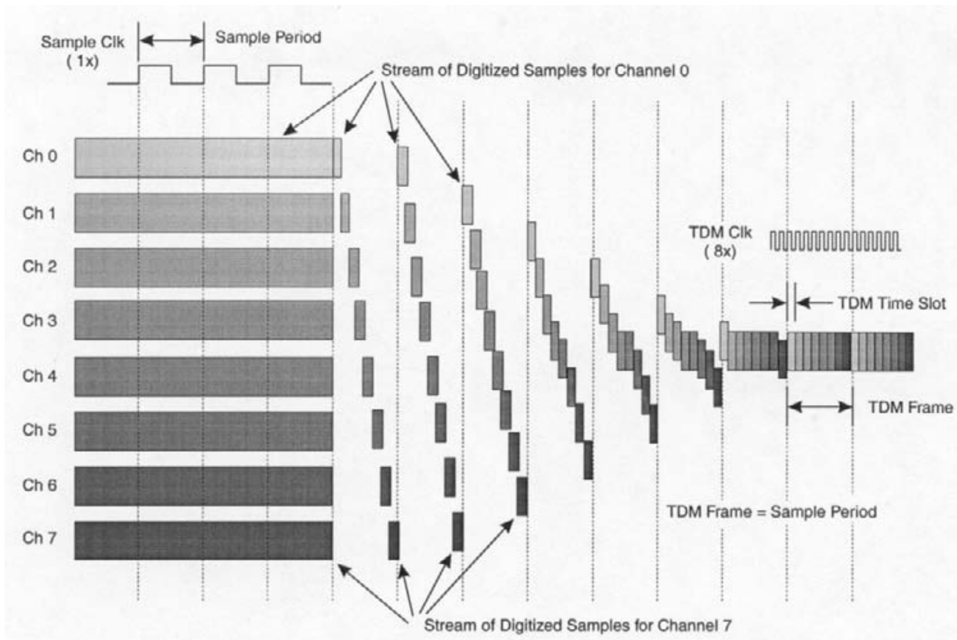
Figure 4.4 A comparison of the 9400 Intercom System to the 9500 Intercom System (see inset). The 9500 represented a tremendous reduction in physical size.

By 1988, the limits of the square architecture were beginning to show. The 350 port McCurdy 9700 matrix intercom systems that NBC commissioned for the 1988 Seoul Olympics required 10 full racks, over 20 kW of power, and weighed in at over 2 tons. The 9700 matrix was the largest matrix intercom of its day. While providing nearly all the features of today's most advanced intercom systems, the limit on size had been reached for traditional architecture.

By the early 1990s, manufacturers in Europe were developing intercoms based on a new architecture. Time Division Multiplexing (TDM) had been deployed in telephone routing and switching systems much earlier, and now it would be applied to matrix intercom systems.

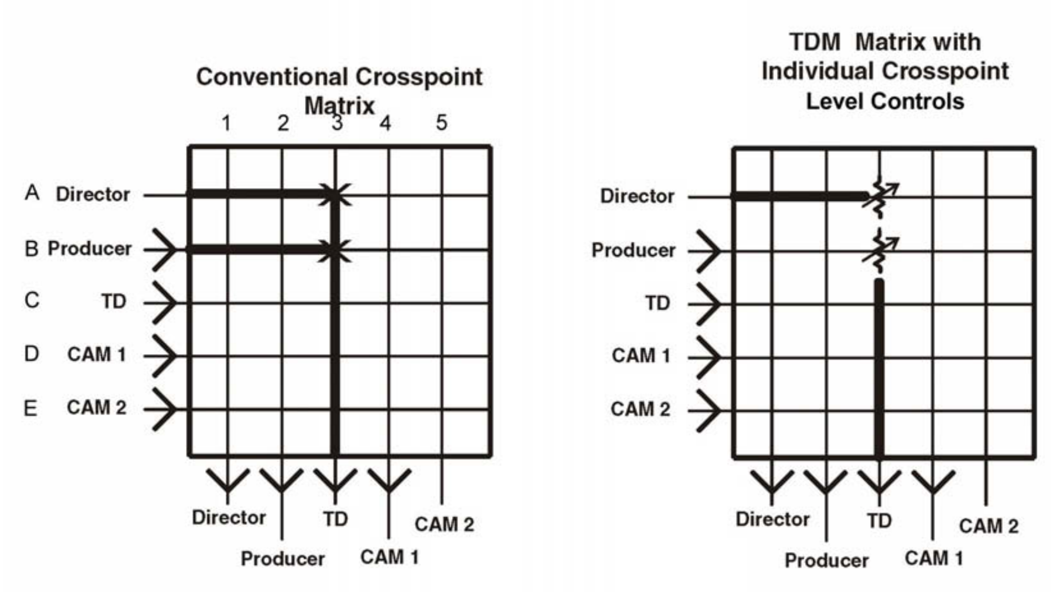
In a TDM matrix, the incoming signals from users (microphones or headsets) are run through an A/D converter and assigned a “time slot” on a TDM backplane. A good (although not strictly accurate) analogy would be the signals on a cable TV system. Whereas on the cable system you might have ESPN, HBO, and MTV, on a TDM backplane you would have the timeslots for Director, Producer, and Camera1. A user can then listen (or be talked to by) any or all of the timeslots. Determining which signal is heard is under software control, and can (generally) be selected by the listener, or pre-programmed. It can also be a function in which other users are calling the listener at that moment.

Figure 4.5 An example of how multiple signals are “time-sliced” for use in a TDM system.



Again, if you use the cable TV analogy, it is easy to understand why the systems do not have to obey the square law. In a conventional square law matrix, adding a single user to a 100-user matrix requires the addition of 201 crosspoints (101^2-100^2). In the TDM world, it requires the addition of two simple bits – a “transmitter” for the already existent time slot, and a receiver to tune in the other time slots for that user to hear.

Figure 4.6 Conventional Matrix vs. TDM Matrix



Today, nearly all matrix intercoms are based on TDM or similar technology. Telex manufactures the RTS™ Zeus™, ADAM™-CS and ADAM™ TDM Matrix intercoms, Clear-Com has the MatrixPlus3, and other manufacturers in Europe offer TDM-based solutions to their markets.

Modern Day Matrix Intercoms

As discussed in the last section, today's matrix intercoms are TDM based. Let's take a closer look at the architecture of such a system, as a prelude to understanding its exact capabilities

As shown in Figure 4.6, one major difference between conventional crosspoint matrices and TDM matrices is that a TDM matrix is comprised not simply of crosspoints, but is a full-fledged audio mixer. The offshoot of this can be understood by the following example:

In the conventional crosspoint matrix shown in Figure 4.6, if the TD wants to listen to both the Director and the Producer, then crosspoints A3 and B3 are turned on (or closed). As these crosspoints are nothing more than switches, the relative levels of the signals are wholly dependent on the speaking level of the Director and Producer.

In the same example through the TDM matrix, the crosspoints are replaced by volume controls – the resulting matrix is referred to as having individual crosspoint level adjustments. In this case, the capability exists for the relative signals levels to be adjusted by volume controls for the Director and Producer as heard by the TD. Various means can be used to make that adjustment, but for now the salient point is that different listeners (or outputs) have the ability to selectively mix the signals from the sources they wish to listen to.

For the most part, this is the major difference between conventional crosspoint intercom matrices and modern TDM (or similar technology) matrix intercoms. There are other differences that are primarily a function of the addition of features and capability which are part of the normal product development process. These details will be discussed in the next chapter when we get into system design issues.

Special Considerations

When considering the type of intercom system to install for a given application, there are many factors to take into account and many of these have been discussed in an earlier chapter. These factors are discussed in detail in the following section on advantages and disadvantages of matrix intercom systems versus the other types of systems available.

Advantages

Matrix intercom systems have numerous advantages over other types of intercoms. These advantages include size, configurability, variety of communication types supported, and ancillary functions available. The following discussion will refer to the RTS™ ADAM™ Intercom System, but a number of the principles may apply to other matrix intercom products.

Size

In this context, size refers to the number of user stations supported. The RTS™ ADAM™ line of intercoms is available in sizes from eight users up to 1,000+ in a single matrix, and can be expanded by means of trunking to include 31 such matrices interconnected. A typical hardwire PL system is no more than four channels – although most modern PL systems can be expanded to a dozen or more, the economics and ergonomics quickly become less desirable with size.

Configurability

In a matrix intercom system, the hardware is typically installed once and not altered day-to-day to accommodate day-to-day operational needs. Since each user station has the electrical capability to be connected to any other user station (via the crosspoints or individual crosspoint adjustments), changing who talks to whom, rules for what happens under certain circumstances, and the assignments of keys are under software control. In matrix intercoms this configuration can be done in many ways. There is usually a computer connected to a port of the matrix with software that allows changes to be entered, activated, and saved. Additionally, changes the users are allowed to make on their panels can be used to configure the system.

The flexibility to make these changes and more without the need for labor intensive wiring changes are a key advantage of matrix intercom systems. It allows a single system to function as three independent intercoms for three studios most of the year and as a single large system during election coverage by the simple act of loading a new file.

Types of Communications Supported

A modern matrix intercom system has the ability to allow any of the user stations to be connected to any of the other stations. Since the connections are under software control, virtually any communications configuration can be accomplished, and as such, there are very few limitations on type of communications supported, without the need for specialized hardware.

A great deal of the capabilities of modern matrix intercom systems is in the ease of which they allow different types of communications to be established. For example, from your home telephone you can establish a four-way conference call. It may involve calls to the operator, or conferencing in two people, one of whom then conferences in a third, but it can be done. At the office, it may be a bit easier. Call Alice, press the “CONF” button on your telephone; Call Bill, press the "Conf" button again; call Chuck, then press the "CONF button, and you have a conference with all parties involved. With a matrix intercom system, you press the talk keys assigned to Alice, Bill and Chuck and say "Meet me on Tech PL". You, Alice, Bill and Chuck each press "Tech PL" on your user station, and instant conference.

Other types of specialized communications can be established as easily (or easier) in a matrix intercom system. These types include the following:

Conference or PL – described above

Isolate or ISO – a temporary private discussion amongst two parties

IFB – Temporary interruption of a program signal with private conversation

Special List or Group Call – Single key to address many individual users (also used as "All Call"

Telephone – single key to answer an incoming telephone call, or to make an outgoing telephone call (requires telephone interface, such as RTS™ TIF-2000).

Relay – pressing a given key activates a relay – a typical use would be to activate a transmitter to send audio communications via wireless.

Ancillary Functions

Warning! Low-key sales pitch …most modern matrices provide some form of ancillary functions. I will describe those which are common to the matrices I have experience with, including competitors of Telex, then I will delve into some functions which I know to be available in the RTS™ line of Matrices including Zeus™, ADAM™, and ADAM™-CS intercom systems.

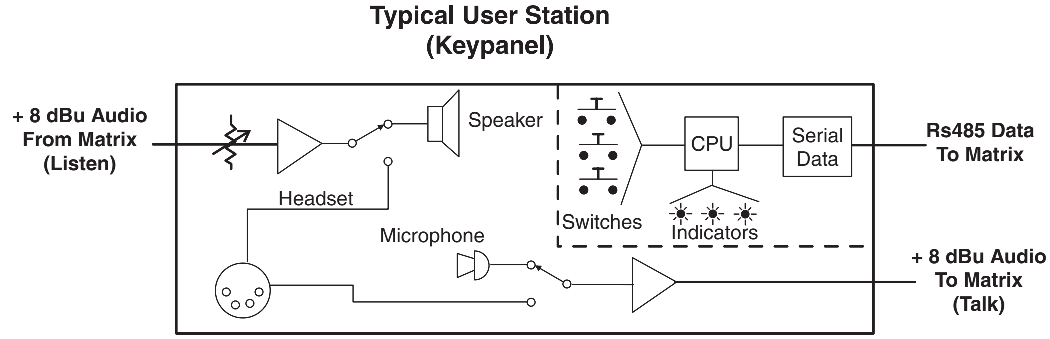
The most common ancillary functions are those referred to under the heading of interfaces or "GPI/O." Quite often, in an intercom system, there is a need to interface to varying degrees with the outside world, and the more complex the intercom system, the greater need for such interfaces. Usually, these methods are quite predictable and the manufacturers provide or recommend a solution. A good example is a telephone interface that allows the intercom system to tie to the public telephone system to allow users to "dial in" or be called by the intercom system.

Basic Ancillary Functions via GPI/O General Purpose Input / Output

Oftentimes the interface needed is not so predictable, a user may have a need for the intercom system to flash a strobe light when calling into a high noise environment or to activate a "gong" signal over a paging system to announce a message. For these purposes, relays (one form of the "O" in GPI/O, which means "output") can be wired from the intercom system to the strobe or gong generator and programmed to activate when required. Relays are not the only form of output available. A given system might instead provide a logic level signal or an open collector signal from a transistor or opto-isolator.

The opposite need might also arise. A need for a signal, external to the matrix system, to cause the intercom system to undertake a certain action. As defined previously, a user station is a device that feeds a "port" of the matrix intercom system. At its most basic, it is a "box" with three basic functions. First, it takes speech through a microphone, amplifies, and processes it to a given signal format (balanced +8 dBu audio in RTS™ Matrices) to feed the matrix. Second, it takes audio signals from the matrix (again. +8 dBu in RTS™ matrices) and converts them to a level suitable for driving a speaker. And third, it provides some degree of signaling and control to the matrix. For example, something which says to the matrix, "the user wishes to send his (or her) voice to FRED."

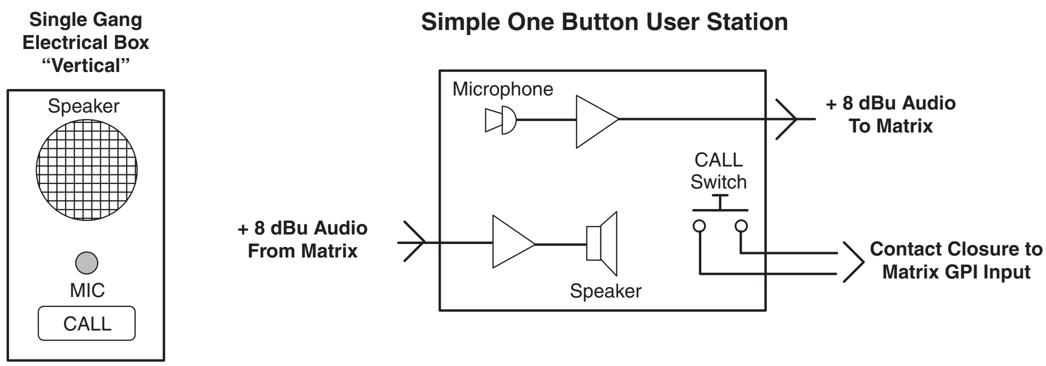
Figure 4.7 Typical Keypanel



Normally, a user station provided by the manufacturer of the intercom performs all of these functions. However, suppose the user requires a user station to be very small, low cost, and mounted in a single gang electrical box, and that the station only needs to call a security desk. The user, dealer, system contractor, or any third party company can build a small box with a microphone, preamplifier, audio amplifier, speaker, and a push button. The only question is, how does the builder easily create the control protocol to notify the matrix that he or she wishes to be heard? Making the situation more difficult is the fact that manufacturers do not publish the details of their control protocols.

The answer is simple. The push button of the user station is connected to a logic input of the matrix (the "I" in GPI/O) and the operating software is instructed to treat the activation of that logic input as the press of a talk key pre-assigned to the security desk.

Figure 4.8 Simplified Low-Cost User Station



A number of other examples with more detail of GPI/O are shown in the next chapter.

More Complex Ancillary Functions

The examples above presume that the interface requirements are very basic, and can be defined as an action which controls or is controlled by a single change in one logical state, a single "bit" of binary information.

There are often cases where the definition is nearly as easy, but multiple conditions must be met. Perhaps, in the previous security desk example, the user needs a certain intercom panel to call the receptionist from 8:30 AM until 4:30 PM, then from 4:30 PM until midnight calls the security desk, and then from Midnight to 8:30 AM sends the signal through the building paging system to wake up the watchman.

Another example, if the "ON AIR" light in studio three is on, DO NOT allow audio to go to the three speaker stations in studio three, unless the panels are feeding headsets AND NOT the built in speakers.

In RTS™ Zeus™, ADAM™-CS and ADAM™ matrices there is a feature called User Programmable Language (UPL), which allows the following conditions to be tested:

Output from a Previous UPL Statement
GPI Input
Local GPI Input
Status of a GPI Output
Status of a Local GPI Output
Talk Key Status
Listen Key Status
UPL Resource
Crosspoint Status
Input Talking
Output Listening
Headset Transfer Switch Status
Current Date
Current Time
IFB Interrupted
Counter

This allows the test to be chained with other conditions via **AND**, **OR**, **NOT** and **XOR** to be tested and cause one of the following (or multiple of the following) actions to take place:

Close Crosspoint
Inhibit Crosspoint
Assert GPI Output
Inhibit GPI Output
Assert GPI Output Local
Inhibit GPI Output Local
Force Talk Key Closed
Force Talk Key Open
Dim Crosspoint Volume
Load Setup File
Force Listen Key Closed
Force Listen Key Open
Clear Counter

The user can construct these statements easily using selections chosen from pull down menus in the operating software. UPL is the answer to the time dependent routing described above.

Getting more difficult, there are cases where the possible actions and situations are much more complex, and an external computer or device of some type is involved.

An example of this is a large television complex where an automation or scheduling system assigns a given control room to a given studio. The routing switches, camera tally matrices, machine control, and intercom systems are expected to make appropriate assignments in support of that configuration.

Another example might be a group of conference rooms that can be combined or used individually as controlled by a system such as manufactured by Panja (AMX) or Crestron. Again, the intercom system must respond to these assignments from the external systems.

For this need, RTS™ has implemented a serial RS-232 control language called "Command Line Protocol" which is standard on the Zeus™, ADAM™-CS and ADAM™ matrices. This protocol allows simple ASCII communications between the intercom matrix and the external computer. The protocol is published, and is contained on the accompanied CD. A typical statement might look like this:

To accomplish the following:

Force the following crosspoints:

input 1 --> outputs 43, 44, 45
input 3 --> output 43
program input 1 --> output 45

also inhibit the following crosspoints:

program input 1 --> output 1

Issue the following ASCII Command String to the Matrix:

IN1FI43F44F45F1IIN3FI43FINPG1FI45F

The simplified ASCII command line protocol still requires some programming to take place external to the matrix to either translate the native language of the external control system to **Telex®** Command Line Protocol, or to modify the internal code of the third party device to speak and understand Command Line Protocol. This effort is likely small when compared to the benefits of such tightly integrated control between systems. Now that we have outlined the advantages of matrix intercom systems over other types of systems, let's go to the opposing viewpoint.

Disadvantages

Matrix intercom's disadvantages over other types are pretty much the opposite of the advantages listed above. Disadvantages include size, cost and complexity. Complexity, in particular, renders them unsuitable for many applications.

Size

Here, size refers to not only the number of ports, but physical size as well. The smallest physical matrix available today is the Zeus™ matrix that is two RU in height. Add in a single RU user station and you now have a minimum of three RU of rack space required. By contrast, **Telex® RTS™**, **AudioCom®**, **and RadioCom™** intercom systems (as well as some competitors) offer systems providing both a multi-channel user station and a system power supply in a single RU. Matrices with larger number of ports become correspondingly larger, physically. There are times when size is of paramount concern such as, travel packages for news crews, remote trucks, cockpits, and Manhattan.

True Story! One customer in NYC justified replacing their 15 year old matrix intercom with a newer system solely on the space and power savings (electricity and cooling), going from more than 18 racks to 2 racks of equipment and increased the number of ports in the process!

Cost

Again, somewhat related to size. If the intercom needs are small, and the complexity of requirements are not great, the overhead of having the matrix is hard to overcome. As an example, (2001 pricing) an intercom system with four users communicating over two channels can be completely for less than $1,600, using a party-line system. Given the relatively high cost of any matrix, four user stations along with a matrix would cost at least $8,000. The matrix system would have tremendous expansion and many extra features, but if that is not required, the cost is a definite negative factor.

Complexity

Complexity is quite often the major negative to matrix intercom systems. Complexity brings a whole world of issues, which can be of major consequence. I'll start with a few examples based on our "friend" the personal computer.

You are in your kitchen – QUICK, multiply 347.2 times 15.8 –

Well let's see, I could go down to the den, turn on the computer, wait for Windows® to boot up (have a cup of coffee), start my spreadsheet program, and type in "=347.2*15.8<enter>," read the answer – "oops, no pencil -%(&#@) Select **File**, **Page Setup**, **Set Print Area** highlight the cell with the answer, **Print**, wait for the Laser Printer to warm up, take the print out, tell the computer to shut down, go back upstairs…." elapsed time 9 minutes.

— OR —

Take the free Time Magazine calculator out of the junk drawer in the kitchen and press 347.2 X 15.8 = and read the answer (5,485.76 for you curious types).

Same example, except now you are not in your home but in a research lab you are visiting, and see a pocket calculator lying next to a turned off monitor for a workstation. Now the considerations become more complex – does the workstation work at all? Is it an operating system I understand? Does it have a spreadsheet program at all? Would turning the monitor on and trying to start a spreadsheet disrupt some important research? Which device would you choose to get the answer?

Last example – you are not computer literate, the only PC in the house belongs to the expert (your 12 year old daughter and she is at a neighbor's working on the web site for their dot.com startup). "Oh, for gosh sakes, just hand me the calculator already!"

Despite the attempted humor, the same considerations apply to matrix versus TW or wireless intercom systems. Matrix systems (like PCs) are good for complex things, and they can also do simple things, but if PCs really were good for the small jobs, why do you still have that calculator, pencil, pad of paper, photocopier, and fax machine in your office? The answer is because, like with an intercom system, sometimes all you need to do is to scribble "call Paul" on a Post-it® note to put on your computer monitor for after lunch.

TW and wireless intercom systems are generally simple to operate, transport, hookup (configure), and do not require an expert to setup. This is especially true if the system in question does not need to change on an hour-to-hour or day-to-day basis. They are very affordable, robust, reliable, and physically small.

Interconnection between components may be a simple as thin air (wireless), microphone cable (PL), coax or twisted pair for matrix, but is more likely to be multi-conductor cable. Again, another layer of complexity.

To change the configuration of a PL system, you can likely just change which units are tied together by changing cables, or by turning some switches on an assignment panel. In a matrix system, you will likely need to connect a PC and run the configuration program.

Figure 4.9 Use of Source Assignment Panels such as this SAP-1626 allow the rapid reconfiguration of PL systems without changing any cables

So now…Quick! You need to setup an intercom on the roof of your facility to cover a local parade. You can go to your matrix intercom, locate two unused ports, assemble appropriate length three pair cables, fish the cables up to the roof. Then fish a power cord to the roof, take two keypanels up there (hope it's not raining), and connect the panels. Now go down to the configuration PC, assign appropriate keys to those panels. Go back to the roof and verify that you have communications. – "What do you mean the parade ended two hours ago?"

— OR —

You can take two beltpacks and two microphone cables to the roof, daisy chain the beltpacks together, drop the single microphone cable down to the equipment room and either connect directly to your existing PL system or to the interface between your PL system and the matrix for (nearly) instant communications. Do the same example using wireless intercom, and it gets even easier!

For all the strength, features, and power of modern matrix intercom systems, there are many situations where they are more of a burden than a solution.

CHAPTER

5

DESIGN OF MATRIX INTERCOM SYSTEMS

RALPH STRADER

Introduction

In this chapter, we will address the major issues and considerations for designing a matrix intercom system. At the end of the chapter, you will *not* know everything to specify, plan, design, and install a matrix intercom system. Nevertheless, you will have a good idea of the basic requirements, pitfalls, and opportunities involved in the design and installation of a matrix intercom system.

Back-to-Basics

As discussed previously, a modern matrix intercom system is very similar to a telephone system. It is comprised of, in its most basic form, a Central office switch (the matrix), interconnect wiring, and telephones (user stations). Most of the concepts and some of the terminology is common to both. Calls can be made, busy signals encountered, “call waiting” exists, conference calling is possible, unlisted numbers can exist, calls can be blocked (incoming and outgoing), and long distance (trunking) is possible.

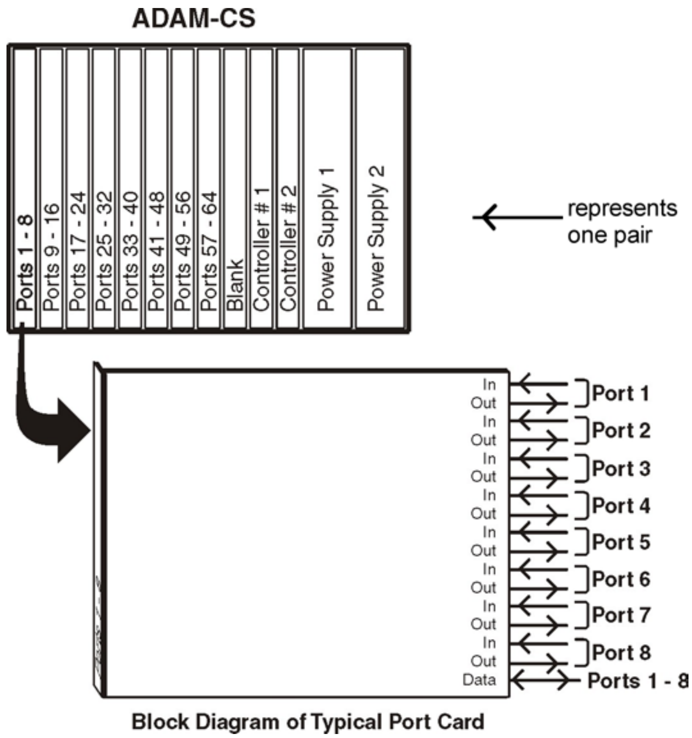
The following examples will use the **Telex® RTS™ ADAM™** intercom matrix, unless otherwise noted. Most matrices on the market today will have similar features, but unlike **Telex®** products, the competitors’ units are not designed to also prevent dandruff, solve the meaning of life, the universe and everything (with apologies to Douglas Adams), and achieve world peace. We would like you to believe that our products will do so. (And in writing that I felt a bit like Dogbert from Dilbert.)

RTS™ Matrix Intercom Systems

Because of design and installation issues specific to the brand of intercom matrix used, it is now necessary to talk in some detail about the specifics of the **RTS™** products, including **Zeus™**, **ADAM™-CS** and **ADAM™** intercom matrices, as well as some accessories. When I refer to the **ADAM™** series of intercoms in the following portions of the chapter, unless otherwise noted, the comments also apply to **ADAM™-CS** and **Zeus™ intercom systems.**

Previously, we discussed the analogy between telephone systems and matrix intercom systems – the analogy is not correct in all cases, here are some exceptions.

Figure 5.1 Typical ADAM™ Matrix Connections



In **ADAM™** matrix intercom systems, the connection between the matrix and keypanel is normally via three twisted pairs of unshielded cable. As shown in Figure 5.1, one pair carries balanced audio from the keypanel to the matrix, one pair does audio in the opposite direction, and one pair is a RS-485 data signal which is shared among 8 panels in a group.

IMPORTANT As eight panels share one physical data line, the matrix must have some means of identifying which panel is sending data to it, and also have some means of addressing messages to one specific panel of the eight. The key word in the previous sentence is "addressing." Each keypanel in the system must be assigned an address by one means or another. On some keypanels this involves setting "dip switches" to select a "one of eight address" via binary code (KP-9x family of panels). On other keypanels, the address is set via rotary switch on the keypanel (KP-32 and Low Cost Series of Panels). And, on others the means is via menus and firmware (KP-12 series of panels). In all cases, the factory set default address has one chance in eight of being set correctly "out of the box."

If a separate keypanel is attached to each of the 8 ports which share a data line, each panel must have a unique address set which matches the physical port to which the panel is connected. Having a panel with an address different from the physical port to which it is connected will render that panel unusable (in a practical sense, even though the panel may receive audio). Having two or more panels in a given group of 8 with the same address will disrupt all eight panels in that group by causing data collisions on the common data line. This is so important that I will repeat it. ***Having two or more panels in a given group of eight with the same address will disrupt all eight panels in that group by causing data collisions on the common data line.***

At time of initial installation, or system modification, the great majority of anomalies can be traced to improper addressing.

As received "out of the box," a matrix intercom system needs to be configured (programmed). This can include how many users are connected, how many conferences are expected, what you wish to name the users, who can talk to whom, and, just as importantly, who cannot talk to whom. In some cases, the default configuration upon first operation is adequate and may allow enough communications to meet your needs, but it is unlikely, and frankly, a tremendous waste of capabilities. We will discuss configuration via software in more detail later.

In this chapter, we will first touch on the design and requirements aspects of the matrix intercom system, then move into installation, and finally operation. On the CD, we have included a full, up-to-date (as of this writing) **AZ™-EDIT** configuration software package which can be installed and ran on your PC. You do not need to have an intercom system connected to run the software.

Note You may not be able to see and/or use some features because they require that an actual intercom system be connected to your PC.

Loading and running the supplied software will add a good amount of "hands on" to your experience, but in the interest of keeping this book a useful reference, regardless of what intercom matrix system you may be exposed to, the examples given will not be specific to the included **RTS™ AZ™-EDIT** configuration software except where absolutely necessary.

To Begin

The first question you should ask, as with all systems design is, "What are you trying to accomplish?" The matrix intercom needed by a small station in Botswana (to avoid offending any US residents of small states who are no doubt tired of being referred to as being suitable for simple, basic, limited products) is considerably different from that required by MegaMedia Corporate Conglomerate Entertainment Enterprises Ltd. with 87 stations, 4 film studios, and a theme park, located on 3 different continents, all engaging in joint productions.

I find that system design is best started from the bottom up, rather than the top down. On that note, figuring the requirements for communications to determine the size of matrix needed and then later deal with informed compromises to meet size or budget requirements.

I will also proceed on the basis that any needs for non-matrix portions of the system will be covered in detail elsewhere in the book, and that we need only concern ourselves with how to interface to them from the matrix.

In this section I use a lot of examples which are television based, owing to my background in television, and the origins of modern matrix intercoms, which have been predominantly TV station driven. The questions and procedures are, however, relevant for all applications, regardless of industry.

Let's get started.

How many individual locations and/or persons need to communicate with one another? Write them down. Organize them by logical grouping or location such as:

Studio A

Floor

Lighting Director
Camera 1

Camera 2
Camera 3
Floor Director
TelePrompTer
Anchor A
Anchor B
Anchor C
Weather

Control Room

Director
Producer
TD
PA 1
PA 2
Segment Producer
Audio Operator
News Computer Operator
Font Operator

Other

Green Room
Makeup

Do this for all locations; it will give you a quick "port count" for your system, which will have a significant impact on size of the matrix, and, as a result, the cost. I presume that even if you do work for MegaMedia Corporate Conglomerate Entertainment Enterprises Ltd., you do not have an unlimited budget (Shame, really).

Next, figure out what external "stuff" you need to deal with, such as:

- Interface to allow access to telephone lines – How many? How capable?
- Interface to TW (party-line) intercom systems.
- Relays and GPI/O for external devices.
- Interface(s) for remote locations such as:
  - Transmitter.
  - News Bureaus in other cities.
  - ENG vans.
- Interface to other matrix intercom system (trunking).

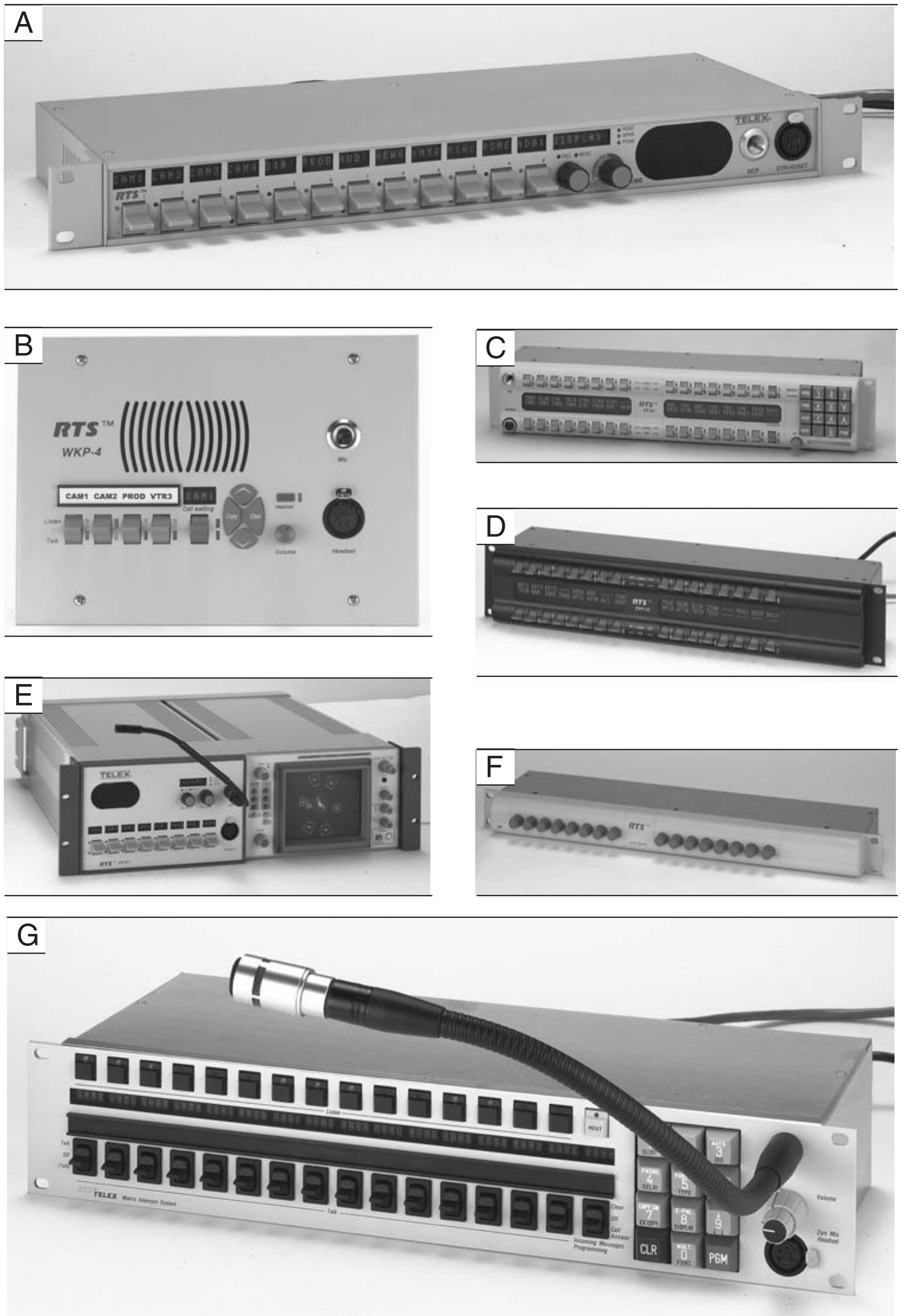
Now, it is time to put some detail on the above requirements. For each identified user, you need to know certain things, such as:

- How many other users will he (or she) need to readily communicate with at one "sitting" – this will determine the number of keys required on the keypanel.
- Does the identity of the key assignments change? If not, a keypanel without displays, which relies on labeling strips, will save money.
- Does the user want, need, or deserve the ability to reprogram their keypanel features, key assignments and defaults? If yes, a more complex panel may be required, and chance for errors is increased, but the user can make changes without involving you or some other expert.

- Does the user regularly need the ability to adjust individual volumes of the keys (not to be confused with the overall volume control which all panels have)? If yes, a Level Control Panel should be added to their station.
- Is space an issue? Can a smaller panel be chosen which meets the other requirements?
- Does the user really just need to be part of a given conference at all times? If yes, then putting that user and the other members of that conference on a TW channel and interfacing that channel to the matrix may make more sense.
- Does the user need to be untethered? If yes, a wireless beltpack is required.
- Is the user really "two-way", or are they listen only – such as the paging speaker in the green room, or the earpiece (IFB) for the talent.
- Is the user of sufficient stature that they will get "the top of the line" regardless? Those of you that have done systems design before have likely encountered this phenomenon. Those of you who haven't encountered this previously would do well to ask yourself if the CEO of MegaMedia Corporate Conglomerate Entertainment Enterprises Ltd. really needs that Pentium VIII 35 GHz computer with the 30 inch monitor on his or her desk just to read weekly reports from the boys in marketing – The answer will enlighten you.

In undertaking this exercise, it helps to have a catalog of available products (see Figure 5.2) from your vendor of choice in front of you to assist you in categorizing which panels you will assume are suited for the intended user. A copy of the current (as of the publication date of this edition) **RTS™ Matrix** catalog, as well as the **RadioCom™**, **AudioCom®**, and **RTS™ TW** catalogs are on the included CD.

Figure 5.2 A wide variety of keypanel options exist. Here we have a selection of RTS™ keypanels that fit a range of needs. Small keypanels such as the (A) KP-12LK and (B) WKP-4 provide an interface for those with limited keypanel needs. The (G) KP-96-7, a medium sized unit, was the workhorse of the RTS™ keypanel line until the 1980’s and 1990’s. The (C) KP-32 is the top of the line keypanel, and can be enhanced through additional options, such as the (D) EKP-32 expansion panel, and the (F) LCP-32/16 level control panel. The
(E) KP-8T is an example of a specialty keypanel that makes use of an empty bay in a Tektronix vectorscope.



Let's proceed on the basis that you have now compiled a list of needed equipment, have gotten approvals, placed the order, and are now ready to begin the installation of your system.

Cable Considerations

Cabling types do vary considerably among the manufacturers of matrix intercom products, as do the signals transported by them. For that reason, the following discussion is somewhat specific to **RTS™ Zeus™**, **ADAM™-CS** and **ADAM™** matrix intercom systems.

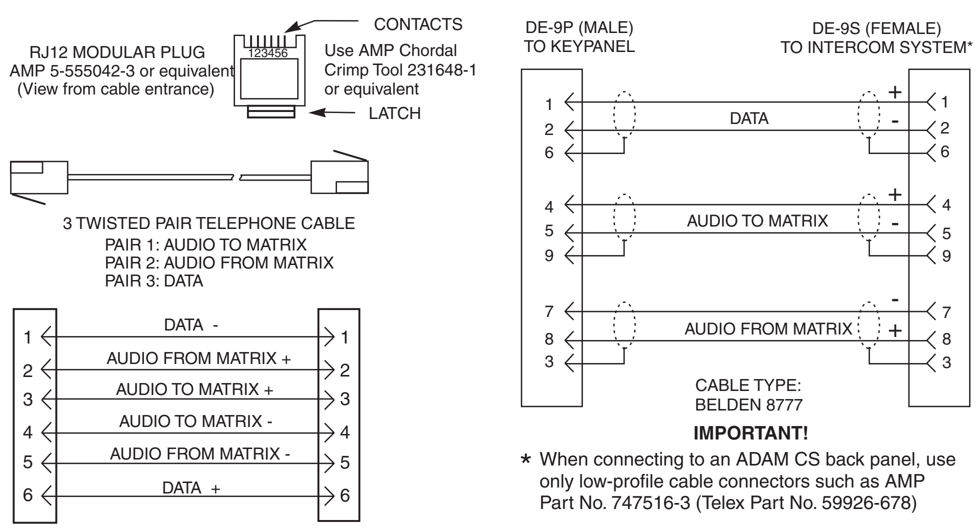
As noted earlier, **RTS™** Matrix Intercom Systems typically use three twisted pair, unshielded cabling for interconnection. I use the word "typically," as coaxial cable adapters are available from Telex to allow keypanels to be connected via coax, but the standard twisted pair methodology is more cost effective in most cases.

Telex allows the user to choose from two different connector styles for the three pair connection. The choice is simply a matter of preference by the user. RJ-12 connectors can be used, and these are readily available, low cost, and quick to assemble.

Note RJ-12 connectors are sometimes incorrectly referred to as RJ-11 connectors. While they are basically the same size, RJ-11 connectors have four conductors and RJ-12 connectors have six conductors.

On the negative side, they are plastic, and not as robust as some installations demand. DB-9 (actually DE-9, the more proper name) connectors are also provided, and can be used. These will be more robust, but are also harder to wire, and more expensive. All **Telex®** keypanels have both types of style connectors on them. The type of connector on the **ADAM™** and **ADAM™-CS** matrices must be specified at the time of order, and can be either the RJ-12 or the DE-9 style. Zeus comes with DE-9 only.

Figure 5.3 **ADAM™** (including ADAM™ CS and Zeus™) Intercom Cable Connections



As seen in Figure 5.3, the wiring takes pin 1 to pin 1, pin 2 to pin 2, and so on, for both style connectors. What the drawing also shows, and is equally important, is that a given twisted pair cable carries both portions of the same signal. If you were to wire pin 1 to pin 1, and pin 2 to pin 2, etc., but had one of the wires in a twisted pair carrying +audio in, and the other wire of that pair carrying -data, the audio would be degraded by having "data buzz" audible in that audio signal. The data signal would not carry for as great of distances. This type of error is second in the top ten of initial installation problems, after addressing mistakes.

ADAM™ and **ADAM™-CS** systems are also available with other wiring schemes, including multi-pin breakout to jackfields for monitoring and rapid changes and for use of 25 pair "Telco cable" for distribution. More information on this can be found in the **ADAM™** and **ADAM™-CS** System Installation manuals on the included CD.

Audio and Data Considerations

One of the benefits of the signal format described above is that generally, it does not matter how the audio and data signals get from the keypanel to the matrix. If you want to have a keypanel used in a Broadcast booth at the top of a football stadium which then is connected to an **ADAM™** matrix in the Sports Truck below, it is perfectly OK to have prewired a small adapter to let you transport the three balanced signals (audio in, audio out, and data) over three microphone cables in the audio harness which is already run between the locations.

Also, if you want to "piggyback" the audio and data on an existing corporate WAN running between two buildings on a campus, there should be no problem. The maker of your WAN hardware, no doubt, has modules available for your system that let you feed the balanced audio and data into an adapter that create appropriate format data to be merged into the WAN data stream, thus, you have eliminated the need to install any cables!

If you have "dark fiber" available to you, Telecast Fiber and others make adapters which can take the audio and data, and run them down the fiber, even while running other audio, video and data down the same fiber for other purposes.

Need to be able to "dial in" with a keypanel from a remote location to a matrix somewhere? Multi-Tech and other modem manufacturers make voice over data modems that can do the job. Intraplex and others make equipment that can take the voice and data signals and send them via ISDN or switched 56.

Do both locations have bi-directional radio equipment? For example, satellite uplinks and downlinks, microwave studio-transmitter links (STLs), or wideband full duplex two-way radios. These will also work with appropriate modulators.

Again, with one possible concern, which is discussed in the next section, it does not matter how you get the signals between keypanel and the matrix, simply that you do.

Polling Issues

Earlier, I mentioned one area of possible concern. In the examples I gave, where the distance between matrix and keypanels is large, the transit time can become problematic. If the distance is great enough, even the speed of light becomes a limiting factor.

Geo-synchronous satellites are 22,000 miles above the earth. To send a signal up to one, and back down again will take on the order of a quarter of a second. To complete a round trip will take half of a second (500 milliseconds), at best. You may have heard this phenomenon on international telephone calls with your own voice coming back to you greatly delayed. While the voice delay can be distracting, the delays in data are the real problem. These data delays can become a problem even when the distance between the matrix and keypanel is "only" 3,000 miles – because the encoders, modems, muxes, etc. in that path also add delay; 30 milliseconds is typical.

We talked earlier about how addressing of keypanels is critical in the matrix intercom system. The way in which addresses work is as follows:

In a given group of 8 panels sharing a common data line, the data gets sent from a keypanel to and from a matrix by a process called **polling**. The matrix will broadcast a signal to all eight panels to the effect of “Panel Number 1, do you have any changes for me to act upon?” These changes could be as simple as a talk key having been pressed or as complex as the user wanting to see a list of all available party-lines. The matrix expects an answer from the panel, either a simple “nope, nothing new to report” or a request for a specific action.

The matrix normally will not wait very long (less than 10 milliseconds) for an answer before deciding that the panel in question is not there, and moving onto panel number 2, and so on, up to panel 8, and then starting all over again at panel number 1. The short wait is mandated in order to assure quick response to panel requests. This 10 milliseconds is the “polling window”, the 30 milliseconds between LA and NYC is the “polling delay”.

To make such a system work without unduly slowing down all panels by globally increasing the system polling delay, you can use the **AZ™-EDIT** configuration software to allow a longer poll delay (say 33 milliseconds) for one panel with no appreciable impact on other panels.

In the case of 250+ milliseconds delay due to satellite transit time, it is common practice to make sure the keypanel associated with the delay is in a group of eight ports where the delay is not important. For example, on ports that are used for paging outputs or IFBs, where there is no other data present.

In these ways, remote keypanels become very manageable and feasible, due in large part to their common format of standard balanced audio and RS-485 data.

Very Large Systems, Split Operation and Trunking

We have used the term trunking earlier and likened it to the long distance telephone system. In the case of **RTS™ ADAM™** matrix intercom systems, that analogy is very close to reality. Before we get deeply into trunking, let’s discuss the different ways available to make large systems.

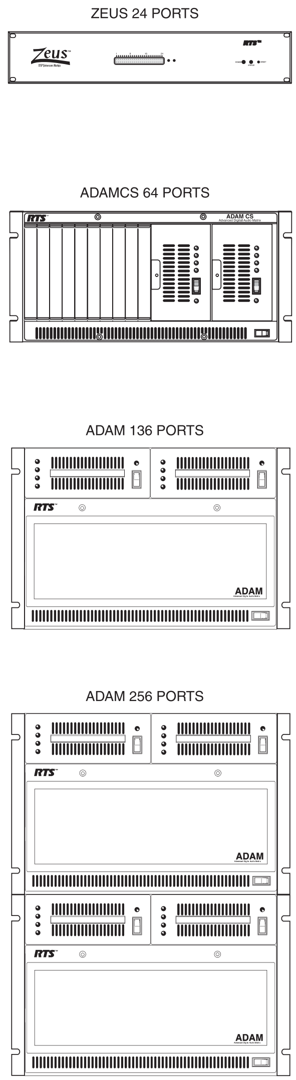
First, exactly what do we mean by a large system? How big is “BIG?” As we discussed earlier, with older technology (pre-TDM), systems were limited to a certain size (as a practical matter, in the “few” hundreds of ports) because of physical size and cost, not because of technological or logistic limitations.

Today, intercom matrices in general, and **RTS™** intercom matrices in particular, have a higher absolute limit, and a larger “typical size”. For example, in the early 1980s, a well appointed high end Sports Truck, the type which would do an NFL game, likely had 12 or so channels of PL, 6 IFB channels and 6 ISO channels. Today, most “network size” trucks carry 64+ ports of **ADAM™** matrix, and in some cases, over 100 ports. The intercoms have grown to carry program audio for monitoring, support 10, 15, or 20+ cameras, a host of graphics operators, and statistics personnel. Clearly, what is typical today was unimaginable less than 20 years ago.

Let’s consider matrix sizes for a moment, again sticking to those I know best:

- **RTS™ Zeus™** Matrix Intercom System: 24 ports fixed.
- **RTS™ ADAM™-CS** Matrix Intercom System: 8 – 64 ports in groups of 8.
- **RTS™ ADAM™** Matrix Intercom Single Frame: 8 – 136 ports in groups of 8.
- **RTS™ ADAM™** Matrix Intercom Multiple Frames: 136 – 1,000 ports in groups of 8.

Figure 5.4 A Comparison of Relative System Sizes



These are the numbers of ports that are available in a single **RTS™** intercom matrix from Telex. Other manufacturers offer systems in sizes from eight to approximately 500 ports. As you can see, size is not a limitation in most cases. At the time of this writing, the largest known single matrix intercom system in service is a **RTS™ ADAM™** system which consists of 784 ports at both ESPN and NBC.

Size and capability are not the limiting factor in most cases. Many factors may guide the design in favor of smaller individual systems. If the system is needed for four separate studios in a facility, which never or very rarely work together, then it may make more sense to use four separate systems. Some very good reasons for doing this might include:

- *Cost*: Four 128-port systems cost less than one 512-port system.
- *Reliability*: A fire in one rack room will not destroy the entire system.
- *Manageability*: Four different control studios have four different crews affecting the setup of their operation.
- *Shorter cable runs*: The matrix for a given group of panels can be physically closer to those panels.
- *Ease of Expansion*: It is easier to expand a single matrix if the needs for one area grow.

Now, let's take the opposite tack; what would be the reasons for going to a single large matrix? Some of the reasons might include:

- Operations require ability for any of the 512 users to communicate with any of the other users.
- Desire for single point of administration, control, troubleshooting and monitoring.
- Design of the facility is highly decentralized operationally, and day to day, different portions of the facility must work together.
- Certain users must work with all the facilities, and giving them four separate keypanels (one per system) is not feasible.

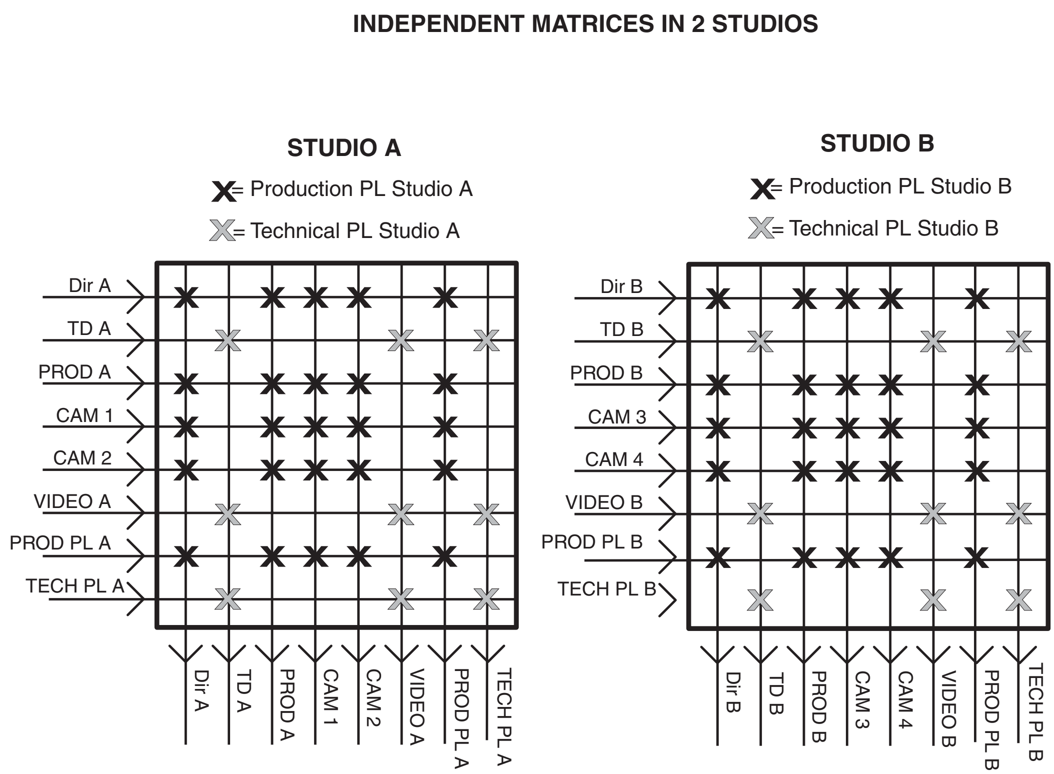
Now we have helped to identify whether to use one large matrix or a number of smaller ones. What happens when you get mixed answers to the questions above? Certain requirements drive you to use separate matrices, but one or two key factors seem to demand a single matrix.

A couple of different options or "hybrid designs" can be used in these cases.

The first and simplest is to define a few common points of contact between the intercom matrices. Take the following example, a television complex has two studios and two control rooms. Normally Control A works with Studio A, and Control B with Studio B. Occasionally, the wall between the two studios opens, (never mind how; that's the architects problem!) and there is a need for Control A to work with the cameras in the combined Studio AB.

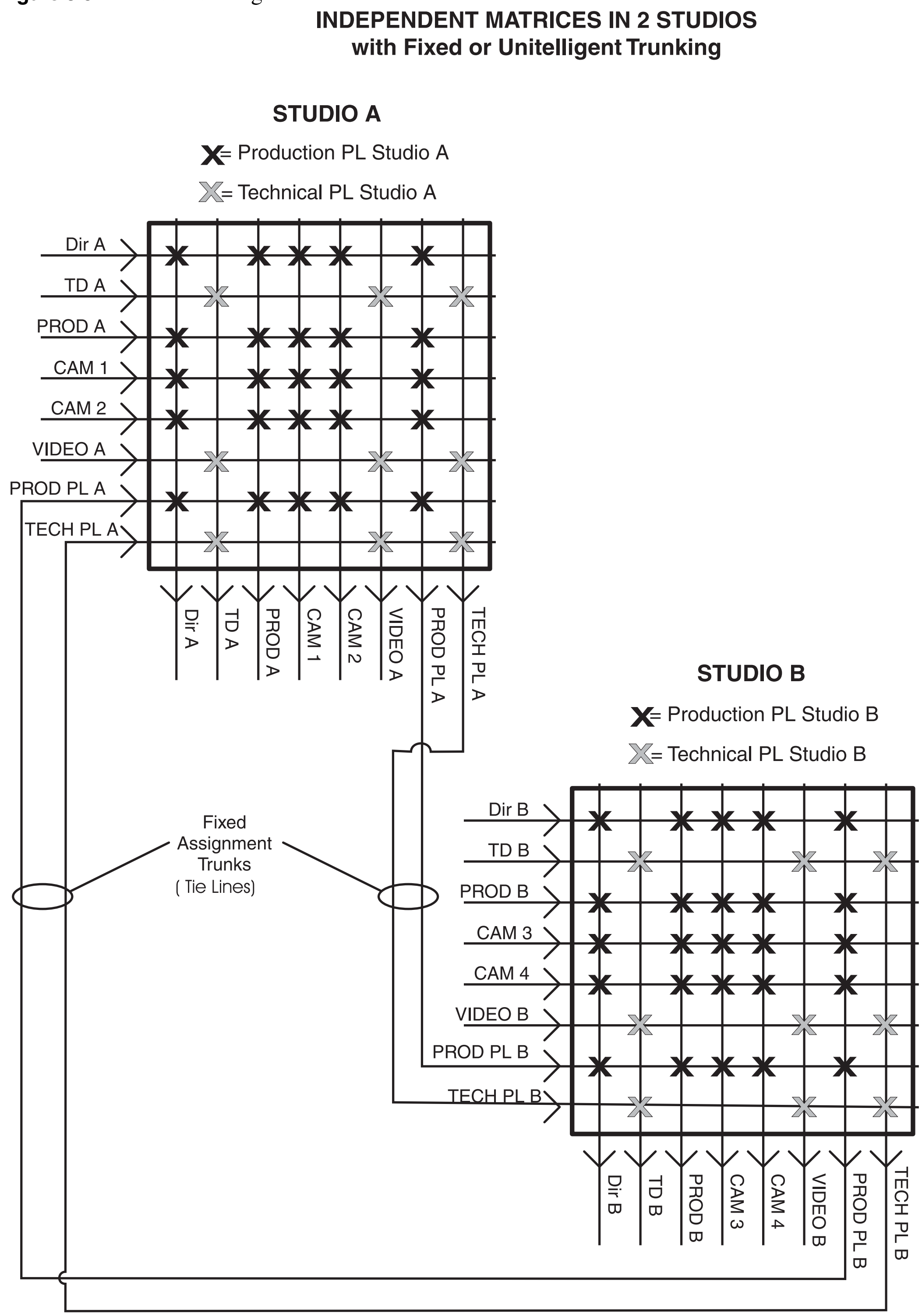
Let's further presume the normal method of operation has the cameras in each studio receiving two channels of intercom; a "Technical PL" created in the intercom configuration, and a "Production PL" also created in the intercom configuration of the respective matrices for Studio A and Studio B.

Figure 5.5 Separate Studios, Separate Intercom



A quick way of allowing the combined operation would be to configure (in **AZ™-EDIT**) the Production and Technical PLs of each matrix to include two available sets of ports on a jackfield. Then, simply connect the output of Production PL from Studio A to the input of Production PL for Studio B, and conversely, connect the output of Production PL from Studio B to the input of Production PL for Studio A. Do the same for Technical PLs.

Figure 5.6 Fixed Trunking



Now, any conversations on Production PL for A control will also be available to the Studio B cameras for both talking and listening, and the same is true for the Technical PL. Our problem is solved.

The technique described is called **trunking**; the two ports of each system assigned to PLs have been “trunked” to one another. For reasons that will become clear later, we refer to this as “dumb” or unintelligent trunking. That isn’t to say that it isn’t a brilliant idea or solution. It means that no system intelligence was employed in establishing the trunks.

To go back to our telephone system analogy, this is early-20th century technology, harkening back to the days of an operator in your hometown asking the long distance operator for a “trunk” to Chicago. That trunk then connected you to your Aunt in Chicago.

But, “wait,” you say, “didn’t the telephone make this much easier back in the fifties by going to long distance area codes and direct distance dialing?” Yes, you are absolutely correct, give the reader a prize!

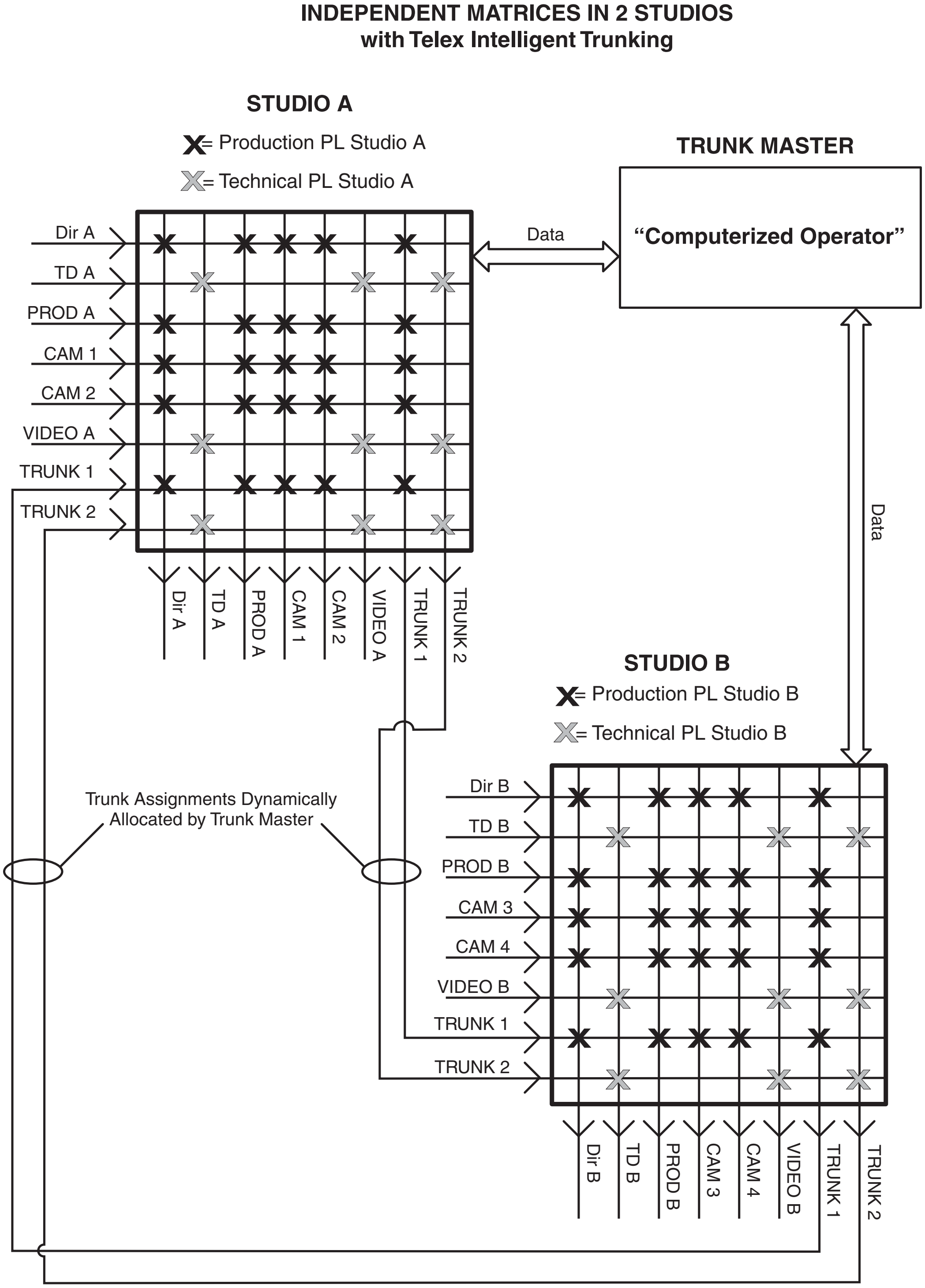
Today, some intercom matrices (including at least one from someone other than Telex) offer varying degrees of improved trunking that eliminates the manual patching described above.

WARNING Sales pitch coming – Telex has the largest, most intelligent, most proven trunking system available today, offering the ability to trunk more than 20 **ADAM™**, **ADAM™-CS,** or **Zues II** systems together.

This can all be done without human intervention and in a system comparable to the long distance telephone system. Let’s look at some of the features and attributes of the system.

Taking the example of the two Production and Technical party-lines manually trunked together given earlier, let’s make a couple small changes. Make the “trunking ports” assignable, and give them the designations “Trunk A” and “Trunk B,” Connect a “computerized operator” between the two systems, communicating via a standard RS-232 serial port with both matrices. Let’s call the computerized operator the “Trunk Master.”

Figure 5.7 Intelligent Trunking



Now, all we need to do is assign "area codes" to identify which matrix has which port. In actuality, in the **Telex®** Intelligent Trunking system, the trunk master figures out which matrix has which ports and keeps track of it for you. If you assign "ADIR" from the Studio A matrix to a panel on the Matrix for Control Room B, the system "knows" that it will have to configure and establish a trunk to allow that conversation to take place. It does so automatically, establishing the trunk, monitoring trunk usage, and releasing the trunk when the conversation is completed.

"Great" you say, "Why not always trunk and avoid HUGE matrices?" I'm glad you asked that question.

First, there is what I refer to as the "Mother's Day Syndrome." Mother's Day rolls around, and all good sons and daughters decide to call their dear, sweet mom and wish her the best, and many of them don't get through. They hear a nice recording of someone saying, "All circuits are busy, please try your call again later." If you think about it, you have probably gotten that message a few times in your life when calling long distance, and never when calling someone down the block. This is because local calls (large metropolitan areas excluded) go through a single matrix (single central office), and there is a dedicated crosspoint (or a close equivalent) for each path. You get the message when calling long distance because there are a limited, finite number of long distance trunks available, and the heavy traffic volume keeps all of them busy at times.

Looking at the last example, imagine what would happen when the first person from Matrix A calls someone in Matrix B, Trunk A (or B) gets assigned, and life is good. A second person (maybe from B calling A this time) initiates a call, the other trunk is assigned and life is good. Now a third person in Matrix A decides to try to call someone in Matrix B. Oops, "All circuits are busy, please try your call again later."

In actuality, no voice is heard, but the calling party does get a busy indication on their panel, and the call does not go through. Therefore, we can see that trunking systems need to be sized appropriately for the anticipated traffic. *Appropriately* is the key. The Telephone Company (actually "companies" in the post-AT&T breakup era) set aside enough trunks to handle all of the traffic most of the time – sounds suspiciously like "You can fool all the people some of the time," doesn't it?

Telex® Intelligent Trunking shares something else in common with the telephone company, the trunk master continuously monitors and reports on status of trunk utilization. The telephone companies do it in great "war rooms" with multi-story maps with lighted paths. Telex does it with a constantly updated and logged report of trunk utilization on a conventional PC. It keeps track of (amongst other things) the maximum number of trunks you use simultaneously in the past x amount of time. With good historical data, you can determine the number of trunks you set aside for trunking.

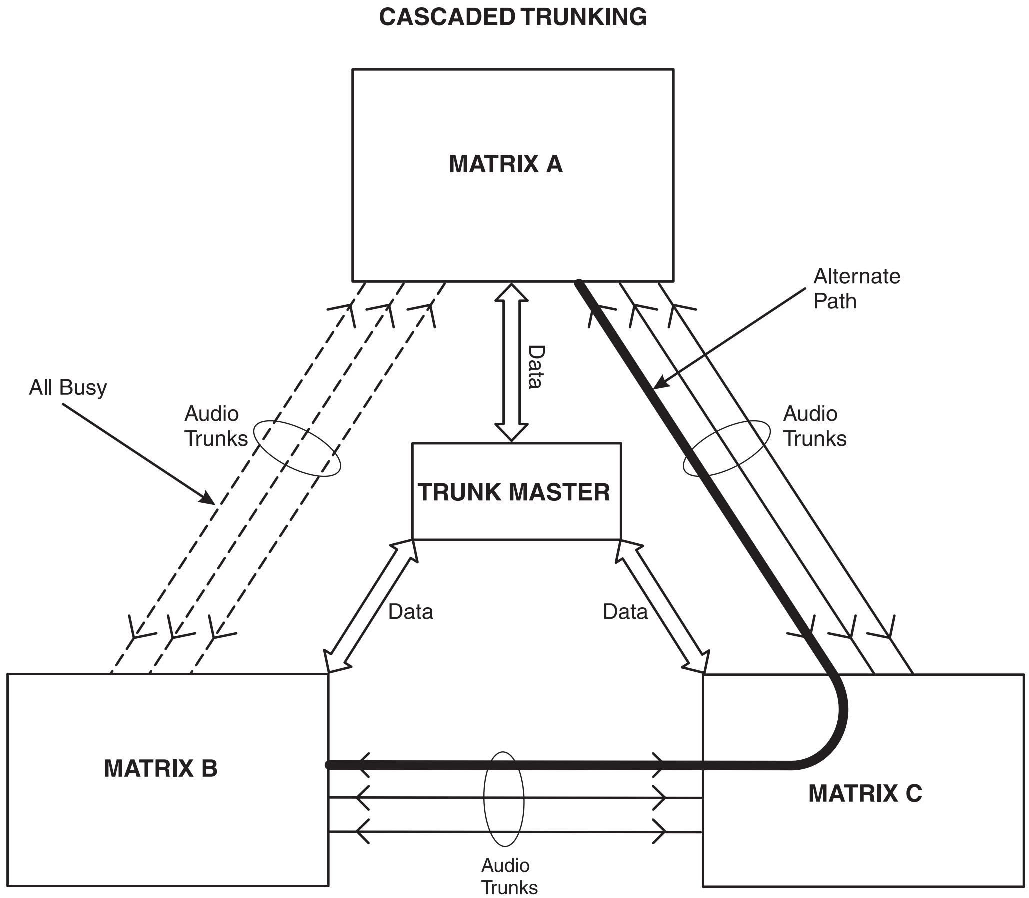
However clever you think you are in setting aside trunks, there will always exist the unforeseen possibility that you may run out of trunks at some point.

For example, you have two studios, trunked together with five trunks, and in the past year have never used more than four at one time. Today, both studios are manned, and in Studio B is a news program being directed by Steven Spielberg, produced by George Lucas, with Tom Brokaw interviewing Madonna and Jerry Falwell (it could happen!). Studio A is busy doing a documentary on the history of dental appliances in South America. Care to take a guess how many of the crew in studio A will decide to listen in to the director, producer, talent IFB, program audio, cameras from B? All at the same time? Know what'll happen? Yep, "All circuits are busy, please try your call again later."

The other significant limitation may be for each trunk you assign (which requires a port), you give up a port that be used for two keypanels (one at each matrix). Make your system too long distance "friendly" by allocating a lot of ports as trunks, and you either limit the number of keypanels on each matrix or spend more money to buy additional ports for each matrix.

All of these limitations aside, trunking can be a very good solution for many applications. Trunking works best when limited numbers of trunks are required to support occasional usage. Trunking works very well when many matrices need to be interconnected. As noted earlier, **Telex®** Intelligent Trunking can simultaneously handle automated routing between more than 30 matrices. A side benefit of such a multiple matrix trunked system is that the trunk master can figure out and establish trunk paths via multiple hops if needed due to trunk usage. If the trunks from Matrix A to B (see Figure 5.8) are all in use, the possibility exists for the trunk master to route a signal from A to C and from C to B, thereby bypassing the bottleneck.

Figure 5.8 Cascaded Trunking



Another advantage of trunking is that there is no requirement for the individual matrices involved to be in close proximity. Systems, which are hundreds or even thousands of miles away, have been successfully trunked using techniques described earlier with respect to remote keypanels. Trunking is nearly identical to those situations, requiring the transmission of a single data signal and the appropriate number of audio signals.

Note A series of articles written by Andy Morris and Ralph Strader, on trunking at NBC, appeared in Broadcast Engineering magazine in 1996. These articles, as well as an article on Trunking Supervisory Systems by Robert Streeter and Thom Drewke of NBC, in PDF format, are included on the CD.

A final methodology for distributing large matrices is a function of the manner in which multiple **ADAM™** frames are interconnected. When two **ADAM™** frames, each 128 ports, are connected together, they become a single 256-port intercom system. The interconnect between the two frames is through a Bus Expander, which transports all 128 ports between the two frames without rendering any of them unusable for keypanels.

The physical interconnect between the frames with bus expanders can either be via a pair of coaxial cables, which can be used for distances up to 1,000 feet, or via a pair of fiber optic cables, which can run for over 1,000 meters. The signal sent over the fiber or coax is a multiplexed data stream, running at approximately 220 megabits/second. Since this data rate is lower than the 270 megabit CCIR-601 serial digital video standard, many of the asynchronous devices that can transport serial digital video can be used for this signal to achieve even greater distances.

By using the Bus Expander with multiple **ADAM™** frames, a single electrical matrix can be located floors or buildings apart within a complex, and yet function as a single large matrix.

Now that we have discussed a number of different methods used to create large intercoms, and how to interconnect smaller intercoms into a single system, let’s move onto interfacing and accessories.

Interfacing

It is rare that an intercom system is an island unto itself. Communications has become such a pervasive need and set of technologies that it’s not a matter of IF you interconnect; it is more a matter of WHEN and HOW you interconnect.

If you doubt this, consider that today in your home, you may have a cable modem connecting your PC to your cable TV system; you may have your PC answering your phone and taking messages with an embedded voice mail system. Soon, you may have your refrigerator talking to the local supermarket over the Internet, ordering tomatoes and milk.

Some of the more common needs for interfacing, which are encountered when installing or modifying an intercom system are presented in this chapter.

Signal Formats

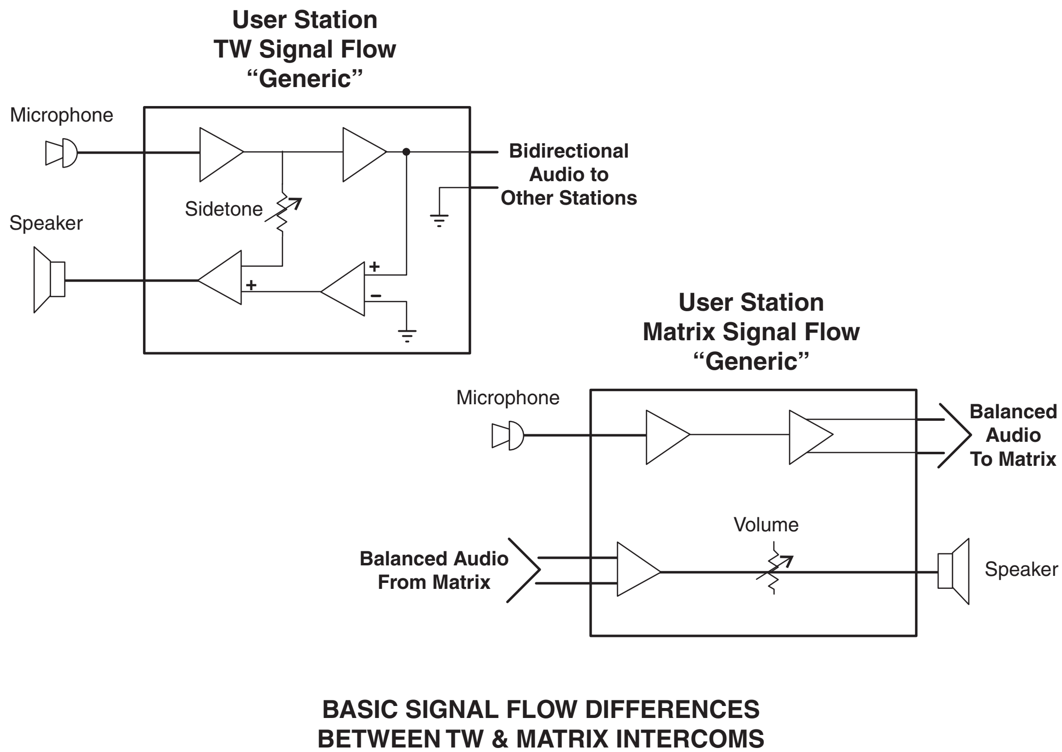
TW and Wireless systems often are tied to matrix intercom systems. A brief description of the signal formats of the various types of intercom systems are helpful.

In any intercommunications (intercom) system, the “inter” refers to two-way communication. For the purposes of this discussion, we label one of the directions as “talk” and the other as “listen”. Obviously, either party in a conversation can be talking or listening at any time, or even at the same time. “Talk” or “Listen” is a matter of perspective. In a given two-way communication, what “talk” is to me is “listen” to you and vice versa. This is only a matter of semantics, as far as this discussion goes; what is key is both sides of the communications can be occurring simultaneously.

In a matrix intercom system of the type that RTS manufactures, the talk and listen signals are full duplex and travel on their individual pairs of wires.

In a TW system, regardless of the manufacturer, the communication is also full duplex, but both sides of the conversation travel on the same pair of wires.

Figure 5.9 TW and Matrix Signal Flows



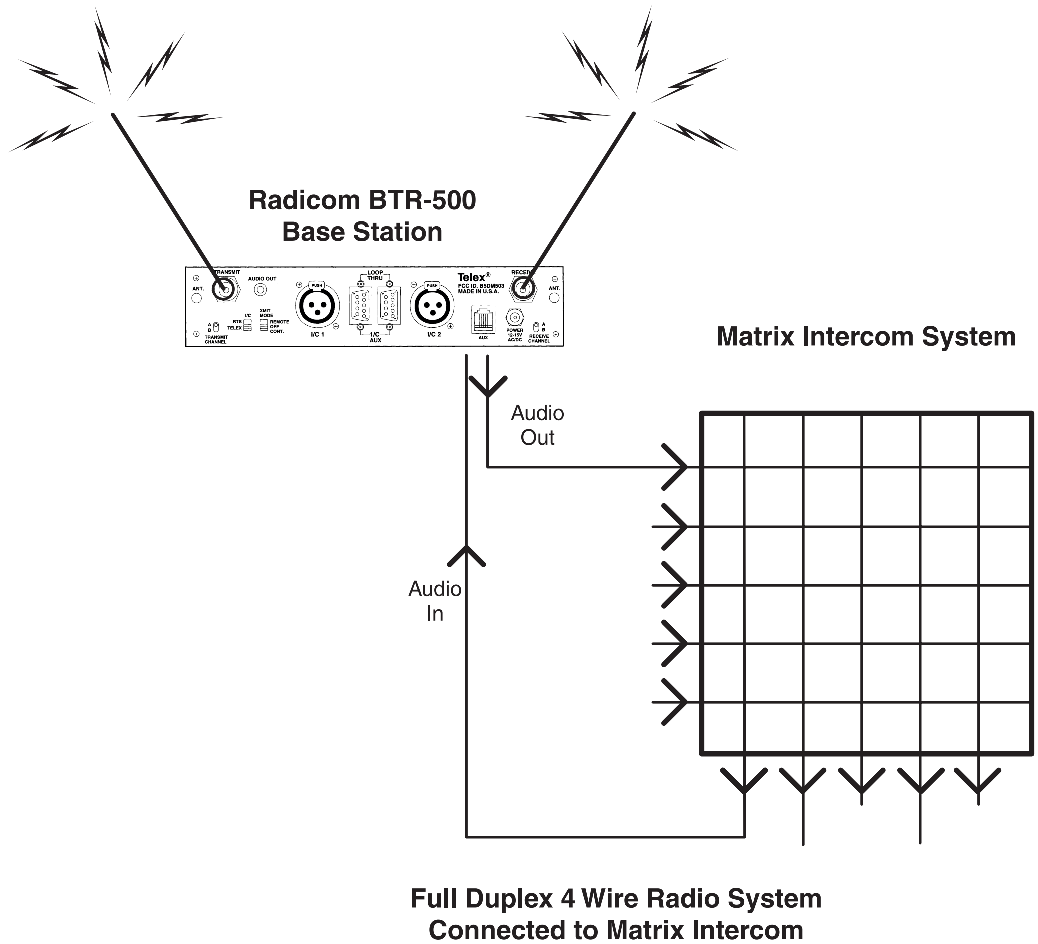
In a wireless intercom system, the communication may be full duplex, with the two sides of the conversation carried on two separate frequencies. This is the case with all the **Telex® RadioCom™** products. In this way, the signal format is essentially the same as the matrix intercom system shown in Figure 5.9.

In some wireless communications systems (two-way radios for example), both talk and listen may share a single frequency, in which case the communication must be half duplex, with the users taking turns between talking and listening. A good example of such a system is low cost walkie-talkies, wherein the speaker you hear audio from doubles as the microphone you speak into when you press the transmit button. In the following discussion, we do not concern ourselves with that variety of two-way radio systems because those systems are rarely encountered in installations with intercom systems. For more detailed information, refer to the chapters on wireless intercom.

Interconnecting Matrix, PL, and Wireless Systems

As discussed earlier, the signal format for **ADAM™**, **ADAM™-CS** and **Zeus™** Matrices is the same as used in the **Telex® RadioCom™** line of wireless intercoms. This makes interfacing between the two systems very easy, and in fact, Telex provides connectors specifically for this on the **RadioCom™** products. To make the two systems work together, you simply connect two audio lines.

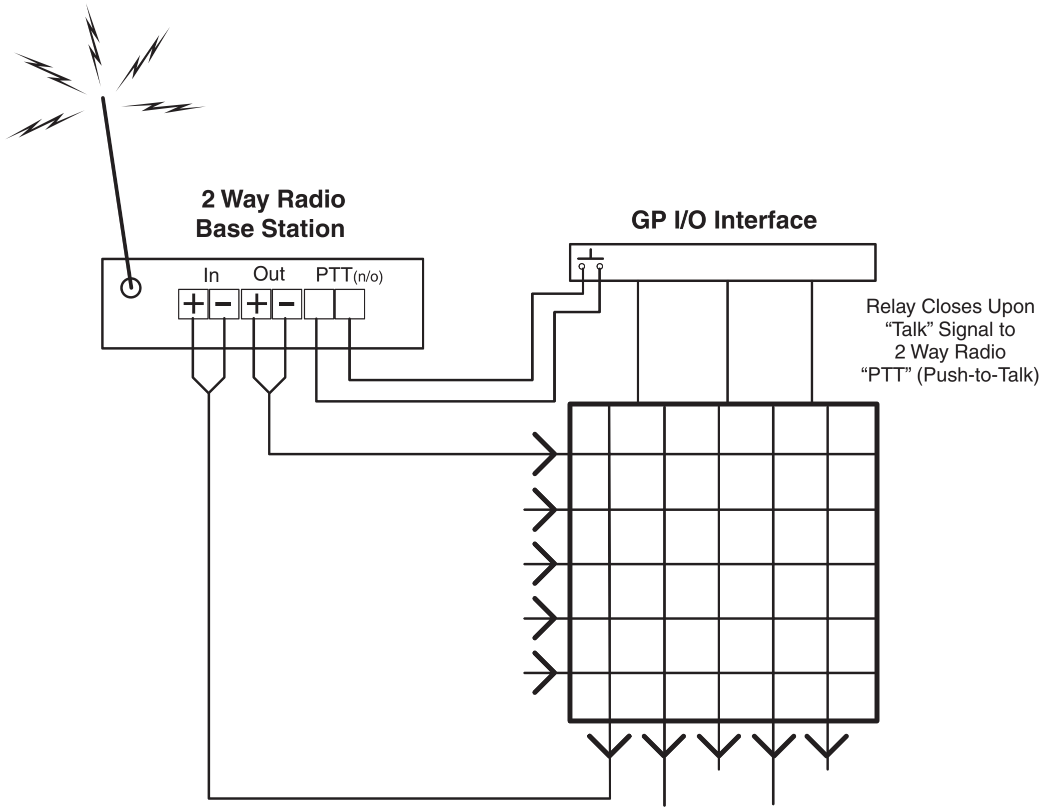
Figure 5.10 Wireless Intercom Interfaced to Matrix Intercom



Since the **RadioCom™** system is full duplex, with the base station transmitting continuously, there is no need for the matrix intercom to provide a PTT (Push To Talk) signal to the base.

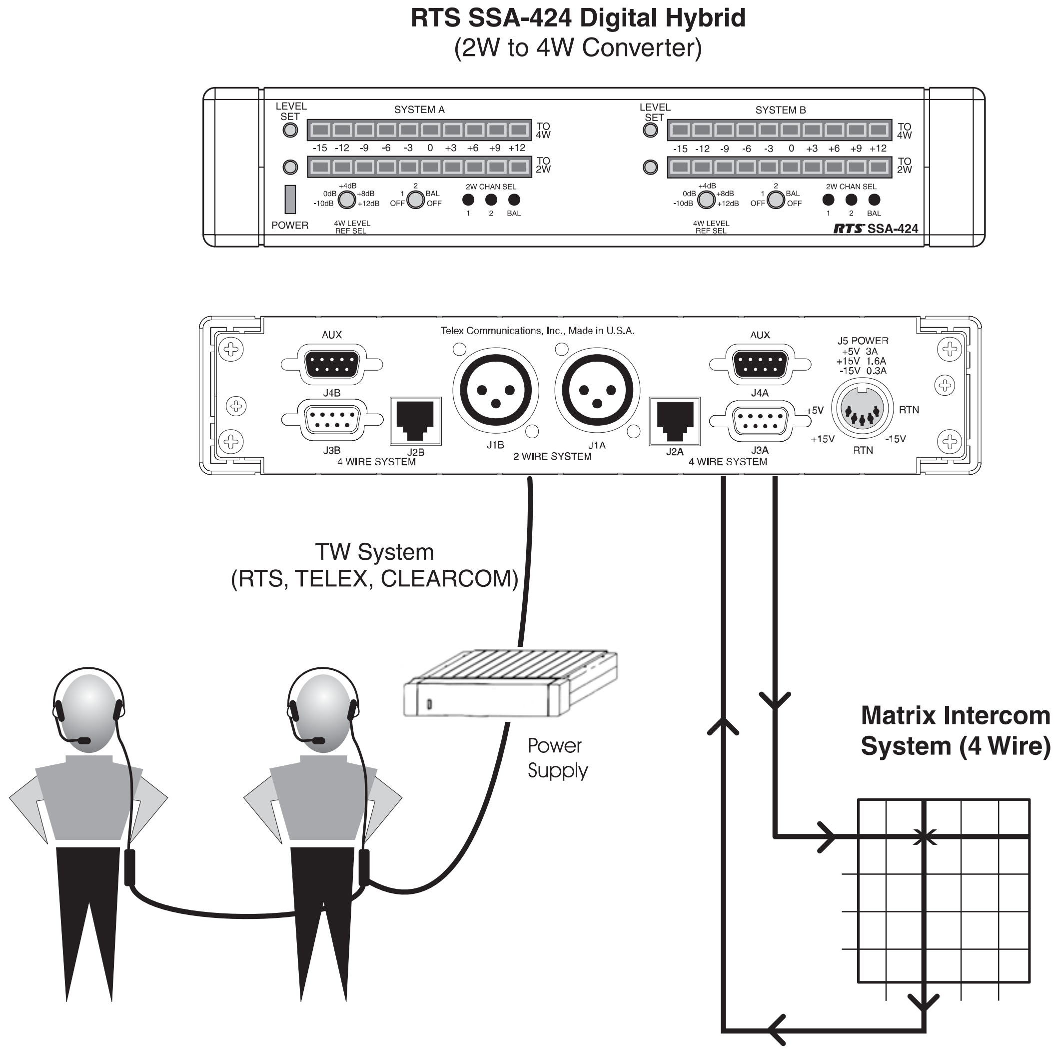
In the case of radio systems where the base station is not transmitting continuously, the matrix must provide a logic signal corresponding to a user pushing an intercom key to talk to that wireless system. ADAM, ADAM-CS, and Zeus all come standard with logic signals, with open collector outputs for this purpose, and have available the UIO-256, as an accessory, which can provide an actual relay closure, if required.

Figure 5.11 GPI/O Implemented PTT (Push-To-Talk)



As mentioned earlier, TW systems are, by definition, two-wire (one pair) communications systems, having both talk and listen present at the same time on the same conductors. In order to connect a TW intercom system to a four-wire system an interface is required. This interface is known by a number of different names, including: **hybrid**, **two-wire to four-wire converter**, and **system interface**. Regardless of the name, the function is simple, although the technology is not.

Figure 5.12 TW to Matrix Interface



The hybrid, in Figure 5.12, acts as a "traffic cop" allowing the talk signal from the matrix to be applied to the bi-directional TW line while blocking its return when the talk signal from the TW is presented to the matrix. The effect of the blocking is termed "nulling", as it cancels of the return signal. The effectiveness of the cancellation is driven by many factors. Hybrids are generally available from many sources, including intercom manufacturers, Gentner, Telos, and others. Telex has two models available, the **RTS**™ SSA-324, and the **RTS**™ SSA-424. Both units are suitable for most applications. The primary difference is the SSA-424 is digital and auto-nulling, eliminating the need for manual setup and calibration.

Software Considerations

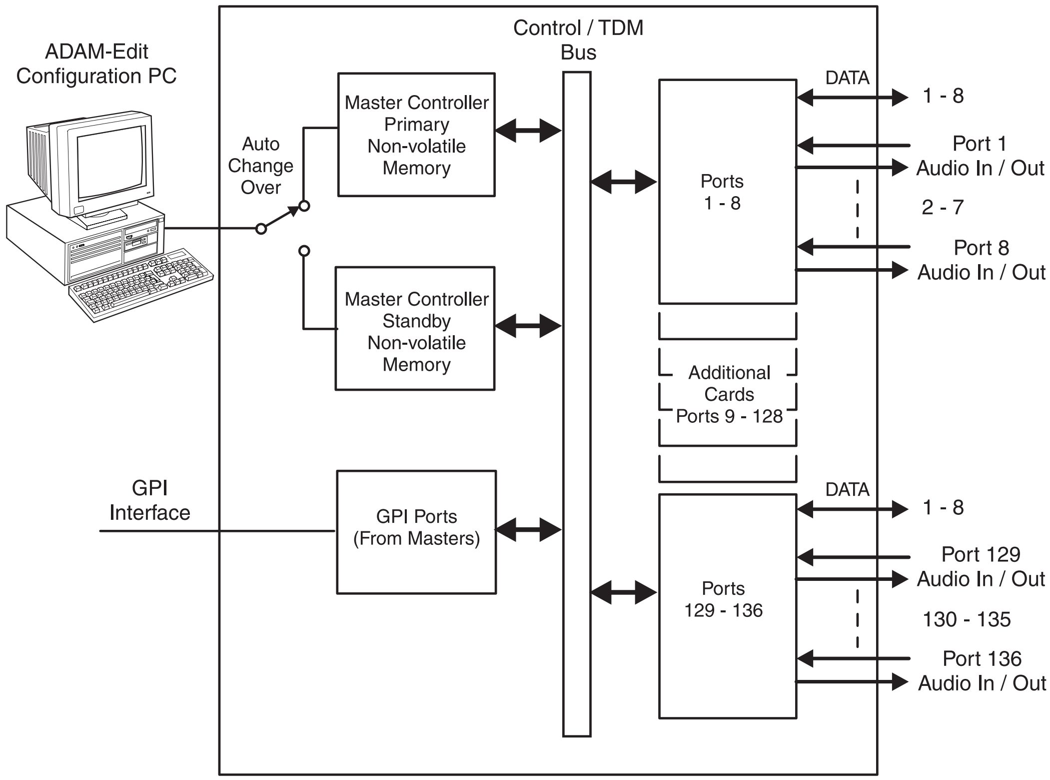
Until now, we have concentrated on the physical and hardware issues for a matrix intercom system. As noted earlier, the intercom matrix itself is a matrix mixer, which is capable of mixing any combination of inputs to any output. A 50-port system is literally a 50-input by 50-output bus digital mixer. Firmware and software are what turns this digital mixer into an intercom system.

For the following discussion, it is helpful to understand the different roles played by the system firmware and software in a matrix intercom system. As system architectures vary, and some information is proprietary to each manufacturer, the information being presented here is specific to **RTS™ ADAM**, **ADAM™-CS** and **Zeus**. The basic concepts hold for other matrix intercoms on the market.

A brief word on the architecture of the **ADAM™**, **ADAM™-CS** and **Zeus™** matrix intercom systems will set the stage. Zeus is the entry-level matrix and is configured as 24 ports, and does not include power supply or controller redundancy. ADAM and ADAM-CS are expandable systems, and are standard with redundant power supplies and redundant auto-switching controllers. Apart from these differences, and the physical characteristics, the three matrices are very similar.

Communications to and from the keypanels is handled by serial data ports, which are RS-485 based, and each port controls a group of eight keypanels. The need for addressing of the keypanels was covered earlier. The information sent to and received from the user stations is stored within the intercom matrix in non-volatile memory.

Figure 5.13 **ADAM™** and **ADAM™ CS** Basic Components



Note The diagram in Figure 5.13 and the discussion that follows can also be applied, with a few minor exceptions, to the **Zeus™** system.

As seen in Figure 5.13, the intercom system has provisions for an external PC, which is used to do initial setups and configurations, including: naming of ports, assigning of PLs, creation of IFBs, creation of ISOs, etc. The PC is also useful in monitoring system status and for other housekeeping functions.

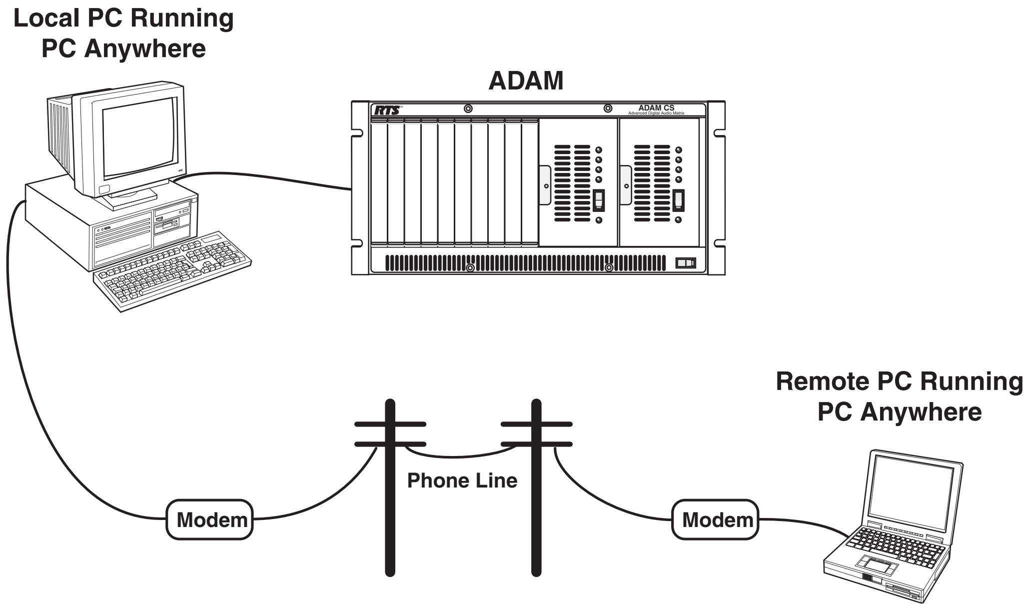
The PC is not required for operation of the matrix, except in certain very rare circumstances where UPL statements need to act on files, or in response to date information. It is perfectly acceptable to use a PC to configure the intercom, and then remove the PC. Even without the PC connected, the intercom will function normally. The intercom recovers from power failures, and in the case of ADAM and ADAM-CS, primary controller failure, all without need for a connected PC.

Included on the enclosed CD is a copy of AZ-EDIT, the windows-based configuration programs for the **RTS™** line of matrix intercom products. These programs can run without a connected matrix and the best way to learn the programs is to install them. An extensive help file is provided and the program is laid out in a logical manner.

Because the configuration software is run on a standard Windows PC, and communicates with the matrix via a standard serial RS-232 port, a number of possibilities exist for remote configuration, control, and monitoring. One option is to replace the PC with an auto-answering modem. This permits the PC, which is running AZ-EDIT, to connect from anywhere via telephone lines and remotely control and diagnose the intercom.

If that notion strikes you as just a bit too insecure, there are a number of available utilities such as PC-Anywhere, which can be used to accomplish the same thing in a different manner. Install a PC running AZ-EDIT and PC-Anywhere at the matrix location. Use another PC, running PC-Anywhere to dial into the PC at the matrix, running PC-Anywhere, then supply the required login information, including security password, and again, you have full ability to control and monitor the matrix remotely.

Figure 5.14 Matrix Intercom Remote Control



As noted earlier, the differences between system architectures for control of matrix intercom systems from different manufacturers are significant. We do not go any further in describing them, except to point out that in the case of **Telex® RTS™** Intercom systems, the supplied software is included on the enclosed CD. You are encouraged to play with it.

CHAPTER

6

INTRODUCTION TO WIRELESS INTERCOM SYSTEMS

TOM TURKINGTON

Introduction to Wireless Intercoms

Wireless intercoms have a long and important history as part of the communication professional's repertoire. They have gone through many changes and technological improvements over the years to bring us to where we are today. The purpose of this chapter is to allow you to become familiar with the history, general workings, and special considerations of wireless intercoms. This includes their advantages and disadvantages so that in the next chapter we may explore the wild, sometimes weird, but almost never boring, world of wireless intercom systems design.

Note The use of the term **RF** is made extensively throughout this chapter and the next. RF is an abbreviation for **Radio Frequency**. If you are unfamiliar with the term and would like a detailed explanation of what RF is, see the definition in the glossary of this book.

History of Wireless Intercoms

In the beginning there was wire, and the wire was good. Soon engineers realized if they could cut the wires and move the audio, video and communications signals around the television venue without encumbering cables, they would have tremendous freedom to accommodate ever-increasing production challenges. They also believed that wireless transmission of signals would make their job easier by not having to run miles of cable for large remote productions. It turned out not to be so simple. Developing wireless microphones, wireless cameras and wireless intercom systems would be a trial and error adventure that has spanned the last 30 years or more, and it is not over yet!

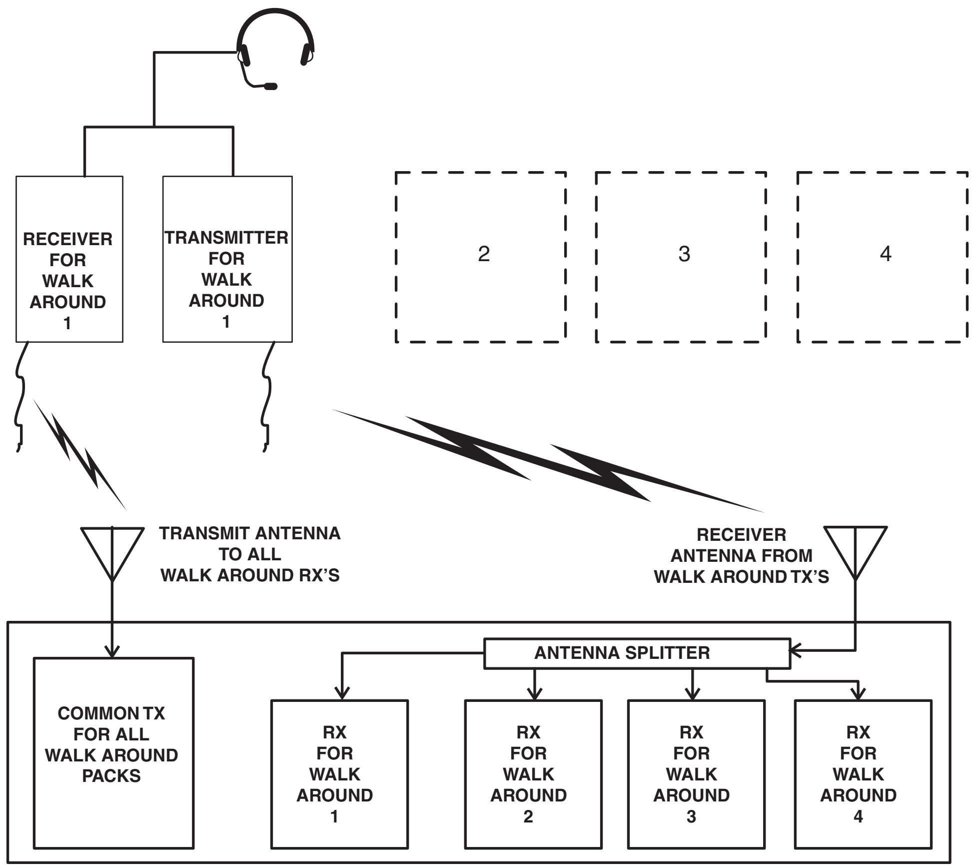
In this section, we look back at the history of wireless intercom systems and see what we have learned about wireless communications in the process. The original "wireless intercom" consisted of two-way radios and (if you were lucky) a headset. The advantages were the technology was readily available and it was relatively inexpensive to use. Two-ways worked well for some applications, such as pre-show setup and post-show teardown where they are still used today in much the same way they were 30 years ago. Two-ways (now often called HTs or Handie-Talkies) have higher operating power which affords substantially increased operating range of over a mile or more in some cases. This range can be increased to cover an entire city by the use of repeater stations located at the top of centrally located buildings.

Two-way radios did not, however, do as well for the rigors of live television production. In live TV, the restrictive nature of HTs was only too evident. First, HTs utilize a half-duplex communication scheme. Half-duplex means that while there is bi-directional conversation, only one user may communicate at a time and all other users must listen until the person who is communicating is finished. During setup this does not pose a huge problem, but during a show, when seconds can seem like hours, this can be a real problem. Imagine a cameraman is transmitting over a half-duplex HT system, while the director is trying to take a new shot or make some other time-critical change. Obviously, a half-duplex system would never do.

Soon after it became apparent that a half-duplex communications system would never satisfy the needs of on-air production, a vast array of new HT-based system configurations emerged. The greatest of these utilized two HTs on each user and multiple base station units in a complex repeater configuration. An interface box allowed users to wear one headset that fed both radios at once. While achieving some of the functionality of the most basic modern day wireless intercom, the system was bulky, heavy and unreliable due to the numerous wires and complexity of setup. While this system was much closer, it still did not offer communications professionals the robust functionality and reliability they needed for day-to-day operations.

The next generation of wireless intercoms to hit the scene was truly a breakthrough. It eliminated much of the complex wiring and minimized the equipment the user was forced to wear. The system consisted of a base station and multiple user beltpack pairs. In the base station there was a single transmitter and multiple receivers (one for each wireless user). The audio coming from each receiver was put on a single intercom channel or audio bus, and was fed to the transmitter as well as an external intercom line. The transmitter was a low power, always on unit that maintained constant outgoing information to all wireless users. See Figure 6.1.

Figure 6.1 The first beltpack based wireless intercom system.



Each user station in the system consisted of two beltpacks, one for transmit and one for receive. Two beltpacks were necessary to combat the phenomenon know as desensing, where a transmitter in close proximity to a receiver causes the receiver to have greatly reduced sensitivity. Desensing is discussed in greater detail in a future section. Each wireless user's transmitter was on a unique frequency which corresponded to their own receiver in the base station. All of the wireless users' receivers were tuned to the same frequency which corresponded to the single base transmitter. A single headset with a split feed cable eliminated the need for an external headset interface box. See Figure 6.1. By utilizing this system, each wireless user could communicate to both hardwired and wireless intercom users in a full duplex mode.

This system, at long last, provided engineers with a reliable and functional solution to the wireless communications problem. Future systems would combine the transmit and receive beltpacks and incorporate numerous interfacing and operational advantages. We look at some of these in the next section.

Modern Day Wireless Intercoms

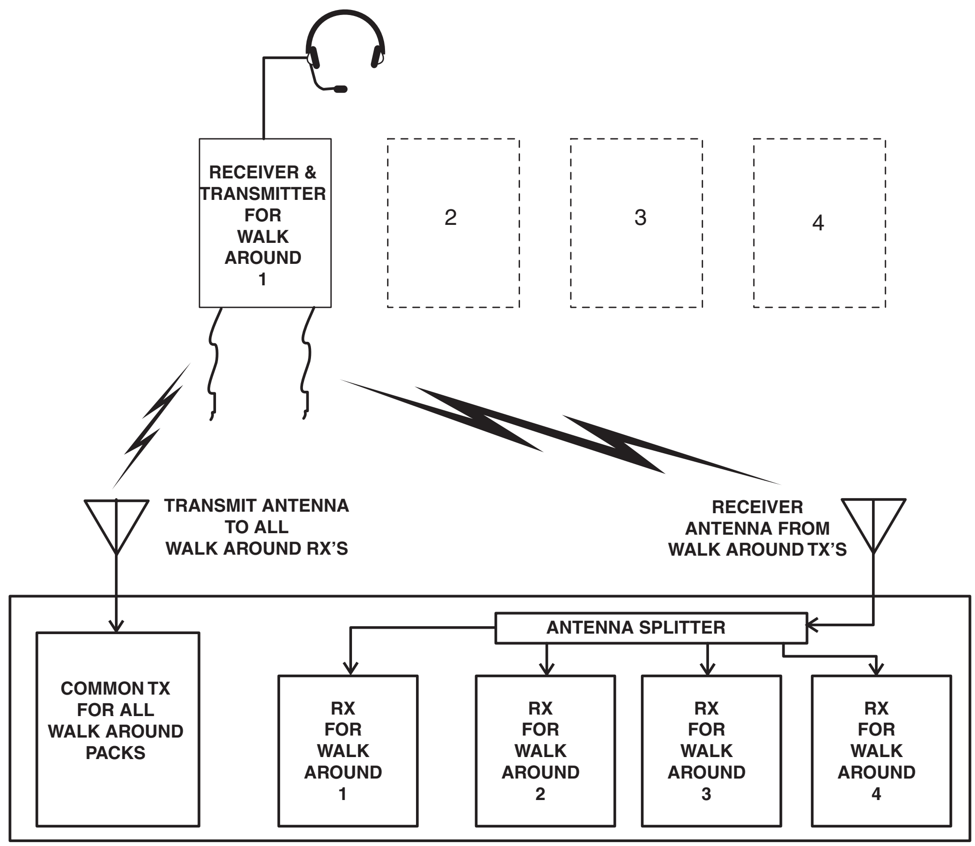
Today's wireless intercom systems are technological giants compared to their earlier predecessors. They allow users to "cut the cable" of hardwired party-line systems and move about freely within the system's operational range. Modern wireless intercoms can be either party-line intercom systems or individual beltpack systems that allow users to operate independently from other wireless users. Good quality systems can be seamlessly attached to existing hardwired communications systems commonly used in broadcast and other facilities. As discussed previously, modern, high quality wireless intercoms offer a distinct advantage over traditional, two-way radios in that they offer a more natural full-duplex operation. This enables all users on the system to speak and hear other users simultaneously without “covering” other users’ transmissions.

The demands of modern broadcast productions make the full-duplex operation of wireless intercom systems an absolute necessity for stage managers, lighting and audio technicians, or any professional who has to deal with the breakneck speed and complexity of television productions.

The spread of digital television (DTV) and the ever-increasing number of wireless users has made the available frequency spectrum a more difficult place in which to find available channels for wireless intercoms. The spectrum has also become a lot smaller, especially considering that four television channels (24 MHz of spectrum) have been reallocated for public safety use and the upcoming reallocation of UHF TV channels 60 through 69 (60 MHz of spectrum). Broadcast professionals now have to consider such factors as the compatibility of frequencies with each other, as well as, how to best avoid interference with local TV transmitters. We discuss these topics in more detail, later.

Unlike wireless microphones that operate only in one direction, wireless intercoms have more specific frequency spectrum requirements because of the relationship between the transmitter and receiver frequencies. Each intercom (if it is to be full-duplex) must have at least one system transmitter frequency that broadcasts to all beltpacks and one receiver frequency for each individual beltpack in the system. For a four beltpack system, also known as a four up, that means it must have a minimum of five total frequencies. See Figure 6.2.

Figure 6.2 An example of a modern day wireless intercom system.



Each beltpack must have a receiver set to the base transmit frequency and a transmitter set to its own unique receiver in the base. Due to a phenomenon mentioned earlier called desensing, these two frequencies must have a fairly large frequency separation, typically at least 12 MHz for VHF systems, and even more for UHF, or the transmitter will interfere with the receiver’s operation.

The answer to the frequency problem is to utilize a digitally synthesized, frequency agile system. That may sound simple enough in theory, but in reality, designing such a product is a totally different matter. A digitally synthesized, frequency agile system must not only incorporate a superior design with high-quality filtering to withstand the rigors of an overcrowded frequency spectrum, but it must also offer an ergonomically designed user interface that allows ease of frequency selection and operation. End users must experience the same ease of operation they get from their existing two-wire beltpacks.

To date, the chief limitation to most wireless intercoms (other than finding available spectrum) has been they are inherently one-channel in nature while the most common hardwired intercom system from RTS (used in virtually all TV broadcast trucks and facilities) is two-channel. Two-channel operation allows users to switch easily from one intercom channel to another. This allows a stage manager, for instance, to communicate with the producer and then switch over to the director circuit as necessary. Two-channel operation has become the hardwired industry standard and users who have increasingly relied on wireless intercoms must be able to employ that technology in wireless form without having to deal with huge racks full of equipment.

Wireless intercom systems that can operate in high RF environments must not only offer interference resistant operation, but must utilize design techniques that will not interfere with other wireless equipment like wireless microphones and IFBs. Another key to a wireless intercom's successful operation and coexistence with DTV is its ability to avoid strong local TV stations, as well as, coordinate multiple system frequencies. This holds true whether the system is VHF or UHF, fixed-frequency, frequency-agile or synthesized. Utilizing the minimum power necessary is absolutely critical if wireless intercoms are to coexist with other low-power wireless equipment. The utilization of intelligent systems that reduce beltpack transmitter power levels as they get closer to the base station can greatly decrease the harmful interference that can be associated with wireless communications gear.

Future wireless intercoms (see Figure 6.3) will need to provide users with frequency agility, high-end filtering, RF power management, ease of use, two-channel operation, extended battery life, small lightweight beltpack and a user interface that allows operational and frequency parameters to be easily set and checked without the use of external equipment, such as a laptop computer or special interface box.

Figure 6.3 The **RadioCom™** BTR-800 System is an outstanding example of the next generation of wireless intercom systems.

As wireless intercom applications for broadcast professionals continue to grow more complex and challenging, the need for products that can meet these challenges will also grow accordingly. All these factors and more, as discussed in this chapter, and in the next chapter, must be considered when looking at the quality and functionality of a modern wireless intercom system.

Special Considerations

Wireless communications are here to stay. They have become an integral part of the total professional communications package. There are, however, many factors associated with wireless that need to be understood and addressed that do not come into play with hardwired communications systems. In this section, we look at the special considerations that must be considered when deciding whether or not to implement a wireless system.

The first area of study is the RF spectrum and how it can be used to implement a wireless intercom system. Traditionally, wireless intercoms have been a function of broadcast television productions, and as such have used, at least in part, a spectrum that falls under FCC Code 47 CFR, Part 74 in addition to itinerant frequencies. The spectrum most commonly used falls into two areas: VHF systems from approximately 154 MHz to 216 MHz, and UHF systems from 460 MHz to 608 MHz and 614 - 806 MHz. As mentioned in an earlier section, large chunks of this spectrum have either been reallocated, or will soon be reallocated. The FCC has found that auctioning spectrum is a good way for the commission to move from an expense center to a profit center, and they are pursuing it with a passion.

Wireless intercoms, like any other wireless system, require at least one transmitter to function. Under FCC rules, all transmitters must be licensed prior to operation (there are some very low power transmitters that can operate under Part 15 and do not need to be licensed, but that doesn't apply to any modern RF intercom systems). There are different forms to obtain various types of licenses depending on what area of the spectrum your system will operate in, who will be operating the system, and what the system will be used for. The law is very clear in that no one is permitted to operate a transmitter typically used for wireless intercom systems without first obtaining an FCC license.

Wireless equipment often operates in areas of the RF spectrum that are designated for TV channels, but are unused in a given area. In all cases low power transmitters used by wireless intercoms and wireless mics must operate on a secondary, non-interfering basis. This means that wireless users must not cause harmful interference to television or other receivers, and must accept all interference sources. In keeping with this, the FCC rules state that VHF systems must not be operated within 50 miles of a television transmitter occupying a similar spectrum. The rules further state that UHF systems must not be operated within 75 miles of a television transmitter occupying a similar spectrum. See Figure 6.4, for a depiction of what a television station's assigned spectrum looks like. Refer to Table 6.1 for the standard frequency allocations of television transmitters.

Figure 6.4 NTSC channel configuration.

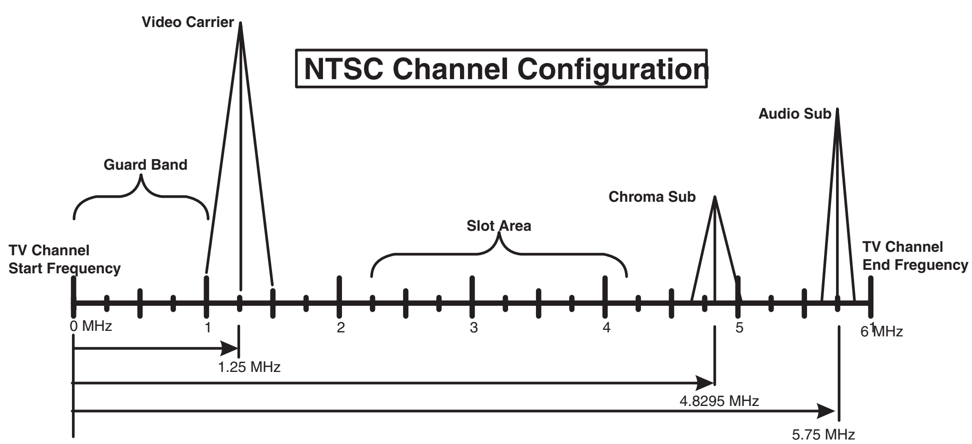


Table 6.1 Standard US television channel allocations.

Chan	Start	Video	Chroma	Audio
2	54	55.250	58.8295	59.750
3	60	61.250	64.8295	65.750
4	66	67.250	70.8295	71.750
5	72	73.250	76.8295	77.750
6	78	79.250	82.8295	83.750
7	174	175.250	178.8295	179.750
8	180	181.250	184.8295	185.750
9	186	187.250	190.8295	191.750
10	192	193.250	196.8295	197.750
11	198	199.250	202.8295	203.750
12	204	205.250	208.8295	209.750
13	210	211.250	214.8295	215.750
14	470	471.250	474.8295	475.750
15	476	477.250	480.8295	481.750
16	482	483.250	486.8295	487.750
17	488	489.250	492.8295	493.750
18	494	495.250	498.8295	499.750
19	500	501.250	504.8295	505.750
20	506	507.250	510.8295	511.750
21	512	513.250	516.8295	517.750
22	518	519.250	522.8295	523.750
23	524	525.250	528.8295	529.750
24	530	531.250	534.8295	535.750
25	536	537.250	540.8295	541.750
26	542	543.250	546.8295	547.750
27	548	549.250	552.8295	553.750
28	554	555.250	558.8295	559.750
29	560	561.250	564.8295	565.750
30	566	567.250	570.8295	571.750
31	572	573.250	576.8295	577.750
32	578	579.250	582.8295	583.750
33	584	585.250	588.8295	589.750
34	590	591.250	594.8295	595.750
35	596	597.250	600.8295	601.750
36	602	603.250	606.8295	607.750
37	608	Radio Astronomy Only		
38	614	615.250	618.8295	619.750
39	620	621.250	624.8295	625.750
40	626	627.250	630.8295	631.750
41	632	633.250	636.8295	637.750
42	638	639.250	642.8295	643.750
43	644	645.250	648.8295	649.750
44	650	651.250	654.8295	655.750
45	656	657.250	660.8295	661.750

Table 6.1 Standard US television channel allocations.

Chan	Start	Video	Chroma	Audio
46	662	663.250	666.8295	667.750
47	668	669.250	672.8295	673.750
48	674	675.250	678.8295	679.750
49	680	681.250	684.8295	685.750
50	686	687.250	690.8295	691.750
51	692	693.250	696.8295	697.750
52	698	699.250	702.8295	703.750
53	704	705.250	708.8295	709.750
54	710	711.250	714.8295	715.750
55	716	717.250	720.8295	721.750
56	722	723.250	726.8295	727.750
57	728	729.250	732.8295	733.750
58	734	735.250	738.8295	739.750
59	740	741.250	744.8295	745.750
60	746	747.250	750.8295	751.750
61	752	753.250	756.8295	757.750
62	758	759.250	762.8295	763.750
63	764	765.250	768.8295	769.750
64	770	771.250	774.8295	775.750
65	776	777.250	780.8295	781.750
66	782	783.250	786.8295	787.750
67	788	789.250	792.8295	793.750
68	794	795.250	798.8295	799.750
69	800	801.250	804.8295	805.750

Having touched briefly on the FCC rules, I must inform you the vast majority of users, not only of wireless intercoms, but of wireless mics and IFBs as well, do not obtain licenses. In fact, historically, many users of UHF wireless gear, outside of television broadcasters or people working with broadcast entities, could not even qualify to get an appropriate license. Right, wrong or indifferent, this has been the case. Telex Communications, Inc. strongly recommends that every wireless system be licensed and operated in strict accordance to FCC rules. Your local wireless dealer can help you understand the requirements and regulations that apply to you.

There has been progress though. The FCC has, as of late, worked with users, other than broadcast, to facilitate a win-win licensing scheme that may, in the future, help to ensure all systems are licensed and operated according to FCC rules. In any case, each user must in all good conscience, research his ability to license and operate wireless equipment, and govern equipment purchase and implementation accordingly.

In addition to the FCC rules, wireless users must also consider how best to avoid harmful interference to ensure uninterrupted and intelligible communications. One of the best ways to go about selecting the area of spectrum you will use is to do a frequency survey. By using a spectrum analyzer or other specialized receiver, it is possible to look at potential interference sources and avoid them. Picking an area of spectrum that is free from external interference sources will go a long way in helping you select frequencies that offer trouble free operation.

In addition to a clear spectrum, you must also consider the intermodulation affects of the specific frequencies you pick. An in depth study of this topic is beyond the scope of this book, but we will touch on the subject to give you a general overview.**Intermodulation**

(IM) or intermod as it is often called, happens when two or more frequencies mix in a non-linear device and produce a number of related different frequencies known as intermodulation products. We look at intermodulation in more detail in the next chapter, but suffice to say, choosing a manufacturer or dealer that is qualified to pick intermodulation free frequencies is a must.

Now let's look at cost. Wireless intercom systems cost substantially more at initial purchase than do hardwired communications systems. For a comparable number of users, wireless systems cost between two to ten times as much as hard wired partyline systems, depending on system type and configuration. Because of this increased cost factor, it is important to determine which members of the production team must be wireless and which can be tethered by a wire. Of course, everyone wants to be wireless and has a great reason why they need a wireless beltpack, but in the end you must make the budget meet the overall production needs.

Generally speaking, the added cost and special consideration factors that wireless communications systems have to be concerned with are far outweighed by the increased flexibility and functionality they offer. It is important to choose a wireless system with all of the facts and considerations in front of you so you can have years of trouble free operation to come.

CHAPTER 7

Design of Wireless Intercom Systems

Tom Turkington

Introduction

The design, and subsequent operation, of a wireless intercom system is, like any wireless network, highly dependent on numerous factors. Some of these factors you will have control over, but many you will not. The key to successful wireless system design, whether it be intercoms, talent audio, or roving camera, is to gather the information related to all of the variables before you get started and then match the system components and architecture to your specific requirements. There is no such thing as a one size fits all wireless system. In this chapter, we explore some Radio Frequency (RF) theory that allow you to have a better understanding of how RF works. We will also look at many of the key components of a wireless communications system and how they go together to create the desired effect.

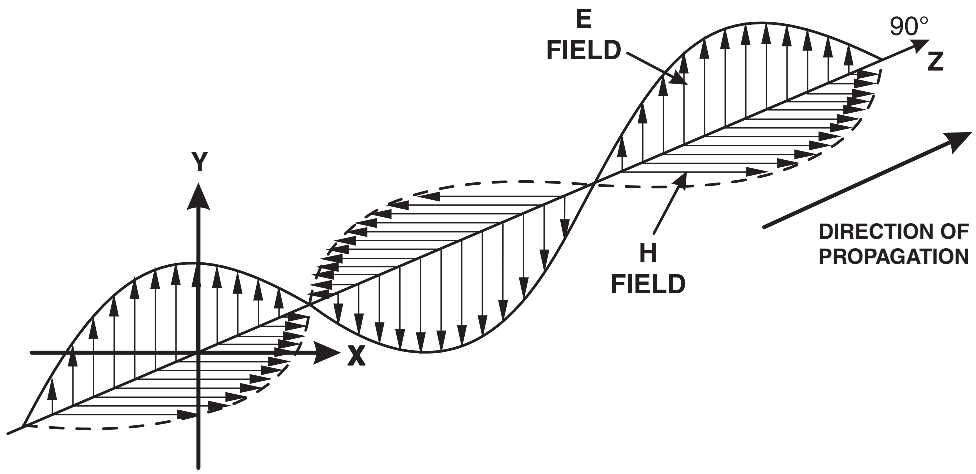
Back-to-Basics

In this section, we discuss the theory of how RF signals act and how they are affected by various conditions. There is some math discussed here, but only enough to convey the principles at hand. The idea is to give you a good working knowledge of RF principles, not make you an expert in Bessel functions. Old RF pros can probably skip this section, although a refresher of this material is almost always appropriate.

First, let’s answer the question, “What is RF?” Contrary to popular belief, the frequency of a signal does not determine whether it is an RF signal or not. The defining factor for RF signals is the medium through which they propagate. All energy that travels in waves propagates through some medium which allows the wave to move from one location to another. In the case of sound, the medium is typically air or water or some other physical mass. RF signals on the other hand, regardless of frequency, always propagate or move through the electromagnetic spectrum. Where as sound needs some physical mass to move, RF signals do not. The electromagnetic spectrum exists everywhere (as far as we know), and enables RF signals to move through the vast vacuum of space where sound waves could never go.

A brief look at the properties of the electromagnetic spectrum can tell us a lot about the RF signals that move through it. As you can see, the name electro-magnetic is really a combination of two words, electron (or electronic) and magnet (or magnetic). The reason for this is that waves that propagate in the electromagnetic spectrum have two separate and distinct components, an electrical and a magnetic. As you can see in Figure 7.1, these two components exist at right angles to each other, as well as, to the direction of propagation. The electrical component, or field as it is called, is represented by the letter E and the magnetic field by the letter H. (No, I don't know why they use H, but they do!)

Figure 7.1 The E an H fields exist in two separate planes, 90° apart from each other.

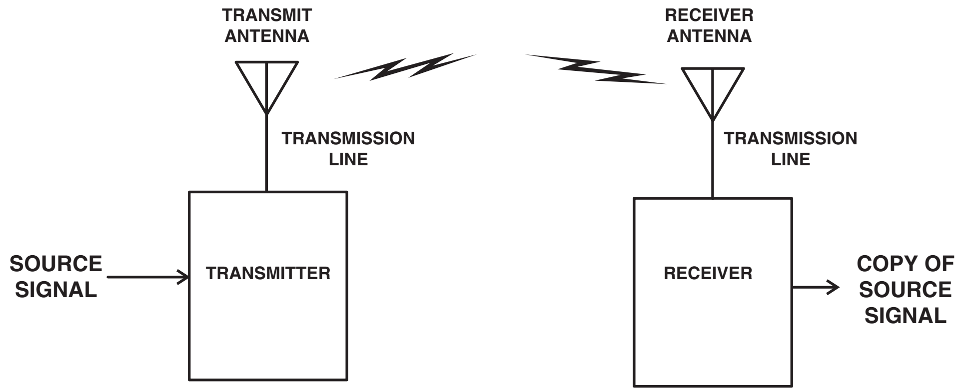


RF signals at different frequencies have different propagation characteristics and are affected by external forces in different ways. The reason for this is the ratio of the magnitudes of the electrical and magnetic components of an RF wave vary dramatically as frequency changes. Generally speaking, the magnetic component of an RF wave is much greater than the electrical component at very low frequencies. As the frequency increases, the electrical component increases and the magnetic component decreases, until, at very high frequencies, the electrical component is much greater than the magnetic.

This is not just "gee whiz" information. The different makeup of RF waves at different frequencies is what allows us to use the signals for different and sometimes unusual applications. For instance, at super low frequencies, such as 5 Hertz, where the magnetic component is extremely dominant, the US Navy has been able to propagate RF signals through the Earth's core to communicate with submarines on the other side of the world. Try that at 13 GHz! In a more pertinent example, at much higher frequencies the highly reflective nature of the mostly electrical component wave can cause self-interference, known as multipath. Multipath can cause an RF signal to be unusable at a very short distance from the transmitter if not properly handled. We will discuss multipath in more detail later in this chapter.

Now that we know what RF is, we can discuss what it does, how we can use it and how it is affected by outside forces. In its most basic form, an RF system puts information on an RF signal, sends it to a remote location and retrieves the information in exactly the same form as it originally existed. Let's take a look at this most basic system and define some terms so we can talk about this process more easily. Refer to Figure 7.2.

Figure 7.2 An example of wireless transmission and reception.

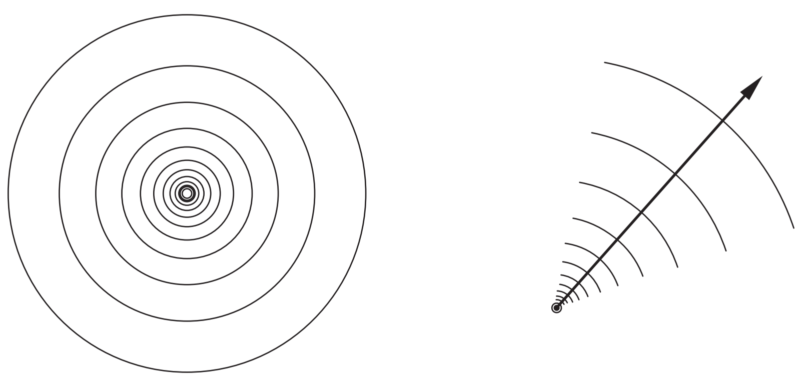


In Figure 7.2, the transmitter is a device that has an input for information, audio, data, or some other form of intelligence called a source signal, that needs to get from here to there. The transmitter then takes that information and puts it onto an RF signal. The RF signal is called a **carrier** because it, in effect, carries the source signal as it propagates. The process of actually putting the source signal onto the carrier is called modulating the carrier, which normally is referred to simply as **modulation**. The carrier which has had the source signal applied is then broadcast into the air (actually the electromagnetic spectrum) via an antenna. The antenna is a transducer that allows the carrier to be efficiently broadcast or received.

Once the signal is broadcast into the air, it propagates out away from the transmit antenna and eventually reaches the receive antenna. The area between the transmit antenna and the receive antenna is called the **propagation path**, or just path. At the receive antenna, the signal, which is now much weaker, is collected and enters into the receiver. The receiver's job is to find the one unique carrier from the transmitter and strip off the source signal so it exactly matches the original information. This process is called **demodulation**.

Now, let's look at the RF wave as it moves along the propagation path. We know that RF propagates or moves from one point to another, and that propagation can be affected by the frequency of the wave. Now we'll find out how RF waves normally act in typical environments. You can think of an RF signal that radiates out into open space from a specific point, such as a transmit antenna, like the waves generated by throwing a pebble into a pond as shown in Figure 7.3. The energy carried by the wave moves away from the original point in all directions equally and each vector that can be drawn from the center point represents RF energy traveling away from the point of origin in a straight line as shown in Figure 7.3.

Figure 7.3 An example of electromagnetic waves being radiated.



In addition, the RF wave continually gets weaker as it moves away from the transmit antenna. The rate at which the wave becomes weaker can be calculated via the inverse square law $1/D^2$ where D = distance traveled by the wave. This is a very important concept because it shows why a wave that travels twice as far as another wave of equal magnitude is not half as strong. Take the following example:

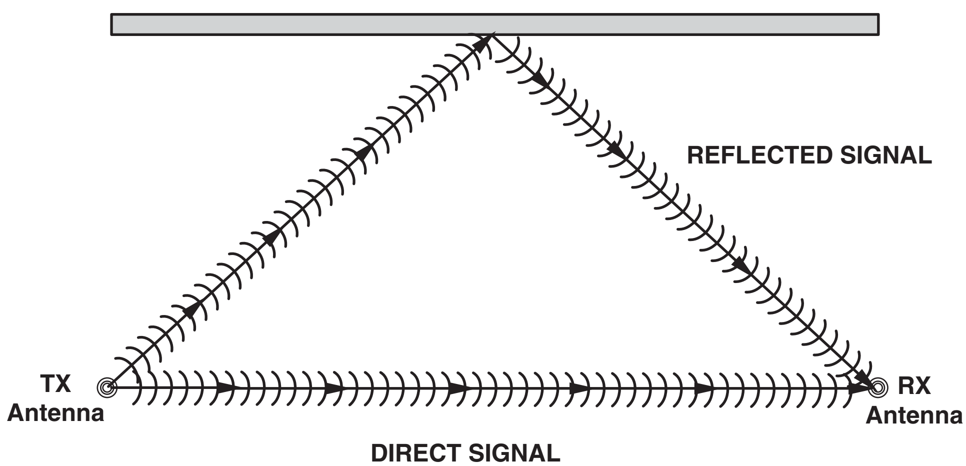
> Two transmitters TXA and TXB both emit signals that are exactly the same at 1 Watt of power. The signal from TXA travels 10 units. Power at that point can be calculated by $1/10^2$ x 1W or 0.01 x 1W. That means there is 0.01W of the TXA signal left after it has traveled 10 units. Now let's say that the TXB signal travels twice the distance of TXA or 20 units. Power at that point can be calculated by $1/20^2$ x 1W or 0.0025 x 1W. That means there is 0.0025W of the TXB signal left after it has traveled 20 units. As you can see, the signal that traveled twice as far was not ½ the power, but ¼ the power of the first signal.

Because of the inverse square law, the effective radiated power (ERP) of a given transmitter must increase by a factor of four times to achieve twice the operating range. This information is important in determining the necessary power for a wireless system for a given range. It is always important to use the minimum power necessary to accomplish the task at hand so that excess power does not affect other systems and cause undue harm.

The theoretical range of an RF system is important to know, but it is the functional range you must be more concerned with. The functional range of a system takes into account a certain cushion factor called **fade margin** that will ensure the signal coming from the transmitter to the receiver will not only be detectable, but will also be usable. This is less of a concern in communications systems as you can tolerate less fade margin than in an on-air wireless microphone system, because a small momentary dropout will not critically affect communications as it would program audio.

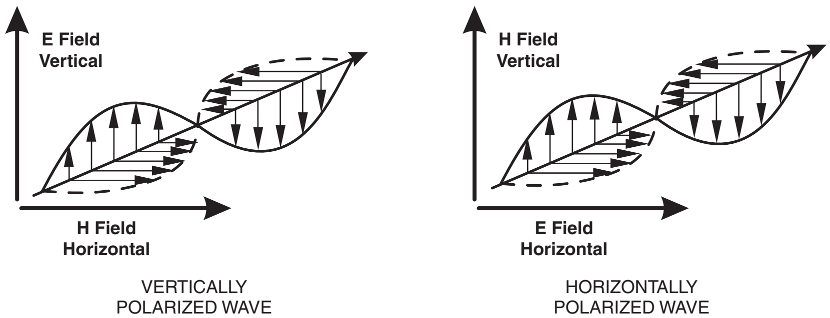
RF waves travel away from the source in a straight line until that path is interrupted or disturbed by some outside influence. Figure 7.4 shows an RF wave being reflected and thus changing the path of some of the RF energy. If you remember earlier when we mentioned multipath, this reflected energy is the cause of that phenomenon. Before we can discuss multipath in more detail, though, we must learn about another aspect of the RF wave and how it changes when it comes in contact with a reflective surface.

Figure 7.4 An example of reflected RF waves.



Polarization is the term that describes the orientation of an RF wave. Remember back to when we discussed the two components that make up an RF wave, the electrical and the magnetic. We said that the E field was the electrical component and that the H field was the magnetic component. The polarization of an RF signal is determined by the orientation of the E field. If the E field is perpendicular to the plane of the Earth, the wave is said to be vertically polarized. If the E field is parallel to the plane of the Earth, the wave is said to be horizontally polarized. See Figure 7.5.

Figure 7.5 The orientation of the radiator (antenna) determines the polarization, and therefore, the orientation of the E and H fields.



Transmit and receive antennas of the same system must be oriented in the same direction (plane) to have a proper transfer of the carrier. In theory, if a transmit antenna is oriented vertically, thus producing a vertically polarized carrier, and the corresponding receive antenna is oriented horizontally, the receive antenna will not be able to see the vertically polarized wave at all. In practice, there will always be some polarization shift in the path and the receiver will see a very small signal if it is close enough to the transmitter. To avoid this problem, antennas in a given RF system should always have similar orientation.

There are other forms of polarization, such as circular polarization, which can be used to help counteract the effect of multipath, but for now we will use horizontal and vertical polarization for our discussion. It is important to note here the difference between polarization and phase, as the two terms are often confused. Phase refers to the relationship of the sinusoidal energy of two or more waves, not to the orientation of the electrical component. See Figure 7.6. Two identical waves that are in phase, and are combined, add to make a larger wave. Two identical waves that are out of phase by exactly 180°, and are combined, cancel each other out. See Figure 7.7. Waves that are not exactly identical in either frequency, amplitude, or phase will have a composite sum that may increase the overall amplitude at some points, and either reduce or eliminate the overall amplitude at others. See Figure 7.8. It is critical to have a good understanding of these two principles as we start to discuss multipath.

Figure 7.6 Waves that are in phase combine to form a larger wave.

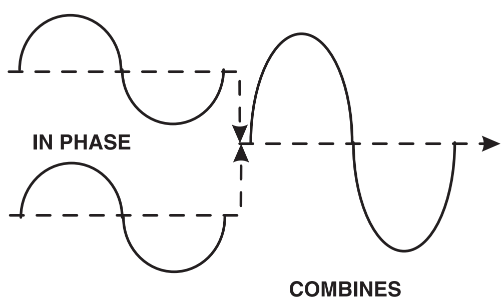


Figure 7.7 Waves that are out of phase cancel each other.

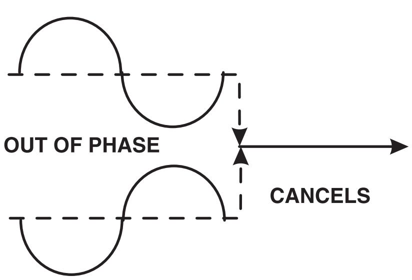
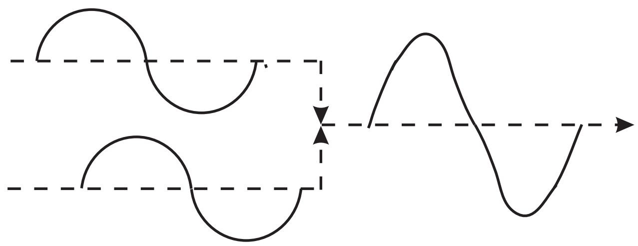


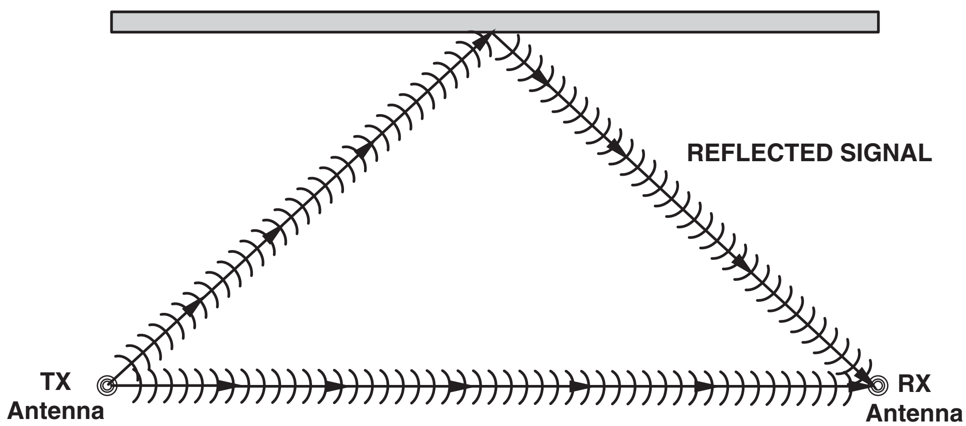
Figure 7.8 An example of combining waves that are not 180° out of phase.



Interference

As mentioned earlier, multipath can be described as a form of self interference caused when a reflected RF carrier arrives at the receive antenna along with an RF carrier that has taken a direct path. See Figure 7.9. The reason multipath is so detrimental to the successful operation of an RF system has to do with the nature of the relationship of the reflected signal to the direct path signal.

Figure 7.9 An example of multipath in its most basic form.



The direct path carrier takes the most direct, and consequently, the shortest path from transmitter to receiver. The reflected carrier, on the other hand, takes a longer path, from the transmitter to the reflective surface, and from the reflective surface to the receiver. The waves leaving the transmitter antenna are all in phase, but because the direct carrier and the reflected carrier travel different distances, thus taking slightly different lengths of time, the two carriers are out of phase, and of different amplitudes (remember the inverse square law), when they reach the receive antenna. The two carriers are combined at the receive antenna and, being out of phase, they cancel each other out so that little or nothing can be detected by the receiver. This causes a momentary interruption in the RF wave, which is called a **dropout**. Dropouts are manifested in audio RF systems by a loud click or pop surrounded by noise. Proper system design and careful antenna placement can go a long way to reducing the effects of multipath on a wireless communications system. We discuss how to avoid multipath later in this chapter.

The next concept that you must be familiar with to move forward in the design of your wireless intercom system is **receiver desensitization** or desensing. As mentioned earlier, desensing happens when a transmitter is in close proximity to a receiver, even if that transmitter is not on or near the receiver's operating frequency. Receiver desensitization happens because receivers must maintain critical voltage and current levels throughout the front end stages, and a strong (i.e. close by) transmitter can cause these levels to vary greatly. As these levels are changed over a wide range, the receiver performance will be greatly degraded. The greater the physical distance between transmitter and receiver, the less the receiver will be affected. Likewise, the greater the frequency separation between the two, the less the receiver performance will be affected.

Selecting frequencies that are "clean," or free from the effects of intermodulation products, is essential to good wireless communications. Intermodulation is often one of the prevalent sources of system interference. We touch on just the basics of intermod here so you can get a sense of what it is and how it works. As stated in an earlier chapter, **intermodulation**, or IM as it is often called, happens when two or more frequencies mix in a non-linear device and produce a number of related different frequencies known as **intermodulation products**. These IM products can cause severe, harmful interference to a wireless intercom system if they fall on or near any of the operating frequencies of that system.

For intermodulation interference to take place, at least two transmitters must be broadcasting at the same time on frequencies that have a definite, calculable relationship with the affected receiver. In many cases of IM interference the receiver can detect and demodulate the IM product almost as cleanly as if one of the interfering transmitters was on the operating frequency of the receiver. Turning off either one of the two (or more) transmitters will cause the IM interference to cease.

Because there is a fixed and calculable relationship between frequencies, intermodulation products can be calculated and avoided. Here is an example of some of the more common IM products that can be calculated:

2A – B = C

2(651.500 MHz) – 650.000 MHz = 653.000 MHz

A – B + C = D

184.000 MHz – 190.600 MHz + 188.200 MHz = 186.400 MHz

3A – 2B = C

3(518.200 MHz) – 2(520.500 MHz) = 513.600 MHz

There are, of course, many other combinations that can cause harmful interference. These examples give you a good idea of how the calculations work, but for comprehensive frequency selection, an advanced computer program must be used.

It is important to note that intermod products are not created in the air, they are the result of the mixing of signals in non-linear devices such as amplifiers or other usually active elements. The most common place for this mixing to take place is in the active receiver RF circuitry. Once RF signals get past the receiver front end and get to the first RF amplifier and beyond, mixing of those signals can and will take place. If the intermod products that are generated fall on or near the operating frequency of the receiver, harmful interference will be heard.

Good quality receivers have front ends that are passive, linear devices that limit the range of frequencies that will enter the rest of the receiver circuitry. Making sure you pick wireless intercoms with well designed front ends, is critical to proper operation in hostile RF environments.

The next most common place for IM products to be generated is in the final amplifier of a transmitter. Because the transmit antenna can and does also act as a receive antenna, strong RF signals from nearby transmitters can make their way into the non-linear, active, final amplifier and produce intermod products. These products can then be broadcast out with the intended signal and cause harmful interference. It is important to note that IM products do not have to end up exactly at a receive frequency. Sometimes, they can be of sufficient power at relatively close frequencies to create a desensing situation.

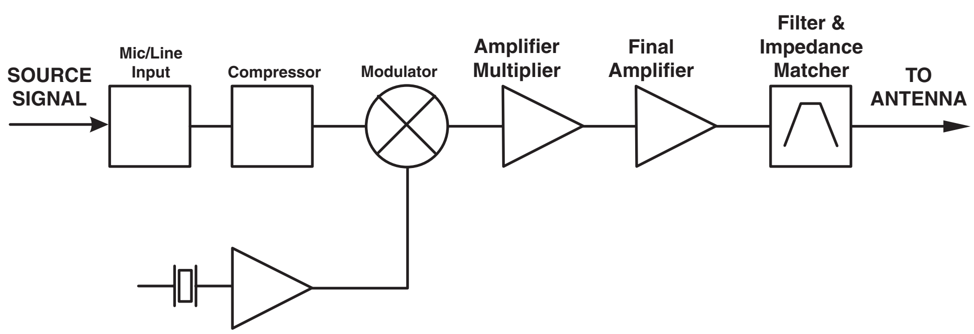
Reducing the effect that intermodulation can have on your wireless intercom system comes down to a few important principles. First, and foremost, you must pick frequencies that are intermod free with each other and with surrounding transmitters. Second, you should pick wireless intercom systems that have well designed receivers and transmitters with appropriate passive filtering. Third, you must manage the positioning of antennas and beltpacks within the system to optimize operational potential.

Transmitters and Receivers

To be able to select the appropriate wireless communications equipment you need to understand the basic operations of transmitters and receivers, and which aspects are important to proper operation. In this section, we cover generic functional block diagrams of transmitters and receivers, and point out the most critical aspects of each. While design variations are great between manufacturers, the block diagrams that follow represent the most basic designs.

Let's start with the transmitter (see Figure 7.10). The primary job of the transmitter is to take in a source signal, modulate it onto an RF carrier, and then deliver it to the transmit antenna for broadcast into the electromagnetic spectrum.

Figure 7.10 Transmitter block diagram.



First, an audio signal is brought in and any necessary audio amplification is done via the Mic/Line Input section. Next, the signal is sent through a Compressor circuit to ensure the levels of the input signal are held within acceptable limits. The signal is then mixed with a reference frequency in the Modulator. This reference frequency can be the main carrier frequency, or (as in most cases) it is a base frequency that results in a composite signal.

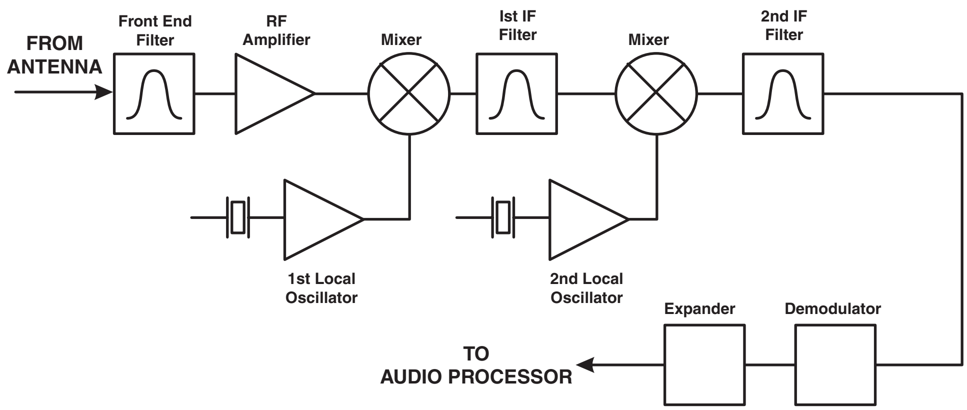
Note There are many different types of Modulators, as well as, many different types of modulation. A detailed discussion of their detailed workings is beyond the scope of this book.

The signal is then sent to the Amplifier/Multiplier. If the signal is already on the desired transmit frequency, it is only further amplified. If, however, the signal is only a composite signal, then it is frequency multiplied to reach the desired operating frequency. The signal is then sent to a Final Amplifier where it reaches its maximum power level. Usually this is slightly more than the actual output power as measured at the output connector. The reason for this is to make up for the losses induced by the Output Filter and Impedance Matching circuit(s).

The Output Filter and Impedance Matching circuits are generally passive and therefore, do not provide any means of amplification. As such, they can only reduce the output signal levels. The Output Filter is a very narrow bandpass filter that removes any unwanted harmonics from the signal. The Impedance Matcher provides the necessary interface between the transmitter and the Antenna/Transmission Line to ensure maximum power transfer. If the Antenna/Transmission Line are not properly matched, significant loss can occur. In some situations, it is possible for this to cause damage to either the transmitter, transmission line, and/or antenna.

Now let's look at the receiver and it's primary functional aspects (see Figure 7.11). The receiver in a wireless system is the exact compliment of the transmitter, but is usually much more sophisticated and complex in design. Its job is to receive the signal from the receive antenna and extract the source signal so that it matches the original exactly. In practice, there will always be some modification or distortion of the source signal in the course of transmission, but good quality wireless systems minimize this to a level that is indistinguishable.

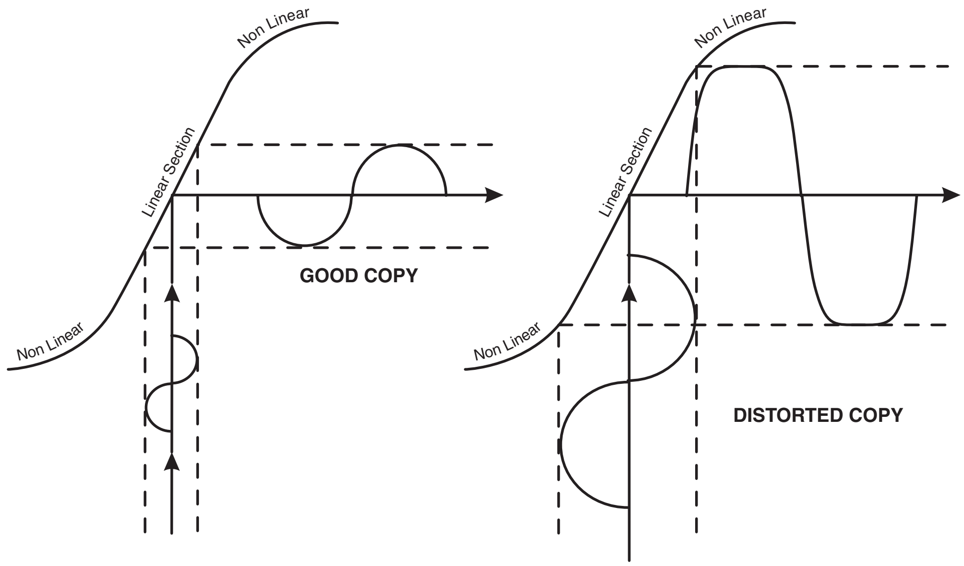
Figure 7.11 Receiver block diagram.



As in the transmitter, the antenna will be covered in the next section. The receiver starts with the front-end filter. The front-end filter is extremely important to successful operation in high RF level environments. The front-end filter is the first line of defense. Its job is to limit the number of potential interfering frequencies that could affect the receiver. It is usually a passive, linear section and it must be impedance matched to the antenna for proper signal transfer. Linearity is the most important factor in a front end, even more so than how tight or narrow the section is. A high degree of linearity will ensure that no intermodulation products are generated in the front end before extraneous RF signals are filtered out. Having a front-end that is relatively tight and that is extremely linear is critical if the system is to work properly under worst-case RF scenarios.

The next section of the receiver is the first RF amplifier. The first RF amp's job is to take the extremely low level RF signal coming through from the front end and bring it up to a usable level. The incoming RF signal at the first RF amp can vary dramatically from less than 0.5 µV to almost the value of the transmitter output. The key for the first RF amp is that it should be able to handle very small, as well as, relatively large incoming signals within it's linear region of operation. See Figure 7.12. To maintain a good linear region, RF amps normally require a high current drain which can negatively impact battery life. A compromise between linearity and effective battery life must be managed carefully.

Figure 7.12 Good linearity is a must for faithful signal reproduction.



The next receiver section we look at is the first local oscillator (LO). The job of the first LO is to provide a reference signal that is a fixed distance from the operating frequency of the system. It is very important the first LO be stable over a wide range of temperatures. In fixed crystal systems, one or more crystals cut to a specific relationship of the operating frequency are used to generate this highly accurate reference signal. A different crystal is necessary for each operating frequency. In synthesized units, a single reference crystal is used in a phase-lock-loop to provide the signal for any operating frequency needed by the receiver.

The First LO feeds the reference signal to the Mixer where the incoming RF carrier is mixed, or beat with the reference signal, to produce the First Intermediate Frequency (IF). The frequency of the First IF is the difference in frequency between the incoming RF carrier and the First LO reference signal. Unfortunately, what comes out of the Mixer is not a just the First IF, it is the algebraic sum and difference of the two signals being mixed plus numerous other harmonic junk. To get to the point where you have a clean First IF consisting of just the desired frequency, the signal is passed through to the First IF Filter. The First IF Filter is extremely important to proper receiver operation. It is a passive, very narrow (often 50 to 250 KHz), and precise filter that eliminates the vast majority of unwanted signals so the true First IF can be processed correctly. It is very important that the First IF Filter be sharp, as well as, very linear. Any non-linearity in the filter will cause unwanted distortion of the demodulated source signal.

Next, the signal is sent to the second Mixer where a second IF frequency is produced in the same way the First IF was obtained. The Second LO is the same frequency for any RF carrier frequency the receiver is capable of because the first LO takes care of the frequency differences and produces an always-constant First IF frequency for the Second IF to handle. Again, the Second IF signal as it leaves the Second Mixer is full of harmonic junk and needs to be filtered by the Second IF Filter. The Second IF filter eliminates unwanted harmonic energy and prepares the signal to be demodulated.

The next phase of the receiver is the Demodulator. There are several types of demodulators used by wireless manufacturers today and it would be beyond the scope of this book to discuss them all in detail. Suffice to say, through a type specific process the Demodulator extracts the source signal from the Second IF carrier. The quality of the Demodulator circuit is critical to good audio quality. Any type of signal distortion or modification that takes place in the demodulation process will cause the final signal to be a less than perfect reproduction of the original source signal.

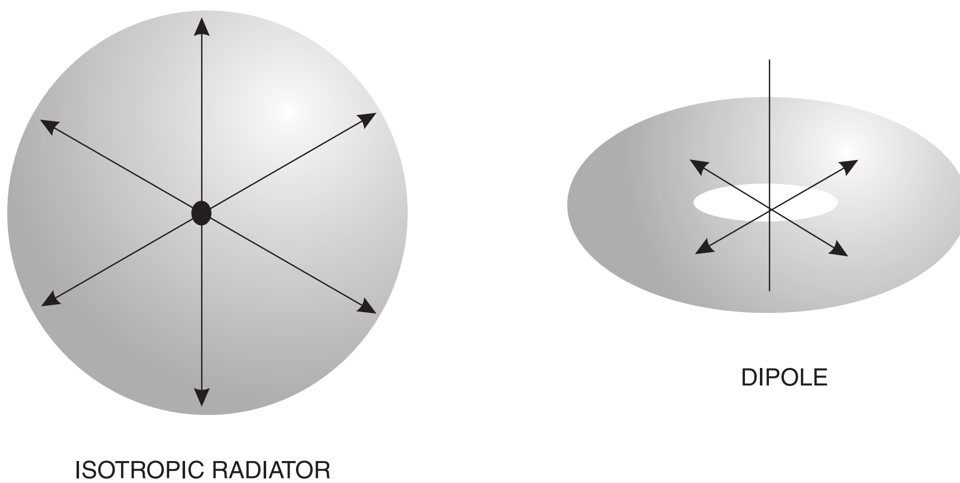
Antenna & Cable Considerations

Antennas and cables (transmission lines) are one of the least thought about aspects of a wireless system among RF novices. Good quality antennas and cables, however, are some of the most important aspects to establishing and maintaining a quality RF link. In addition, because antennas and cables are more easily changed and in general are less expensive than other system components, they can be a “quick fix” for many RF problems found in common wireless communications systems. In this section, we cover some of the more common types of antennas and the operating characteristics of each. In addition, we take a look at coaxial cable and what is required when selecting cable for your system.

To adequately look at and evaluate the strengths and weaknesses of some common antenna types, we first must have a very general knowledge of antenna theory. Antenna theory is a course of study unto itself and we will not even scratch the surface in this brief section, but it should be enough to understand some basic principles. To start, let’s ask the question, “What is an antenna?” To answer that question we must look at what an antenna does. In a transmitter, the antenna takes electrical energy and allows it to be propagated out into the electromagnetic spectrum. In a receiver the antenna “gathers” the RF signal and converts it back into electrical voltages and currents. In either case, the antenna acts as a transducer to change the form of the RF energy.

All real world antennas have a pattern or specific shape with which the RF energy is released or captured. There is no such thing as an antenna that sends energy out equally in all directions. The primary reason for this is that you have to get the signal to the antenna via a transmission line and that line must be connected to the antenna some how. That connection will always cause a disruption or altering of RF propagation in some direction. In theory though, it is nice to talk about a perfect antenna. This perfect antenna radiates equally in all directions and is called an isotropic radiator. See Figure 7.13.

Figure 7.13 A comparison of the radiation patterns for an Isotropic Radiator (theoretical) vs. a Dipole (practical).



We can look at how all other antennas emit RF energy as a comparison to our perfect antenna. The isotropic radiator is said to have zero antenna gain. Antenna gain is an often misunderstood term, so we will cover it here. Let’s start by saying that a passive antenna is not an amplifier and cannot increase to total RF energy being emitted or received. Having said that, an antenna can and does focus the RF energy in a specific direction or directions. This focusing of energy causes greater RF energy levels in those directions and weaker energy levels in the remaining areas as shown in Figure 7.13.

We can think about this by looking at a water balloon. If we had a water balloon that was a perfect sphere, it would accurately represent the pattern of an isotropic radiator located in the balloon’s center. All of the RF energy is equally dispersed in all directions. If you squeezed the balloon's center with your hands, a corresponding bulge would appear on either end. The balloon is not any larger or smaller than it was, it has only changed shape. This is how a real world antenna works. When energy is focused in one direction, it must always be at the expense of energy going in another direction.

The most basic form of real world antenna is the dipole. The dipole has 2.15 dBi of antenna gain over an isotropic radiator. That means there is 2.15 dB more signal in the direction that the energy is focused than there would be if the antenna were an isotropic radiator. Antenna gain is specified in one of two ways: dBi or dBd. It is very important to know which specification is being used when comparing antennas. dBi, as stated above, is referenced to the uniform radiation of an isotropic radiator. dBd, on the other hand, is referenced to a dipole. Most antenna manufacturers like the dBi spec because the number is bigger, but since there is no real world antenna that represents the 0 dB mark, many engineers prefer dBd. In reality, either specification is fine as long as you are comparing apples to apples. In the remainder of this book all antenna gain references will be in dBd, referenced to a dipole, unless otherwise specified.

Important: If you need to convert from dBi to dBd, simply subtract 2.15 dB from the dBi number. If you need to convert from dBd to dBi, simply add 2.15 dB to the dBd number.

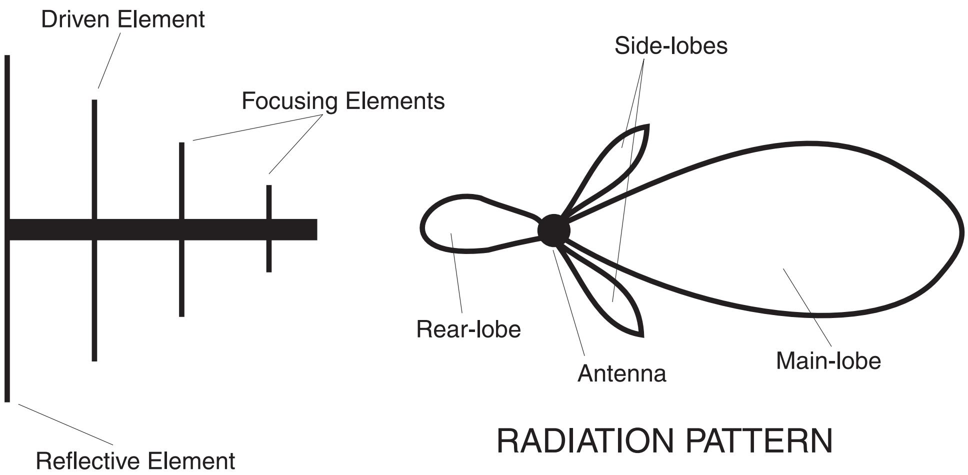
There are two basic groups of antennas, omni directional and directional antennas. **Omni directional** antennas are generally low gain antennas used in the center of operational areas. Because the RF energy in omni directional antennas is in 360° and not in one specific direction, the antenna gain must always be low. The isotropic radiator and dipole antennas are both examples of omni directional antennas. Normally, omni directional antennas will be found with antenna gain less that 5 dBd. Gain in omni directional antennas is achieved by flattening the vertical angle of the pattern as shown in the dipole example in Figure 7.13.

For proper propagation to take place, the length of an omni directional in critical. The theoretical minimum length for an omni directional antenna is ½ the wavelength of the RF carrier to be served. In many cases this ½ wave length is too long to be practical so a ¼ wave antenna is used instead. It is extremely important to note that for a ¼ wave antenna to work properly, it must have a corresponding ground plane that is equal to or greater than the length of the antenna itself. It is for this reason that a ¼ wave antenna that works just fine when it is attached directly to the back of a wireless receiver has very poor coverage when operated at the end of a length of coaxial cable. The cable does not provide the necessary ground plane for proper ¼ operation as the receiver does. This is a very common mistake made by RF novices who are trying to improve RF performance and end up killing it instead!

Directional antennas, on the other hand, seek to focus the area of coverage to something less than 360° to form a flashlight like coverage pattern. Directional antennas are normally used on the edge of a coverage area. They can have very high antenna gain factors in excess of 20 dBd. Normally though, in conventional wireless communications systems, size and cost limit directional antenna gain to less than 12 dBd.

Directional antennas have the advantage of not only focusing the RF energy in a given direction, but also attenuating energy from undesired areas. This is very important for receive antennas in areas with high levels of RF. If positioned properly, a directional receive antenna can increase the desired RF energy while attenuating unwanted, potentially interfering RF energy from other areas. See Figure 7.14.

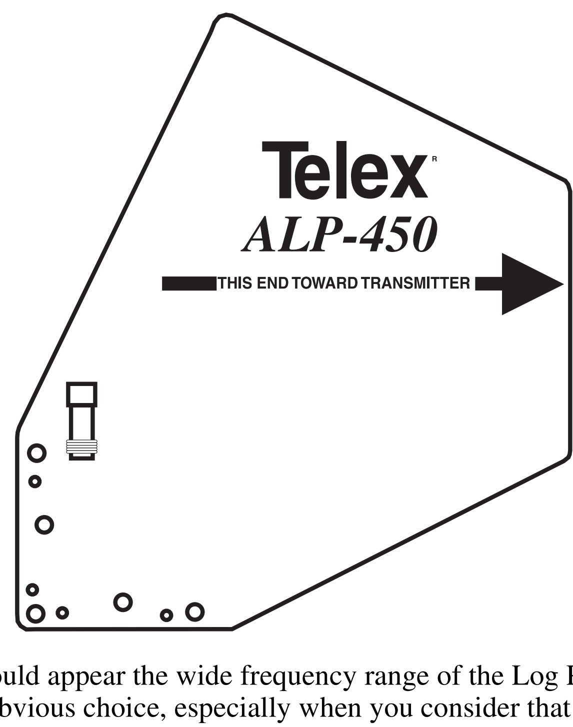
Figure 7.14 An example of a Yagi antenna.



ANTENNA CONSTRUCTION

There are two very commonly used directional antennas in wireless communications systems today, Yagi and Log Periodic antennas. We will not cover the technical differences of these antennas here, but we will discuss the functional differences. Just as in omni directional antennas, directional antennas must be tuned or “cut” to a specific frequency range. This is all well and good when there is only one RF frequency, but if you are using a range of frequencies through a single antenna, it is important to ensure that all of the RF signals will be in the effective range of that antenna. The primary difference of Yagi and Log periodic antennas is the range of frequencies they can handle. Yagi antennas normally handle a relatively narrow range of RF frequencies, while Log Periodic antennas can achieve much larger effective frequency ranges.

Figure 7.15 **Telex®**’s ALP-450 is an example of a Log Periodic antenna.



On the surface it would appear the wide frequency range of the Log Periodic antenna would make it the obvious choice, especially when you consider that Log Periodics are generally also much smaller than Yagis. This however, is not always the case. Consider the application where there are strong off frequency interference sources (virtually all high RF level applications!). In these situations, the off frequency rejection of a Yagi antenna can greatly improve system performance and decrease harmful interference. In general, it is a good idea to choose an antenna that is just wide enough to handle the desired operating frequencies.

One more note on directional antennas. Because FCC rules concerning transmit power (Effective Radiated Power or ERP) take into account the antenna gain of the transmit antenna, high gain transmit antennas may not be used on transmitters in most wireless communications applications. The good news is that high gain antennas on the receive side of an RF system are also very effective for increasing system range and are commonly used.

We reiterate one more important antenna concept. As stated in an earlier section, antenna polarization is critical to proper system operation. Transmit and receive antennas of the same system must be oriented in the same direction to have a proper transfer of the carrier. In theory, if a transmit antenna is oriented vertically, thus producing a vertically polarized carrier, and the corresponding receive antenna is oriented horizontally, the receive antenna will not be able to see the vertically polarized wave at all. In practice, there will always be some polarization shift in the path and the receiver will see a very small signal if it is close enough to the transmitter, but system range will be greatly reduced. To avoid this problem, antennas in a given RF system should always have similar orientation.

Now, lets take a brief look at the role coaxial cable (transmission line, feedline) plays in the big picture. See Figure 7.16. Unless an antenna is attached directly to the receiver or transmitter in an RF system, coaxial cable is the usual means used to span the gap. The importance of choosing the right coaxial cable cannot be over-stressed. When choosing cable to use in your RF system three main factors must be considered:

1 The cable must be properly impedance matched (correct characteristic impedance). Most wireless systems today are 50 ohm impedance systems. That means the final amplifier and filters in the transmitter, the front end of the receiver and both transmit and receive antennas, are designed to work using 50 ohms as the nominal impedance. It is extremely important to choose coaxial cable that is also 50 ohms. Coaxial cable that is used in video applications is normally 75 ohms, not 50 ohms. Don’t ever use video cable in RF transmit applications. An explanation of why this is bad is beyond the scope of this book, but trust me on this one, it is a bad thing, don’t do it.

2 Consider the loss per foot of coaxial cable at your system’s operating frequency. In VHF systems it is usually easy to select cable with acceptable loss for runs of 100 feet or more. In UHF applications however, it gets a little tougher. See the coax loss chart below in Table 7.1. In general, it is a good idea to never have more loss in the transmission cable than you have antenna gain in the system. This is a good rule of thumb that will keep you out of trouble most of the time.

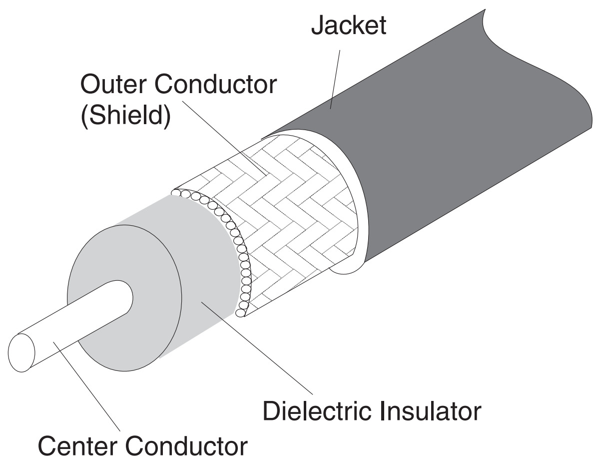
3 Consider how the system is used. Is this a fixed installation, or a mobile one. If the system is being moved frequently you want to use coaxial cable that has a stranded center conductor. Just like other types of wire, coaxial cable with a stranded center conductor will tolerate being flexed repeatedly without a degradation in performance. However, this doesn’t mean you can tie a knot in the cable, or crimp it in a door and expect it to work perfectly

Table 7.1 Coaxial Cable Loss Chart

Attenuation (dB per 100 feet) at the frequency given				
	220 MHz	450MHz	700MHz	900MHz
Times LMR-400	1.8	2.7	3.4	3.9
RG-8/U	2.9	4.5	5.8	6.7
RG-213/U	3.5	5.2	6.7	8.0
Times LMR-240	3.7	5.3	6.6	7.6
RG-8/X	6.0	8.6	10.7	12.8

Results are calculated and can vary.

Figure 7.16 The typical parts of coaxial cable.



Installation

Having all the right gear and all the proper frequencies selected is a good first step to having a top notch, highly effective, wireless communications system. However, having the right stuff is not enough, it has to be installed properly or it is all for not. In this section we take the time to cover the most common do’s and don’ts of installing a wireless system that actually works!

We’ll start with the general conceptual strategy for selecting a location for the RF equipment to live. Unlike hardwired communications systems, that can be tucked away almost anywhere, wireless systems must have prime real estate locations due to the extremely limited length of the coaxial cables that connect the transmitter and receiver to their respective antennas. As discussed in the previous section, the length of the antenna cables in a wireless system should rarely exceed 100 feet, and in some cases they should be kept much shorter due to frequency and cable loss. Because of this, selecting the location of transmitter and receiver equipment is absolutely critical to system performance.

First of all, it is necessary to determine all of areas where coverage is absolutely necessary. These are the ‘no compromise’ areas, and your system must be designed and installed to consistently meet or exceed these minimum operational requirements. Anything you can get after these areas is gravy. Select a location for the wireless base station that is centrally located in the “must work” area whenever possible. Obstacles like buildings, cars, trees outside, walls, cameras, lighting, and equipment racks inside all act as factors to limit range. If they are in the direct line of site between the base station antennas and the wireless beltpacks, it is important to locate the base antennas as high as possible. Getting a few extra feet up will often make a large difference in overall system performance.

When installing the wireless base station it is important to avoid locating it near computer or other microprocessor controlled equipment. All computer type equipment radiates RF energy that can cause harmful interference in even the best wireless equipment. Likewise, your RF equipment may interfere with the operation of the computer equipment. Try not to have your wireless base station in the same rack as lighting controllers, audio processors or other highly RF radiant electronic equipment. Whenever possible, locate the wireless base station in its own enclosure to avoid harmful interference to or from other gear.

The specific location of the antennas is also extremely important. Antennas should never be placed in close proximity to large metal surfaces that are parallel to the active element of the antenna. A large metal surface can cause numerous problems in most wireless systems. Try to locate antennas in the middle of a room (when inside) as far away from reflective surfaces as possible. When using omni directional antennas inside, try hanging them upside down with the ground plane on top, near the ceiling. This will provide the most effective radiation pattern. It is also a very good idea to get as much space between the transmit antenna and the receive antenna as is practically possible. As was covered in an earlier section, this will avoid the phenomenon known as desensing and increase system range. It is important to note that having the receive and/or transmit antennas as high up as practical is often more beneficial than increasing the output power of the transmitter(s).

As covered in the previous section, it is critical to make sure that antennas are polarized the same on both the transmit and receive end of the wireless system. In wireless communications systems, the deciding factor lies with the beltpacks. Since the antennas on the beltpack will normally be vertically polarized it is important to ensure that the base station antennas are also vertically placed. Never mount an antenna using part of the working or active elements. Most antennas that are designed to be mounted come with specific mounting hardware. Trying to rig some other “innovative” mounting solution will almost always result in reduced system range and performance. Also, don’t paint or otherwise cover you antennas. Some paint has a metallic component and will greatly impact system performance in a negative way.

In some extreme applications, it may be desirable to have multiple antenna locations for an individual wireless base station. This technique can greatly increase the effective range of the system if the everything is done just right. If not, it will most assuredly degrade overall system performance. The first thing to remember is that splitting the transmit antenna is never permissible! This is tantamount to setting up a frequency interference source right next to your operational area of coverage.

Splitting receive antennas can sometimes be a good idea. The key thing to remember here is the line impedance must be properly maintained. This means you should not use a standard “T” connector to perform the split. The most common device used for this function is a Mini Circuits splitter. This device maintains the 50 ohm impedance on all legs. It is important to remember there is no such thing as a free lunch. Splitting antennas comes at a price. When you add a second antenna the signal from each antenna is reduced by at least 3dB. This loss then needs to be figured into the total loss/gain calculation for proper system performance. Multiple antenna configurations can be very challenging. When faced with the need to do so, it may be time to consult with an RF professional for help.

As covered in the previous section, make sure you have selected a coaxial cable that is of the correct impedance and has a loss per foot low enough to support the length necessary without having more loss than you have antenna gain. Be sure to note with omni directional antennas, it is sometimes acceptable to have a dB or two more cable loss than you have antenna gain. When running RF coaxial cable, make sure the cable is not bent sharply as to crimp the cable. The magic of coaxial transmission lines is a direct result of the relationship between the center conductor, the dielectric and the shield. If a cable is pinched in a door, or bent sharply around a corner, the characteristics of the cable can be changed dramatically and have a significant negative affect on system performance.

Electromagnetic fields generated by other radios, AC power, arc welders or…, well you get the idea, can also have a negative affect on your wireless communications system. Avoid placing antennas near any device that has a strong electromagnetic field associated with it. Also, do not route antenna cables in the same runs as high voltage AC lines. Whenever possible, try to keep antenna cables by themselves. The thing to remember is the RF signal at the receive antenna of a typical wireless intercom system can be less than 0.5µV, that's 0.0000005 of a volt! It does not take much to disrupt such a small signal and anything we can do in the installation process to prevent that disruption is time well spent.

CHAPTER

8

DETERMINING INTERCOM NEEDS

DAVE RICHARDSON

Conference Versus Point-to-Point Requirements

As previously discussed in this book, there are at least two types of wired intercom systems: **conference (two-wire)** and **point-to-point (four-wire)**. Although the conference style provides sufficient communications capabilities for some facilities, the point-to-point four-wire matrix offers not only functions of the conference style, but, also other advantageous modes. The static two-wire conference system, often seen as the back-end of a good matrix system, is usually comprised of belt packs, power supplies, system adapters, and some method of assigning channels to the ports of the matrix, such as the SAP612 Source Assign Panel. This scenario provides the best combination of resources to cover most requirements of the medium to large modern broadcast facility.

A TW intercom circuit transmits and receives audio on two wires. This format is conference by nature, with each station paralleled to each other. The TW system, originally manufactured by **RTS**™ Systems, was the first professional two-wire intercom to include two conference channels, call signaling, and microphone-cancel, all on three wires using ordinary microphone cable! Communication on a TW system may be full or half duplex. Operation in either of these two modes is dependent upon factors such as ambient noise, congestion, etc.

Forms of communication, known as conference, party-line (PL), point-to-point (PP), interrupt fold back (IFB), and isolate (ISO) may be introduced by adding subsystem modules to the base system. The two-wire conference system, as we shall see in larger systems, usually requires more wiring than a digitally controlled matrix. This increase may be a financial and engineering consideration when choosing such a system.

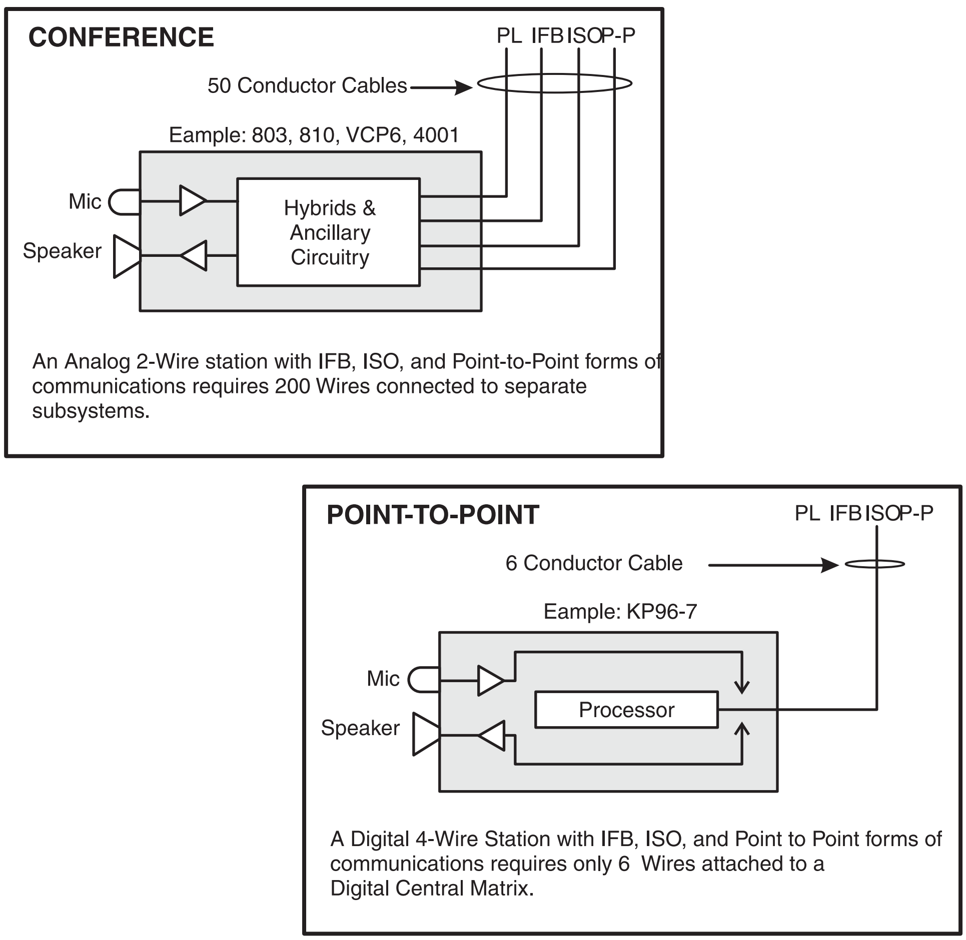
A matrix (four-wire) intercom circuit transmits audio on one pair of wires and receives on a second pair. This format is point-to-point by nature and can be pictured as a star configuration - each station connects to the center through its own multi-conductor link. Instead of subsystems to achieve different functions, the central processor and software permit the system to be dynamically configured for different forms of communication. Because of the digital control inherent in most modern matrix intercoms, this type of system usually requires much less wiring.

Choosing either of these systems for a given facility is a blend of budget, existing wiring considerations, and potential intercom size. But another, less obvious reason, that studio control operators might prefer a two-wire intercom, in spite of wiring difficulties, is subconscious panel differentiation. With a separate module used for every form of communication, a Director's station may have separate panels for PL, ISO, point-to point, and IFB control. A minimal link exists between the control stations and sub-systems. The link is provided by an un-switched microphone connection on the control station. Therefore, in the heat of a production, the director knows which panel does what, and may quickly access IFB to a specified talent.

In contrast, a digital four-wire control station emulates each form of communication from different keys on one panel. Because panel differentiation is gone from the digital station, a concern for operator confusion arises. To solve the problem, the forms of communication can be easily be grouped into specific areas of the station panel or expansion panels.

Based on wiring complexities, a two-wire intercom is often preferred for less complicated applications that require quick setup and teardown. A small two-wire system is easy to install, because it uses simply a power supply, the cable and the stations themselves. This configuration provides one or two channels, which is often enough.

Figure 8.1 Wiring differences between larger conference and point-to-point styles.



If a medium or large permanent installation is under study or system expansion is expected, a digital, four-wire, point-to-point system offers advantages. It inherently produces all forms of communications without the subsystems. The **ADAM**™ and **Zeus**™ matrices require minimal rack space, which is a welcome factor in tight quarters, such as TV mobile units.

A static two-wire conference system can be tied to a four-wire, point-to-point matrix through interfaces such as the SSA324 and SSA424. It is best to avoid interfaces, if given the choice, but digital interfaces, such as the SSA424, yield good intersystem transparency. As a result, an initial two-wire purchase can interconnect to an ADAM or Zeus four-wire host later without significant trans-hybrid losses.

Fixed vs. Mobile Requirements

A General Overview

In the age of the portable control room, fixed and mobile requirements in larger systems are surprisingly similar to each other. Within television production vehicles, such as those used for sports and by major networks, all of the intercom forms of communication must be present to produce from small, to very large shows. Notwithstanding, some contrasts between fixed and mobile requirements could include quantities of cameras, belt packs, and IFBs. There are other minor differences, with the majority of them mechanical and weight related. For example, placing a matrix system in a truck might be better served with the more secure DE9 connector rather than the quick disconnect RJ12.

In large television production vehicles, it is common to see 12 to 16 cameras covering a major event. This is typically more than the average television news station where three to six cameras would be more common. Traditionally, cameras were voltage based two-wire, and most of them were not directly compatible with the **RTS**™ **TW** intercom. A belt pack was used instead and connected to a SAP1026 or SAP1626 Source Assign Panel to set up the desired conferences.

Along with the increase in number of cameras came the boost in IFB channels in both large mobile and fixed installations. Today, eight to 16 channels of IFB output in a large mobile vehicle are common and have matched the corresponding IFB increase in large fixed installations. The trend toward more IFB circuits in fixed installations has been fueled, of course, by the advent of the portable videotape camera and microwave links that permit multiple reporters to contribute to live newscasts simultaneously.

Curiously, the requirement for vast numbers of belt packs and other two-wire devices has diminished slightly over the years for fixed installations and, to a lesser extent, in large mobile vehicles. One reason for this tendency is that camera intercoms, operating in the four-wire mode, can be attached directly to a digital matrix without an interface. The desired conference channels may be dynamically assembled in the central matrix to establish the required conferences.

Also of note is that the trend toward matrix intercom for mobiles has increased over the last ten years when customarily, trucks would only contain two-wire intercoms with appropriate subsystems. The primary reason for this development is better reliability in matrix systems, as well as, easier reconfiguration of the intercom.

In smaller truck systems, such as satellite and electronic news gathering (ENG) vehicles, a minimal intercom system is often required. These vehicles usually consist of just the cameraman, local director, and talent. The two forms of communication that are required for this kind of remote operation are conference channels and IFB circuits.

IFB used in the ENG situation consists of both local IFB and studio director IFB with the local IFB downstream from the studio IFB. A system configured this way simultaneously allows the mobile director to communicate with the cameraman and IFB. At the same time, the main studio conference channel may be superimposed on the local conference net via either a microwave channel or a telephone interface such as the Telos Link.

The smaller television studio can be similarly equipped. The addition of more control stations, a method of source assignment, and more two-wire belt packs or wireless PL systems is usually more than enough to provide a powerful and easily reconfigurable intercom for the smaller market.

Determining Intercom Needs, two-wire, four-wire, or both?

In determining intercom needs for a specific application, we begin by first giving examples of intercom requirements. Then, we will attempt to specify an **RTS**™ intercom system to fill that requirement. Although specific applications are presented, we finish with a general discussion of how to determine whether a given system should be two-wire, four-wire, or some combination of each system.

Before we begin, we should again mention that although analog two-wire conference intercoms continue to carry basic communications for many smaller production facilities, digitally controlled matrix systems offer a range of flexible alternatives. By integrating various forms of communications features, the four-wire matrix system for larger applications makes the difference between confusion and a successful production session.

Small Studio or ENG Vehicle

Previously, we presented the example of the ENG vehicle with its small two-wire system. We have also determined this application requires at least two forms of communications: conference and IFB

To fill the need of a small studio or ENG vehicle, we propose the MCE325 system with its inherent simple IFB (meaning no tally or priority). As shown below, the MCE325 and its associated components make an excellent choice for this application. The powerful MCE325 intercom panel has the capability to produce two conference channels and two IFB channels, which is perfect for the electronic news gatherer. There are many other configurations available in the highly versatile and user programmable MCE325, including four-wire, vox, relays for radios, and many others.

Note A small four-wire matrix system such as the Zeus could have been specified, but probably would have been overkill for this application.

In essence, the client now has saved several thousand dollars using a small, but extremely versatile intercom system.

Figure 2. The Model MCE325 provides the intercom backbone for the typical ENG vehicle or small studio.

MCE325 Modular Programmable Station

This unit is available in many physical configurations. For this application, we specify the MCE325-K. The 'K' designator includes the MCE325, MCP1 (1U rack kit), MCS325 modular speaker, and the MCP6 removable panel microphone. The four talk buttons on the panel are Talk1, Talk2, IFB1, and IFB2. Talk 1 and 2 operate the two conference channels. The rear panel channel 1-2 connector ties to the TW5W 1x5 Splitter while its loop-through connects to the PS15. The channel 3-4 connector ties to the IFB325 talent stations. The MCE325 has a headset connector to allow the director to use the system privately, if desired. Other MCE325's can be interconnected to the director's panel. IFB capabilities are retained on other MCE325 panels via a special keying circuit that allows the slave units to operate the master unit IFB.

PS15 Power Supply/MCP2 Rack Kit

The PS-15 power supply and its MCP2 single component rack kit (not shown) provide operating power for both the BP325 programmable belt packs and IFB talent stations.

TW5W Splitter

This rugged unit allows up to five BP-325's to connect to a central location. It can be used outside the truck and placed directly on the ground. Using it, only one microphone cable needs to be connected from the truck to the cameramen, which is useful when the shoot is several hundred feet away. In the small studio scenario, the TW5W can be replaced with the TW7W 1x7 splitter mounted in the rack.

IFB325 Talent User Station

The IFB325 talent belt pack is worn on the talent's belt or can be set on the ground if desired. It has a volume control, a XLR3 line jack, and an earset jack. A substitute for the IFB325 is the TT44 and TR34 wireless IFB transmitter/receiver to provide more freedom for the reporter, or in the case of the studio, the weather position.

BP325 Programmable Belt Pack

Used by the cameramen (or floor managers in the small studio), this unit features microphone kill, call light, and has two conference channels. It has provision for both 4 and 5-pin dynamic binaural-headset and an additional carbon headset. A substitute can be the BP318 single channel belt pack.

Headsets and Earsets (not shown)

The PH1-R Single Muff Headset offers moderate outside noise attenuation, and allows ENG camera operators to use a carry-cam more efficiently. The lightweight version is the PH-88R for small studio use. If binaural operation is desired for the director, the PH3-R5 Dual Muff Headset and PH44-R5 lightweight headset can be used. They offer the ability to maintain separate conference channels in each ear. The PH44-R5 may also be used with BP325 belt pack for positions such as a crane operator, who needs to communicate with the director, the cameraman, and hear program audio, all with separate volume controls. The talent earset is the model 2234, which is on-camera invisible.

Medium Sized Studio and Mobile Intercom

In many ways, the medium sized studio intercom is the most difficult to specify, for it is here we reach a thin dividing line: the extended two-wire system sometimes makes the most sense for such an application, until we run out fixed assets (i.e. quantities of forms of communications). Also, when we have more than five multi-channel, two-wire stations such as the **RTS**™ Model 803 or 810-CL, we should think about the four-wire matrix as a sensible alternative

We briefly touched upon the primary forms of communication used in television broadcast. These are PL, IFB, ISO, and PP, with the first two (PL and IFB) being the most important in order to produce a television program such as the news. Let's take the example of the medium size system and specify first a two-wire intercom and later a four-wire intercom. The positional needs for our example medium studio intercom are shown in Table 8.1.

Table 8.1 An example of the positional needs for a typical medium sized intercom.

Director	**Producer**	**Audio**	**Video**
Technical Director	Tape 1	Tape 2	Tape 3
Chyron 1	Chyron 2	Camera 1	Camera 2
Camera 3	Camera 4	Camera 5	Camera 6
Teleprompter	Floor Director	Roof Access	Telco
Talent 1	Talent 2	Talent 3	Talent 4
Talent 5	Talent 6	Talent 7	Talent 8

Two-wire Case (Medium Intercom)

In Figure 3, we show a drawing of our proposed two-wire medium studio system. A medium mobile vehicle would be similarly equipped. It demonstrates both PL and IFB forms of communication.

803-G1G5 Master Station

In this example, we are using what we call the TW approach. Historically, TW is an acronym for two-wire, and has been used with reference to an RTS™ PL (conference or Party-Line) system for almost 25 years. We have used the model 803-G1G5 two-wire master stations for the more important intercom stations in the control room. This unit allows the operators to access up to six conference circuits in any combination with the additional ability to selectively listen without talking to all, some, or none of them. Because of the 8-channel IFB requirement, the 803 stations have been fitted with the G1G5 option. This option effectively moves stand-alone Model 4002 8-channel IFB control panel into the 803 itself, a compact, space saving feature. This G1G5 option is a high-end IFB system providing both tally and override with other control stations. Two rack units are all that is required for the full-featured 803 Master Station.

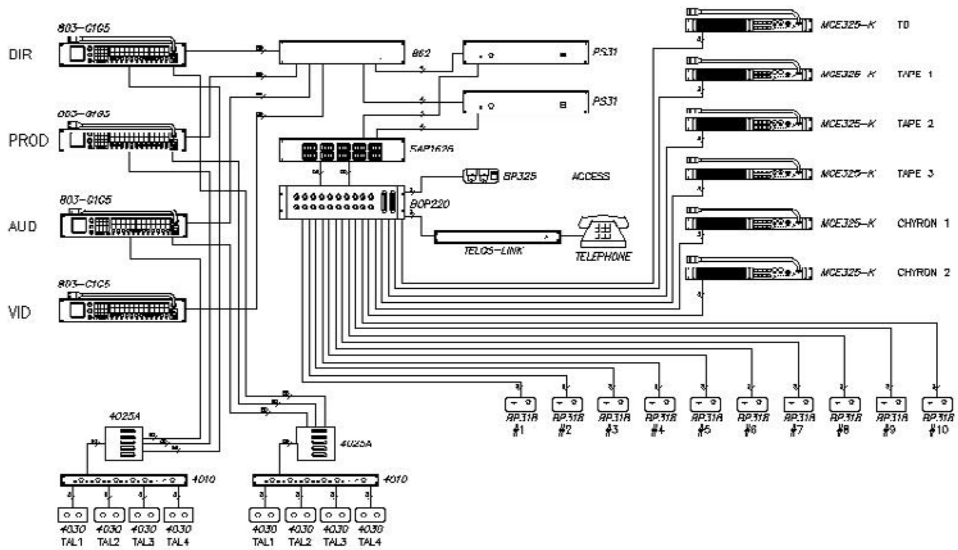


Figure 8.2 Figure 3. Block diagram of a medium sized intercom system using two-wire. The forms of communications depicted here are six conference lines and eight IFB circuits.

862 System Interconnect

The Model 862 System Interconnect Panel is the bridge between the Series 800 two-wire balanced portion and the unbalanced TW section.

PS31 Power Supply

The PS31 TW Power Supplies provide six powered conference channels (three each). All **RTS**™ power supplies contain active impedance generators to allow two-wire intercom audio to be superimposed on the powered channels.

SAP1626 Source Assign Panel

The SAP1626 Source Assignment Panel provides the TW system with its own intercom circuit switcher. In our particular application, the SAP1626 panel provides two functions: it assigns each of the 2 TW channels (stations on the right) to 1 of 12 conferences amongst themselves; and, it fixes each of these conferences to one of the 12-talk/listen buttons on the 803 stations. This provides maximum routing capability, which would be more useful in a mobile vehicle than in the fixed installation.

BOP220 Connector Translation Assembly

The BOP220 is a breakout panel that allows the TW stations to be attached to the system. The BOP220 (Break Out Panel, 2 Channel, 20 Jacks) connects to the SAP1626 on two short 25 pair ribbon cables. There are 20 positions available for user stations. Microphone cable or **Belden**™ 8723 is all that is required for interconnection.

4010 IFB Central Electronics Unit

Located in the audio room, the 4010 unit provides the logic switching and power and program volume adjustments for the 4030 talent user stations. One of the 4010 units handles the first group IFB 1-4, while the other controls the second group IFB 1-4. The program sources for the IFB system are introduced to the 4010 units and then fed to the talent stations on ordinary microphone cable.

4025A Splitter

The 4025A combining device parallels up to four, 25 pair cables to yield one cable to connect to the 4010 IFB Central Electronics Unit.

4030 Talent User Station

The 4030 talent user station is a distributed amplifier with interrupt and non-interrupt volume controls. Studio talent personnel use a Model 2234 earset (previously described), which plugs into the 4030. Sports commentators wearing headphones or headsets use the stereo jack on the 4030. This allows IFB in one ear and a separate program feed to the other ear.

MCE325-K Programmable User Station

In our medium sized intercom, we have specified six MCE325-K units. Although described earlier in the small mobile unit example, these powerful stations are configured differently here. As you recall, we programmed the MCE325-K for two PL and two IFB circuits for the director. In this case, however, the panel assumes the role of a two-channel conference station, which contains a host of features such as individual talk/listen buttons and levels, footswitch control, and call light.

BP319 Belt Pack

The BP319 Belt Pack is a single channel, distributed amplifier (assignable to any of 12 conference circuits via the SAP1626). It provides simple operation with its electronic talk switch and volume control.

BP325 Programmable Belt Pack

Used by the Roof Access position in our scenario, this unit was chosen for its binaural headset capability and high SPL, needed for a high noise area such a helipad.

Telos Link

This interface is a vendor unit that provides a no-fuss link to the telephone for studio-mobile production coordination.

Headsets and Earsets (not shown)

The PH-88R Lightweight Headset was chosen for the BP319 positions, the aircraft noise rated PH10-R5 was selected for the roof access position, and the Model 2234 talent earset was chosen for the news anchors because of its invisibility on camera.

Four-wire Case (Medium Intercom)

Referring to the position table (see Table 8.1), we now specify an equivalent four-wire system for our sample medium intercom. Using the four-wire matrix, we gain two more forms of communication, namely ISO and Point-to-Point:

Referencing figure 3, first we notice the four-wire alternative for our medium system is simpler and uses less cable. Using a laptop or desktop computer (not shown), we configure our system as we did our application demands. In essence, we are the painters on a blank canvas. Attached the back end of our matrix system is a static conference line system linked to the **Zeus**™ matrix via four interface lines. The SAP612 provides a two-wire channel assignment to any of these four lines.

Zeus™ DSP2400 Matrix

The **Zeus**™ Matrix is lightweight, rugged, powerful, and easy to interconnect. It also comes with a wonderful manual and has terrific specifications. It contains 24 ports for connecting to the four-wire devices. The connectors are already installed on the back plane of the unit. All forms of communication are integrated within the Zeus; no subsystems are required.

KP96-7 Keypanel

This master station has 15 talk keys and 15 listen keys, and it is the perfect choice for our application, which requires six conference channels and eight IFB circuits. Being a programmable unit, which can be set up differently for each position, the Video position is set to have 6 conference channels and 6 camera ISO channels. This allows the Video operator to work privately with each camera on a case-by-case basis, if required. On all of the KP96-7 keypanels, we add point-to-point functions between panels themselves and the roof access position on the remaining talk/listen keys.

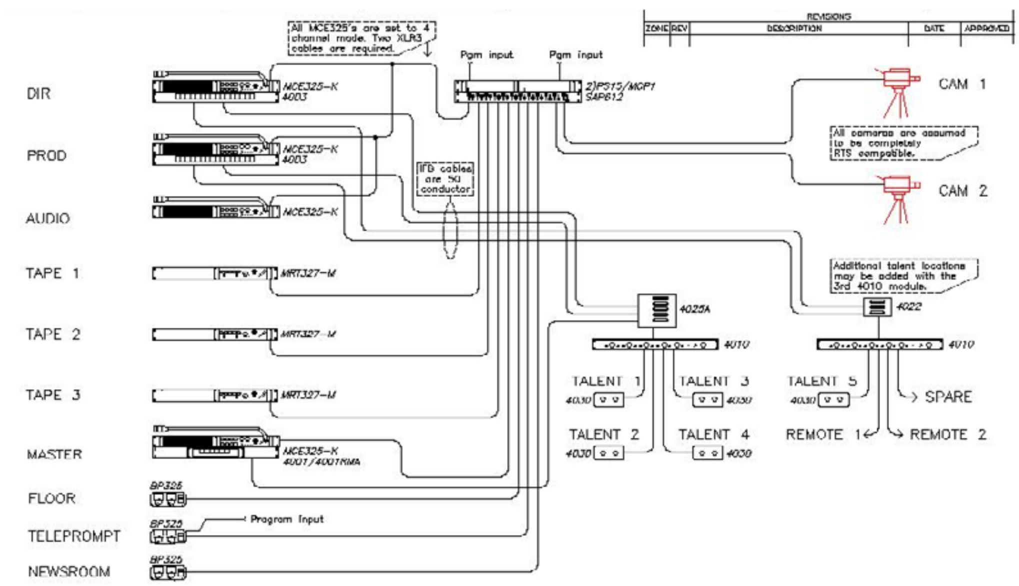


Figure 8.3 Block diagram of a medium sized intercom system using the **Zeus**™ four-wire matrix. The forms of communications depicted have increased to include point-to-point and ISO.

TIF-2000 Intelligent Telco Interface

It is intelligent because it works seamlessly with the KP96-7 panels allowing the operators to use their keypads as telephone dialers. It also can be programmed to ring a keypanel or just silently tally it. This tally, or flashing alphanumeric display, continues whenever the telephone line has been seized, either via auto-answer by the TIF-2000 or manual answer by another keypanel.

MKP4-K Modular Keypanel

This matrix panel can be programmed just like the KP96-7. The K package designator, which applies to all panels in the modular series, means that a speaker, panel microphone, and rack kit assembly are included. The Technical Director in our scenario needs point-to-point capability with the Director, one IFB circuit, and two conference lines. Like all matrix panels, these keys can be programmed either at the panel itself, or from the configuration computer (not shown) that is attached to the Zeus.

IFB828 IFB Power Supply

This unit is the Model 4010's matrix brother. Since all priority, tally, and interrupt chores and handled by the **Zeus**™ matrix, the IFB828 acts simply as a power center for the 4030 talent stations. There are eight of these circuits available. The program sources from the audio console are connected directly to the Zeus on the same ports that are used by the IFB828.

SSA324 System-to-System Adapter

The SSA324 changes a two-wire circuit to a four-wire circuit for introduction into the matrix. Each SSA324 is capable of two of these circuits. We have four ports in our scenario.

PS15 Power Supply

Provides operating voltage, with active impedance generator, to the BP325, BP318's, and the MRT327-K.

SAP612 Source Assign Panel

Since a matrix typically assumes more of the traffic management role than our previous two-wire consoles such as the 803, less outside source assignment is required. Therefore, we can specify the smaller SAP612 for our application to achieve the desired circuit assignment. The SAP612 assigns 12 positions (TW two-wire stations) to one of six conference circuits. As with the larger SAP1626, the SAP612 in our system does two things: it assigns each of the 2 TW channels to 1 of 6 conferences amongst themselves, and it fixes each of these conferences to up to 4-talk/listen keys on the matrix stations.

MRT327-K Modular User Station

The MRT327-K was chosen for the Chyron positions in our medium system for its simplicity. These stations are capable of two-channel operation, one channel at a time.

PAP951 Program Assign Panel and UIO256 GPI

These ancillary devices attach to a special connector on the matrix. No extra ports are required, preserving the maximum size of the matrix. Meaning, more of these units can be added to the system. The PAP951, located in the Audio position, allows the operator to quickly assign programs to IFB circuits. The unit attaches to the matrix on a digital pair, and no actual audio flows through this controller device.

The UIO256 is a 16x16 General Purpose Interface unit. It is included in our intercom for keying a two-way radio in future expansion of the system. If a squelch relay contact closure is available from the radio, the UIO256 can cause a flashing tally to appear at any given matrix keypanel when there is activity on the radio.

Cameras in the Medium Intercom

If the cameras can be set to four-wire intercom operation, they can be tied directly to the matrix. They may be established as any number of conferences. Operating in this manner gains the advantage of camera ISO ability from any matrix station **(on the left side in figure 4)**. Also, the audio levels to and from the cameras can be set internally within the matrix. No extra outside audio amplifiers are necessary. The reason for this is the **RTS**™ **Zeus** is a mixer, not just a matrix of physical crosspoints like those we had in the past.

Large Studio or Mobile Vehicle

Before the advent of the digitally controlled matrix system, large intercom systems were cumbersome to specify. The engineer had to know precisely how many forms of communication to obtain for the system. Any increase in these after the initial sale would mean major physical changes in equipment. Also, to accommodate the special needs of customers, manufacturers were driven to produce special, one of a kind intercoms that were difficult to test, install, and provide support.

The arrival of the first digitally controlled four-wire matrix systems changed customers thinking. Wary at first, because of early reliability problems, customers were slowly purchasing matrix systems to test the waters. The arrival of the **ADAM**™ system in the late 1990's brought many welcome changes. The most important of these were two enhancements: the ability to change individual audio levels at the stations, and matrix linear expansion, instead of logarithmic expansion, which effectively lowered the price of systems (which previously cost thousands of dollars more).

To determine the needs for a large intercom system is, in many ways, easier than of the small and medium systems. First, we assume a four-wire matrix will most likely be a better choice for such an application than an extended two-wire system. The **ADAM**™ 136x136 expandable matrix and its small cousin ADAM™-CS 64x64 make excellent choices for these larger systems.

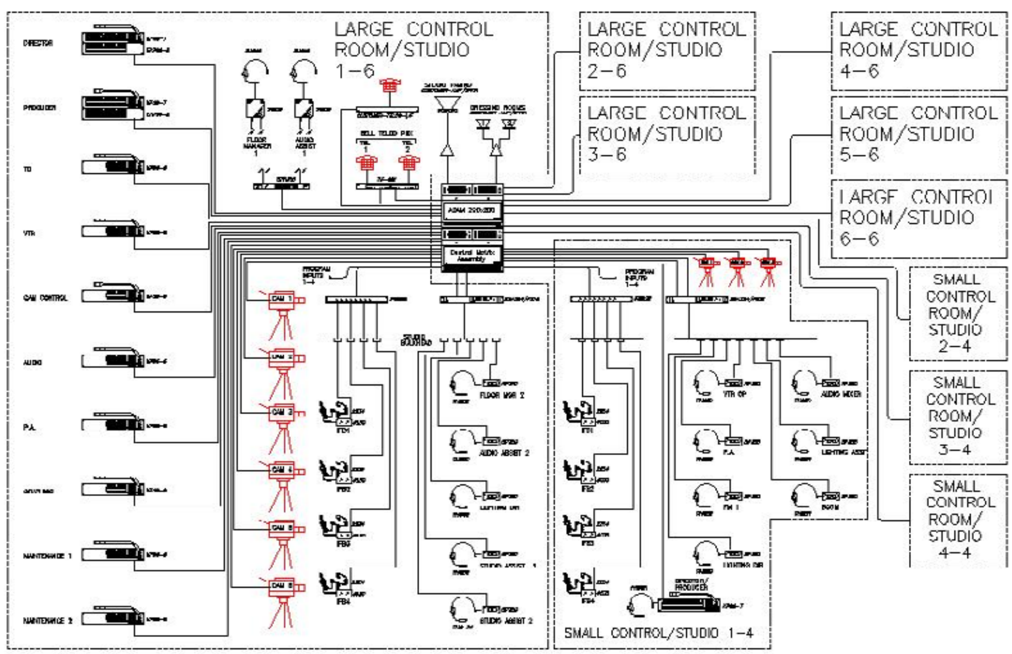


Figure 8.4 Figure 5. Block diagram of a large size intercom system using a twin ADAM™ configured as a 200x200 matrix.

In very general terms, we only need to know two things to get into the ballpark when we specify the equipment list: the size of the matrix; and, the type of keypanels in each location. We cover that exact process the next section, but for now let's look at a typical large system.

After we determine the specific needs of the control rooms and studios in the figure 5, we duplicate these areas to get the total equipment count. A large two-wire system like our sample, would leave us facing such issues such as excessive cabling, mixing loses, routing, and a host of other problems. Working with a digitally controlled matrix makes a large design effortless. Apparent in our sample system is the ADAM here is a storehouse for 10 separate intercom systems! These internal intercom systems are configured to work separately or concurrently with each other. With the new automated server feature in AZ-Edit, files can be downloaded without human intervention by the powerful new UPL (User Programmable Language). This set of useful Boolean algebra expressions allows an almost endless chain of events to be introduced to the system to solve any problem. In short, the days of custom intercoms are over.

Determining the Makeup of the Intercom Matrix

First Step--Determine the Size

The matrix is composed of audio in/out ports (four-wire). These are further classified as panel ports (6 wire) and non-panel ports (four-wire). A **non-panel port** is simply a regular port without the data lines tied to the matrix leaving only the four-wire audio in and audio out. A typical broadcast intercom system consists of Users, IFB Circuits, Cameras, and Miscellaneous ports (to include static Party-Line systems, 2-way radios, telephones, etc.). The first step is to determine the size of the central matrix by counting everything that is attached it. We will use our large sample for this exercise.

Users

The users of the matrix are operators with keypanels. Going down the list of stations (derived from a source-destination table, block diagram, or position list), we count them one by one. In **figure 5**, we have 10 keypanels x 6 large control rooms plus 1 keypanel x 4 small control rooms. Thus, in this example, we have a total 64 users. The 64 users narrow our deciding matrix down to either an ADAM or ADAM™-CS (barely) depending on subsequent port counts. A Zeus (24x24) would definitely be too small.

IFB Circuits

The next port count we need to add is the number of IFB circuits. All **RTS**™ matrices have the unique ability to use a port delegated for IFB in a split fashion. What this means is a port counted for IFB automatically yields an input port for the program feed from the audio console. Therefore, program sources do not typically become a factor in the count unless there are more of them than IFB circuits. The situation is rare, though.

The IFB circuit, used in virtually all television facilities, is usually a one direction audio cue to on-air talent. The signal interrupts a predefined audio source, such as program audio, to inject a directive from the director, producer or audio. In its simplest form, IFB uses an earpiece, an external headphone box (to permit the talent to control the audio foldback level), a program source and a control station. A common IFB application is the live TV newscast where a director wishes to advise the talent a cut-in is starting.

Since all routing of the programs to IFBs is performed outside the matrix in our example, no large matrix assignment panel, such as the LCP102 (64x64 switcher) or PAP950-50 (50x50 switcher), is required. Unlike the PAP951 and PAP952, these panels sometimes find use in larger systems because of their ability to switch to any part of the matrix and thus, not fixed to any section.

With the influx of ENG vehicles, many of today's IFB circuits are telephone dial-in. These particular circuits are sometimes left out of the count for various reasons. Including all IFB circuits not only insures correct matrix size, but also helps specify ancillary equipment such as the program assignment panel as described above.

In our large intercom system, we find 4 IFB x 6 large control rooms plus 4 IFB x 4 small control rooms. This makes our IFB count 40, which brings our total count so far to 100. This means we cannot use an ADAM™-CS 64x64 for this application.

Cameras

Most high-end cameras are capable of operating in the four-wire intercom mode. In our sample large intercom system, the engineering staff has chosen the best cameras available and purchased a great quantity of them. Continuing our count of points in the system, we find 6 cameras x 6 large studios plus 3 cameras x 4 small studios. The total number of cameras is 48. This makes our total count so far 148, which means our ADAM will consist of at least a twin frame combination using bus expanders to connect two matrices together. Each frame alone is capable of 136.

Miscellaneous

Static Party-Lines

On the back end of our matrix, we find a number of belt pack rings. As discussed earlier, these static Party-Lines (2 in each studio) are fixed to dynamic Party-Lines (created by the matrix amongst panels). Under the AZ™-Edit configuration program, the static Party-Line is set as permanent talker and listener on the dynamic PL. Since each static Party-Line must be counted as a port, we find 20 in our large system (2 PL's x 6 large studios and 2 PL's x 4 small studios) bringing our total matrix count to 168.

Wireless Intercom

In each studio, we have added a wireless intercom to consist of a BTR300 base and two TR300 transceivers. Each of these is counted as one port. There are 10 studios total, so our matrix count is now 178.

Telephones

We have three telco lines for each of the large studios, two for production coordination (TIF951) and one provided by a vender for talent dial-in and other general use. Our total telephone circuits are 18, (3 telco's x 6 large studios). Our matrix total now stands at 188.

Studio Announce and Dressing Room Paging

We have one stage announce amplifier and one dressing room page amplifier in each large studio. That is 2 amplifiers x 6 large studios equals 12. This brings our total matrix count to 200.

Second Step--Determine the Panels

Now that we have found the matrix size (200x200) for our large application, it is time to establish the type of keypanel to specify for each of the control positions. The **RTS™** matrix intercom product line has almost 25 different panels from which to choose, from four keys to 64. Generally, we are interested in the quantity of these keys that are needed for each control position. These keys (talk and listen on each panel) are programmed to emulate the four forms of communications. Discussions with operators might be of help in determining the type of intercom console to assist them in performing their jobs.

In our large system scenario, we find two styles of matrix panels are more than enough to fulfill the needs of all 10 positions in each large studio and the director/producer position in the small studios.

KP96-7 Keypanel

The KP96-7 matrix intercom panel has 15 talk and 15 listen keys with alphanumeric displays. This is a perfect quantity for the Director and Producer positions. In the large control rooms, we have added the EKP96-8 Expansion Panel to these positions which gives us an additional 16 talk and 16 listen keys with alphanumeric displays. These extra keys are used for the IFB circuits. We have specified these panels, instead of the new KP32 (32 Talk and Listen Keys, 2U), because of the customer's desire for panel differentiation between the IFB circuits and the other forms of communications (PL, ISO, and PP) that will be programmed into the main KP96-7 panels.

KP96-6 Keypanel

All other positions in the large control rooms will have the KP96-6 keypanel. This unit has seven talk and seven listen keys with alphanumeric displays. Most of the **RTS**™ matrix keypanels have the ability to adjust individual volumes of the point-to-point and conference lines.

Other Considerations in Determining Intercom Needs

Physical Constraints

In mobile and fixed applications, there are times when the client does not have the luxury to choose a given panel because there is not enough room to mount the panel. We will cover alternative panels and other devices that can be substituted for both two-wire and four-wire systems.

two-wire Conference Systems

For rack mount speaker stations, consider using the MRT327-K (1U) rather than the older RMS300 (2U). You will gain call light ability with the MRT327, a removable panel microphone, and a speaker that sounds great despite its small size.

In a tight fit, when using multi-channel master stations, the 810-CL may be substituted for the model 803 Master Station. While not possessing all the features of the 803, this compact 1U station still features 10 conferences.

In the central equipment rack, when considering a SAP1626 Source Assignment Panel to add to an 803 system, you may consider the 4012 Break Out Panel instead. Though the ability to quickly assign conference channels is lost, you will gain 6U in space. The reason for this is that the 862 and SAP1626 are eliminated from your system, and the 4012 mounts in the back of the rack. Conferences are set to 803 buttons by connecting the TW XLR3 cable to one of 12 jacks that determines the button assignment (1-12).

Four-Wire Point-to Point Systems

The **Zeus**™ matrix may be specified if the port count is 24x24 or less. It is lightweight, only 2U high, and all the connectors are on the back.

In a tight area where a 2U matrix intercom panel such as the KP96-7 (15T/15L) has been specified, a substitute could be the KP12. The KP12 (2U) has 12 keys, alphanumeric displays, speaker, and optional panel microphone. If a KP96-6 (7T/7L) has been specified, the substitute in this case might be the MKP4-K. This 1U panel has four keys, alphanumeric display for the call waiting window, panel microphone, and speaker.

In the case where a high-end station such as the KP96-7 with EKP96-8 expansion (4U total) is earmarked, a KP32 keypanel (2U) could be specified gaining 2U of rack space. This is particularly valuable in the large mobile situation.

How old is Too Old?

Should you completely replace your existing system? Probably not! The **RTS™ TW** system has been around for a long time. As such, there are products that have been in constant service for 20-25 years. The good news is the newer two-wire conference products are completely compatible with the older two-wire products, as long as the older units are Phase 3. **Phase 3** (circa 1979) means that operating power is required on only one channel, not both. Adding a source assignment panel and more user stations can be done rather easily as long as the existing power supply (a PS10 or PS50) can handle the increased capacity. Even an 803 system can be added to an old system with little or no modification, as well as the addition of a front-end matrix to the older system.

In the case of older matrix systems, such as those currently in use worldwide, existing keypanels can be used with a new system. Even those panels used as far back as the CS9400+ disk based system, can be used with a new **ADAM™** matrix with a little modification.

Expandability

As discussed earlier, one of problems with the extended medium and larger two-wire conference systems is you will need to know precisely what you need in advance of the purchase. This is more apparent when specifying IFB, ISO, and Point-to-Point forms of communications. If you have purchased the 4001 4-channel IFB system, you will find yourself in a bit of a pickle the day you need six IFBs! In the case of conference channels, you can certainly add more belt packs or master stations without much of a problem. This is because the **RTS™ TW** system is a current based system rather than a voltage based one. All TW stations exhibit high impedance to the line, and hence, do not load it down.

In terms of the matrix systems and their expandability, it depends on the application. A **Zeus™** matrix is expandable up to a 24x24, and indeed comes with the maximum 24 ports. Therefore, additional panels or other devices can simply be added without having to do anything to the frame. The ADAM™-CS can be ordered in groups of eight ports to a maximum of 64. It is a good choice for medium applications where a Zeus might work initially, but expansion is foreseen. Finally, the standard ADAM can be used in larger applications from 8x8 to 136x136 and beyond via frame expansion.

Interoperability

One of the key things to consider in determining intercom needs is when a television station is owned by a network or has a mobile vehicle already with an **RTS™** matrix. Additionally, if there is a need for trunking to other intercom systems, which are of the ADAM or Series 9000 vintage, a similar system should be specified. With trunking, as described in other sections of this book, up to 20 intercom systems can work intelligently with each other, as if they were one very large system. This sets up the ability to use the scroll list on a given matrix keypanel to access the Director in another intercom system, either locally or halfway around the world.

Maintenance

Less is more, so they say, and so it is with the intercom (i.e. less wiring the better). In the age of the digital matrix, even with its fewer wires, comes another welcome arrival, AZ-Edit. Virtually, anything regarding the health of the system may be determined through this intuitive program. Gone are the days of troubleshooting audio because any keypanel can produce a tone to follow. The crosspoint screen can be displayed to show why people are hearing at any given point. You can even show what keys are activated on a miniature keypanel at the configuration terminal. Recent advances also include software upgrades to keypanels and matrix via active download, audio level control, and other interesting features.

Keypanels in **RTS**™ matrix systems do not store configurations. Therefore, if a panel needs to be replaced at any location, the new panel will assume the identity of the old one. On the matrix, there are diagnostic LEDs on the hardware that show if there is a problem or fault.

Budget

Getting back to reality, budget is always the determining factor whether you will purchase a two-wire, four-wire, or some combination of both. Generally, if your budget is limited, you can start with a small two-wire system with idea of making this system the back end of a future matrix system. Tailoring a purchase in this way takes a bit of doing, because of the increased level of planning that must be done. Telex Communications, Inc will work with you to help you in this early stage of equipment specification. Whatever your decision, you will have a new system that works flawlessly and becomes an invisible part of your work life.

GLOSSARY

A

Acoustics The science of sound.

Acoustical A term used to differentiate a sound signal from its electrical signal counterpart or representation. For example: A microphone converts an acoustical signal (from music or speech) to an electrical signal. A loudspeaker converts an electrical signal to an acoustical signal.

Active Devices Devices requiring operating power (battery or other) in addition to the signal. Examples are transistors, integrated circuits, amplifiers, and intercoms.

AF Audio Frequency. Within the range of 20 hertz to 20,000 hertz.

AGC Automatic Gain Control.

All Call For talk key assignment only. Activating an All Call key will also activate all talk keys to the left of the All Call key (up to, but not including another All Call key).

Alpha Alphas are the user-changeable names which identify destinations (intercom ports, Party-Lines, etc.). Change Alpha names for intercom ports using the Port Alpha button in AZ™EDIT Change Alpha names for everything else using the Other Alpha button. When you assign a destination to a talk key, the alpha name will appear in the alphanumeric display for that key (on keypanels so equipped).

AM Amplitude Modulation.

Ambient Conditions existing at a location. Example: ambient temperature.

Ambience Background noise or sounds.

Ampere The amount of electrical current when one volt is applied to one ohm. Also equal to one coulomb of electrical charge passing a point in one second.

Amplitude The size of analog electrical signal as opposed to its frequency or other parameters. Magnitude also indicates a size. Amplitudes focus more from the measurement viewpoint, for example: a one volt peak sine wave amplitude, a one volt average amplitude.

Amplifier Usually an electronic device that increases the amplitude of an electrical signal. Examples include a microphone preamplifier that brings millivolt signals to volt levels. A power amplifier that makes a one milliwatt signal into a 10, 100, 1000, or more watt signal.

Analog vs. Digital Analog (as opposed to digital) here refers to the way information is put onto an electrical signal. An analog signal varies in voltage or current in step with the signal it represents. In the case of the acoustic pressure wave from speech, the pressure wave is converted to an electrical signal by a microphone. The voltage from the microphone varies as the sound pressure from the acoustic wave. A digital electrical signal either represents a binary number 0 or binary number 1. Combinations of numbers represent the amplitude of the pressure wave. The pressure wave is sampled at a rate two or more times the highest frequency to be transmitted. Therefore, there are a sequence of digital numbers representing the speech over a period of time. The advantage of analog circuitry is that it is conceptually simple and relatively easy to create. The disadvantage of analog circuitry is that it is sensitive to distortion and the quality of the circuit design and fabrication must be very high. Advantages of digital circuitry include 1. Frequency response, and distortion are constant and independent of the circuitry (either it works or doesn't, the circuitry doesn't change the frequency response or distortion). 2. Physical aging, wear, and tear have little effect on the quality of the signal. 3. The circuit design and fabrication are very straightforward. The disadvantages of digital are that a substantial investment in system / circuit design must be made, the circuitry tends to produce and radiate interfering signals, more circuitry than analog is required in small units.

Attenuation The decrease in magnitude of a wave as it passes through a transmitting medium (including air, cables, circuitry). Attenuation is also used to indicate a numerical value of the attenuation through an electrical attenuator, for example: a 10 decibel attenuator (or "pad").

Attenuator (Loss Pad) A device, usually passive, that decreases the amplitude of an electrical signal. For example: to prevent the overload of a sensitive microphone input when the signal is much larger than a microphone signal.

Audio 1. A term used to describe sounds within the frequency range of human hearing. Also used to describe devices that are designed to process signals generated from audio (acoustic) energy or to be used to generate audio (acoustic) energy. 2. In television, the sound portion of the program.

Audio Frequency Range of frequencies lying within the range of human hearing, often 20 hertz to 20,000 hertz, where hertz is cycles per second.

Auto Follow (AF) A key assignment for listen keys only. Auto follow causes a key's listen assignment to always be the same as the talk assignment. Thus, if you change the talk assignment, you do not also have to change the listen assignment. You can manually activate an auto-follow listen key independently of the talk key. If you want auto-activation (or deactivation) of listen during talk, use one of the other auto key assignments, such as auto listen or auto mute.

Auto Functions Auto functions are special key assignments that work with other key assignments. For further information, see the glossary descriptions of individual auto functions: auto-follow, auto-listen, auto-reciprocal, auto-mute, auto-table, all-call, DIM.

Auto Listen (AL) A key assignment for listen keys only. This assignment works like auto follow, except that listen automatically activates during talk, Auto listen is sometimes a good assignment for use with Party-Lines or other non-keypanel devices that do not have talk-back control of matrix crosspoints.

Auto Mute (AM) A key assignment for listen keys only. This assignment works like auto follow, except that listen automatically mutes during talk. Auto mute can help prevent feedback or echo when talking to certain destinations. In some cases, you may find it works better to disable talk latching for this type of key, because if you accidentally leave talk latched on you will never be able to hear the destination. To disable latching, in the Keypanels / Ports menu of AZ™EDIT, check the "D" check box for any talk key that has auto mute selected as the listen assignment.

Auto Reciprocal (AR) A key assignment for listen keys only. This assignment forces you to continuously listen to whatever is assigned to the talk key. It is used commonly on keypanels which are not equipped with listen keys, to allow listening to Party-Lines. It is also useful to force listening when it is desirable to have an operator continuously hear a Party-Line or other source.

Auto Table (AT) A key assignment for listen keys only, when the corresponding talk key is assigned to an IFB. Auto Table causes a listen key's assignment to always be the same as the Listen Source for whatever IFB is currently assigned to the talk key. (You define the Listen Source in AZ™EDIT during IFB setup.) Auto Table is convenient in a broadcast environment when a director needs 2-way communication with the IFB talent, AND the IFB keys are frequently reassigned during the course of a program to talk to new talent locations. Using AZ™EDIT, several IFBs can be set up in advance, and their Listen Sources can also be defined during setup. Then every time an IFB talk key is reassigned on a keypanel, the Listen Source for each new IFB will automatically become the listen key assignment for that key. For further information about Auto Tables, Listen Sources, and IFBs, search for "IFB" in AZ™EDIT help.

AWG American Wire Gage. For example: the AWG wire size recommended by RTS™ Systems for intercom wiring is 22 gage.

B

Balanced line A balanced two conductor line carries audio that is differentially driven and balanced to ground. Neither conductor is tied to circuit common. Circuit common is either tied to a transformer center tap, or is an electrical center point, or not tied at all. The signal (with respect to ground) on one conductor is equal and 180° out of phase with the other conductor. Balance Adjustment or Control In stereo audio, an adjustment to balance the left channel versus right channel. At RTS™ Systems a name given to the null adjustment control. The null adjustment or balance control is used in two lines of products: user stations and interfaces — two to four-wire interfaces and two-wire to two-wire interfaces. In the headset user stations the balance control (called sidetone in this case) is adjusted until the side tone heard in a headset is optimum. In a speaker user station, the control is adjusted for the best null so that a full duplex conversation can be held with both the panel microphone and loudspeaker enabled at the same time (without feedback).

Bathtub Curve A curve showing failure rate versus elapsed time. Typically this curve is bathtub shaped. Initially a high failure rate occurs when active and passive parts fail (“infant mortality”). The parts fail under initial turn-on and burn-in stress. The flat part of the curve is the normal life of the equipment. The curve rises again when the equipment ages and is in the wear-out part of its life. Inspecting parts before they are used reduces the initial failure rate. Using higher quality components and proper derating of components in the equipment design, lengthens the equipment operating life. RTS™ Systems burns-in power supplies and other equipment to catch early failures before the equipment goes to the end user.

Bel Originally a unit of measurement that meant that a sound was twice as loud. A more convenient unit for other reasons is a decibel, which is a tenth of a bel. Therefore an increase of 10 decibels is twice as loud (and a decrease of 10 decibels is half as loud). Mathematically a unit that represents the logarithm of the ratio of two powers. See also decibel.

Beltpack Portable headset user station. This station is designed to be worn on a user’s belt, but is also fastened to the underside of consoles, taped to a structure near the user, or mounted on a piece of equipment.

Binaural 1. A special process of using an artificial head and two microphones to closely emulate the spatial and frequency hearing of a human. 2. Two earphones, two signals, may be stereo or may be two different signals.

Biscuit A portable speaker station.

Bit Binary Digit. One eighth of a byte. One eighth of a dollar.

Block Diagram / Single Line Diagram A diagram to show the basic concepts of a device or system. Often the block diagram has system parameters such as transfer functions, gain, loss, level, DC voltage, inputs, outputs, and so on. I.) Block Diagrams are used on the Product Data Sheet to clarify the functions, show performance capabilities, and to show the input, output, control, and interconnection points. This diagram also defines and clarifies the specifications called out on the data sheet. II.) The diagrams called Block Diagrams are often, in fact, “Single Line Diagrams”. Single Line Diagrams are similar to the Block Diagrams but show more detail such as number of conductors in a cable, connector designations, connector details such as male/female, Equipment Model Numbers, and Equipment Designation Names/Numbers. At RTS™ Systems, these single line diagrams are called “System Block Diagrams” and are used for several purposes: 1. Act as a check list against the customer requirements. 2. Demonstrate to the customer the meeting of the customer’s requirements. 3. Are used to develop the equipment list (Lists the Quantities and Models numbers of equipment required to make the system). 4. Provide the information necessary to perform a system test. 5. Provide information to estimate wire and cable requirements. 6. Provide information to aid installation at the customer site. (Wiring Diagrams and Wire List can be generated from this information). 7. Graphically give a measure of the size and complexity of a given communications system. 8. Provide a means of troubleshooting system problems during commissioning, during operation, and during maintenance. 9. Provide a documentation basis to expand the system in the future. 10. Provide documentation for telephone support of the customer from the factory.

Blocking A communication system blocks a requested call or access usually by a busy signal.

Bridging Bridging impedance means an impedance that when paralleled with a nominal impedance will have an non-significant effect on a circuit. For example: for a nominal impedance of 600 ohms, a parallel impedance of 3,000 ohms (5 times) would make the net impedance 500 ohms, 17 percent less than 600 ohms or 1.6 dB. A parallel impedance of 6000 ohms (10 times) would cause about a 9 percent change or about a 0.82 dB difference. A parallel impedance of 12,000 ohms (20 times) results in about a 3 percent change or 0.26 dB. In present day audio systems, line level and power amplifiers have input impedances specified at 600 ohms or 15,000 ohms. Microphone preamplifiers are usually ten times the expected source impedances. For example: for 150 ohm microphones the input impedance is 1,500 ohms or greater. Earlier RTS™ Systems intercoms also had this microphone input value, but a compromise value of 470 ohms was necessary because of the crosstalk in headset cords. However, RTS™ Systems professional audio equipment generally adheres to current audio standards.

BW Bandwidth.

Byte Eight binary digits or bits.

C

Call light A feature in intercoms that is used for two different purposes: 1) To get a user to put his headset back on (blinking call light). This method is the standard way for RTS™ Systems equipment. 2) To generate a cue (steady call light). The usage in this case is often as follows: light on means standby, light off after light on means execute. This method is used by other manufacturers and IS optional with RTS™ Systems equipment. In some user stations, the call light feature is standard (BP325, MCE325), in other stations, it is an option.

Capacitance The ability to store electrical charge between two conductors. Measured in farads (Named after Michael Faraday). A capacitance value of one farad can store one coulomb of charge at one volt. One farad permits one ampere of current when the voltage changes at the rate of one volt per second. Typical sizes are measured in: millifarad one-thousandth of a farad microfarad one-millionth of a farad nanofarad one-thousandth of a millionth of a farad picofarad one-millionth of a millionth of a farad.

Capacitive Reactance The opposition to alternating current through a capacitor. Capacitive reactance, Xc is measured in ohms and is equal to: 1 / [2 * π * frequency * capacitance].

Capacitor Two conducting surfaces separated by a dielectric. The dielectric could be a material, air, or a vacuum. The capacitance of the capacitor is a function of the area of the surfaces, dielectric, and spacing between the conducting surfaces.

Cardioid Pick-up Pattern The pick-up pattern of a directional microphone is frequently of cardioid (heart) shape. The maximum cancellation (minimum pick-up) occurs at an angle of 180°. The sound power concentration is approximately three times.

CCU Camera Control Unit. Usually located in an equipment room (studio) or a bay in a mobile truck (mobile). The CCU is connected to the camera "Camera Head" via a cable. The cable is either wire "multicore," triaxial cable "triax," or coaxial cable "coax."

Channels and Buses Channels and Buses are pathways for signals to travel. There are more than one channel or bus to allow for multiple conversations or information flows to occur simultaneously. Multiple buses separate signals using space and the process is sometimes called space multiplexing.

Analog Channels and Buses In the discussions here, analog Channels or buses carry signals representing audio. There is an exception, the call light signal is superimposed over the signals representing audio. This signal is not heard by humans because of its 20.0 kilohertz frequency. In this case the voice audio and the call light signal are multiplexed using frequency separation. The words "channels" and "buses" are often used interchangeably. In a twelve channel or bus system, it is possible for a user station to be tied to say system bus 5 and system bus 3 when the user station channel selector switch reads 1 and 2 respectively. For purposes of distinction, discussions that talk about system channels or buses and user station channels, the word bus will refer to system buses and the word channel will refer to user station channels.

Digital Buses In the microprocessor units, addresses and data are moved on digital data buses. These buses vary in width from three to 16 bits. Some buses are bi-directional and the logic transmitting and receiving data on these buses is usually of the "three state" variety. Data is multiplexed on these buses using time division or separation.

Characteristic Sound Pressure Level The Characteristic Sound Pressure Level of a headphone is the sound pressure level that an electrical output of 1 milliwatt generates.

Circuit 1. A complete path for electrical power or an electrical signal (usually two conductors). 2. In a system, a channel for one or two way conversation may be called a circuit.

Circumaural Headset A headset where the earpieces surround the ear usually providing some isolation of outside noises from the ear.

Clipping A type of distortion resulting from overdriving an amplifier.

Close-Up Effects Found in a pressure gradient microphone, this effect causes strong low frequency pickup at near distances.

Coil Effect The inductance exhibited by a spiral-wrap shield at audio frequencies.

.Communication between Ports (Point-to-Point, or P-P) The audio signal from any input port can be routed to any output port. For example: during keypanel setup, you assign keypanel keys so that keypanel operators can talk and listen to other intercom ports. Communication of this type is called point-to-point communication. You can also route signals between intercom ports without keypanels. One way to do this is to force crosspoints in the Crosspoint Status screen of AZ™EDIT. Another way to do it is with a GPI input.

Compression Force Headset wearing comfort is affected by weight and the force of the earpieces on the head. This "compression force" is measured in newtons, N. One newton is about the weight exhibited by a mass of 100 grams.

Condenser Microphone A microphone using a capacitor as the sound pressure sensing element. Condenser microphones require a polarizing voltage. Condenser microphones outputs are high impedance and need to be buffered by an active device. The active device(s) needs power, so various phantom and A-B powering schemes are used to buffer the active device(s).

Conductivity The ease by which a material will support an electrical current. Mathematically the reciprocal of resistivity.

Conductor A material that will support an electrical current.

Conference Intercom Systems, Conference Line Intercom Systems, Party-Line (PL) Systems A conference system allows a group of people to intercommunicate. For example, one person can talk and all the others on the bus or channel can hear. When the system is full duplex, anyone can talk and the rest can hear or interrupt the speaker at any time. The conference and distributed matrix systems presently sold by RTS™ Systems are full duplex and are non-blocking, which means that access to the channel is immediate and there is no busy signal. Conversations on conference systems are in general, non-private. A conference system can be two-wire or four-wire. RTS™ Systems sells both two- and four- wire conference systems. The two-wire conference system (RTS™ Systems "TW" system) is simple, economical, and very convenient to use. The four-wire conference system performs as well as the two-wire system, is easier to interface to other systems, but requires more equipment and is more costly. Conference systems can be distributed or centralized. Most of the systems that RTS™ Systems makes are distributed conference systems. Distributed means that a station can be plugged-in at any arbitrary point along the bus or channel. Centralized means that all stations are tied to a central point where the conferencing function is actually accomplished. Note: Sometimes the conference intercom system is called an interphone or headphone / headset system.

Control Room A room, usually adjacent to a studio, where the production is controlled by the producer, director, technical director, (and sometimes the audio mixer, lighting director, assistant director, production assistant, and Chyron operator). In remote pickups the control room is in the mobile unit, which may be several kilometers from the televised action.

Coupling with the Ear Basically differentiation between ear-pieces that are worn on the auricle (supra aural headsets), and those that envelop the auricle (circumaural headsets).

CPS Cycles Per Second. Obsolete designation replaced by Hertz (Hz).

Crosspoint The term "Crosspoint," like the term "Matrix" is inherited from intercom systems, such as the RTS™ CS9500, CS9600, and CS9700, that use a switching matrix to route intercom audio. In those systems, the crosspoints are the actual switches that close or open to connect or disconnect talk and listen paths. RTS™ ADAM™, ADAM™ CS, and Zeus™ Intercom Systems do not actually use crosspoint switches, but use a technique called time division multiplexing (TDM), in which communications are routed as digital packets. However, use of the term “crosspoint” persists since packet routing basically accomplishes the same thing as conventional crosspoints: namely, connecting distinct talkers and listeners. In this sense, a crosspoint can be thought of simply as a communication link between any two points in the intercom system.

Crosstalk Interference caused by audio energy from one line coupling (“leaking”) into adjacent or nearby lines.

Current A current is a flow of electrons past a point in a circuit and is measured in amperes (coulombs per second). Practical currents in electronics are measured in: amperes, milliamperes one-thousandth of an ampere, microamperes one-millionth of an ampere, nanoamperes one-thousandth of one millionth of an ampere, picoamperes one-millionth of a millionth of an ampere, femtoamperes one-thousandth of a picoampere

Current Sources RTS™ Systems uses “Current Source” technology in many of its communications products. This technology allows the summing of signals on a single pair of conductors across a single system bus termination. This allows a distributed conference line system. Stations can be added arbitrarily anyplace in the system. The system allows two to 75 stations to be put on the system with only a maximum level difference of six decibels. The current source allows a signal to be put on the bus without shorting out the other signals.

D

Daisy Chain Some TW user stations allow the stringing together (or daisy chaining) of user stations. These stations have a “loop through” or “extension” connector as well as a ‘line” or “line input” connector. Connecting up a TW system by connecting one user station to another via the line and loop through or “ext” connectors. This is as opposed to “home running,” which is running a cable from each user station to a central point (“home”).

dB Decibel, see definition for decibel.

dBm A reference level where 0 dBm equals 1 milliwatt. In a 600 ohm system 1 milliwatt corresponds to a voltage of 0.775 volts.

dBu A reference level where 0 dBu equals the voltage as a dBm (0.775 volts) but without the 600 ohms in the circuit.

DC Direct Current. Example: current as from a battery.

decibel (dB) 1. One-tenth of a bel. It is equal to 10 times the logarithm of the power ratio, 20 times the log of the ratio of voltages or currents. Three decibels increase represents a doubling of power, six decibels increase represents four times the power or a doubling of the voltage in a circuit. 2. A derived unit of loudness. The human ear perceives a 10 decibel increase as twice as loud, and a 10 decibel decrease as half as loud.

Dedicated Line 1. A term used by some to indicate a single path in a point-to-point system. 2. A term used instead of point-to-point or matrix system, for example: a dedicated line system. (This term seems to be more marketing than engineering oriented).

Destination A destination is anything that a talk key talks to or a listen key listens to. A destination can therefore be any port, Party-Line, IFB, etc.

Dielectric An insulating (nonconducting) medium or material.

Dim “Dim” occurs in two contexts in RTS™ Digital Matrix Intercom Systems.First, there is the Dim Table feature. Dim tables are used to correct a feedback problem that can occur between two keypanels operating in close proximity that have keys assigned to talk/listen to a common destination. Dim tables are set up in AZ™EDIT (search for keyword “dim” in AZ™EDIT help. Once a dim table is set up, it can be assigned as a level 2 talk assignment for those keys that are causing the feedback problem. For information about how to make this assignment from a programmable keypanel, search for “Dim Table” in the keypanel manual index.There is also an adjustable speaker dim feature available on the KP-32 Keypanel. This causes the speaker or headphone volume to diminish by a preset amount whenever a talk key is activated. This can help to prevent occasional feedback between the speaker and microphone due to volume settings, microphone placement, etc. For setup and usage, search for “Speaker Dim” in the keypanel manual index.

Distortion Distortion is the effect when the output of an electronic device contains undesired signals that were not present at the input. This is assuming that the electronic device is supposed to be a linear device. The undesired signals have a frequency or frequencies that are related to the input signal. If the frequency(ies) is/are harmonically related to a single frequency input, then the undesired signal is ‘harmonic” distortion. If the signal is the sum or difference of two input frequencies, then the distortion is called “intermodulation” distortion. If the distortion is the result of a pulse or step input, and the frequency(ies) is/are related to sums and differences of the frequencies determined by the Fourier transform of the input pulse or step input, the distortion is called “transient intermodulation distortion”. Distortion can occur both in active devices (e.g. amplifiers) or passive devices (e.g. transformers). Harmonic Distortion is measured in percentages or decibels below the fundamental signal. For example: a distortion of 0.1 percent is “60 dB down”. Intermodulation Distortion requires two input signals (say 1000 and 400 hertz) to be inserted and the sum and difference to be measured.

Double Headset Headset with intercom in one ear and program in the other.

Double-Muff Headset Headset with two earphones plus a microphone. It can be connected monaurally (same information, both ears) or binaurally (separate feed each ear). In binaural operation, the feed can be intercom in one ear and program in the other, or intercom channel A in one ear and intercom channel B in the other ear. Channels A and B are either conference line channels or other intercom feeds. To get a binaural feed requires a binaural/stereo capable user station such as BP320, BP325, Model 802, Model MCE325, or any station so optioned.

Drain Wire An uninsulated wire in contact with a shield throughout its length, and used for connecting (“terminating”) the shield.

Dry Pair / Dry Line A dry pair or dry line is a communications line that has audio signals but no direct current (DC) voltage or current.

DSP Digital Signal Processor. Usually a microprocessor with two memory addressing capability. One memory is the program memory which tells the microprocessor what to do, and the second memory contains: data to be processed, intermediate results, and final results. The advantage of a DSP is its speed. It is fast enough to process analog (or audio) signals in real time, and is often used in that application. Some applications are system to system interfaces (e.g. Telos “Link” for interfacing a standard telephone line to an RTS™ Systems TW Intercom line).

Dual Listen This is either an option or feature of intercom user stations. Dual listen permits an operator to listen to two channels at once. This may be a mix of two channels to one ear, or in a binaural or stereo user station, one channel can be assigned to one ear and the other channel to the other ear. Dual Listen could also be an intercom channel and a program audio source. The dual listen pots are functionally configured in one of three ways: 1. One pot controls the audio of the channel actively used, and the second pot controls the audio of a monitored channel. 2. One pot is always one channel and the other pot is always the other channel. 3. On three channel systems, operation is similar to 1. except if the active channel and the monitored channel coincide, the monitor feed is blanked out to prevent a 6 dB increase in volume and feedback.

Dual Listen Option An option for user stations that allows a monaural mix of two channels. Usually the station has two volume controls, sometimes two concentric volume controls.

Duplex / Simplex See Full Duplex, Half Duplex, or Simplex.

Dynamic Microphone Converts sound pressure waves to electrical signals by means of a coil attached to a diaphragm moving in a magnetic field.

E

E A symbol for voltage used in electronics, and engineering. Also used as the symbol for the electric field (volts per meter).

Earth British term for a reference ground. Earth may mean power line ground or a facility zero-reference ground.

Earphone A device used to hear an electrical audio signal. The earphone converts electrical signals to acoustic signals that can be heard.

EFP Electronic Field Production. Production of television programming using field equipment (mobile trucks, portable gear, et cetera).

EIA Electronic Industries Association (formerly RMS or RETMA).

EIA Sensitivity Also called Gm rating. Adding the EIA sensitivity to the SPL at the microphone gives the microphone power output in dBm into a matched load. Sensitivities for open-circuit can be considered as follows: -65 dB re 1 volt / microbar = high sensitivity (usually results in better signal to noise ratio). -75 dB re 1 volt I microbar = medium sensitivity -85 dB re 1 volt / microbar = low sensitivity.

Electret Microphone A microphone using a capacitor as the sound pressure sensing element. Electret microphones are a special case of condenser microphones in that they are permanently polarized and require no special polarizing voltage. Electret microphone outputs are high impedance and need to be buffered by an active device. The active device needs power so various battery, phantom, and A-B powering schemes are used to buffer the active device, (which is very close to or on the microphone diaphragm).

Electronic Switching versus Mechanical Switching Audio and other signals can be switched either electronically or mechanically. The electronic switching is generally faster and quieter, but usually has some losses. Mechanical switching is generally slower, noisier, but has less or little loss. In switching signals from current sources, electronic switching prevents loss of termination for a significant amount of time.

EMF Electromotive Force (voltage).

Energy The capability of doing work.

Energy Dissipation Loss of energy by conversion to other forms, usually heat.

ENG Electronic News Gathering. Accomplished using television and accessory equipment in a small van, with the capability of relaying pictures and sound back to a broadcast station or network control center. The equipment used may be of special design, for example smaller “ENG type” television cameras.

EMI Electromagnetic Interference. Interference caused by the radiation of electrical or magnetic fields from sources such as radio transmitters, light dimmers, computers, and transformers.

Equalization (EQ) The ability to correct or adjust non-uniform frequency response in a sound system. The equalization may be applied to a signal to be recorded, that has been previously recorded or to a real time (“live”) signal.

Equalizer An electronic device or circuit that allows for the adjustment of a signals frequency response.

F

Farad A measure of the ability to store electrical charge between two conductors. Farad is named after Michael Faraday. A capacitance value of one farad can store one coulomb of charge at one volt. One farad permits one ampere of current when the voltage changes at the rate of one volt per second. Practical sizes are: millifarad one-thousandth of a farad, microfarad one-millionth of a farad, nanofarad one-thousandth millionth of a farad, picofarad one-millionth of a millionth of a farad.

Feedback 1. Audio deliberately fed back to a user, for example a monitor for a musician to hear his own instrument or voice, 2. Audio feedback to a headset or earset as in IFB operations (see IFB), 3. An unintentional return of an electrical or acoustic signal to a microphone or amplifier input, the result of which is an oscillation.

Filter A circuit that is sensitive to signal frequency and is capable of attenuating some signal frequencies and not attenuating others.

Film-Style Directing Directing separate takes or scenes that are to be later edited in postproduction. These takes or scenes are not necessarily in the same sequence as they will appear in the film or tape.

FM Frequency Modulation. A method of adding audio to a radio frequency carrier. FM signals are usually more noise free than amplitude modulation (AM) signals. Wireless intercom units usually use FM.

Follow Spot Used to accent or light action on stage, the follow spot is a focused high power light that focuses a beam from a large circle to a small spot. The operator of the follow spot is usually on the lighting intercom line, and sometimes in small productions on the primary intercom line. The follow spot operators have been known to tape their beltpack to the spotlight or a nearby metal structure. This practice can cause hum and noise in the intercom line because of the large currents involved in lighting. Some of the currents are induced into the metallic structure of the facility causing large "ground" currents. If a belt pack is to be taped to something, a layer of tape should be put around the belt pack first to insulate it from any metal. The newer BP325 has a nonmetallic case, so adding tape to the case is unnecessary, but it is necessary to prevent contact of connector shells and other metal objects with ground or metallic structures.

Four-Wire A communications system where the path is different for talk and listen. In electrical pathways there are, in fact, four wires (two paths). Four-wire systems can be four- wire balanced and four-wire unbalanced.

Four-Wire Balanced Four-wire balanced is similar to four-wire unbalanced except that conductors are not tied to circuit common. Circuit common is either tied to a transformer center tap, or is an electrical center point, or not tied at all.

Four-Wire Unbalanced A four-wire system that uses a circuit common and two additional conductors. The talk pathway consists of one conductor plus circuit common. The listen pathway consists of another conductor and circuit common.

Full Duplex Duplex communication allows simultaneous two-way conversations, that is one person can interrupt the other. In data communications, full duplex permits confirmation of sent data by the receiving terminal echoing or sending back the same data or confirming data.

Frequency The number of times per second a periodic action occurs. Frequency is measured in Hertz (formerly cycles per second).

Frequency Response The range of useful frequencies for a particular device, circuit, or system. For example: a microphone frequency response of 20 Hertz to 20,000 Hertz ±3 dB would be considered excellent. The design goal of the TW system is 75 Hertz to 20,000 Hertz (system), 75 Hertz to 10,000 Hertz (microphone preamplifier), and 75 Hertz to 8,000 Hertz (headphone/speaker amplifiers). The response on an actual system will vary according to the amount of cable in the system, various trade-offs, and the number of stations in the system.

G

Gain 1. Level of amplification for audio/video signals. Operators may need to periodically adjust these levels during production (especially those gain controls on the audio mixer board). 2. An important parameter of a functional block or a circuit device. The gain is the output voltage divided by the input voltage, the output current divided by the input current, or the output power divided by the input power. For example: a microphone preamplifier in a TW user station may have a maximum gain of 54 dB (a voltage ratio of 500). Note that, in the case of the bilateral current source, it is a voltage controlled current source, and is characterized not by gain, but by transconductance. Transconductance is given by the output amperes divided by the input volts. The units of transconductance are siemens (formerly the units were mhos). The bilateral current source used in RTS™ Systems user stations usually has a transconductance of 5 milliamperes divided by 1.5 volts or 3.3 millisiemens.

GND An abbreviation for ground.

GPIO General Purpose Input / Output. (You may also see this referred to simply as "GPI".) GPIO is a means of controlling devices using switch contact closures, DC voltages, or similar methods. For example: you can control a lighting system from keypanel keys, or key a transmitter from a talk key during transmit. Or, simply operate a light or buzzer for cueing. In ADAM™, ADAM™ CS, and Zeus™ intercom systems, you can also control intercom events from external switches. For example: you can activate key assignments, close or open crosspoints, activate GPI outputs, etc. In CS9000 Series intercom systems, general purpose control outputs are provided by optional FR9528 Relay Frames (8 relays each). In those systems, a relay may be assigned to an intercom key on a keypanel using the Relay key assignment type. Pressing the intercom key activates the relay. ADAM™, ADAM™ CS, and Zeus™ intercom systems all have a dedicated GPIO connector (J27 on a Zeus™ Frame, J903 on an ADAM™ CS Frame, and J11 on the XCP-ADAM™-MC Master Controller Breakout Panel in an ADAM™ Intercom System). This connector supports 8 control inputs and 8 control outputs. Additionally, one or more UIO-256 Universal Input/Output frames may be connected to the intercom system. Each UIO-256 provides another 16 control inputs and 16 control outputs. Control outputs may be assigned to intercom keys using the Relay key assignment type, and the intercom keys can then control external devices the same as the FR9528. Control inputs can be assigned to activate “virtual” key assignments. (A virtual key assignment is a key assignment at an intercom port where there is not actually any keypanel connected. Basically, you use an external switch to act like a talk or listen key.) The control inputs and outputs can also be used as conditions for UPL statements in AZ™EDIT.Finally, there is a GPIO option available for the KP-12 keypanel, and a connector module option for the KP-32, which includes GPIO. These are referred to as “Local” GPIO, since they are assigned and used locally at the keypanel. Each local GPIO includes 4 control inputs and 4 control outputs.

Green Room A room for performers / talent to stay just before making their appearance on stage. This room is usually close to the stage, and has amenities plus a video and audio monitor.

Ground The term ground has several meanings. One meaning is a circuit common point potential. Another meaning is a 0 volts point. Another meaning is a connection to the earth. Another meaning is the chassis of radio equipment. Radio Frequency engineers almost always connect circuit return to the chassis. This can cause a ground loop in systems if the chassis is connected to earth ground as well, and the circuit return in the system encounters another earth ground. The TW system circuit return is bypassed to earth ground and tied to earth ground through a 10,000 or 22,000 ohm resistor, in order to prevent ground noises or hums from being introduced into the intercom system. Connection of a chassis grounded device to the TW System should be done through an audio isolation transformer.

Ground Loop A ground loop occurs when a system circuit common is tied to earth ground or another ground or another conductor at two places in the system. This allows “ground currents” to be superimposed on the intercom system circuit common, causing hum and spurious noises.

Ground Potential Often the potential of the earth, but also the potential at a zero voltage point in a system or an electrical/electronic circuit.

H

Half Duplex Half Duplex communication allows two-way conversations, one-way at a time, such that one person cannot interrupt the other. In data communications, half duplex means sent data is not confirmable by the receiving end on a continuous basis.

Harmonic Distortion A distortion at the output of a device where the amplified input signal is accompanied by the sum of unwanted signals that are harmonics o the input signal. Harmonic distortion can be expressed as a percentage of the total output intensity, or in decibels. See also Distortion.

Headphones / Headsets Headsets are headphones with microphones added. Headphones and headsets are available in a wide range of variations. Some of the variations include: Lightweight, Heavyweight, Medium Weight: Lightweight can often be used or worn for a ten hour shift with only mild discomfort; medium weight usage ranges from two to six hours continuous, and heavy weight usage ranges from 15 minutes to 2 hours. Acoustic Isolation: which varies from 0 dB to 40 dB. Usually more isolation means a heavier headset. Acoustic Isolation (30 to 40 dB) is required in high ambient noise environments such as concerts, auto racing, construction areas, aircraft engine run-up, near machinery such as printing presses. Medium isolation (10 to 20 dB) is required in quieter concerts, near crowds, near quieter machinery. Low acoustic isolation can be tolerated in environments such as television studios (news). Impedances: Impedances of headphones range typically from 2000 ohms to 2 ohms. Common impedances per earphone are 300 ohms, 150 ohms, 50 ohms, 25 ohms. Headphone total impedances depend on the earphone impedance and whether they are connected in series or parallel. The headphones in standard headsets sold by RTS™ Systems ranges from 25 ohms to 300 ohms. Military headphones may be very low impedance, 10 ohms or less. A fuel tank entry system sold by RTS™ Systems has 2000 ohm headphones. Lower impedance headphones allow a louder sound (up to 110 dB SPL) to be generated with relatively low voltage in the user station (say 12 volts DC). Microphone Types: (for headsets) The microphones types may be carbon, carbon emulate, dynamic, electret. The carbon types produce high output levels but have higher distortion, the carbon emulate types, put out high levels with low to moderate distortion but require special electronics and a way to power the electronics. The electrets usually have electronics built on the microphone, but there is no voltage gain from this electronics, just impedance matching (from megohms to kilohms). Electrets have about 10 dB more level than dynamics, but are very prone to "popping". To prevent popping, windscreens need to be installed or placed over the microphone element, and the following circuit should have a circa 500 hertz high pass roll off. Dynamic and electret microphones usually have low distortion and good frequency response (100 to 8,000 hertz). Some dynamic microphones made with low technology may have poor frequency response. Some typical microphones impedances are as follows: carbon: small button 600 ohms, large button, 50 ohms; dynamic: 2 ohm (military), 150-200 ohm (RTS™ Systems recommends), 600-1000 (lower cost push-to-talk and others). Microphone impedances can also be higher such as 50 kilohms, but these usually are not on headsets. Most RTS™ Systems User Stations microphone inputs allow for an impedance range of 50 to 1000 ohms for dynamic microphones, 1000 to 2000 ohms for electrets, 50 to 200 ohms for carbon or carbon emulate.

Headroom The difference between the instantaneous level of a signal and the peak signal possible in a given system. Headroom is often expressed in decibels. System headroom in the TW system is about eight to ten dB. Headroom for the microphone input is an apparent 40 dB because of the 30 dB limiter compression ratio. Because of the design of the TW and 800 series systems, and the consistency of levels, the peak to average speech ratio is close to 10 dB.

Hertz The unit of frequency, cycles per second. One thousand hertz equals one kilohertz equals one thousand cycles per second.

Home Run Running the user station system connection cables to a central point (as opposed to Daisy Chaining).

Hot 1. A wire actively carrying power or signals. 2. Equipment that is turned on, for example a "hot" microphone.

Hum Hum is an interfering addition to audio. Its frequency is within that of human hearing and it is at the frequency of the power line or its harmonics. For example: a pickup of the fundamental will result in a 50, 60, or 400 hertz tone in the audio. If the hum is due to excess ripple in a full wave rectified supply the frequency will be 100, 120, 800 hertz. If the power line waveform is distorted (which it often is), other harmonics will be heard. Hum is induced electrostatically via unshielded wires in high impedance circuits, or electromagnetically via unshielded dynamic microphones, transformers, tape recorder heads, or ground loops.

Hypercardioid A microphone pick-up pattern. This pattern has its maximum rejection at 100° off axis. This pattern has good rejection of far field sound and room reverberation. Good in house speaker systems.

I

I Symbol used to designate current.

IBEW International Brotherhood of Electrical Workers

IFB The IFB * System is a special intercom system used for television shows with highly flexible formats or where important program changes are likely, for example, newscasts or special events telecasts. The IFB system connects control room personnel such as the director, producer, audio mixer, and technical director directly with the performers or "talent". The performer wears a either a small earpiece or headset ** that carries the program sound unless the director or another member of the production team operates the IFB and interrupts the program sound with special instructions. * IFB means Interrupted Feedback, or Interrupted Fold-Back. This system is also called Interrupted Return Feed (IRF), program Interrupt, or prompt-mute. ** In sports, stadium, and parade remotes, a double muff headset is used. Impedance is the resistance to an alternating current.

Impedance Impedance is composed of resistance and reactance (rectangular coordinate representation). Impedance can also be viewed as a vector quantity with a magnitude and a phase angle. Impedance can be measured with an impedance meter. Impedance may vary with frequency. In the discussions in other sections, the impedance is usually that at one kilohertz, unless otherwise specified. The unit of impedance is the ohm. An impedance stated in rectangular coordinates is a complex number. In RTS™ Systems equipment the impedance is important over a band of frequencies and this band is normally stated in the specifications.

Inductance A property of a conductor or circuit that resists a change in current. Transformers, coils, chokes, wires, and printed circuits have inductance. Inductance is measured in henries. The symbol for inductance is L.

Insertion Loss A measure of the attenuation of a device by determining the output of a system before and after the device is inserted into the system.

Intercom 1. A means of organizational communications. The design of the intercoms systems produced by RTS™ Systems focuses on the concept of team communications. A team is an organization of members who perform individual tasks to accomplish a team goal or objective. The intercom is the pathway or means for the voice communications used to coordinate the team activity. 2. In larger systems, intercom refers to the matrix or point-to- point communications equipment, and interphone refers to the conference type equipment.

Intercom Data Groups and Port Number Calculation For data routing purposes, port numbers are arranged in groups of 8 sequential intercom ports. In an ADAM™ or ADAM™ CS Intercom System, each Audio I/O card comprises one data group. In a Zeus™ Intercom System, each group of 8 port connectors comprises a data group. Within each data group, each keypanel is uniquely identified by its address setting. Whenever you display the Panel ID, the intercom system determines which data group the keypanel is connected to, and also the address setting. It then reports the calculated address. For example: suppose a keypanel is connected to data group 3 and the keypanel address is set to 5. Since each data group consists of 8 sequential intercom ports, the calculated port number for this keypanel will be (2*8) + 5, or 21. This is the total of all intercom port numbers on the first 2 data groups, plus the offset of 5 ports into the third data group.

Interconnect A cable, device, or method of connecting one device to another, or one system to another.

Interface The place where two systems or a system and a subsystem meet. Also the device that adjusts levels and other parameters such that one system appears to the other system as a compatible extension.

Intermodulation Distortion See Distortion.

Inverse Square Law The decrease in level as a listener moves away from a loudspeaker, or a microphone is moved away from an acoustic source. The law says that the sound pressure will decrease six dB every time the distance is doubled. This law applies to the outdoors, and to the indoors where reverberation and room effects are negligible.

IR Drop Applies to the voltage drop along a wire as a function of the current (I) and the resistance of the wire (R). For example, the resistance of 10,000 feet of a number 22 gage pair is 320 ohms. The DC voltage drop at the end of the wire due to a user station using 50 milliamperes of current is 0.040 amperes times 320 ohms equals 12.8 volts. If the power supply is 32 volts and the drop is 12.8 volts, the voltage at the end of the wire is 19.2 volts. The minimum operating voltage for a user station operating in the high impedance mode is 18 volts. So a belt pack user station such as the BP317, or BP300 has enough DC voltage to work at the end of 10,000 feet of a 22 AWG wire pair.

Isolation The ability of a circuit or component to reject interference, usually expressed in dB.

ISO (Camera ISO) ISO is a means for a keypanel operator to isolate a particular intercom port for private communication. While the intercom port is isolated, it can only hear audio from the keypanel operator. ISO is frequently used in television broadcasting to temporarily isolate a member of a camera Party-Line. The isolated camera operator can then receive directions without interference from other audio traffic on the Party-Line. ISOs are setup using the intercom system configuration software. Each ISO can also be given a name which is meaningful to keypanel operators. Once an ISO has been set up and named, it can be assigned to any keypanel key (provided that ISO assignment has not been restricted or disabled in the intercom system configuration software). For further information about ISOs, search for “ISO” in AZ™EDIT help.

J

K

k (kilo) metric prefix symbol for 1000.

K (Kilobyte) prefix symbol for 1024 (common usage).

kilo A prefix meaning 1000.

L

L Symbol for inductance.

Lavaliere A small microphone. There are two types: 1) a very miniature type that clips onto clothing on the front of a performer below his head, and 2) a larger microphone on a cord worn around the neck of the performer with the microphone hanging below the neck on the chest.

Leakage The undesired leakage of a current or signal into another path.

Level The amplitude of power of a signal. If in decibels, the level has to be stated relative to a reference, and the reference has to be made clear.

Light Signaling See Signaling. Accomplished on the intercom line, using DC levels (Clear-Com®, HME, Theatre Visions) or a 20 kilohertz tone (RTS™ Systems, Telex® AudioCom®).

Limiter An effective communications system needs to limit dynamic range to ensure adequate intelligibility to the listener. The limiter/compressor in the TW system user stations has three functions: I) It helps loud talkers and soft talkers to be heard equally well, 2) It prevents a loud voice from being severely distorted, 3) It keeps the voltage levels from exceeding system limits. Function 3 is important because the user station must operate over a wide range of power voltages, and the limiter makes a practical system possible.

Line A single communication path.

Line Level Line Level depends on the system and the reference. It is often used to differentiate microphone level (-40 to -60 dBu) and a higher level (often 0 dBu).

Local Power Option Local Power Source is a small AC converter that converts AC line power to low voltage in order to power a user station --a separate connector is provided. User stations usually get DC from the converter, although occasionally low voltage AC power is used.

Loop-Through See “Daisy Chain”.

Loudspeaker A transducer that converts the electrical output of an amplifier to a audible sound.

M

mA Shorthand for milliamperes or thousandths of an ampere.

Main Station A user station where a user station and a system power supply are combined into one package.

Master Station A multichannel user station. There may be one or more of these stations in a system. Another definition is the primary station in a system.

Matrix “Matrix” is a term inherited from earlier point-to-point intercom systems, where all point-to-point communication was accomplished by closing specific switches in a switching matrix. Examples include the RTS™ CS9500, CS9600, and CS9700 Intercom Systems. In many instances, “Matrix” is used interchangeably with “Intercom System”. RTS™ ADAM™, ADAM™ CS, and Zeus™ Intercom Systems, on the other hand, do not use a switching matrix, but use a method called Time Division Multiplexing (TDM), in which communications are routed as digital packets. However, use of the term “matrix” persists since packet routing basically accomplishes the same thing as a conventional switching matrix: namely, connecting distinct talkers and listeners.

Maximum SPL The acoustic level above which operation changes from linear to nonlinear. This is a specification usually for microphones.

Mho The old unit of conductance or transconductance, now a siemen. The more familiar units of transconductance are amperes (output) per volts (input). For electronic devices, the units are usually millisiemens or milliamperes per volt.

Mic Short for Microphone

Micro (μ) Micro is a prefix meaning one millionth. For example a one microfarad capacitor has a capacitance of a millionth of a farad.

Microcontroller A Microprocessor that has a built in RAM (Random Access Memory), built in ROM (Read Only Memory), parallel type inputs / outputs, and often a serial input / output.

Microphone A transducer that converts sound into an electrical output or voltage.

Microprocessor The heart of a computer on a chip. Has inputs and outputs and can read RAM (Random Access Memory) and ROM (Read Only Memory). Used in the Model 802 to process stimuli such as button pushing and incoming tally signals and produce reactions such as blinking or steady lamp illumination, crosspoint closure, tally generation, relay control, or audible chime signal.

milli (m) A prefix that means one-thousandth.

Mixer An electronic device used to combine several signals inputs to a single output or to stereo outputs. Often other features are added to make it easy to achieve the basic goal.

Mix-Minus Bus / feed 1. In the studio, a mix-minus feed can be fed to a singer on stage. The mix- minus consists of a prerecorded orchestra. The performers microphone signal and the mix-minus feed are combined in another mixer output for the final air or recorded feed. This method is used for reasons of economy and to simplify production. 2. In ENG operations, a mix-minus feed is used for the IFB. The mix-minus allows the talent to hear the program audio that includes the voices of other talents at other venues, but not the talent’s own voice. The effect is to allow more normal conversations, on air, among the performers. The bus feed refers to the mixer mix-minus feed available to one or more IFB program inputs.

Monaural Containing one source of audio although the source may be a summation of two or more original sources.

Monitor An audio speaker used to supply program audio to the control room, audio mixer, and to others who need an acoustic audio feed. Some special monitors are used for musicians to hear their own instrument or voice. Usually a monitor is placed in the Green Room.

Mu (μ) Greek letter, mu, symbol for permeability (magnetic), amplification factor, prefix for micro (one-millionth).

Mu Metal Shield A highly effective magnetic shielding material.

Multiplexing A method of carrying more than one signal on a single “path.” Multiplexing may be by means of frequency, time division, and / or space. The TW system frequency multiplexes DC power, speech signals, and a 20 kilohertz call signal on a single pair of wires. The Models 848A / DC848 use time multiplexing in a digital RS485 signal to send data for all 24 stations down a single path.

mV Millivolt or one thousandth of a volt.

mW Milliwatt or one thousandth of a watt.

N

NAB National Association of Broadcasters.

NABET National Association of Broadcast Employees and Technicians.

NEC National Electrical Code

NEMA National Electrical Manufacturers Association

Nibble A nibble is half a byte or four bits.

Noise Usually an unwanted sound or signal that interferes with a sound or signal normally present in a system, device, or circuit. Sometimes a special noise source such as a pink noise source or a white noise source is used to test a system or acoustically test a room.

O

Ohm The electrical unit of resistance. One volt will maintain one ampere of current through one ohm.

Ohm's Law This law relates the electrical parameters of voltage, current, and resistance. The symbol for voltage is V*, current is I, and resistance is R. Voltage, V=I*R. Current, I=V/R. Resistance, R=V/I. *An older symbol or term for voltage is E. E is also the symbol for the Electric field, which has units of volts per meter.

Omega (Ω) Symbol for ohm.

Omnidirectional Microphone A microphone that picks up sound from all directions with the same amplitude.

Option Options are extra features available (for a price) on intercom and pro-audio equipment.

Output The useful signal (voltage, current, power) produced by a system, device, or circuit.

P

Paging Making a voice announcement over a sound system. The sound system is "P.A." in the sound contractor world, and "SA", Stage Announce, in the television / theater world.

Parallel Circuit or Connection In a circuit, the paralleled elements would be across the same voltage and the currents would divide amongst the elements. This kind of connection can apply to circuits, devices, or systems. For example: two RTS™ Systems TW Intercom Systems can be paralleled, by coupling with the appropriate capacitors, and switching each system power supply(ies) from 200 ohms to 400 ohms on the channels to be paralleled.

Party-Line (PL) A Party-Line (also called a conference line) is a group of intercom ports which can always talk and/or listen to each other. Party-Lines have default names PL01, PL02 etc. These names can be changed to more meaningful names using Other Alpha setup in AZ™EDIT. Members are assigned to a Party-Line using Party-Line setup in AZ™EDIT. Once a Party-Line has been set up, it can also be assigned to a keypanel key either from the configuration software or at a programmable keypanel. This allows the keypanel operator to talk and/or listen to the Party-Line without being a member. IMPORTANT: Do not confuse special lists and Party-Lines. A special list is used when a keypanel operator needs to occasionally talk or listen to a group of intercom ports that are otherwise unrelated. A Party-Line is typically used when several users of non-keypanel devices (such as belt packs or camera intercoms) are engaged in a specific common activity and they need to talk and/or listen to each other all the time. Keypanels are almost never members of Party-Lines (although they can be). However, a keypanel key can be assigned to occasionally talk or listen to a Party-Line if desired. Just remember: Party-Lines are primarily set up for Party-Line members, with occasional access by keypanel operators, while special lists are set up exclusively for keypanel operators to talk or listen to several unrelated intercom poRTS™.For specific information about Party-Line setup, search for "PL" or "Party-Line" in AZ™EDIT help.

Patch Bay / Patchboard A system of interconnecting audio signals. Consists of fixed connectors interconnected with flexible "patch" cords that are cords usually with a male connector on each end.

Peak The crest value of a voltage, current, or power.

Phantom Power There are three standard voltages: 12, 24, and 48, according to DIN 45 596. Voltage is applied to the circuit in a balanced fashion using a center tapped transformer or two resistors. The size of the resistors depend on the voltage.

Phase, Phase Shift Comparing the reference of one waveform to the reference point of another waveform. With periodic waveforms, the phase varies from 0 degrees (waveforms line up), to 180 degrees (waveforms opposite), to 360 degrees (waveforms line up but one is delayed by the time of one waveform).

Pi (π) Symbol for the quantity 3.14159...

Pickup Pattern Refers to the sensitivity of a microphone to acoustic audio signals originating from different spatial directions.

Pink Noise Equal noise energy per octave.

PL See Conference Intercom Systems.

PLL Phase Lock Loop. A tone decoder utilizing a phase lock loop is used in many of the TW System stations.

Point-to-Point (Matrix) Systems A point-to-point system allows two or more people to intercommunicate. But the conversations are limited to those selected by the originator of the call. This system normally includes a “tally” subsystem. The “tally” subsystem tells the called station where the originator is so that the called station operator can press a button to answer. Some systems automatically press the button and complete the return path. Most systems made by RTS™ Systems are full duplex (one can interrupt the speaker), and non- blocking (access to the channel is immediate and there is no busy signal). Conversations on point-to-point systems are in general, private. There are two kinds of point-to-point systems available. One is a distributed matrix system; the other is a central matrix system. The Models 848A and DC848 are modules in a distributed matrix system. The McCurdy Models 9500 and 9400 are examples of central matrix system.

Pop An undesired effect on a microphone output when a puff of air hits the microphone diaphragm. The effect sounds like a thump or pop. The effect is noticed with the following sounds: “p”, “b”, “t”.

Pop Filter Material placed between a sound source and a microphone that reduces the “pop” effects. It slightly affects microphone performance.

Port Ports are the individual channels that devices are connected to. Devices include: 2-way communication devices, such as keypanels, belt packs etc. Audio sources, such as broadcast feeds or background music. Miscellaneous audio output devices, such as powered loudspeakers, PA systems etc.

Port Gains RTS™ Keypanels are calibrated to send and receive audio at the standard operating levels of the intercom system. No audio gain adjustment is normally required when connecting these. However, many other types of devices may not operate at the standard intercom system levels. To assure signal level compatibility between the various types of audio devices connected to the intercom system, there are separate analog input and output gain adjustments for each intercom port. It is also possible to adjust the listen gain for any specific intercom port when listening to any other specific intercom port. This is called the point-to-point listen gain, or crosspoint gain. For example, a keypanel operator might want to monitor a music source connected at some intercom port, but at a reduced audio level so that it does not interfere with normal intercom communications. The crosspoint gain can be reduced for the keypanel port listening to the port where the music source is connected.Analog gain adjustment is only available using AZ™EDIT. Crosspoint gains can be adjusted either within AZ™EDIT or from a programmable keypanel. For further information on any gain adjustment in AZ™EDIT, search for keyword “gain” in AZ™EDIT help. For procedures to adjust gain from a programmable keypanel, look for “gain” in the manual index.

Port ID Numbers and Alphas Intercom ports have identification numbers 001, 002 etc. These numbers cannot be changed, but may not be commonly known to intercom system users. Each intercom port also has a default name, called an “alpha”, because this name appears in the alphanumeric displays on keypanels when you assign the ports to keys for talking and listening. The default alpha names are N001, N002 etc. These default alpha names can be changed to ones that are meaningful to keypanel operators using Port Alpha setup In AZ™EDIT. (Click the "Port Alpha" button in AZ™EDIT, then press F1 on the computer keyboard if you need help.)

Postproduction Production activity that occurs after the actual production phase. For example the editing of a television or motion picture production.

Postproduction Editing The process of making decisions and actually manipulating the media (film or tape) to change action sequences, delete, insert, and modify images and sound.

Pot (Potentiometer) A device to electrically change audio or video levels. Potting up means increasing a level from a control panel. An audio mixing console is an audio control panel.

Power Amplifier An amplifier used for driving lower impedance (8 to 500 ohms) headphones or speakers (2 to 45 ohms).

Power Supply 1) The source of electrical power ("power outlet"). In North America this source is generally 120 volts AC, 60 hertz. In Japan the source is generally 100 volts, 50 or 60 hertz. In Britain the source is 240 volts, 50 hertz. In Europe the power is usually 220 volts, 50 hertz. There are exceptions in every location, and there are still isolated odd systems throughout the world. RTS™ Systems equipment has been designed to operate at these various voltages. In addition, some equipment is operable off of DC sources such as batteries, automobile 12 volt power, aircraft 28 volts, and aircraft 120 volts, 400 hertz. 2) A unit used for converting power outlet power to DC power.

Power Supply, TW A special power supply to run user stations on the RTS™ Systems TW system. This supply provides low noise DC power (nominally 32 VDC) and an audio impedance of 200 or 400 ohms. This impedance extends from 100 hertz to 20000 hertz.

Power Ratio See decibel.

Preamplifier An amplifier usually used to raise the small signal from a microphone to a "line level" sized signal.

Presence Peak A rise in the response of a microphone in the range of 2000 to 10, 000 hertz. In circuits, a deliberate alteration of the frequency response in the range of 1000 to 10,000 hertz. RTS™ Systems Model 802 has a small presence boost in the speaker amplifier change in the 1000 to 2000 hertz range.

Pressure Zone Microphone (PZM) Used to pick up audiences or groups. A microphone with a reflecting surface such that the sound waves arrive in phase at the microphone element, providing good frequency response. Also used for orchestral pickup.

Program, Program Audio In television, the audio signal that is being sent out with the picture to be broadcast.

Push- To- Talk (PTT) Usually used on handsets or push-to-talk microphones. Pushing the button enables the microphone and often also enables an electronic switch in an intercom station. The electronic switch prevents amplifier and cable pickup from going on the intercom line as undesired noise.

Q

R

R Abbreviation for resistance, and the symbol for a resistor.

Rack Unit(s) (RU) A standard unit of measure used when dealing with electronic equipment racks. 1 RU = 1.75" (44.45 mm). For example: a particular piece of equipment is described as being 3 RU in height. This means that it is 5.25" (3 x 1.75") in height. Detailed information on the specification of standard electronic equipment racks can be found in EIA RS-310-D (See the references section).

Reactance A property of an inductor or capacitor that is frequency dependent. Capacitive Reactance is opposite to Inductive Reactance. Inductive reactance increases with frequency. Capacitive reactance decreases with frequency. See Capacitance and Inductance.

Relay Relay is used interchangeable with GPI output. The relay feature works with the 16 GPI outputs of an optional UIO-256 Universal Input / Output Frame, and with the relay outputs of an FR9528

Relay Frame. The relay feature also works with the 8 GPI outputs of an ADAM™, ADAM™ CS, or Zeus™ intercom system (J27 on a Zeus™ Frame, J903 on an ADAM™ CS Frame, and J11 on the XCP-ADAM™-MC Master Controller Breakout Panel in an ADAM™ Intercom System). You can assign a keypanel key to control a GPI output from any of these devices, and then use that key and output to control an external device. For example: you could use a keypanel key to control lighting. Or, you could assign a relay as a level 2 talk key assignment in a stacked talk key arrangement to both send audio and key a device, such as a paging amplifier or a 2-way radio.

Remote Station A user station located at a distance from the master station.

Remote Truck, Remote Unit A mobile television studio. Supports television productions with equipment and production personnel operating positions. Carries cameras, CCUs, switcher, monitors, audio console and ancillary equipment, VTRs, and last, but not least, intercom systems.

Resistance In DC circuits, the opposition to current, in AC circuits, the real part of the opposition to current.

Resonance A condition where an applied signal's frequency coincides with a natural response frequency of a circuit, device, or system. There must be reactances in the circuit for a resonance, a pure resistance doesn't resonate.

Retractile A cord whose jacket is treated and formed to retract as a spring.

RF Radio Frequency. Frequencies generally ranging from 15 kilohertz to 150 gigahertz. The electrical energy at these frequencies is often converted to electromagnetic waves that are propagated through space. These waves are the basis of the wireless intercom, radio broadcasting, television broadcasting, microwave ovens, industrial processes. Frequency Name of Band Examples: 3 to 30 kilohertz Very Low Frequencies (VLF) Underwater Communications 30 to 300 kilohertz Low Frequency (LF) Navigation 300 to 300 kilohertz Medium Frequency (MF) AM Radio Broadcasting 3 to 30 megahertz High Frequency (HP) Short Wave Radio, Long Range Terrestrial Communications 30 to 300 megahertz Very High Frequency (VHF) TV, PM Broadcast, Fixed I Mobile Communications, Walkie Talkie, Wireless IFB, Wireless Intercom, Airborne Communications 300 to 300 megahertz Ultra High Frequency (UHF) TV Broadcast, Mobile Communications, Walkie Talkie, 3 to 30 gigahertz Super High Frequency (SHF) Satellite Uplink/Downlink 30 to 300 gigahertz Extremely High Frequency (EHF) Satellite Uplink/ Downlink kilohertz = 1000 hertz megahertz = 1000000 hertz gigahertz = 1000000000 hertz

RFI Radio Frequency Interference. This interference may originate from AM and FM radio stations, television stations, light dimmers, electric motors, intermittent incandescent or fluorescent lamps, doorbells, et cetera. It results in either direct demodulation into audio circuits or position sensitive effects.

RMS The abbreviation for root-mean-square. The effective value of an alternating current waveform. The root-mean-square current, power, or voltage is as follows: Square the amplitude, so that the positive and negative halves of a waveform are the same polarity. Then the value is averaged over time. Finally the square root of the average or mean is taken. The RMS value of a one volt peak sine wave is 0.707

Roll-off The frequency at which the response of a filter, circuit, network, device, or system changes from its center value by 3 dB.

RU See Rack Unit(s).

S

SA or S.A. Stage Announce. A public address system originating in a control room and ending up on the stage. The SA may also have the functionality of an IFB. This allows the director to interrupt dance music and address the dancers. The SA function is a standard function on the Series 4000 IFB system, the Model 801 / Model 860 system, and the Model 802/ Model 862 system. The SA function is available as a special option on almost any RTS™ Systems user station. In the option case, the SA does not necessarily include the IFB function. An IFB option has to ordered to assure IFB functionality.

Semiconductor A material whose conductivity falls between a conductor and an insulator. Semiconductors include the elements carbon, silicon, and germanium in crystal form. Other semiconductors include compounds. Some examples are semiconductor diodes, transistors, integrated circuits, transector overvoltage devices, thyristors, and carbonized substances.

Sensitivity 1. The electrical output of a microphone for a given SPL input. Open circuit sensitivity is the output voltage a microphone produces (in dB relative to 1 volt) into an open-circuit load, at a sound pressure of 74 dB SPL (1 microbar or 0.1 pascal). In practice, this is hard to do because of ambient noise, so readings are taken at 94 dB and the calculation adjusted accordingly. 2. Defined as the on-axis sound pressure level at a distance 1 meter in front of a speaker driven by 1 watt continuous average input power of a specified waveform. 3. Defined for headsets to indicate the SPL generated for a given set of conditions. The Beyer DT 108 headphone has a sensitivity of 94 dB SPL produced for one milliwatt input. The Telex® PH10 headphone has a sensitivity of 105 dB SPL produced for one milliwatt input. A Telex® EMV -2 announcer earset has a sensitivity of 120 dB.

Series Circuit An arrangement of circuit elements, circuits, devices or systems such that the components are arranged and connected end to end to form a single current path.

Shield An electrostatic shield prevents crosstalk from one conductor to another by blocking the electric field. For example: by shielding wires in a cable, crosstalk can be dramatically reduced. A magnetic shield prevents electromagnetic coupling of undesired signals from one transformer to another. For example: the magnetic field from a power transformer to an audio output transformer can be attenuated by mu metal shielding.

Shotgun Microphone A highly directional microphone for long distance pickup. Looks like a shotgun.

Sidetone In the truest sense, sidetone is a small amount of microphone signal that is fed back to the earphone of the individual speaking into the microphone. In RTS™ TW user stations, the null balance control is sometimes used to adjust the amount of sidetone the user hears. This control is sometimes (technically erroneously) called the sidetone control. Other RTS™ TW equipment have both null balance adjustments and a true sidetone adjustment (Models 802, 848 for example).

Signal The name applied to visual, audible, or electrical energy that carries information.

Signal to Noise Ratio (S/N or [S+N]/N) In a given circuit, device, or system, the ratio of the Signal plus Noise to Noise. The noise spoken of is the residual noise in the system when no signal is present. The signal is a signal at a reference level representing typical operating conditions of the system. This ratio is usually specified across a frequency band. The S/N ratio of a microphone is specified at a given SPL ratio. For example, a S/N ratio of 60 dB at 94 dB SPL is considered good, 65 dB, very good, 70 dB is excellent.

Sigma Symbol for summation, also used to indicate a summing amplifier or summing function.

Signaling Signaling in these intercoms has several meanings. In the TW system, signaling is accomplished by a blinking light initiated from any Call Light equipped station on that channel. Signaling is often used to get attention such as getting someone to put their headset back on, or getting a sound mixer person to turn down the monitor speaker and talk on the intercom. Signaling can also be used as a visual cue. The blinking light can be used for a cue, but theatre productions often prefer a steady light. The light coming on is a “Standby” signal. The light going from on to off is an “Execute” signal.

Simplex Simplex communication is one person at a time. There is only one communication channel available and it is unidirectional at a time, usually in the direction determined by the call initiator. An example of this is a CB or Citizens Band radio.

Single-ended Unbalanced, using circuit common or “ground” as a return lead.

Single Channel / Two Channel In the RTS™ Systems TW Intercom system, most user stations are two channel. Other channel numbers available (some optional some standard) are 1, 3, 4, 5, 6, 10, 12, 24. The term channels usually is applied to conference style user stations. The TW system can carry two channels on a standard microphone cable. Other systems carry one balanced channel (Telex® AudioCom®), or one unbalanced channel (Clear-Com®, Telex® AudioCom® in unbalanced mode, HME, Theatre Vision).

SMPTE Society of Motion Picture and Television Engineers. An organization that (among other things) pioneers standards used in the television industry.

SMPTE/EBU Time Code Recorded on videotape or audiotape. Provides a time address for each video frame in hours, minutes, seconds, and frame numbers. Requires a SMPTE code generator to create. Some RTS™ Systems audio equipment can be used to distribute SMPTE time code. Consult RTS™ Systems Engineering Department for details.

Sound Variations in pressure (usually air) caused by vibrating bodies. Examples are air columns (pipe organs), strings (violins), vocal cords and larynxes (humans).

Sound System A combination of transducers, amplifiers, and interconnections. A simple sound system could consist of a microphone or pickup and an amplifier and a headphone or speaker.

Special List A special list is a means for a keypanel operator to talk and/or listen to several unrelated destinations using a single key. Special lists are useful for group call or zone paging. Special list members are defined in the intercom configuration software. Once a special list has been configured, it can be assigned to a keypanel key. A special list is a group of intercom ports that a keypanel operator can talk or listen to by activating a single key. Special lists are typically used for paging, all call, group call etc. Special lists have default names SL01, SL02 etc. These names can be changed using Other Alpha setup. You define the members of the special list using Special List setup. Once a special list has been set up, you typically assign it to a keypanel key using Keypanel setup. The keypanel operator can then activate the special list key to talk or listen to all members of the special list. IMPORTANT: Do not confuse special lists and Party-Lines. A special list is used when a keypanel operator needs to occasionally talk or listen to a group of intercom ports that are otherwise unrelated. A Party-Line is typically used when several users of non-keypanel devices (such as belt packs or camera intercoms) are engaged in a specific common activity and they need to talk and/or listen to each other all the time. Keypanels are almost never members of Party-Lines (although they can be). However, a keypanel key can be assigned to occasionally talk or listen to a Party-Line if desired. Just remember: Party-Lines are primarily set up for Party-Line members, with occasional access by keypanel operators, while special lists are set up exclusively for keypanel operators to talk or listen to several unrelated intercom ports. For specific information about special list setup, search for "special list" in AZ™EDIT help.

SPL Sound Pressure Level. Sound is alternating pressure waves. Sound Pressures (amplitudes) are measured in pascals. The amplitudes can be converted into decibels using an equation. These new numbers are Sound Pressure Levels.

Squawk (Versus Matrix) A term to differentiate two kinds of point-to-point intercom. A squawk type intercom allows instantaneous momentary communication. The Model 810 in the "squawk" configuration (momentary buttons only) is a pure squawk system. The Model 810 is available in a "matrix" configuration (alternate action buttons, electrically the same).

Stacked Key See the descriptions for talk level, talk level 2.

Stereo Recording, transmitting or reproducing a sound source with two or more separate sound pickups. The recording has two or more separate channels.

Studio Camera Used both in studios and remote locations. A heavier, larger camera mounted on a pedestal or other suitable mount. Generally has better quality pictures than a lighter, smaller field camera.

Studio Talkback A loudspeaker system from the control room to the studio. Also called SA (Stage Announce, or Studio Announce).

Supra Aural Headset A headset where the earpieces rest on the ear.

Supercardioid A microphone pattern with a maximum rejection at 125 degrees off axis. A compromise design between the cardioid and the hypercardioid.

Switcher (usually the Technical Director) Person who operates the switcher to switch the video; the switching device itself.

System 1. An assemblage or combination of things or parts forming a complex or unitary whole. 2. The structure or organization of society, business, or politics. 3. The interrelationship of a collection of interdependent elements and processes.

T

Talent Collective name for all performers and actors who appear regularly on television.

Talk Level 1 Talk level 1 is the normal talk key assignment. This is the assignment that normally appears in the alphanumeric display (on keypanels so equipped). You may add a talk level 2 assignment to activate a second device along with talk level 1.

Talk Level 2 Talk level 2 is used with stacked talk keys. A stacked talk key activates two types of communication at once. For example: a stacked talk key could simultaneously activate audio output to a transmitter and key the transmitter using a relay. The audio output is called the level l assignment and the relay is called the level 2 assignment.

Tally An intercom's tally usually identifies a calling party to a called party, tally can also indicate a call waiting.

Telco Abbreviation for Telephone Company. Refers to communication lines owned and operated as part of a standard telephone system.

Tone Signaling An audio tone distributed on the intercom line (may be associated with light signaling). This tone is often used for alerting an operator.

Transducer In sound, a device that converts acoustic energy to electrical (e.g.: microphone) or vice versa (e.g.: loudspeaker).

Transformer Audio transformer: a device that can isolate two circuits, and, in addition, match impedances, step up or down voltages or currents. A microphone transformer can be used to optimize the signal to noise ratio of a microphone preamplifier combination. Power transformer: a device that isolates electronic circuitry from a direct connection with the power line, and provides power at a convenient voltage for the electronic equipment (usually rectified and filtered first).

Transient In acoustics, a sudden change such as that from a percussive instrument (drum or piano). In electronics, a sudden change such as a stepped signal or spiked signal.

Transient Intermodulation Distortion See Distortion.

Transient Response Equipment needs to be designed to be damage proof from power line transients. Good audio equipment needs to handle well signal transients that occur normally. Transient handling may require headrooms of 20 to 60 decibels. The headphone amplifiers used in RTS™ Systems intercoms generally have good transient handling capability.

Trunking Trunking is a method of interconnecting two or more independent intercom systems. The connection is accomplished by reserving one or more audio ports in each of the intercom systems for use as audio links between the systems. A special device, called a Trunking Master Controller, is required to control access and usage for the trunked intercom ports. A configuration utility, called CStrunk, is used to set up the Trunking Master Controller.

Two-Wire A communications system where the path is the same for both talk and listen. In electrical pathways there are, in fact, two wires (one path). Two-wire systems can be two- wire balanced or two-wire unbalanced.

Two-Wire Balanced Two-wire balanced is similar to two wire unbalanced except that neither conductor is tied to circuit common. Circuit common is either tied to a transformer center tap, or is an electrical center point, or not tied at all.

Two-Wire Unbalanced A two-wire system that uses a circuit common and one additional conductor for a pathway. This system allows easy addition of DC power as well. Although a balanced two wire system has less sensitivity to outside electrical interference.

Unbalanced Line A communications line that uses circuit common or "ground" as a return path. In the case of intercom systems, in general, an unbalanced line is more susceptible to noise interference. But practically speaking, a low impedance unbalanced line (200 ohms) works well in the real world, and has the added advantage of being able to operate on just two wires, carrying the DC power, the full duplex audio signal, the 20.000 kilohertz call light signal, and the 24.000 kilohertz microphone reset signal. Many installations have been previously wired for telephone operation and the two wire intercom works on these pairs. A special "golf' system application of the TW System has the advantages of the TW system and balanced line operation. A standard TW power supply operates a string of belt pack or other user stations in an isolated, balanced mode. The balanced system is transformer coupled into the regular unbalanced system.

V

V The symbol for volt.

VA The symbol for volt-ampere. A volt-ampere describes the demand on the power line without regard to power factor or true power. This figure is more helpful in determining the maximum load on a circuit, that has a given ampere rating.

Voltage The term for electrical potential of electromotive force.

Voltage Drop See IR drop.

VU Volume Units.

VU Meter Used to show the relative levels of signals. A change of one VU is a change of one dB. VU meters have their 0 VU point referenced to different levels. This level may be 0 dBm, +4 dBm (Recording Industry), +8 dBm (Broadcast Industry). In addition, correct calibration is further involved. The VU Meter has certain characteristics tailored for monitoring sound. Generally speaking the audio peaks are 10 dB above the indications shown on a standard VU meter due to the lag of the meter. Some VU meters are combined with peak reading meters. Levels stated in “VU”s generally apply to program material. Readings in dBm generally apply to a steady state sine-wave.

W

W The symbol for the unit of power, the watt.

Watt A unit of electrical power. One watt is the power delivered by one volt at a current of one ampere in a DC circuit, and one volt at one ampere with a phase angle of 0 degrees in an AC circuit.

Wave Form A graphical representation of a varying quantity. The x (horizontal) axis usually represents time and the y (vertical) axis usually represents amplitude, energy, or power.

White Noise Equal noise energy per hertz.

Windscreen A microphone cover designed to reduce extraneous noises caused by gusts of wind. Also useful for reducing pop from speech plosives such as “b”, “p”, and “t”.

Wet Line An intercom or telephone line that carries both audio and DC voltage / current. As opposed to a dry line that carries only the audio.

X

Y

Z

INDEX

E

F

G

H

I

R

S

T