

Atlas  
**AVM**

**Installer's Manual**

Key System US  
West Palm Beach, Florida

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# *Overview of Installation and Integration*

## About this Manual

The purpose of this manual is to guide you through the installation and integration of your Atlas AVM system. This manual is intended for use by certified Atlas AVM installers who are proficient in DOS, telephony, and voice processing.

The chapters in this manual are as follows:

- **Chapter 1** gives a brief description of each chapter, provides a summary of installation, and explains the conventions used in this manual.
- **Chapter 2** is a quick guide to installation for experienced installers.
- **Chapter 3** describes evaluating the customer's voice processing needs and capabilities.
- **Chapter 4** offers an overview of the Atlas AVM installation and integration process.
- **Chapter 5** tells how to plan the basic design of an Atlas AVM system.
- **Chapter 6** describes the functions of voice boards in your Atlas AVM system and tells which models of the Dialogic and Rhetorex boards can be used.
- **Chapter 7** gives general guidelines and recommendations for building an Atlas AVM system.
- **Chapter 8** gives suggestions for connecting Atlas AVM to your telephone system.
- **Chapter 9** tells how to program the Atlas AVM integration screens.
- **Chapter 10** explains how to add fax capability in your Atlas AVM system.

- **Chapter 11** covers the system administration issues that need to be addressed as part of the installation process.
- **Chapter 12** discusses the final steps required to make the system operational.
- **Glossary**
- **Index**

## Conventions

This manual uses the following conventions:

CONVENTION		KEY INFORMATION
lowercase x		indicates a variable in a file, version, etc.
ALL CAPS		indicates a file or directory
< <i>italics</i> >		indicates a variable or place holder
<b>Boxes→Renumber</b>		example of a 'path' that a user can follow to get to a particular screen or field.
< >		indicates a keyboard command entry such as < Enter >
<b>Lowercase bold</b>		used for commands or data to be typed at the keyboard
	<b>Note:</b>	Information that is vital to the success of a process
	<b>Tip:</b>	A shortcut or helpful hint
	<b>Caution:</b>	Possible damage to equipment

## Evaluating the Customer

Before installing an Atlas AVM system, you need to gather information about the prospective customer. This will help you assess what that customer needs to build a voicemail system.

1. Collect pertinent information about the company.

2. Decide which telephony features and applications the customer desires such as:
  - Voicemail
  - Auto attendant
  - Audiotext
  - AMIS-Analog Networking
  - Fax
3. Determine whether the customer can or does meet the hardware and software requirements. (*See* Chapter 9, “Building the System.”)
4. Use the Customer Engineering Form in Chapter 3, “Evaluating the Customer,” as you compile information about the customer.

## Gathering Integration Information

In an Atlas AVM system, integration means enabling communication between Atlas AVM and a specific telephone system.

Telephone systems differ, and Atlas AVM must be programmed to communicate with the customer’s specific telephone system. To accomplish this you need to gather information about the customer’s telephone system.

1. Collect the telephone system information suggested on the Integration Form found in Chapter 4, “Integration Investigation.” This includes:
  - Manufacturer, model, and software version
  - Name of the person who maintains and services the telephone system
  - Necessary hardware for voice messaging
  - Information about telephone system operations
  - Data packet information

Resources for doing this include the *Atlas Telephone System Compatibility Listing*, the customer’s interconnect, the telephone system manufacturer, and the telephone system documentation.

2. Gather any other needed information through on-site testing.
3. If you have affirmed that Atlas AVM can integrate acceptably with the customer’s telephone system, save the information you have compiled for later use in the installation/integration process.

## Designing the Basic System

### Planning the Design

You will need to draw up a basic design from the information you have obtained from the customer. This will lay the groundwork for the system and help you determine sizing issues (*see* “Sizing the System” below).

With the customer, you can help make decisions about how the overall system will work. The following steps will help you do this.

1. Design how the Auto Attendant will handle calls. This includes:
  - Designing greetings which can be set for different times (business hours vs. non-business hours), days and ports
  - Designing the options available for transferring from the Auto Attendant
2. Plan audiotext boxes in addition to the Auto Attendant. This includes:
  - Determining the total number of audiotext boxes
  - Planning audio menus
  - Mapping the levels of audiotext boxes
3. Determine the number of voicemail, etc. boxes needed.



#### Note

Customizing and programming boxes is explained in the *Atlas AVM System Administrator's*

### Sizing the System

After you have calculated the number of audiotext and voicemail boxes required for an Atlas AVM system, you also need to determine the number of voice board ports and the amount of disk storage required.

For worksheets and information regarding sizing, *see* Chapter 5, “Designing the Atlas AVM System.”

## Building the System

Keep in mind the following as you build your Atlas AVM system:

1. Software requirements
  - Atlas AVM installation disks
  - DOS 5.0 or later, 6.2 recommended
  - Fax Driver Disk (optional—available from Key System US)
  - CO/Session 7.0 or later

2. Hardware requirements
  - Minimum hardware specifications (memory for CPU and ports)
  - Suggested hardware configuration
  - Recommended hard drive size
  - Voice boards to handle number of ports required for the system

For information on system requirements, *see* Chapter 9, “Building the System.”

## Installing Voice Boards

You need to consider the following:

1. Familiarize yourself with the voice boards Atlas AVM supports. You can help the customer decide which voice boards to buy based on the number of ports required, the number of available slots, and the cost of different boards.
2. Prior to installing voice boards, determine the following for your system (Atlas AVM will suggest defaults during Atlas AVM installation):
  - I/O port address
  - Hardware interrupt
  - Base memory
3. Use the Rhetorex utility SHOWJUMP or the Dialogic voice board documentation to determine the jumper settings for the I/O port address before installing the voice boards. You can do this during or after installation of the Atlas AVM software.
4. Finally, physically connect the voice board(s) to the telephone system using a telephone line. Determine the correct jack connection for the voice board and make sure it matches the telephone jack connections on the wall.

For information on setting up and installing voice boards, *see* Chapters 6, 7, and 8.

## Installing Atlas AVM

The following steps are required for installing Atlas AVM:

2. Make sure DOS is running.

3. Install the voice board(s).
4. Attach the hardware lock.
5. Install the Atlas AVM disks.
6. Install and configure CO/Session (optional).
7. Create a VXNIGHT.BAT file (optional).
8. Reboot the system.

For information on installing the Atlas AVM software, *see* Chapter 10.

## **Connecting Atlas AVM to a Telephone System**

Determine the type of integration desired and what will be needed to complete the connection.

For information on Connecting Atlas AVM to a Telephone System, *see* Chapter 11.

## **Integrating with the Telephone System**

This process involves taking the information you have collected and inputting it into the Atlas AVM installation screens.

1. Refer to the Integration Form you filled out earlier as you program the Atlas AVM integration screens and parameters.
2. Plan to test the integration after you have completed “System Administration Procedures” below.

For instructions on programming the Atlas AVM integration screens, *see* Chapter 12.

## **Installing and Configuring Fax (Optional)**

Fax capability is an add-on feature that must be purchased separately.

Install and configure a GammaFax MLCP-4/AEB board, and install and configure the fax drivers provided by Key System US.

For more information, *see* Chapter 13, “Fax Installation.”

## **System Administration Considerations**

Now that Atlas AVM is installed and integrated with the telephone system, you need to instruct or assist the system administrator in the following areas:

1. Configuring system boxes—voicemail, audiotext, etc.
2. Recording the audio portion of company greetings and other audiotext boxes.
3. Testing the system.
  - System design
  - Integration with the telephone system(For detailed administration information, see the *Atlas AVM System Administrator's Manual*.)
4. Training users on how to use the Atlas AVM telephone interface (can be done by the installer or the system administrator).

The *Atlas AVM Pocket Reference Guide* provides information on using the telephone interface. Atlas AVM also provides a tutorial over the telephone when box owners first set up their voicemail boxes.

## Finalizing the Installation

Now that you have Atlas AVM installed, configured, and tested, you can take the final steps necessary to make the system operational. These steps, called the final cut over, include:

1. Connecting single line extensions to Atlas AVM.
2. Programming Atlas AVM lines at the telephone switch.
3. Organizing lines going to Atlas AVM into a hunt group.
4. Forwarding trunk calls (outside calls) to Atlas AVM.
5. Programming telephone handsets.

Some of these steps may require the expertise of the customer's telephone interconnect.

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## ***Quick Installation Guide***

This chapter is an abbreviated version of Chapter 10, “Installing Atlas AVM.” For more detailed installation information, *see* Chapter 10.

### **What is Atlas AVM?**

Atlas AVM is a DOS-based voice processing system that provides a suite of telephony features, as well as the means of integrating with a telephone switch.

### **Installation Procedure**

Briefly, installing and setting up Atlas AVM involves the following:

1. Installing and setting up your voice boards.
2. Installing and configuring CO/Session to allow remote maintenance.
3. Installing the Atlas AVM disks and configuring your voice boards.
4. (optional) Creating a VXNIGHT.BAT file, a custom batch file Atlas AVM runs daily at 4:00 AM.
5. Integrating with the specific telephone system, if necessary, through the Atlas AVM installation screens.
6. Using the *Atlas AVM System Administrator's Manual* to set up Atlas AVM system parameters.

## Installation Steps

### Step 1: Equipment Inventory

Verify that you have all the needed equipment and software to complete a total installation of the Atlas AVM voice processing system:

- A computer equipped with the hardware listed in Step 2 below
- DOS (version 5.0 minimum, 6.2 or later preferred)
- Atlas AVM installation package:
  - Installation disks
  - Hardware lock
  - *Atlas AVM System Administrator's Manual*
  - *Atlas AVM Installer's Manual*
  - *Atlas AVM User Guide* pamphlets (25 copies)
- Dialogic or Rhetorex voice board(s)
- Miscellaneous telephone equipment for connecting to the telephone system.

### Step 2: Verify Hardware Requirements

Certain hardware specifications and configurations are required before a PC can be utilized for Atlas AVM voice processing.

Component	4-12 Port System	16-20 Port System	24 Port System
<b>CPU</b>	386, 486, or Pentium	486DX to Pentium	486DX4 to Pentium
<b>RAM</b>	4 MB	8 MB	12 MB
<b>Bus</b>	ISA	ISA	ISA
<b>Speed</b>	25-66 MHz or higher	33-66 MHz or higher	66 MHz or higher
<b>Hard drive</b>	IDE or SCSI	IDE or SCSI	IDE or SCSI
<b>BIOS</b>	AMIBIOS	AMIBIOS	AMIBIOS
<b>Power supply</b>	200 watt UL approved	250 watt UL approved	300 watt UL approved

**Figure 2 -1: Minimum Hardware Specifications**

### Step 3: Voice Board Setup and Installation

Install your voice boards. Information on installing voice boards is provided in Chapters 7 and 8.

#### **Step 4: Attach the Hardware Lock**

Attach the hardware lock you received with your installation disks to LPT1.

#### **Step 5: Have DOS running**

You need DOS version 5.0 or later.

#### **Step 6: Load the Atlas AVM Disks**

In this step you will load the Atlas AVM software, load the voice board drivers, install QEMM, and configure your voice board.

The following disks are included in your Atlas AVM installation package:

- Engine Disk 1
- Engine Disk 2
- Switch Integration Disk
- Rhetorex Driver Disk
- Dialogic Driver Disk 1
- Dialogic Driver Disk 2
- System Prompts Disk 1
- System Prompts Disk 2
- System Prompts Disk 3
- QEMM Disk 1
- QEMM Disk 2
- Fax Driver Disk (if you have purchased the optional fax capability)

To begin installing, insert Engine Disk 1. Use **a: install**. Follow the prompts. For a complete guide to all the installation screens, *see* Chapter 10, "Installing Atlas AVM."

#### **Step 7: Create VXNIGHT.BAT (Optional)**

Daily at 4:00 A.M. Atlas AVM offers the option of running a batch file of your own making. For Atlas AVM to run this file, it must be named VXNIGHT.BAT and must be located in the Atlas directory.

You can use your custom VXNIGHT.BAT file to perform automated tasks related to your Atlas AVM system. For example, the following would back up and then reboot/purge your Atlas AVM system:

```
xcopy c:\Atlas\*.* /e/s/v g:\  
c:\coldboot.com
```

### **Step 8: Install CO/Session**

CO/Session allows remote maintenance of Atlas AVM, and also will allow Key System US technical support personnel to access your Atlas AVM system if you need assistance.

For more information, *see* Step 6, “Setting Up CO/Session,” in Chapter 10.

### **Step 9: Install Fax Capability (Optional)**

You can purchase optional fax capability for your Atlas AVM system. For more information, *see* Chapter 13, “Fax Installation.”

### **Step 10: Reboot the System to Verify all Settings**

Reboot your system and look for error messages.

Your Atlas AVM system is now running on system defaults. If your system is working with the default telephone system settings, refer to the *Atlas AVM System Administrator's Manual* for information on port setup and programming system boxes (audiotext, voicemail, etc.).

If your system is not working with the defaults, *see* Chapter 12, “Integration with the Telephone System,” for information on interfacing with your specific telephone system.



## *Evaluating the Customer*

When you meet with customers who are interested in Atlas AVM, determine what their companies need to build a voicemail system.

Evaluating customers' situations and determining their needs will help you guide them in developing an Atlas AVM system. The final decisions should be based on considerations such as:

- Type of telephone system
- Telephony needs of the customer
- Cost of implementing everything required

Some customers may have telephone systems that do not have voicemail capability, especially if they have older, smaller systems. If that is the case, the customer will need to upgrade the telephone system in order to have voicemail. Atlas AVM requires only basic voicemail capabilities from the telephone system.

## **Gathering Information**

Begin the evaluation process by collecting information about the prospective customer and recording it on a form such as the Customer Engineering Form given in this chapter. Use this as the basis for building a file on the customer.

## **The Customer Engineering Form**

Some of the information you need to gather relates to your role as a salesperson; other parts relate to your role as an Atlas AVM installer.

Use the descriptions below as you fill out the Customer Engineering form.

## **Company Information**

Record the company name and address, the name of your contact person at the company, and also the person who will make the final decisions about purchasing an Atlas AVM system.

## **Site Information**

Determine the approximate number of people who will be using voicemail and make an estimate of how many boxes this system will need. This will help you get a rough estimate of the size and cost of the Atlas AVM system.

Find out how many telephone lines there are from the central office to the company so you will know if additional lines will be required.

## **Telephone System Information**

Collect information about the telephone system, including the manufacturer, model number, and what software version is running. You may need to get some of this information from the company's interconnect (telephone maintenance person).

Make a record of the name, address, and telephone number of the company's interconnect. The interconnect's knowledge about the telephone system can benefit you throughout the sales and installation process.

## **Telephony Applications/Features Desired**

Find out if the company wants its voicemail system to have an Auto Attendant, voicemail, audiotext boxes, fax boxes or a combination of these features.

## Customer Engineering Form

<b>Company Information</b>		
1. Company name		
2. Company information	Address	
	Phone	
	Fax	
3. Contact person	Name	
	Phone	
4. Final decision-maker	Name	
	Phone	
<b>Site Information</b>		
5. Number of employees who will use the Atlas AVM system		
6. Estimated number of boxes		
Voicemail boxes (boxes for voicemail users)		
Auto Attendant boxes (boxes for company greetings)		
Audiotext boxes (boxes providing information)		
Estimated total number of boxes		
7. Number of lines from central office		
<b>Telephone system Information</b>		
8. Telephone system manufacturer		
9. Telephone system model		
10. Telephone system software version		
11. Interconnect company	Name	
	Address	
	Telephone	
<b>Telephony Applications/ Features Desired</b>		
12. Voicemail		Yes / No
13. Auto Attendant		Yes / No
14. Audiotext		Yes / No

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## *Integration Investigation*

During the integration investigation, you will determine if Atlas AVM can be integrated with the customer's telephone system and what voicemail features will be available.

### **Integration Requirements**

Atlas AVM requires the following basic features from a telephone system in order to have voicemail:

- Necessary software version for voicemail  
The telephone system software may need to be upgraded if it is not a voicemail capable version.
- Single-line extensions  
The voice boards in Atlas AVM are analog boards and need single-line analog extensions. The user's telephone does not need this type of extension, only the lines going into Atlas AVM.
- Ringing capability on the single-line extensions  
The telephone system must have ringing capability because the voice boards and voice board drivers used in Atlas AVM require the use of ringing as their method of notification.
- End-to-end DTMF  
The Atlas AVM system sends and receives address information for calls in the form of DTMF (Dual Tone Multi-Frequency) signals. The phrase 'end to end' refers to the ability of a device to both send and receive DTMF signals.

The following section, "Investigation Steps," gives some possible ideas for bringing a system that lacks some of the necessary requirements up to standard.

## Investigation Steps

The information required to determine if successful integration is possible can be obtained from the telephone system's documentation or manufacturer, the customer's interconnect vendor, or through on-site testing. (Some methods of on-site testing are covered in the *Atlas Certified Network Telephony Engineer Course Manual*.)

As you gather the integration information listed in the following steps, enter the data in the corresponding sections of the Integration Form beginning on Page 4-7.

1. Document general information about the customer and the telephone system.
2. Find out if the telephone system software supports voicemail.

You will probably need to consult the company's telephone representative to find out what software version is installed on the telephone system, and what features this software has.

3. Find out if the telephone system has the necessary hardware for voicemail.

### Locate single line extensions

If the customer has a fax machine or modem lines that go through the telephone system, those are single line extensions. Find out how many single lines there are.

If there are currently no single line extensions, you can add a single line card (SLC) to the telephone system. These are also called industry standard telephone cards (IST), or off-premise extension cards (OPX).

### Check for ringing on single line extensions

Dial a single line extension from another single line extension to see if you hear ringing over the telephone.

The voice boards and their drivers used in Atlas AVM necessitate the use of ringing as the method of notification. Therefore, the telephone system has to have ringing capability on the extensions going to Atlas AVM.

If the telephone system does not currently have ringing capability, you can add a 'ring generator' card.

### Check for end-to-end DTMF

- Attendant console
- Single line handset
- Digital handset

The voice boards in the Atlas AVM machine, being analog devices, necessitate the use of DTMF as the method of receiving control information.

If the telephone system cannot provide DTMF on the extensions going to Atlas AVM, then an additional DTMF generator board must be added to the telephone system.

4. Find out how the telephone system operates concerning the following:
  - hunt groups
  - disconnects
  - supervised transfers
  - event signaling
  - off-premise transfers
5. Make a list of equipment or software that needs to be upgraded.
6. Obtain information about the data packets that are sent by the telephone system.

You can gather this information from the customer's interconnect or from the telephone system vendor. If you cannot collect this information through these avenues, or you want to confirm the information you have collected, you will need to use a digit grabber. For an example of how to use a digit grabber, *see* Page 4-16.

You will enter the data packet information you have gathered when you are ready to program the Atlas AVM integration screens covered in Chapter 12, "Integrating With the Telephone System."

You do not need to input every data packet into Atlas AVM—only the ones that will be relevant for Atlas AVM operations. These would include such packets as 'forward, busy,' 'forward, no answer,' 'retrieve message,' and possibly a 'disconnect.'

Read the information in the next section, "Understanding Data Packets," before you fill out the data packet portion of the Integration Form.

## Understanding Data Packets

The data packet investigation involves the following three parts:

- Events and Packet Identifiers
- Packet Formats
- Atlas AVM Packet Definitions

**Note**

Contact your telephone system installer or manufacturer for the correct codes to use in defining types of calls in the integration data packets. If this is not possible, you may be able to recreate the call type and monitor it on the Diagnostic Trace screen. (See the Atlas AVM System Administrator's Manual

## Events and Data Packet Identifiers

You must learn the 'events' the telephone system sends data packets on and their corresponding 'packet identifiers' (i.e. the code used to denote that event).

**Example:**

<u>Event</u>	<u>Packet Identifier</u>
Forward, busy	2
Forward, no answer	3
Retrieve message	4
Disconnect	7

**Note**

The telephone system may not send a data packet for each of the Atlas AVM data packet definitions.

## Data Packet Formats

You must learn the packet format of the data packets. This involves knowing what part of the data stream pertains to such things as the packet identifier, the called party extension, the calling party extension, and possibly other information.

**Example:**

The telephone system could send a string of digits to Atlas AVM that communicates the called party extension and the packet identifier:

1332

Atlas AVM must know what the packet format is in order to understand this communication. Since there is no standard default packet format, Atlas AVM does not know whether the packet identifier is '1' and the called party extension is '332,' or if the called party extension is '133' and the packet identifier is '2.'

However, if we tell Atlas AVM what positions in the data packet relate to what information, then Atlas AVM can understand this data packet.

If the called party was '133,' and the packet identifier was '2' (for example, a forward busy), then the packet format would be:

Data Packet: 1 3 3 2

Positions: 0 1 2 3 4 5 6 7 8 9

Packet Identifier: 2

Packet Identifier Offset (position) 3

Packet identifier Length: 1

Called Party Offset (position): 0

Called Party Length: 3

### **Atlas AVM Definition of a Data Packet**

For every data packet that you input into Atlas AVM, you must tell Atlas AVM how it should 'define' that packet. The following section contains the current possible data packet definitions and how Atlas AVM acts for each definition.

### **Atlas AVM Data Packet Definitions**

#### ***Direct to Box***

The call goes directly to a box owner's mailbox, without ringing the extension, based on the 'called party' information. Atlas AVM plays the box owner's personal greeting without any additional prompts.

#### ***Forwarded***

Atlas AVM plays the following prompt:

"< *Name of box owner*> is not available to take your call."

Atlas AVM then sends this call to the box number of the 'called party' and carries out whatever is in the Busy Option field (e.g., Take Message, Alternate Extension, Caller Options).

Atlas AVM plays the following prompt:

"I'm sorry, that extension is busy."

Atlas AVM then sends this call to the box number of the 'called party' and carries out whatever is in the Busy Option field (e.g., Take Message, Alternate Extension, Caller Options).

### **Forwarded No Answer**

Atlas AVM plays the following prompt:

“I’m sorry, that extension does not answer.”

Atlas AVM then sends this call to the box number of the ‘called party’ and carries out whatever is in the No Answer Option field (e.g., Take Message, Alternate Extension, Caller Options).

### **Retrieve Message**

Atlas AVM recognizes the caller as a box owner, sends the call to a box based on the ‘called party’ information, and then asks for the passcode for that box.

### **Trunk**

Atlas AVM recognizes the call as being a ‘trunk’ call. This means that the Trunk Port field on the Integration Data Packet Parameters screen is valid. Atlas AVM plays the port greeting or Auto Attendant for outside calls that are identified as the trunk listed in the Trunk Port field.

## **The Integration Form**

The Integration Form, found on the next page, can be used by the Atlas AVM installer to help determine if a telephone switch can be integrated with, and to document the details of that integration.

If the particular telephone system in question is listed in the *Atlas Telephone System Compatibility Integration Listings*, use this form just for your own records.

# Integration Form

1. General Information	
(This information can be obtained from the Customer Engineering Form in Chapter 3.)	
a. Customer	
b. Contact	
c. Telephone system manufacturer	
d. Telephone system model	
e. Software version	

2. Telephone System Software Version
Is there a particular telephone system software version that is needed for voicemail?
a. If Yes, what is that version?
<p>Compare 2a. with 1e. If 1e. is the same or a higher version (check with the manufacturer to make sure), then continue on with this form.</p> <p>If 1e. is a lower version, then the telephone system software must be upgraded. Make note of this on the Telephone system Upgrade Form.</p>

3. Necessary Hardware for Voice Messaging	
Does the telephone system have the necessary hardware for voice messaging? This entails the following items:	
a. Available single-line extensions	Yes / No
If Yes, how many?	
b. Ringing on the single line extensions that will be used by Atlas AVM	Yes / No
c. DTMF tones on the single line extensions that will be used by Atlas AVM	
1. From an attendant console	Yes / No
2. From a 2500 set	Yes / No
3. From a digital set	Yes / No
If any of the items in a-c are not in place, those items must be added to the Telephone system Upgrade Form.	

4. Telephone system Operations	
a. Does this telephone system allow for hunt groups? If Yes:	Yes / No
When all single line extensions in a hunt group are busy, what happens to the next:	
Trunk caller (outside caller):	
Station caller (inside caller):	
b. If hunt groups are not available, can all lines be chained to forward to the next available line and eventually back to a live attendant?	
c. How does the telephone system provide notification of a disconnect to a single line extension?  Dial tone (IST port):  Data packet (IST port programmed as VM):  Reorder:  Loop current drop:	
d. Does the telephone system allow supervised transfers? (This is only relevant if there are no data packets.)	
e. What type of signaling, from a single line extension, does the telephone system need for the following events?	
1. Call transfer	
2. Recall busy (supervised transfer only)	
3. Recall no answer (supervised transfer only)	
4. Call connected (supervised transfer only)	
5. Message waiting light on	
6. Message waiting light off	
7. Call screening park (screened calls only)	
8. Call screening remove park (screened calls only)	
f. Does this telephone system allow for off-premise transfers?	Yes / No
If Yes, what type of signaling is used?	

5. Telephone system Upgrade Form	
List the telephone system equipment or software that needs to be upgraded in order to accomplish voice messaging.	
1.	
2.	
3.	
4.	
5.	

**6. Data Packet Events and Formats**

As you fill out the data packet forms, it is recommended that you first complete the second column for all relevant telephony scenarios based upon the information you learn through using a digit grabber. Then go back and complete the additional relevant information in the other columns. The first scenario is an example of a data packet that a telephone system might send for a forward, busy call.

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (Atlas AVM Definition)
<b>EXAMPLE</b>		Packet example: <u>1 3 3 2</u>	
1. Forward, Busy:	Packet example: 1 3 3 2	Positions: 0 1 2 3 4 5 6 7 8 9	(Choose only one)
	Identifier: x x x 2 (Event code)	Packet identifier: 2 (ASCII)	Retrieve msg.: <input type="checkbox"/>
	x = Called Party	Packet ident. offset: 3	Forwarded: <input type="checkbox"/>
	y = Calling Party	Packet ident. length: 1	Forward, busy: <input type="checkbox"/>
		Called party ext. offset: 0	Fwd, no answer: <input type="checkbox"/>
		Called party ext. length: 3	Direct to Box: <input type="checkbox"/>
		Calling Party ext. offset: _____	Trunk: <input type="checkbox"/>
		Calling party ext. length: _____	Terminate call: <input type="checkbox"/>
			Ignore: <input type="checkbox"/>
			DTMF: <input type="checkbox"/>
			Etc.
		Trunk Port: _____	



**Note**

Not all telephone systems will send a data packet for each scenario. Some systems send the same packet for different scenarios.

## Telephone Scenarios

The following are scenarios that you should create to identify the data packets that are sent by the telephone system.

### Forwarded Busy

A station is programmed to call forward on a busy state to AVM.

### Forwarded No Answer

A station is programmed to call forward on a no answer state to AVM.

### Forward All Calls

A station is programmed to call forward all calls to AVM.

### Retrieve Message

A station calls or reaches AVM to retrieve messages. Four different methods could accomplish a station call to AVM to retrieve messages with an accompanying data packet.

1. Call the voicemail hunt group.
2. Call the analog extension.
3. Press a message waiting indicator button.
4. Call yourself when your station is programmed to call forward to AVM.

### Trunk Call

Incoming CO/trunk call is routed to AVM to be answered.

### Voicemail Transferred Recall

AVM transfers a caller to a station that is not programmed to call forward. A busy or no answer state on that station, the telephone system will recall (send back) to the originating transferring extension (AVM).

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
1. Forward, Busy:	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one) Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
2. Forward, No answer	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one) Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
3. Retrieve Message:	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one) Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
4. Trunk Call:	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one) Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
5. Forwarded all calls:	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one)  Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
6. Voicemail transferred recall:	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one)  Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

Telephony Scenario	Packet Identifier (Event Code)	Packet Format	Packet Definition (AVM Definition)
7. (Other):	Packet: _____  Identifier: _____ (Event code)  x = Called Party y = Calling Party	Packet: _____ Positions: 0 1 2 3 4 5 6 7 8 9  Packet identifier: _____ Packet ident. offset: _____ Packet ident. length: _____  Called party ext. offset: _____ Called party ext. length: _____  Calling party ext. offset: _____ Calling party ext. length: _____  Trunk Port: _____	(Choose only one)  Direct to Box: _____ Forwarded: _____ Forward, busy: _____ Fwd, no answer: _____ Retrieve msg: _____ Trunk: _____

The following packets are called 'in-band supervisory' packets. They do not have called party or calling party information. They are packets the telephone system uses to denote certain telephony situation.

Enter the sequences that AVM recognizes for disconnect and off-hook. The AVM menu path tells how to get to the AVM screen where you need to enter this information.

7. Disconnect	Disconnect Sequence	AVM Menu Path
	_____	System Installation Options Port Telephone System Select Port Select Port Telephone System Port Telephone Parameter Options General Parameters  Field = Disconnect Call Sequence

8.	Off-hook Sequence	AVM Menu Path
	_____	System Installation Options Port Telephone System Select Port Select Port Telephone System Port Telephone Parameter Options General
		Field = Outbound Dial Tone Digit

## Data Packet Timeout and Termination

Is there a data packet timeout? Yes / No

If 'Yes,' what is it? \_\_\_\_\_

Data packet timeout is the length of time AVM will wait after going off hook to receive a data packet. The timeout should be as short as possible. Any DTMF received will be assumed to be a data packet from the telephone system.

Is there any 'character' that the telephone system uses as a packet terminator? Yes / No

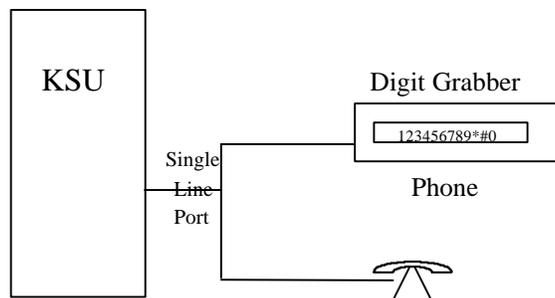
If 'Yes,' what is it? \_\_\_\_\_

## Using a Digit Grabber

You can use a Digit Grabber to manually gather data packets.

Equipment Needed	
------------------	--

Quantity	Item
1	Single line extension programmed for voicemail (sends data packets)
1	Line splitter (this will allow us to connect two devices, 2500 handset and the digit grabber, to the same extension thereby allowing us to view the data packets sent to this extension)
2	Line cords (for the connection from the 2500 handset and digit grabber to the line splitter)
1	2500 handset (this will take the role of a Atlas AVM machine)
1	Digit grabber (this will display the digits, sent via DTMF, that comprise the data packets sent to the extension the digit grabber is connected to via the line splitter)
2	Digital handset programmed to 'call forward on busy' to the extension that our 2500 handset is at (these handsets will be used to place calls to each other in order for the telephone system to generate data packets)



**Figure 4 -1: Using a Digit Grabber**

## Digit Grabber Set-up Procedure

1. Pick a single line extension (NOTE: This extension must be programmed for data packets/in-band signaling). Connect a cable from that extension's wall jack to the line splitter.
2. Connect the digit grabber to the line splitter with either a cable or directly.
3. Connect the 2500 handset to the line splitter.
4. Turn on the digit grabber.

## Test Procedure

1. Test 2500 handset and digit grabber:
  - At 2500 handset, dial an extension.
  - You should see the digits dialed on the digit grabber. (RESET)

## Data Packet Collection

1. Decide which extensions to use for each instrument.

Instrument	Ext.
Digital handset 'A' – call forward to Atlas AVM (made to be busy; no answer; etc.)	
Digital handset 'B' – instrument that calls handset 'A'	
Digit grabber	

2. To learn of a 'forward busy' event:
3. Program digital handset 'A' to call forward to the extension that the digit grabber is attached to (the pseudo Atlas AVM machine). Take digital handset 'A' off-hook and create a busy (press intercom).
4. At digital handset 'B' dial digital handset 'A.' The telephone system will attempt to connect this call to handset 'A' but will not be able to because of the busy state. Because handset 'A' is programmed to 'forward calls on busy' to the extension that is our 2500 handset (Atlas AVM), the telephone system will then forward the call to that extension. When the telephone system does the transfer of the call it will also send along a data packet.
5. When the 2500 set rings, pick up the handset.
6. At the digit grabber, read the incoming data packet.



**Note**

In-band data packet information can also be seen on the AVM Diagnostic Trace screens.

7. To learn of the other events do the same basic procedure except recreate the other telephony scenarios (e.g., no answer, disconnect, CO line call, retrieve message, Atlas AVM initiating a call, etc.).
8. Document what you learn on the *Integration Form*.



## ***Designing the Atlas AVM System***

After you have determined that you can integrate with the customer's telephone system, you should work with a company representative to draw up a basic design for the Atlas AVM system. This will help you in several areas:

- Finding out what type of system the customer wants so you can begin to put it together.
- Closing the sale. Giving the customer a more complete picture of what Atlas AVM can do for them helps to secure the sale.
- Determining the Atlas AVM system size required by the company. You made an estimate of how many boxes this company will need in the evaluation phase. Now you need to come up with a more accurate count.

### **Planning the Basic Design**

Considerations for designing a system are outlined below.

1. Design how the Auto Attendant will handle calls. This includes:

Primary or Secondary Auto Attendant—The Auto Attendant answers all incoming calls, or a live operator answers calls except when all calls cannot be accommodated by the operator.

Greetings—which can be set for different times (e.g. business hours vs. non-business hours), days, or ports.

Caller Options—which give callers options for transferring from the Auto Attendant box to an extension, the operator, or another audiotext box. Callers also hear caller options when they enter an invalid digit or fail to respond to a voice prompt.

2. Planning audiotext boxes. This includes the following:

Mapping the levels of audiotext boxes—which are designed to allow callers to transfer from one audiotext box to another.

Planning audio menus—which are the transfer options callers hear in audiotext boxes.

3. Find out what the customers want their greeting and other audiotext boxes to look like.

Help the customers determine what they want their system to sound like.

Create some rough scripts based on the customer's responses.

4. Planning individual voicemail boxes. This includes:

Obtaining a list of extension numbers of the people who will be using the Atlas AVM system.

Deciding which range of Atlas AVM boxes to use for voicemail boxes.

Renumbering the Atlas AVM boxes to match the extension numbers of their owners.

After you have mapped out the basic design of the system, you should know approximately how many total boxes are required for the Atlas AVM system.

## Sizing the System

The size of the Atlas AVM system purchased is based on the number of boxes the company needs. Atlas AVM systems are available in the following box group sizes:

50, 300, 500, 750, 1000, 2500, 5000, 7500, 10,000

Calculate the number of boxes based on the basic design of the system, allowing for modifications or growth.

You also need to determine the number of ports and the disk space required for the system. Use the Erlang Table and the following worksheets:

- Auto Attendant
- Audiotext Calls
- Voicemail
- Ports Required
- Voice Storage
- Voice Storage Required

## **Erlang Measurement**

Erlang is a unit of measurement for telephone traffic. (Total call seconds per hour divided by 3600 equals an Erlang.) The Erlang measurement is used to ensure that Atlas AVM is sized large enough to minimize the number of blocked calls (busy signals) during the peak hours of calls.

Because all calls that come into a system are not evenly dispersed, you cannot merely estimate the number of call seconds and then plug that number into a chart. The Erlang measurement accounts for calls that are concentrated in the same time period (i.e. overlap).

Use the Erlang Table when you fill out the Ports Required worksheet to determine the number of ports needed. Blockage levels of 2% to 3% are acceptable.

Number of Ports	Blockage Levels						Number of Ports
	10%	5%	3%	2%	1%	0.1%	
1	0.01	0.05	0.02	0.01	0.01	0.00	1
2	0.54	0.36	0.22	0.15	0.10	0.10	2
3	1.14	0.85	0.59	0.45	0.35	0.19	3
4	1.84	1.45	1.07	0.86	0.70	0.44	4
5	2.59	2.11	1.62	1.35	1.13	0.76	5
6	3.38	2.81	2.23	1.89	1.61	1.14	6
7	4.20	3.55	2.88	2.48	2.15	1.58	7
8	5.04	4.32	3.55	3.10	2.72	2.05	8
9	5.89	5.10	4.26	3.74	3.32	2.55	9
10	6.76	5.91	4.98	4.42	3.94	3.09	10
11	7.64	6.72	5.72	5.11	4.59	3.65	11
12	8.53	7.55	6.48	5.82	5.25	4.23	12
13	9.42	8.39	7.25	6.54	5.93	4.83	13
14	10.33	9.24	8.04	7.28	6.63	5.44	14
15	11.24	10.10	8.83	8.03	7.34	6.07	15
16	12.15	10.97	9.63	8.79	8.06	6.71	16
17	13.07	11.84	10.44	9.56	8.79	7.37	17
18	13.99	12.72	11.26	10.33	9.53	8.04	18
19	14.92	13.60	12.09	11.12	10.28	8.72	19
20	15.86	14.49	12.92	11.91	11.04	9.40	20
21	16.79	15.38	13.76	12.71	11.80	10.10	21
22	17.72	16.28	14.60	13.51	12.57	10.80	22
23	18.66	17.18	15.45	14.32	13.35	11.51	23
24	19.61	18.08	16.30	15.14	14.13	12.23	24
25	20.55	18.99	17.15	15.96	14.92	12.96	25
26	21.50	19.90	18.02	16.79	15.72	13.69	26
27	22.45	20.81	18.88	17.62	16.52	14.42	27
28	23.40	21.72	19.75	18.45	17.32	15.17	28
29	24.35	22.64	20.62	19.29	18.13	15.91	29
30	25.30	23.56	21.49	20.13	18.94	16.67	30
31	26.26	24.48	22.37	20.98	19.75	17.42	31
32	27.21	25.41	23.25	21.83	20.57	18.19	32
33	28.17	26.33	24.13	22.68	21.40	18.95	33
34	29.13	27.26	25.02	23.54	22.22	19.72	34
35	30.09	28.19	25.91	24.39	23.05	20.50	35
36	31.05	29.12	26.80	25.25	23.89	21.27	36
37	32.02	30.06	27.69	26.12	24.72	22.06	37
38	32.98	30.99	28.58	26.98	25.56	22.84	38
39	33.94	31.93	29.48	27.85	26.40	23.63	39
40	34.91	32.86	30.38	28.72	27.24	24.42	40

Figure 5 -1: Erlang Table

## Auto Attendant

**AA = Auto Attendant**

**VR = Voice Response (calls coming back to the Auto Attendant)**

1. \_\_\_\_\_ Number of calls to the Auto Attendant during peak hours
2. x \_\_\_\_\_ Length of prompts for AA (in seconds)
3. = \_\_\_\_\_ **AA seconds per hour**

4. \_\_\_\_\_ VR calls
5. x \_\_\_\_\_ Average VR time (in seconds)
6. = \_\_\_\_\_ **VR seconds per hour**

7. \_\_\_\_\_ AA seconds per hour (from line 3)
8. + \_\_\_\_\_ VR seconds per hour (from line 6)
9. = \_\_\_\_\_ **Total call seconds per hour for the Auto Attendant (AA + VR)**

## Audiotext Calls

10. \_\_\_\_\_ Number of audiotext calls per peak hour
11. x \_\_\_\_\_ Number of boxes listened to during a call
12. = \_\_\_\_\_ Number of calls and boxes
  
13. x \_\_\_\_\_ Length of average audiotext message (in seconds)
14. = \_\_\_\_\_ **Total call seconds in peak hour for audiotext**

## Voicemail

### Message Calls (leaving a message)

15. \_\_\_\_\_ Number of employees
16. x \_\_\_\_\_ Number of messages per day per employee
17. = \_\_\_\_\_ Total message calls per day
18. x \_\_\_\_\_ Length of entire message sequence (in seconds)  
(prompts + messages + editing)
19. = \_\_\_\_\_ Total message call seconds per day
20. /8 \_\_\_\_\_ Hours in business day
21. = \_\_\_\_\_ **Message call seconds per hour**

### Subscriber Calls (listening to messages)

22. \_\_\_\_\_ Number of employees
23. x \_\_\_\_\_ Number of calls per day to check messages
24. = \_\_\_\_\_ Total subscriber calls per day
25. x \_\_\_\_\_ Length (seconds) of a subscriber call
26. = \_\_\_\_\_ Total subscriber call seconds per day
27. /8 \_\_\_\_\_ Hours in business day
28. = \_\_\_\_\_ **Subscriber call seconds per hour**

## Ports Required

Step 1: Add the following lines from the previous worksheets

Line 9, Auto Attendant \_\_\_\_\_

Line 14, Audiotext	_____
Line 21, Message calls	_____
Line 28, Subscriber calls	_____
<b>Total</b>	_____

Step 2: Take the total from Step 1 and divide it by 3600 to determine the Erlang number. Go to the Erlang Table and find the line closest to that number in the 2% to 3% blockage range. Find the number of ports associated with that line and enter it here. \_\_\_\_\_

**Total Ports Needed** \_\_\_\_\_

## Voice Storage

1 hour of voice storage = 10 MB of disk storage

### Voicemail

1. \_\_\_\_\_ Numbers of employees
2. x \_\_\_\_\_ Number of messages left per employee per day
3. = \_\_\_\_\_ Total messages per day
  
4. x \_\_\_\_\_ Length (seconds) of messages left by callers
5. = \_\_\_\_\_ **Total seconds of voicemail storage per day**

### Audiotext

6. \_\_\_\_\_ Length (seconds) of average audiotext message
7. x \_\_\_\_\_ Number of audiotext boxes
8. = \_\_\_\_\_ **Total seconds for audiotext messages**

# Voice Storage Required

Step 1: Add the following lines from the Voice Storage worksheet:

Line 5, Voicemail \_\_\_\_\_

Line 6, Audiotext \_\_\_\_\_

Total = \_\_\_\_\_

Step 2: Divide the total from Step 1 by 60 (i.e. minutes divided by seconds) ÷ 60

Total = \_\_\_\_\_

Step 3: Divide the total from Step 2 by 60 (i.e. hours divided by minutes) ÷ 60

Total = \_\_\_\_\_

Step 4: If 1 hour of voice storage = 10 MB of disk storage

Then \_\_\_\_\_ hours of voice storage (Step 3) = \_\_\_\_\_ MB of disk storage

Plus 10 MB for the Atlas AVM files = + 10 MB for Atlas AVM

**Total Disk Storage Needed** \_\_\_\_\_



## ***Installing Voice Boards***

### **Understanding Voice Boards**

Voice boards perform the following functions:

- Digit Processing: detecting and generating DTMF tones
- Voice Processing: playing and recording messages
- Call Processing: listening for and characterizing tones and frequencies, and reporting them back to Atlas AVM

Atlas AVM can use certain voice boards manufactured by Dialogic and Rhetorex. These boards and their setup are described in sections of this chapter and in Chapters 7 and 8. Read the material that applies to the voice boards used in your system.

### **Voice Board Installation**

#### **Unpacking and Handling the Boards**

Before you unpack your voice boards, make sure you have a clear, clean, dust-free space to work.

Printed circuit boards can be damaged easily if subjected to rough handling or electrostatic conditions. Wear an anti-static wrist strap and use electrostatic-dissipative mats whenever you handle PC boards.

#### **Installing the Voice Boards**

See Chapters 7 and 8 for setup and strapping options for your voice boards.

When all boards have been prepared, follow the instructions below to install them in the PC:

1. Unplug the power cords to the PC and to any connected peripheral devices.



**Note**

If you are installing a one-board system it is recommended that you install the voice board in the slot that is farthest away from the power supply. If there is more than one board in your system, install the number one board in the slot farthest from the power supply, the number two board in the slot next to the number one board, and so on.

2. Remove the PC cover and set it aside, along with the screws, in a safe place.
3. Remove the first voice board from its anti-static bag. Set the jumpers according to the information in Chapters 7 or 8.
4. Slide the board into the slot farthest from the power supply, seating it firmly in the connector strips.
5. Tighten the screw connecting the board to the chassis. (This grounds the board.)
6. Remove the second board (if any) from its anti-static bag and install it in the slot next to the number one board. Repeat for all remaining boards.
7. When all boards have been installed in their proper slots, return the cover to the PC and attach it securely.
8. Reattach all power cords and power up the PC and any peripheral equipment.



**Note**

RJ-11 and RJ-14 jacks look similar; however, RJ-11s provide only one line while RJ-14s provide two lines.

## Attaching the Cables

To attach the voice boards to the telephone system, follow the instructions below.

Insert a plug into the jack in the rear bracket of the voice board. There is a snap or click when the plug is properly connected.

Continue to connect the RJ-11 or RJ-14 plugs to all the voice boards.

## Dialogic Voice Boards

The Dialogic voice boards that you can use with your Atlas AVM system include those shown in Table 6 -1.

Model	Ports	Comments	Board Jack
D/41E	4		2 each/RJ11
D/41D	4		2 each/RJ14
D/41SX	4		2 each/RJ14
D/41SL	4		2 each/RJ14
D/41NS	4		4 each/RJ11
D/121B	12	Must work with LSI/120	NA
LSI/120	12		DB25

**Table 6 -1: Dialogic Board Specifications**

### **D/41E**

A 4 port digital store-and-forward voice board with world approvability, access to resource modules, and additional power for future firmware features.

### **D/41D**

A 4 port digital board used to connect to single line station ports from a PBX or telephone company. Typical 2500 telephone interface, i.e. Centrex/SMDI, loop station interface.

### **D/41SX**

A 4 port digital board used with the Mitel PBX using COV or Super Set integration.

### **D/41SL**

A 4 port digital board used with the Northern Telecom SL-1 PBX using SL-1 station integration.

### **D/41NS**

A 4 port digital board used with the NorStar TCM digital station interface.

### **D/121B-LSI/120**

Used together to give 12 ports – same as the D/41D but supports 12 analog ports.

## **Basic Analog Configuration**

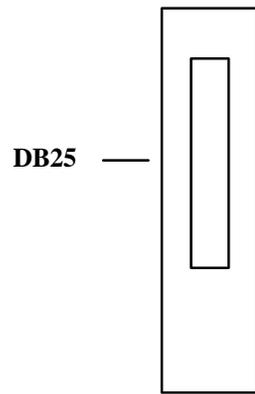
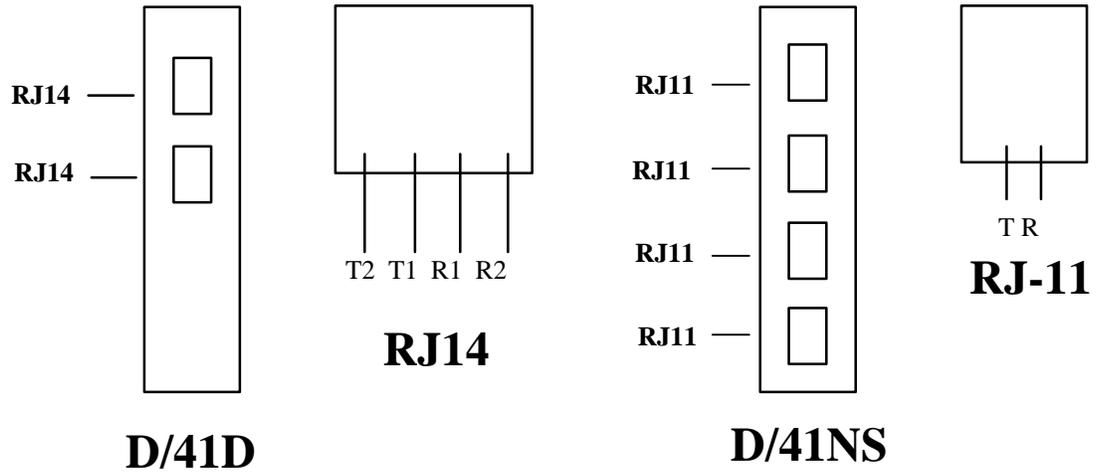
The D/41D board contains everything needed for the basic 4-port analog configuration. Higher density 8-port and 12-port boards use a SpringBoard connected via PEB cable to an LSI board.

The basic 12-port analog application uses a D/121B board connected via a PEB cable to one LSI/120 board. This configuration will handle up to 12 telephone lines. A PEB terminator must be installed on both the voice board and the LSI/120.

## **Multiple Board Configurations**

The number of boards that can be used in a system varies according to the type of boards in use and the number of free slots available in the PC. Each D/41D board requires one 8-bit slot. D/121B and LSI/120 boards must be used together, requiring two 8-bit slots per set.

## Dialogic Telephone Connectors



**D121/LSI/120**

Pin	Term.	Pin	Term.
1	T1	13	T7
2	R1	14	R7
3	T2	15	T8
4	R2	16	R8
5	T3	17	T9
6	R3	18	R9
7	T4	19	T10
8	R4	20	R10
9	T5	21	T11
10	R5	22	R11
11	T6	23	T12
12	R6	24	R12
		25	

Figure 6 -1: Cord Connections for Dialogic Boards

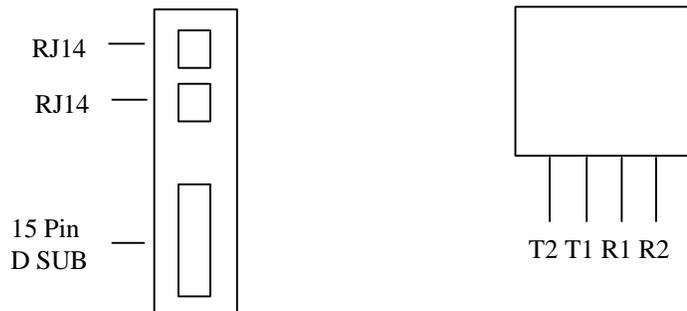
## Rhetorex Voice Boards

The Rhetorex voice boards that you can use with your Atlas AVM system include those shown in the following table.

Models	Card Size	Ports	Jack/Connection
208 Duet	Half	2	1 each – RJ14
408 Quartet	Half	4	2 each – RJ14
432 / 232	Half	4	2 each – RJ14
2132	Full	2	1 each – RJ11
4132	Full	4	2 each – RJ14
9432	Full	4	2 each – RJ14

**Table 6 -2: Rhetorex Board Specifications**

## Rhetorex Telephone Connectors



### 432/2132/4132/9432

**Figure 6 -2: Rhetorex Telephone Connectors**

#### RDSP/432

A 4 port, analog, half length board that supports 4 analog telephone interface ports.

#### RDSP/208

A 2 port, analog, half length card. This board supports 2 analog single line telephone interface ports. Only one board allowed per system.

**RDSP/408**

A 4 port, analog, half length card. This board supports 4 analog single line telephone interface ports. Only one board allowed per system.

**RDSP/2132**

A 2 port, analog board that supports 2 analog telephone interface ports.

**RDSP/4132**

A 4 port analog board that supports 4 analog telephone interface ports.

**RDSP/9432**

A 4 port analog board that is internationally approved. It supports 4 analog telephone interface ports.

## Call Progress Tones

Your telephone system uses audible tones to indicate the progress of a call. These call progression tones include four basic tones:

- dial tones
- busy tones
- ring tones
- reorder tones

**Note**

Although silence is not considered a tone, it is also recognized and acted on by the voice board.

**Dial tone:** This tone tells the caller that the central office or the telephone system is ready to receive address signals (i.e. telephone number or extension).

**Busy tone:** This tone indicates that the number being called is busy.

**Ring tone:** When a caller places a call, a ring tone is returned to the caller. This ring tone, also called a ring back, indicates that a path has been established and the called instrument is being rung.

**Reorder tone:** The tone that sounds when a telephone has been left off hook longer than the maximum length of time.

By listening to the call progression tones, a caller can recognize the state of a call. Similarly, the voice board can listen to call progression tones and recognize the state of a call. However, there is no standard way for a telephone system to express these tones. Consequently, the specific tone characteristics used by the telephone system must be defined in order for the voice board and driver to integrate with the telephone system.

## Tone Characteristics

Tone characteristics relate to three main areas:

**Frequency:** The number of vibrations, or oscillations, per time unit of an alternating current.

**Duration:** The complete signal sequence consisting of 'on time' (the signal) and 'off time' (silence).

**Cadence:** The length of the 'on time' and the length of the 'off time' in a one tone cycle.

These three tone characteristics (frequency, duration, cadence) must be determined for each of the four call progress tones (dial tone, busy tone, ring tone, and reorder).

The driver can recognize a tone if the tone's characteristics match the parameters for a tone in the tone table.

Utility programs such as Call Analysis for Dialogic or AccuCall Plus for Rhetorex, allows users to determine, test and define the frequency, duration and cadence elements of the environment's call progression tones.

## Supervised Transfers

If the telephone system does **not** provide in-band signaling (i.e., data packets), then Atlas AVM will have to supervise all its transfers in order to ascertain the result of those transfers (e.g., busy, no answer, etc.). This is called a 'supervised' transfer. Atlas AVM must therefore be able to accurately understand the tones the telephone system uses to denote these different events, which are called 'call progress tones.' This is done by building a database file of the characteristics of a particular telephone system's call progress tones.

The following is an example of a supervised call:

1. An outside call comes into a PBX. The PBX is programmed to send all calls to Atlas AVM.
2. Atlas AVM plays the port greeting for the caller and prompts the caller to enter the desired extension.
3. The caller enters the number 100.
4. Atlas AVM looks in its voicemail box configuration database for box 100. It transfers the call to extension 100.
5. Atlas AVM attempts to transfer the call to extension 100. It does this by signaling (e.g., flash hook then extension number) the telephone

system to transfer the call to extension 100. (At this point the outside caller is 'parked' at the telephone system.)

6. Since this telephone system does not use in-band signaling (i.e., no data packets), Atlas AVM is going to 'stay on the line' during the transfer in order to learn the result of this transfer.
7. In this case, extension 100 is busy. The telephone system sends a particular tone to denote a busy event.
8. Atlas AVM, which is still 'listening on the line,' references its tone table to see what this tone denotes.
9. Once Atlas AVM realizes that extension 100 is busy, it looks in its voicemail box configuration database for what to do when extension 100 is busy—in this case, take a message.
10. Atlas AVM then connects back to the parked call and plays the user's box greeting for voicemail box 100. The caller can then leave a voicemail message.

For step 8 to work, a tone table must be created for this particular telephone system. One way to do this with a Rhetorex voice board is to use the program AccuCall Plus.

## AccuCall Plus (Rhetorex only)

AccuCall Plus is a program developed by the vendor Rhetorex. This program can be used with a Rhetorex voice board to analyze the tone characteristics of a particular telephone system.

In this regard, it is like a protocol analyzer in the data communication field, except that AccuCall Plus analyzes tone characteristics.

AccuCall Plus can analyze the tone characteristics for each of the call progress events (e.g., dial tone, busy tone, ring tone). AccuCall Plus then puts the information on the specific tone characteristics into a tone table (i.e., a database file). This tone table can then be referenced by the Rhetorex driver and Atlas AVM.

## Installation and Use of AccuCall Plus

AccuCall Plus is a DOS executable program. It must be run on the PC that will have the Rhetorex voice board in it. Follow the steps below to run AccuCall.



### Note

Begin these steps before you have installed the Rhetorex voice board in the computer. You will be installing the board in the third step.

### Copy files (from AccuCall Plus disk).

1. MD c:\rhetorex
2. COPY a:\\*.\* c:\rhetorex

### **View graphical depiction of possible voice board configurations.**

1. \rhorex\SHOWJUMP.EXE

Explanation: This utility shows you a graphical depiction of the board settings for different configuration options. You must pick a setting that will not conflict with the other devices in the machine being used to run AccuCall Plus, and ultimately that does not conflict with anything in the Atlas AVM computer. Please refer to the Server Configuration Worksheet for the configurations already in use by your Atlas AVM computer.

Write down the configuration parameters and the matching board settings.

### **Install the voice board.**

1. Set the switches on the voice board to match your chosen configuration.
2. Turn off machine.
3. Install board in machine. (To reduce interference, put the voice board in an expansion slot that is not near the power supply.)
4. Attach single-line analog lines into the ports of the voice board.

### **Configure AccuCall Plus to match the voice board.**

1. Turn on machine
2. At the DOS-prompt, type: **cd\rhorex**
3. CONFIGUR.EXE

Explanation: Set the following parameters to match the voice board.

Port = \_\_\_\_\_ (also called I/O memory address)

DualPort = \_\_\_\_\_ (also called base memory address)

Entry point = \_\_\_\_\_

### **Execute the Rhorex DOS driver.**

1. Type: **c:\rhorex\rhetrov.exe**

When the driver has successfully loaded, a log file called voicecnf.log will be displayed on the screen. The information shown covers ports, channels, status, serial number, number of voice boards found, number of channels, memory allocation, firmware level, revision, and date.

If voicecnf.log does not come up (if it hangs), then rename CONFIG.SYS and AUTOEXEC.BAT and reboot the machine.

**Execute AccuCall Plus.**

1. Type: `cd\rhetorex\accucall`.
2. Press <F6> to do an auto-run.
3. Enter the extension number you want AccuCall to call.
4. Make that target extension busy by going off-hook.
5. Press <F9> to start the AccuCall analysis of the tone characteristics.
6. When the cycle is completed, it will prompt you to press <F8>.
7. Enter a name to describe the tone characteristics it has just analyzed, e.g. busy.
8. Select the event type by pressing <spacebar>, and then busy.
9. Arrow down to PCPMCODE and enter the number 7. This is the Rhetorex code for a busy event.
10. Press <F8> to add the preceding information to the table.
11. Press <F10> to return to the main menu.
12. Repeat letters c through h for a 'ring no answer' event. Its PCPMCODE code is the number 8.
13. Press <F7> to give a name to the tone table you have just created. Give it a descriptive name, e.g., COMDIAL.TON.
14. Press <F8> to save the tone table.
15. Press <F10> to save the tone table to a disk and exit.

For a complete AccuCall manual Please contact Rhetorex or Key System US

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## ***Building the System***

### **Environmental Considerations**

The following environmental conditions are recommended:

- Operating temperature: 0 to 50 °C
- Humidity: 8% to 80% non-condensing
- Storage temperature for voice boards: -10 to 70 °C

Refer to your PC and voice board owner's manuals for specific recommendations.

### **Power Supply Considerations**

Voicemail system customers are responsible for obtaining appropriate power backup equipment for their systems. The Atlas AVM manufacturer recommends use of an Uninterruptible Power Supply unit (UPS).

The UPS is a battery backup system that ensures continuous operation for the voice messaging system during periods of minor AC power irregularity. In the event of a power loss of longer duration, some UPS units can initiate a graceful shutdown of the system so that no data is lost.

The capacity of the UPS unit required can vary with the number of pieces of equipment to be protected, the current drawn by such equipment, the desired duration of battery backup power, and other factors. The customer is responsible for ensuring that the capacity of the UPS unit is sufficient to provide the required protection.

Not all UPS units provide surge protection. The Atlas AVM manufacturer recommends the use of a UPS unit that provides this feature.

## Hardware Specifications

Certain hardware specifications and configurations are required before a PC can be utilized for Atlas AVM voice processing.

Component	4-12 Port System	16-20 Port System	24 Port System
CPU	386, 486DX, or Pentium – Intel	486DX to Pentium – Intel	486dx2 to Pentium – Intel
RAM	4 MB	8 MB	12 MB
Bus	ISA	ISA	ISA
Speed	25-66 MHz or better	33-66 MHz or better	66 MHz or better
Hard drive	IDE or SCSI	IDE or SCSI	IDE or SCSI
BIOS	AMIBIOS	AMIBIOS	AMIBIOS
Power supply	200 watt UL approved	250 watt UL approved	300 watt UL approved

**Figure 7-1: Minimum Hardware Specifications**



**Note:** The maximum number of ports for an Atlas AVM system is 64.

Device/Port	Usage	IRQ	Address
Serial Port (COM1)	SMDI Serial Integration	4	3F8H
Serial Port 2 (COM2)	Modem	3	2F8H
Serial Port 3 (COM3)	DISABLED	N/A	N/A
Serial Port 4 (COM4)	DISABLED	N/A	N/A
Parallel Port 1 (LPT1)	Hardware Lock/Printer	7	378H
Parallel Port 2 (LPT2)	DISABLED	N/A	N/A
Game Port (If on I/O Card)	DISABLED	N/A	N/A
Voice Board	Call Processing	5	D000* (memory address)
Video Board	Display	N/A	N/A**

**Figure 7-2: Suggested Hardware Configuration**

\*The first voice board should be set to base memory address D000H and offset 0H. Each board requires a total offset of 200H. So, for a two board installation, the first should be set at address D000H-D1FFH and the second should be set at address D200H-D3FFH.

\*\*The video card memory and BIOS must not conflict with the voice board. Most video cards do not access memory above BFFFH, but the BIOS shadowing can create problems if an attempt is made to shadow into the voice board's address segment. This is true for system and other BIOS shadowing as well as other reserved areas of memory like the EMS page frame.

It is recommended that you provide a 9600 baud modem (or higher) for maintenance and remote support.

## Hard Drive Size

VoiceMail Users	Hours of Storage	Recommended Size	Hard Drive Size
25	2	2 port/ 2 hour	100 MB
100	5	4 port/ 10 hour	200 MB
250	12	8 port/ 12 hour	250 MB
500	25	16 port/ 25 hour	500 MB

**Figure 7-3: Recommended Hard Drive Size**



**Note**  
 10 Mb equals  
 approximately 55  
 minutes of voice  
 storage

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## Connecting Atlas AVM to a Telephone System

Since voicemail is an add-on component, it is dependent on the telephone system for all integration features. Some telephone system manufacturers have programmed their systems to work well with voicemail. Some, however, have made their systems work only with certain voicemail systems (they sell a specific voicemail system with their telephone system), and some manufacturers have done nothing to enhance their systems.

**Note**

Remember—if a single line telephone (2500 set) can do it, so can the Atlas AVM.

This chapter explains alternatives to achieve the best possible integration your telephone switch can provide.

Some telephone systems may be cumbersome to operate or cause users to do extra functions in order to achieve their goals, but the main objective is to ensure that all voicemail features possible are available to users.

## Telephone System Integration Definitions

These definitions explain the references in the *Atlas Telephone System Compatibility Listings*.

### Full Integration

- Call forward station to personal greeting
- Message waiting indication on telephone set
- Call transfer to attendant or any extension
- Outdial notification

### Partial Integration

- Call forward station to generic greeting (customer to enter mailbox number)
- May or may not have message waiting indication on station
- Call transfer to attendant or any extension
- Outdial notification

## Restricted Integration

Call forward station to generic greeting (customer to enter mailbox number)

No message waiting

No transfers allowed

No outdial notification

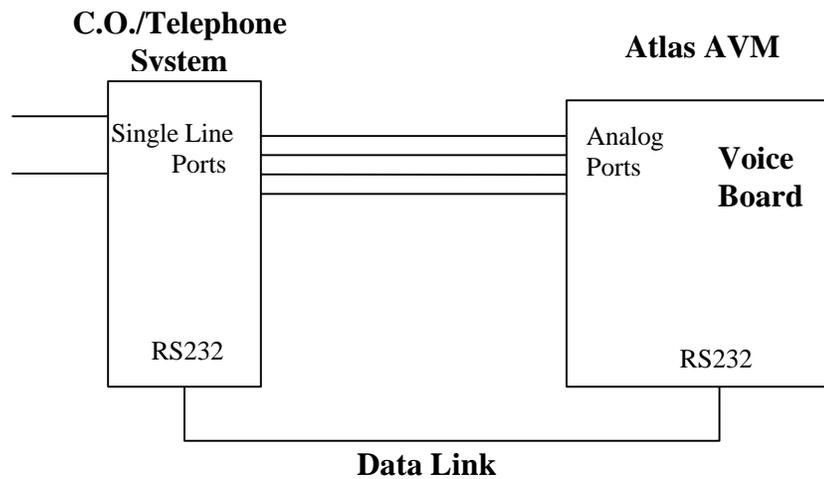
All other features, like group messages, date and time stamp etc. are dependent on the voicemail system.

## Integration Types

There are many types of integration that are designed to offer the most in voicemail features. Listed below are some ways to achieve this.

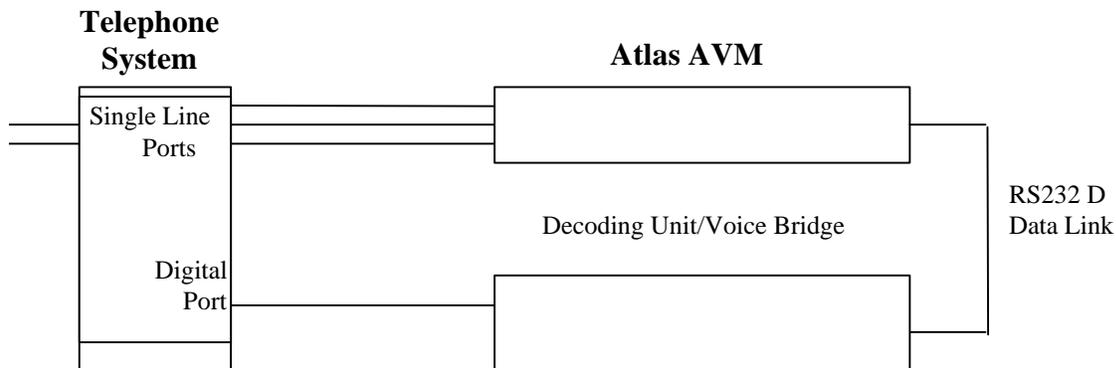
### Serial Integration

This is where the telephone system or C.O. is connected to the voicemail unit via an RS232 link which passes call party information. This type of integration offers all available voicemail features.



**Figure 8-1: Example of Serial Integration**

## Serial Integration with Additional Equipment



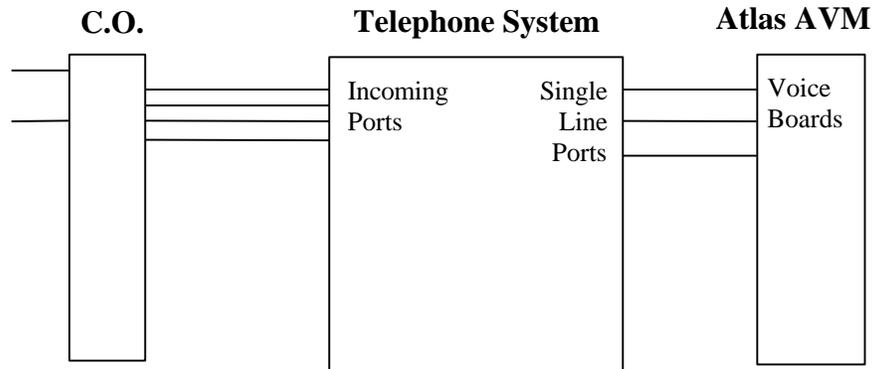
**Figure 8-2: Serial Integration with a Voice Bridge**

The voice bridge unit gives the customer a full featured system by hooking up to an electronic station and “reading the screen.” The voice bridge then converts this information to SMDI protocol and sends the call data over the RS232 link to the voicemail system. The voice bridge unit is used on the AT&T system 75/85, Rolm CBX and Northern Telecom SL-1.

**In-band integration:** In this type of integration voicemail is connected to the telephone system via voice lines. The telephone system passes data via DTMF keys, giving the voicemail information about the type of call that it is handling. For example:

- The telephone system rings voicemail.
- Voicemail goes off-hook.
- The telephone system sends DTMF keys stating:
  - Call type
  - Calling Number
  - Called Number

This enables voicemail to know what to play with each call. The personal greeting, company greeting or auto attendant greeting. This type can offer some or all features, depending on the sophistication of the telephone system.



**Figure 8-3: Example of In-Band Integration**

## Possible Integration Features

Due to the many telephone systems on the market today, it would be impossible to show every connection and program setting available. Below is a list of features given as a reference for you to investigate every possible means of connecting Atlas AVM to a telephone system.

### Voice Connections

This entails having a voice channel from the telephone system to Atlas AVM.

Using a Single Line Station: This is a typical way for Atlas AVM to interface with a telephone system. It offers a talk path that generally can hook flash transfer a caller. DTMF is dependent on the calling extension. The telephone system may allow turning on and off a message waiting indicator.

Using an OPX port on a hybrid system: Most telephone systems that offer OPX ports have the same privileges as a single line set.

### Call Transfers

The telephone system has the ability to hook flash and dial an extension and hear ring back, busy or error tones.

Single line ports usually allow hook flash transfers to the attendant or any extension.

OPX ports usually allow hook flash transfers, but be cautious of transferring on intercom paths. Check to make sure multiple calls can transfer to actual stations and not just light up an intercom path.

## **Outdials**

The telephone system must have at least one port to do outdials on. This could be a single line telephone port from the telephone company which would be used for outdial notification.

## **Call Forward to Personal Greeting**

A station user can forward a telephone to Atlas AVM and have it play the box owner's personal greeting.

The telephone system can forward a station to a voice port, then pause, wait for answer, then out pulse DTMF digits (the called extension).

The telephone system can forward a station to a speed dial number (bin). That speed dial dials the voicemail extension, pauses, then out dials DTMF digits (the called extension).

Depending on its capabilities, the telephone system may be able to forward a station off-premise. This allows you the ability to forward to a trunk, pause, then outdial the called extension number. This is usually done by taking the tip and ring from an out-going loop-start trunk and tying it down on a single line extension number. When the trunk is seized you get internal dial-tone instead of C.O. dial tone. Then you dial the voicemail number, pause, and dial in the called extension number. You hear the personal greeting.

## **Message Waiting Indication**

The telephone system allows a voicemail port to dial an access code, then the extension number, in order to light a message waiting light.

The availability of this feature is totally dependent on the telephone system. Atlas AVM turns on and off the message waiting light by going off hook, dialing a feature code, then the extension number with the message, and going back on hook.

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## ***Integrating with the Telephone System***

### **System Default Configuration**

The Atlas AVM system comes on-line using default parameters that work with a Standard Telephone System. This means that if you are connecting the system to analog single line station ports and the telephone system uses North American standards for ringing, tones and hook-flash intervals, you may not need to make any changes listed in this chapter.

Follow the instructions in this chapter only if you wish to modify some integration information. Check specifically the parameters concerning:

- Message Waiting Light/Indicator
- In-Band Data Packets
- Call Transfer Sequence

If your system is standard please go to the *System Administrator's Manual* to set up mailboxes and record greetings. You need not make any changes in this section.

For non-standard types of telephone systems, *refer* to the individual document for that specific telephone system in the *Atlas Telephony Integration Guide*. Go through all of the integration screens in this chapter and modify them to match your telephone system.

Refer to the specific sections in this chapter, or call Key System US Technical Support Services if you need help completing your telephone system integration.

**Key System US Technical Support Services**

(800) 833-3446

8:30 A.M. to 5:00 P.M. Eastern Standard Time

Monday through Friday (excluding holidays)

**24hr** technical support is also available upon request.

## Understanding the Integration Screens



### Note

Port and global parameters should only be changed under the direction of a qualified Atlas AVM installer or technician.

### Parameters

The screens in this manual are shown with the field default parameters. Each field description explains the type of data to be entered, including range and resolution if necessary.

### Voice Board Considerations

Some of the integration parameters must be set according to what type of voice board you are using in your Atlas AVM system. When that is the case, the screens for both Dialogic and Rhetorex voice boards are shown in this manual. Read the information that pertains to your voice board type and disregard the information for other types.

## Accessing the Integration Screens

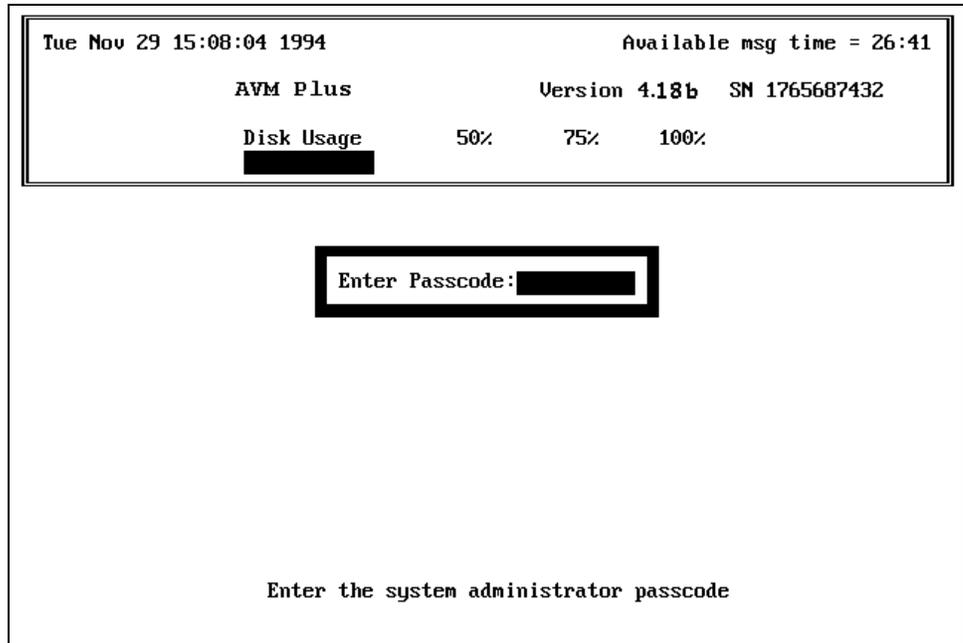
This section will walk you step by step through the installation screens. Use this section for full explanations of all screen languages and system defaults.

Access to the Atlas AVM integration screens is limited to authorized Atlas AVM installers. Entering the *installation* passcode (default is **Atlas**) at the Main Status screen allows access to all screens, including the Integration screens. Entering the *administration* passcode (default is **1234**) does not provide access to the Integration screens.

The installation passcode can be changed in the Administration General Parameters menu if accessed using the installation passcode. If the menus are accessed using the administration passcode, the installation passcode is not displayed and cannot be changed.

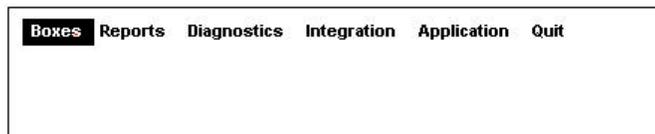
Do not give the installation passcode to system administrators or other unauthorized personnel. Attempting to change, set, or reprogram integration field information by unauthorized personnel could cause the system to malfunction.

From the Main Status screen, press <F10>. The passcode entry box appears, where you must enter the default installation passcode —Atlas. During programming please change this passcode and guard it. Damage to the system can occur if values are changed improperly.



**Figure 9-1: The Passcode Entry Box on the Main Status Screen**

After you have entered the correct passcode, the System Administration Menu appears where you can select integration options.



**Figure 9-2: The System Administration Menu**

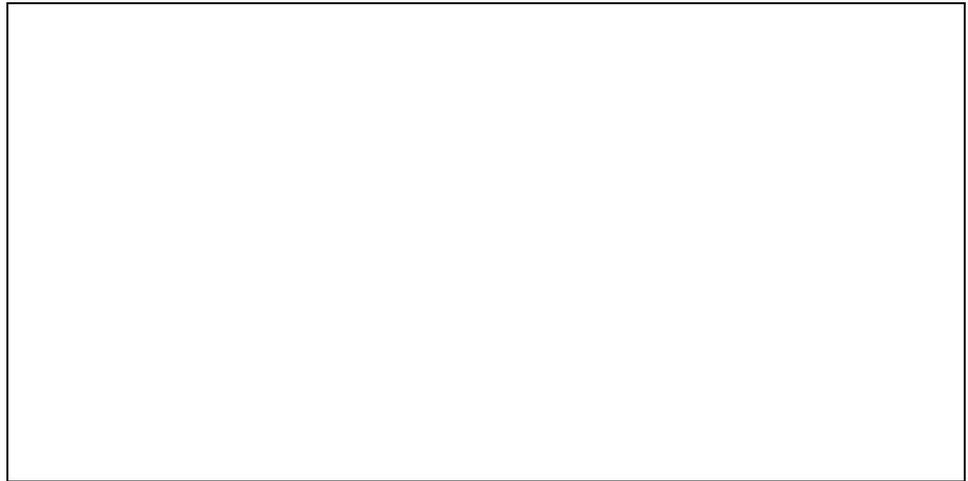
The two Integration options are Global and Port.

The Global Parameters screens set duration and control parameters for all channels on the voice board.

The Port Parameters screens set duration and control parameters for individual channels on the voice board.

## Global Parameters

Select Integration→Global→Edit to access the Edit Global Telephone Parameters screens.



**Figure 9-3: Global Options**

The Global Parameters screens differ depending on which type of voice boards you have.

The following screens and panels are available under Integration→Global→Edit

Edit Global Parameters Screen 1

- Telephone System
- Dialogic Control Block or Rhetorex Global Parameters
- Control (Dialogic only)

Edit Global Parameters Screen 2

- Serial Port
- Integration Control

Edit Global Parameters Screen 3

- Serial Packet Separation (4 screens)

## The Edit Global Parameters Screen 1

The first screen that appears is Edit Global Parameters Screen 1. This screen contains the Telephone System panel, either the Dialogic Control Block or the Rhetorex Global Parameters panel and the Control panel (Dialogic only).



**Note**  
The Rhetorex Global Parameters panel is shown on page 9-9.

```

EDIT GLOBAL PARAMETERS SCREEN 1
-----
TELEPHONE SYSTEM
Name: STANDARD TELEPHONE SYSTEM TYPE
-----
DIALOGIC CONTROL BLOCK

flashchr:8          flashtn:50          pausetm:200
digrate:6053       sch_tm:20          p_bk:6
p_mk:4             p_idd:100         t_idd:5
oh_dly:50          r_on:3            r_off:5
r_ird:80          s_bnc:4           ttdata:10
minpdn:2          minpdoff:2        minipd:25
minlcoff:-1       maxpdoff:50
-----
CONTROL

System start delay time:0 seconds
-----
ESC to exit  F10 to save  PgDn next screen
    
```

*Figure 9-4: The Telephone System and, Dialogic Control Block and Control Panels*

## Telephone System Panel

The Telephone System panel displays Standard Telephone System Type as the default. Select your telephone system name, if it is available, from the list that appears. If it is not available, use the default.

At this point, the default parameters for the telephone system you have chosen are automatically loaded on the Edit Global Parameters screens.

## Dialogic Control Block Panel

Do not change any of these parameter settings unless instructed by a technician.

### **flashchr**

Flash Character: Character in the dialing string that causes a hook flash to occur.

Length: 1. Default: &. Units: ASCII

### **digrate**

Digitization Rate: Play/record digitization rate. Default play/record digitization rate for the system. 6000 Hz or 8000 Hz on DSP voice boards; 6053 Hz or 8117 Hz on non-DSP voice boards. On DSP voice boards you can override digrate per channel by specifying the sampling rate for the play/record function.

Length: 2. Default: 6000 (DSP voice boards) or 6053 (non-DSP voice boards). Units: Hz.

### **p\_mk**

Pulse Make: Duration of the pulse dial on-hook interval.

Length: 2. Default: 4. Units: 10 ms.

### **oh\_dly**

Off-Hook Delay: Time delay that occurs after executing a channel off-hook command. DTMF digits are not recognized while going off-hook. If your application requires immediate DTMF, wink, or loop current drop detection upon going off-hook, set **oh\_dly** to 1.

Length: 1. Default: 50. Units: 10 ms.

### **r\_ird**

Ring Interring Delay: Maximum time between incoming rings. The ring counter is reset after this time has expired; rings that occur after this interval are considered the start of a new call cycle.

Length: 2. Default: 80. Units: 100 ms.

### **minpdon**

Minimum Pulse Digit On: Minimum make interval to validate an incoming loop pulse digit.

Length: 1. Default: 2. Units: 10 ms.

**minlcoff**

Minimum Loop Current Off: Minimum time that loop current must be off.

-1: Disable this condition.

Length: 2. Default: -1/ Units: 10 ms/

**flashtm**

Flash Time: Length of time on hook during a hook flash.

Length: 2. Default: 50. Units: 10 ms.

**sch\_tm**

Scheduler Time: Maximum number of clock ticks in which to perform scheduler activities.

Length: 2. Default: 20. Units: 1 clock tick (1/18 sec).

**p\_idd**

Pulse Interdigit Delay: Delay between digits during pulse dialing.

Length: 2. Default: 100. Units: 10 ms.

**r\_on**

Ring On: Minimum ring voltage on duration for an incoming ring.

Length: 2. Default: 3. Units: 100 ms.

**s\_bnc**

Silence Debounce: Minimum time that silence must be present to produce a T\_SILON or T\_SILOFF CST event.

**minpdoff**

Minimum Pulse Digit Off: Minimum break interval to validate an incoming loop pulse digit.

Length: 1. Default: 2. Units: 10 ms.

**maxpdoff**

Maximum Pulse Digit Off: Maximum make interval to validate an incoming loop pulse digit.

Length: 2. Default: 50. Units: 10 ms.

Pause Time: Delay caused by a comma (,) in the dialing string.

Length: 2. Default: 200. Units: 10 ms.

**p\_bk**

Pulse Break: Duration of the pulse dial off-hook interval.

Length: 2. Default: 6. Units: 10 ms.

**t\_idd**

DTMF Interdigit Delay: Delay between digits during DTMF dialing.

Length: 2. Default: 5. Units: 10 ms.

**r\_off**

Ring Off: Minimum ring voltage off duration for an incoming ring.

Length: 2. Default: 5. Units: 100 ms.

**ttdata**

DTMF Time Data: Duration of the DTMF digits in out-bound tone dialing.

Length: 1. Default: 10. Units: 10 ms.

**minipd**

Minimum Interpulse Digit: Minimum time between loop pulse digits.

Length: 2. Default: 25. Units: 10 ms.

**Control Panel**

**System start delay time**

The system start delay time can be increased from the default, 0, to allow time for the voice boards to load during start up.

If there are not enough ports displayed on the Port Status screen, it may be that the voice boards did not have time to load and you should increase the system start delay time. This is especially true for systems with a high number of ports.

To increase the system start delay time, enter a number between 0 and 99 in this field.

**Rhetorex Global Parameters Panel**

The global parameters for Rhetorex voice boards are found below the Telephone System panel on the Edit Global Parameters Screen 1.

EDIT GLOBAL PARAMETERS SCREEN 1		
TELEPHONE SYSTEM		
Name: STANDARD TELEPHONE SYSTEM TYPE		
RHETOREX GLOBAL PARAMETERS		
FlashChar: 8	FlashDur: 50	PauseDur: 200
PulseBreak: 6	PulseMake: 4	PulseDly: 100
DtmfDly: 5	OffDly: 50	MinRing: 3
MinOff: 5	RingReset: 30	MinSilence: 4
DtmfOn: 10	MinPulseOn: 2	MinPulseOff: 2
MinDigitSep: 25	LcOff: -1	RecallDelay: 50
MaxPulseOn: 50		
ESC to exit F10 to save PgDn next screen		

**Figure 9-5: The Telephone System and Rhetorex Global Parameters Panels**

### FlashChar

Flash Character: the character that causes a flash hook when dialing.

Usage: character. Use the default value—&.

### PulseBreak

The duration of the break interval during pulse dialing.

Usage: Numeric, Units = 10 msec. Use the default value—6 (60 msec.)

DTMF Delay: the interval between DTMF digits during tone dialing.

Usage: Numeric, Units = 10 msec. Use the default value—5 (50 msec.).  
When the HOST parameter in CONFIG is UK, the default value is 8 (80 msec.).

### MinOff

(Minimum off—msec.) The minimum necessary duration of a valid inbound ring's on period.

Usage: Numeric, Units = 100 msec. Use the default value—5 (500 msec.).

### DtmfOn

The DTMF tone duration when dialing.

Usage: Numeric, Units = 10 msec. Use the default value—10 (100 msec.).

### MinDigitSep

Minimum Digit Separation: the minimum necessary off period duration for a valid loop pulse digit.

Usage: Numeric, Units = 10 msec. Use the default value—25 (250 msec.).

### **MaxPulseOn**

Maximum Pulse On: the maximum allowable on period for a valid loop pulse digit.

Usage: Numeric, Units = 10 msec. Use the default value–50 (500 msec.)

### **FlashDur**

Flash duration: the duration of the on–hook state during a flash hook.

Usage: Numeric, Units = 10 msec. Use the default value–50 (500 msec.).

### **PulseMake**

The duration of the make interval during pulse dialing.

Usage: Numeric, Units = 10 msec. Use the default value–4 (40 msec.)

### **OffDly**

Off Delay: the length of time that the driver delays before posting event 8 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Use the default value–50 (500 msec.).

### **RingReset**

The maximum allowable no ring signal before the driver reset rings.

Usage: Numeric, Units 100 msec. Use the default value–80 (8 sec).

### **MinPulseOn**

Minimum Pulse On: the minimum necessary on period duration for a valid loop pulse digit.

Usage: Numeric, Units = 10 msec. Use the default–2 (20 msec.).

### **LcOff**

Loop Current Off: the minimum necessary duration of loop current off before the driver posts event 20 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Use the default value, –1 (Disabled or none).

### **PulseDur**

Pulse Duration: the duration of pulse digits during pulse dialing.

Usage: Numeric

Pulse Delay: the interval between pulse digits during pulse dialing.

Usage: Numeric, Units = 10 msec. Use the default value–100 (1 sec.).  
When the HOST parameter in CONFIG is UK, the default value is 82  
(820 msec.).

### **MinRing**

Minimum Ring: the minimum necessary duration of a valid inbound  
ring's on period.

Usage: Numeric, Units = 100 msec. Use the default value–3 (300 msec.).

### **MinSilence**

Minimum Silence: the minimum necessary silence that must occur  
before the driver posts event 23 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Use the default value–4 (40 msec.)

### **MinPulseOff**

Minimum Pulse Off: the minimum necessary duration off period for a  
valid loop pulse digit.

Usage: Numeric, Units = 10 msec. Use the default value–2 (20 msec.).

Earth Recall Delay.

Usage: Numeric, Units = 10 msec. Use the default value–50 (500 msec.)

When you have verified the fields on this screen, press <PgDn> to go to  
the second screen.

## The Serial Port /Integration Control Panels

```

EDIT GLOBAL PARAMETERS SCREEN 2
SERIAL PORT
Enabled:< * >
Com port:1
Interrupt:4
Data transfer rate:300 bits per second
Data bits:7
Parity:Even
Stop bits:2
INTEGRATION CONTROL
Data packet timeout:1 seconds
Data packet terminator:00 hexadecimal
Binary data packets:< * >
Alert extension number offset:0
Alert extension number length:0
ESC to exit F10 to save PgUp previous screen PgDn next screen

```

Figure 9-6: The Serial Port and Integration Control Panels

### The Serial Port Panel

#### Enabled

If this field is enabled, Atlas AVM expects call information from the telephone system to be sent over the serial port, i.e. SMDI or other serial integrated systems.

Use the <Spacebar> to toggle the field on or off. (<\*> indicates enabled.)

#### COM port

This is the number of the Com port used for serial integration. COM1 is recommended.

Enter a number between 1 and 4.

#### Interrupt

This is the IRQ for the serial integration Com port. IRQ4 for COM1 is recommended.

Enter a number between 1 and 7.

### **Data transfer rate**

This is the average number of bits, characters, or blocks per unit time passing in a data transmission system. This must match your telephone system.

300  
1200  
2400  
4800  
9600

Select the desired rate and press <Enter>.

### **Data bits**

This is the data bit length that your telephone system will send. Generally this is set to 8.

Enter a number between 5 and 8.

### **Parity**

Parity is a process for detecting whether bits of data have been altered during transmission of that data.

This field specifies the type of Parity setting chosen for your system.

When you highlight this field, a submenu appears with the following setting options:

- None:** tells the system to ignore parity bits.
- Odd:** sets the parity bit at 1 if the number of parity bits in the data bits is even and at 0 if the number of 1s is odd.
- Even:** sets the parity bit at 0 if the number of 1's in the data bits is even and at 1 if the number of 1s is odd.
- Mark:** sets parity at 1.
- Space:** sets parity at 0.

Select the desired option and press <Enter>.

### **Stop bits**

This is a special signal that indicates the end of that character.

Enter a number between 0 and 2.

## The Integration Control Panel

### Data packet timeout

This is the number of seconds you want Atlas AVM to wait for a data packet, after receiving a data packet, before assuming that the reception is complete.

Enter a number between 1 and 9.

### Data packet terminator

Enter the character that signifies the end of a packet. If there is a terminating character, Atlas AVM does not have to wait for timeout.

Valid characters for this field are a,b,c,d,#,\* and digits 0-9. (See Figure 9-12 for an explanation of these characters.)

### Binary data packets

Use the <Spacebar> to toggle this field on or off. (<\*> indicates on.)

### Alert extension number offset

The alert extension is the physical extension that Port 1 is tied to. The offset is the location, given as a number, that signifies the order in the data packet where the characters begin.

Enter a number between 0 and 20.

### Alert extension number length

This is the number of digits in the alert extension number.

Enter a number between 0 and 10.

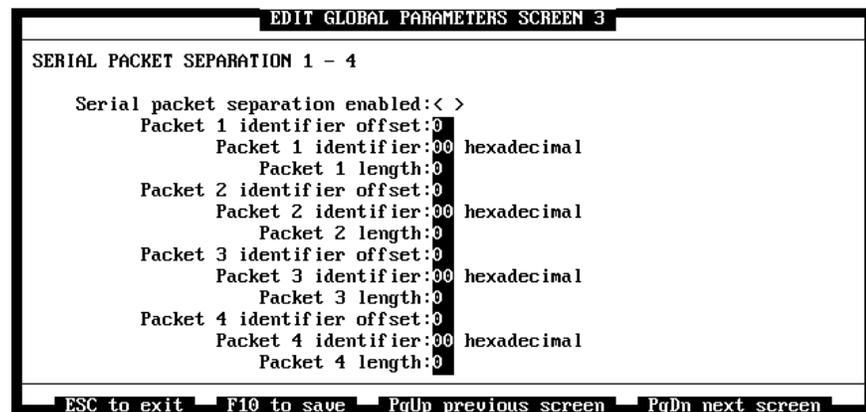


Figure 9-7: The Serial Packet Separation Screens

Figure 9-7 is an example of the Serial Packet Separation screens. Each screen covers 4 serial packet separations. The Serial Packet Separation Enabled field appears once at the top of each screen.

**Serial packet separation enabled**

Press the <Spacebar> to toggle this field on or off. (<\*> indicates enabled.)

The three fields explained below appear for each serial packet separation.

**Packet identifier offset**

Specify where the called packet identifier begins.

Enter a number between 0 and 20.

**Packet identifier**

Each character in this field must be entered as a hex character. (*See* the conversion chart in Figure 9-8.)

**Packet length**

Specify the length of the called packet identifier.

Enter a number between 0 and 20.

The chart in Figure 12-8 shows a list of ASCII characters and their corresponding hex character codes.

ASCII Character	Hex Character	ASCII Character	Hex Character	ASCII Character	Hex Character
0	30	Q	51	q	71
1	31	R	52	r	72
2	32	S	53	s	73
3	33	T	54	t	74
4	34	U	55	u	75
5	35	V	56	v	76
6	36	W	57	w	77
7	37	X	58	x	78
8	38	Y	59	y	79
9	39	Z	5A	z	7A
A	41	a (DTMF)	61	@	40
B	42	b (DTMF)	62	#	23
C	43	c (DTMF)	63	&	26
D	44	d (DTMF)	64	*	2A
E	45	E	65	,	2C
F	46	F	66		
G	47	G	67		
H	48	H	68		
I	49	I	69		
J	4A	J	6A		
K	4B	K	6B		
L	4C	L	6C		
M	4D	M	6D		
N	4E	N	6E		
O	4F	O	6F		
P	50	P	70		

**Figure 9-8: Hex Conversion Chart**

## Port Parameters

Select Integration→Port→Edit to access the Edit Port Telephone Parameters screens.

The following screens and panels are available under Integration→Port→Edit:

Edit Port Telephone Parameters Screen 1

- Telephone System
- Dialogic Channel Parameter Block or Rhetorex Channel Parameters

Edit Port Telephone Parameters Screen 2

- Sequences
- Message Waiting Light Sequences

Edit Port Telephone Parameters Screen 3

- Control

Edit Port Telephone Parameters Screen 4

- Tone Termination (Dialogic only)
- Recording Termination

Edit Port Telephone Parameters Screen 5

- Integration
- In-Band Supervision

Edit Port Telephone Parameters Screen 6

- In-Band Signaling

Edit Port Telephone Parameters Screen 7

- Integration Data Packet (16 available)



**Note**

The Rhetorex Channel Parameters panel is shown on Page 9-25.

### Edit Port Telephone Parameters Screen 1

The first screen that appears under is Edit Global Parameters Screen 1. This screen contains the Telephone System panel and either the Dialogic or Rhetorex Channel Parameter Block panel.

```

EDIT PORT 01 TELEPHONE PARAMETERS SCREEN 01
-----
TELEPHONE SYSTEM
Name: STANDARD TELEPHONE SYSTEM TYPE
-----
DIALOGIC CHANNEL PARAMETER BLOCK
-----
dtp1_dly:5      dt_edge:2      dtrc_dly:0      sb_siz:200
stdely:25      cnosig:4000   lcdly:400       lcdly1:10
hedge:2        cnosil:650   lo1tola:13      lo1tolb:13
lo2tola:13     lo2tolb:13   hi1tola:13      hi1tolb:13
lo1bmax:90     lo2bmax:90   hi1bmax:90      nsbusy:0
logltch:15     higtch:19    lo1rmax:90      lo2rmin:225
intflg:1       intfltr:10   spdeb:10        hisiz:90
alowmax:700    blowmax:530  nbrbeg:1        hi1ceil:78
lo1ceil:58     lowerfrq:900 upperfrq:1000   timefrq:5
re_jctfrq:20   maxansr:1000 ansrdgl:-1      pudmxper:0
pudszwnd:0     puddly:0     mxtimefrq:0     lower2frq:0
upper2frq:0    time2frq:0   mxtime2frq:0   lower3frq:0
upper3frq:0    time3frq:0   mxtime3frq:0
-----
ESC to exit  F10 to save  PgDn next screen  F5 prev port  F6 next port

```

**Figure 9-9: The Telephone System and Dialogic Channel Parameter Block Panels**

## Telephone System Name

The Telephone System panel displays Standard Telephone System Type as the default. Select your telephone system name, if it is available, from the list that appears. If it is not available, use the default.

At this point, the default parameters for the telephone system you have chosen are automatically loaded on the Edit Port Telephone Parameters screens.

## Dialogic Channel Parameter Block

The Channel Parameter Block is used for input to the setcparm() function to modify channel parameters.

Most fields in the Channel Parameter Block are used for Call Progress Analysis (CPA). The following fields are not used for Call Progress Analysis:

- dtp1\_dly
- sb\_siz
- dt\_edge
- spdeb
- dtrc\_dly

**DTMF Play Delay:** Minimum time for DTMF to be present during playback to be considered valid. When a suspected DTMF signal is detected, playback is turned off for the specified duration to determine the source of the signal and prevent termination of playback by incorrectly interpreting voice data as DTMF. If the suspected DTMF

signal stops, playback resumes. Only in effect during playback. Minimum value to use is 5. Increase for more play-off immunity and less sensitivity to DTMF. *See also* **dtrc\_dly**.

-1: Disable this condition.

Length: 1. Default: 5. Units: 10 ms. Restrictions: Supported on non-DSP voice boards only.

### **cnosig**

CPA - Continuous No Signal: The maximum time of silence (no signal) allowed immediately after Cadence Detection begins. If exceeded, a *no ringback* is returned.

Length: 2. Default: 4000. Units: 10 ms.

### **lo1tola**

CPA - Low 1 Tolerance Above: Percent acceptable positive deviation of short low signal.

Length: 1. Default: 13. Units: %

### **hi1tolb**

CPA - High 1 Tolerance Below: Percent acceptable negative deviation of high signal.

Length: 1. Default: 13. Units: %

### **higlthc**

CPA - High Glitch: The maximum nonsilence period to ignore. Used to help eliminate spurious nonsilence intervals.

Length: 2. Default: 19. Units: 10 ms.

### **spdeb**

Silence Pointer Debounce: When recording with silence compression, this is the amount of silence that is recorded on the trailing edge of audio. *See also* **sb\_siz**.

Length: 2. Default: 10. Units: 10 ms.

### **hi1ceil**

CPA - High 1 Ceiling: Maximum 2nd high duration for a retrain.

Length: 2. Default: 78. Units: 10 ms.

### **rejectfrq**

CPA - Reject Frequency: The percentage of time the audio signal is allowed to be outside the chosen frequency range, based on **timefrq**. When the parameter **rejectfrq** is increased, it allows a signal with more noise to be accepted as a valid frequency.

When a frequency within the chosen range is detected, Call Progress Analysis compares the length of time the signal remains inside the chosen range to **timefrq**. The signal is still considered valid if it is outside the chosen range not more than **rejectfrq** percent of **timefrq**.

For example: Using the default values, in order to be a valid tone, the signal must stay in bounds for at least 5 x 10 ms, or be out of bounds 20 percent of the time: for a total of 10 milliseconds.

If the signal goes outside the chosen range more than **rejectfrq** percent, Frequency Detection immediately starts over, as if the tone just started.

Length: 1. Default: 20. Units: %. Restrictions: Supported on non-DSP voice boards only.

### **dt\_edge**

DTMF Detection Edge: The edge in which the DTMF digit will be reported:

1: Trailing edge.

2: Leading edge.

Length: 1. Default: 2. Units: edge.

### **lcdly**

CPA - Loop Current Delay: The delay after dialing has been completed and before beginning Loop Current Detection.

-1: Disable Loop Current Detection.

Length: 2. Default: 400. Units: 10 ms.

### **lo1tolb**

CPA - Low 1 Tolerance Below: Percent acceptable negative deviation of short low signal.

Length: 1. Default: 13. Units: %.

### **lo1bmax**

CPA - Low 1 Busy Maximum: Maximum interval for short low for busy.

Length: 2. Default: 90. Units: 10 ms.

### **hisiz**

CPA - High Size: Used to determine whether to use **alowmax** or **blowmax**.

Length: 2. Default: 90. Units: 10 ms.

### **lo1ceil**

CPA - Low 1 Ceiling: Maximum 1st low duration for a retrain.

Length: 2. Default: 58. Units: 10 ms.

### **maxansr**

CPA - Maximum Answer: The maximum allowable length of **ansrsize**. When **ansrsize** exceeds **maxansr**, a *connect* is returned to the application.

Length: 2. Default: 1000. Units: 10 ms.

### **dtrc\_dly**

DTMF Record Delay (DTMF Debouncer): Minimum time for DTMF to be present to be considered valid. In effect at all times. Used to remove false DTMF signals during recording. For most applications, you rarely need to change the default value. If this value is larger than **dtpl\_dly**, this value overrides **dtpl\_dly** for playback. Minimum value to use is 6. Increase for more talk-off immunity and less sensitivity to DTMF.

Length: 2. Default: 0. Units: 10 ms. Restrictions: Supported on non-DSP voice boards only.

### **lcdly1**

CPA - Loop Current Delay 1: The delay after Loop Current Detection detects a transient drop in loop current and before Call Progress Analysis returns a *connect* to the application.

Length: 2. Default: 10. Units: 10 ms.

### **lo2tola**

CPA - Low 2 Tolerance Above: Percent acceptable positive deviation of long low signal.

Length: 1. Default: 13. Units: %

### **lo2bmax**

CPA - Low 2 Busy Maximum: Maximum interval for long low for busy.

Length: 2. Default: 90/ Units: 10 ms.

### **lo2rmin**

CPA - Low 2 Ring Minimum: Maximum long low duration of double ring.

Length: 2. Default: 225. Units: 10 ms.

### **alowmax**

CPA - A Low Maximum: Maximum low before connect if high > **hisiz**.

Length: 2. Default: 700. Units: 10 ms.

### **lowerfrq**

CPA - Lower Frequency: Lower bound for 1st tone in an SIT.

Length: 2. Default: 900. Units: Hz.

### **ansrdgl**

CPA - Answer Deglitcher: The maximum silence period allowed between words in a salutation. This parameter should be enabled only when you are interested in measuring the length of the salutation.

-1: Disable this condition.

Length: 2. Default: -1. Units: 10 ms.

### **sb\_siz**

Silence Buffer Size: When recording with silence compression, this is the amount of silence that is recorded on the leading edge of an audio. You can use the following equation to convert the size in bytes to the length in seconds:  $\text{Kbps/bytes} = \text{time}$ . See also **spdeb**.

Length: 2. Default: 200. Units: bytes. Range: 1-400 (see D40DRV -C option to override the upper limit of the range).

### **hedge**

CPA - Hello Edge: The point at which a *connect* will be returned to the application.

1: Rising Edge (immediately when a connect is detected).

2: Falling Edge (after the end of the salutation).

Length: 1. Default: 2. Units: edge.

### **lo2tolb**

CPA - Low 2 Tolerance Below: Percent acceptable negative deviation of long low signal.

Length: 1. Default: 13. Units: %.

### **hi1bmax**

CPA - High 1 Busy Maximum: Maximum interval for first high for busy.

Length: 2. Default: 90. Units: 10 ms.

### **intflg**

CPA - Intercept Mode Flag: This parameter enables or disables Frequency Detection and/or Positive Voice Detection and selects the mode of operation for Frequency Detection.

OPTEN: Enable Frequency Detection and wait for detection of a connect using Cadence Detection, Loop Current Detection, or Positive Voice Detection before returning an *intercept*.

OPTDIS: Disable Frequency Detection and PVD.

OPTCON: Enable Frequency Detection return an *intercept* immediately after detecting a valid frequency.

PVD\_ENABLE: Enable PVD.

PVD\_OPTEN: Enable PVD and OPTEN.

PVD\_OPTCON: Enable PVD and OPTCON.

Length: 1. Default: OPTEN.

### **blowmax**

CPA - B Low Maximum: Maximum low before connect if high < **hisiz**.

Length: 2. Default: 530. Units: 10 ms.

### **upperfrq**

CPA - Upper Frequency: Upper bound for 1st tone in an SIT.

Length: 2. Default: 1000. Units: Hz.

CPA - Start Delay: The delay after dialing has been completed and before starting analysis for Cadence Detection, Frequency Detection, and Positive Voice Detection.

Length: 2. Default: 25. Units: 10 ms.

### **cnosil**

CPA - Continuous No Signal: The maximum time of silence (no signal) allowed immediately after Cadence Detection begins. If exceeded, a *no ringback* is returned.

Length: 2. Default: 4000. Units: 10 ms.

### **hi1tol**

CPA - High 1 Tolerance Above: Percent acceptable positive deviation of high signal.

Length: 1. Default: 13. Units: %.

### **logltch**

CPA - Low Glitch: The maximum silence period to ignore. Used to help eliminate spurious silence intervals.

Length: 2. Default: 15. Units: 10 ms.

### **intfltr**

CPA - Intercept Filter: The minimum signal to qualify frequency detection. This parameter specifies how much of the last part of the audio signal is ignored during frequency averaging.

-1: Disable this condition.

Length: 1. Default: 10. Units: 10 ms. Restrictions: Supported on non-DSP voice boards only.

### **nbrbeg**

CPA - Number Before Beginning: Number of nonsilence periods before analysis begins.

Length: 1. Default: 1. Units: rings.

CPA - Time Frequency: Minimum time for 1st tone in an SIT to remain in bounds. The minimum amount of time required for the audio signal to remain within the frequency detection range specified by upperfrq and lowerfrq for it to be considered valid.

Length: 1. Default: 5. Units: 10 ms.

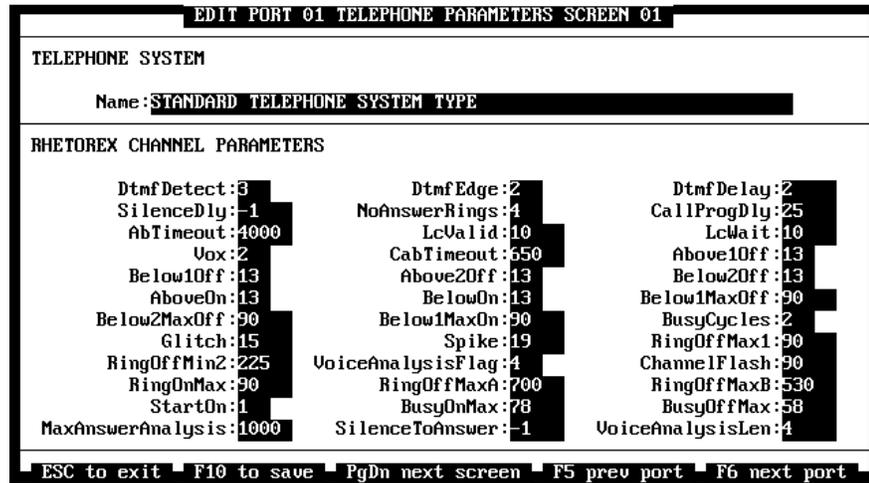


Figure 9-10: The Telephone System and Rhetorex Channel Parameters Panels

**Dtmf Detect**

DTMF Detection: the minimum DTMF tone duration necessary for the driver to detect a DTMF tone during play.

Usage: Numeric, Units = 16 msec. Default value = 3 (48 msec.)

**SilenceDly**

Silence Delay: the maximum allowable silence duration before the driver discards the silence during record.

Usage: Numeric, Units = 16 msec. Default value = 1 (Never).

**AbTimeout**

Absence Timeout: the length of time that must pass during which the driver does not detect a signal before the driver posts event 18 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Default value -4000 (40 sec.).

**Vox**

Usage: Enumerated. Default value -2 (optimize voice).

Possible values:

- 0 Do not change the current value.
- 1 Optimize signal.
- 2 Optimize voice.
- 3 Create PCPMtone structures and PCPMfreqs.

In this mode, the driver uses the following fields to create a PCPM tone table. The driver will use this table instead of the PCPM tone and filter tables specified in the CONFIG file.

Below1Max Off  
Below1MaxOn  
RingOffMax1  
RingOffMin2  
RingOnMax  
RingOnMaxA  
RingOffMaxB  
BusyOnMax  
BusyOffMax  
MaxAnswerAnalysis  
SilenceToAnswer

### **Below1Off**

The maximum variance that the driver allows during an off period compared to the previous off period, with the current cycle's off period below the previous cycle's off period.

For a double cycle, these off periods are the first in each cycle.

Usage: Numeric, percentage. Default value–13 (13%).

### **AboveOn**

The maximum variance that the driver allows during an on period compared to the previous on period, with the current cycle's on period above the previous cycle's on period.

Usage: Numeric, percentage. Default value–13 (13%).

### **Below2MaxOff**

Busy Off Maximum Two: the maximum allowable second off period for a busy call progression tone that has a double-cycle cadence. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default value–90 (900 msec.)

### **Glitch**

The minimum necessary non-signal duration for the driver to detect a valid call progression tone.

Usage: Numeric, Units = 10 msec. Default value–15 (150 msec.)

### **RingOffMin2**

Ring Off Minimum Two: the minimum necessary off period for a double cadence's interring. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default value–225 (2.25 sec.)

### **RingOnMax**

Ring On Maximum: the minimum necessary on period for a short ring cadence. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default value–90 (900 msec.).

### **StartOn**

The call progression tone cycle on which the driver begins PCPM in cadence mode.

Usage: Enumerated. Default value–1.

Possible Values:

- 0 Do not change the current value.
- 1 Cycle #1
- 2 Cycle #2
- 3 Cycle #3

### **MaxAnswerAnalysis**

Maximum Answer Analysis: the maximum allowable duration of answer analysis after an answer has been detected.

Usage: Numeric, Units = 10 msec. Default value–1000 (10 sec.)

### **DtmfEdge2**

Specifies whether the driver detects a DTMF tone on its leading or trailing edge.

When changing this value, the application program needs to make sure that the channel is on hook. Changing this value when the channel is detecting DTMF can cause a double detection.

Usage: Enumerated. Default value–2.

Possible values:

- 0 Do not change the current value
- 1 Trailing edge
- 2 Leading edge

### **NoAnswerRings**

Number of Answer Rings: the number of ring back cycles that the driver must detect before posting event 18 to the System Event Queue.

Usage: Numeric, Units = cycles. Default value–4.

### **LcValid**

Loop Current Valid: the delay that must occur after dialing a digit string before the driver considers the loop current drop to be answered.

Usage: Numeric, Units 10 msec. Default–10 (100 msec.)

### **CabTimeout**

Continuous Timeout: the minimum necessary duration of a continuous signal before the driver posts event 18 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Default value–650 (6.5 sec.).

### **Above2Off**

The maximum variance that the driver allows during an off period compared to the previous off period, with the current cycle's off period above the previous cycle's off period.

For a double cycle, these off periods are the second in each cycle.

Usage: Numeric, percentage. Default value–13 (13%).

### **BelowOn**

The maximum variance that the driver allows during an on period compared to the previous on period, with the current cycle's on period below the previous cycle's on period.

Usage: Numeric, percentage. Default value–13 (13%).

### **Below1MaxOn**

Busy On Maximum One: the maximum allowable on period for a busy call progression tone that has a double-cycle cadence. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default value–90 (900 msec.)

### **Spike**

The minimum necessary signal duration for the driver to detect a valid call progression tone.

Usage: Numeric, Units = 10 msec. Default value—19 (190 msec.)

### **VoiceAnalysisFlag**

Enables or disables voice analysis detection as a condition that can terminate PCPM.

### **RingOffMaxA**

Ring Off Maximum A: the maximum allowable off period for a long ring cadence. The driver uses this field only when Vox is 3.

### **BusyOnMax**

Busy On Maximum: the maximum allowable on period for a single busy cadence. The driver uses this field only when Vox is 3.

### **SilenceToAnswer**

The maximum duration of continuous silence that the driver allows before posting event 10 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Default value—1 (Disabled).

### **DtmfDelay**

The minimum necessary DTMF tone duration for the driver to detect a DTMF tone during functions other than play.

Usage: Numeric, Units = 16 msec. Default value—2 (32 msec.)

### **CallProgDly**

Call Progress Delay: the length of time that the driver delays after dialing a digit string before beginning PCPM.

Usage: Numeric, Units = 10 msec. Default value—25 (250 msec.).

### **LcWait**

Loop Current Wait: the delay that must occur after loop current drops before the driver posts event 18 to the System Event Queue.

Usage: Numeric, Units = 10 msec. Default value—10 (100 msec.)

The maximum variance that the driver allows during an off period compared to the previous off period, with the current cycle's off period above the previous cycle's off period.

For a double cycle, these off periods are the first in each cycle.

Usage: Numeric, percentage. Default value–13 (13%).

### **Below2Off**

The maximum variance that the driver allows during an off period compared to the previous off period, with the current cycle's off period below the previous cycle's off period.

For a double cycle, these off periods are the second in each cycle.

Usage: Numeric, percentage. Default value–13 (13%).

### **Below1MaxOff**

Busy Off Maximum One: the maximum allowable first off period for a busy call progression tone that has a double-cycle cadence. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default value–90 (900 msec.).

### **BusyCycles**

The number of non-ring back cycles that the driver must detect before posting event 18 to the System Event Queue.

Usage: Numeric, Units = cycles. Default value–2.

### **RingOffMax1**

Ring Off Maximum One: the maximum allowable first off period for a ring tone that has a double-cycle cadence. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default–90 (900 msec.).

### **ChannelFlash**

The duration of the on hook state during a flash hook caused by an “f” or “F” character unless “f” or “F” is the character specified in G\_Params.FlashChar. If G\_Params.FlashChar specifies “f” or “F,” the flash duration is G\_Params.FlashDur.

Usage: Numeric, Units = 10 msec. Default value–90 (900 msec.).

### **RingOffMaxB**

Ring Off Maximum B: the maximum allowable off period for a short ring cadence. The driver uses this field only when Vox is 3.

Usage: Numeric, Units = 10 msec. Default value–530 (5.3 sec.).

### **BusyOffMax**

Busy Off Maximum: the maximum allowable off period for a single busy cadence. The driver uses this field only when Vox is 3.



Character	Explanation
Digits 0–9	Numeric value.
&	Flash. Obtains dial tone or transfer tone.
,	Pause. Default value: 2 seconds.
@	Initiates transfer to box programmed in Extension field on the User screen.
abcd, #, *	Dual Tone Multi-Frequency tones.
P	Pulse is used to call out.
T	Tones are used to call out. This is the default.

**Figure 9-12: Valid Characters for the Sequences Panel**



**Note**

Contact your telephone installer and/or manufacturer for the codes you need to enter in the fields on this screen.

**Call Transfer**

Specify the characters required to transfer a call.

**Call Busy**

Specify the characters the telephone system uses to denote a “busy extension” event.

**Call No Answer**

Specify the characters the telephone system uses to denote a “no answer” event.

**Call Connect**

Specify the characters the telephone system uses to denote a “call connected” event.

**Call Screening Park**

If you are using the call screening feature, specify the characters required to initiate a call park over a single line extension.

**Call Screening Remove Park**

If you are using the call screening feature, specify the characters required to remove a call park over a single line extension.

## Message Waiting Light Sequences Panel

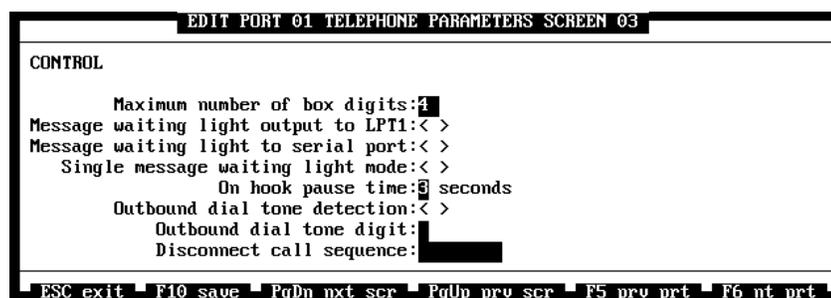
### On

Enter the sequence for activating a message waiting light.

### Off

Enter the sequence for deactivating a message waiting light.

## The Edit Port Telephone Parameters Screen 3



*Figure 9-13: The Control Panel*

## The Control Panel

### Maximum number of box digits

This is the maximum number of digits you are allowing for box numbers on this port. If box numbers on your system consist of 4 digits, this field should be set to 4.

Enter a number between 1 and 10.

### Message waiting light output to LPT1

If this field is enabled, Atlas AVM will print out message waiting light information at a line printer near an attendant's console. The console operator can monitor the message waiting light information.

Use the <Spacebar> to toggle this field on or off. (<\*> indicates on.)

### Message waiting light to serial port

If this field is enabled, Atlas AVM can send message waiting light information through a serial port.

Use the <Spacebar> to toggle this field on or off. (<\*> indicates on.)

### On hook pause time

This is the number of seconds you want Atlas AVM to pause, after a caller hangs up, in order for Atlas AVM to be ready for another call. This amount of time depends on how long it takes the telephone system to make a port available for another call.

Enter a number between 0 and 9.

### Outbound dial tone detection

AVM has the ability to look for dial tone or simply dial what has been programmed into the outdial string. Toggle this field to On if you want AVM to look for dial tone on every out dial.

Use the <Spacebar> to toggle this field on or off. (<\*> indicates On.)

### Outbound dial tone digit

An entry in this field allows AVM to wait for a dial tone digit confirmation before doing any outbound calls. If the digit is not received within one second, the system assumes that there is a call collision and will proceed to answer the call.

If this DTMF pattern is detected by AVM, the call is immediately disconnected.

Enter a DTMF pattern from 1-8 digits.

## Edit Port Telephone Parameters Screen 4

```

EDIT PORT 01 TELEPHONE PARAMETERS SCREEN 04
-----
TONE TERMINATION
      Enabled:< >
      Tone frequency:405 Hz
      Tone deviation:25 Hz
      Cadence on time:25 10ms
      Cadence on time deviation:-25 10ms
      Cadence off time:0 10ms
      Cadence off time deviation:0 10ms
      Cadence repeat count:0
-----
RECORDING TERMINATION
      Maximum silence:5 seconds
      Maximum non-silence:3 seconds
-----
ESC exit  F10 save  PgDn nxt scr  PgUp prv scr  F5 prv prt  F6 nt prt

```

**Figure 9-14: The Tone Termination (Dialogic Only) and Recording Termination Panels**

## **The Tone Termination Panel (Dialogic only)**

This panel allows you to define a tone that signals a disconnect from the CO in case your telephone system does not provide disconnect supervision.

### **Enabled**

Enable this field if your telephone system doesn't have disconnect supervision but it can provide a distinct tone on CO disconnect.

### **Tone Frequency**

Specify the frequency of the tone given by the telephone system to define a termination.

### **Tone Deviation**

Specify the range allowed for variance in the termination tone.

### **Cadence On Time**

Specify the length of time that the termination tone sounds.

### **Cadence On Time Deviation**

Specify the range allowed for variance in the cadence on time.

### **Cadence Off Time**

Specify the length of time between the termination tones.

### **Cadence Off Time Deviation**

Specify the range allowed for variance in the cadence off time.

### **Cadence Repeat Count**

Specify how many times the termination tone repeats to signal a disconnect.

## **The Recording Termination Panel**

Atlas AVM has the ability to distinguish between voice, silence (no voice or noise), and non-silence (background noise) during a recording. When a recording is terminated due to silence or non-silence, Atlas AVM stops the recording, rewinds, and deletes the silence or non-silence.

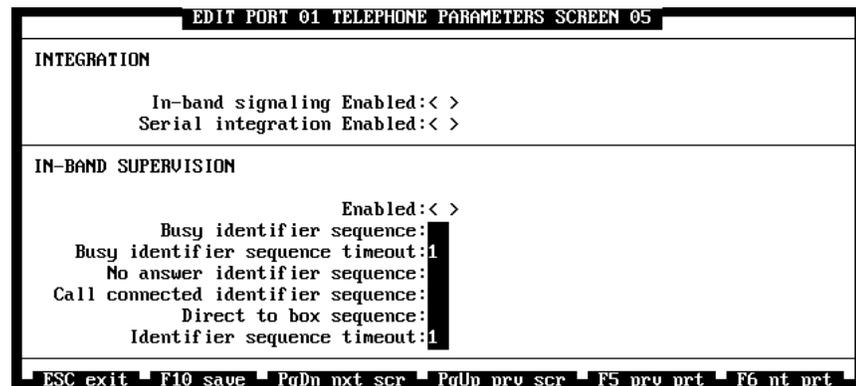
### **Maximum Silence**

The maximum amount of silence allowed before termination of a recording.

## Maximum Non-silence

The maximum amount of non-silence allowed before termination of a recording.

## The Edit Port Telephone Parameters Screen 5



*Figure 9-15: The Integration and In-Band Supervision Panels*

## Integration

Enable one of the following integration types.

### In-band signaling Enabled

If your telephone system uses in-band signaling (DTMF keys) to pass call information to AVM, enable this field.

Use the <Spacebar> to toggle in-band integration on or off. (<\*> indicates on.)

### Serial integration Enabled

If your telephone system uses serial integration (via RS232 serial link) to pass caller information to AVM, enable this field.

Use the <Spacebar> to toggle serial integration on or off. (<\*> indicates on.)

## In-Band Supervision

If you are using in-band signaling for caller identification, program the parameters in this panel.

**Enabled**

Enable this *only* if your telephone system provides in-band supervision or offers sequences for call progression (other than standard tones).

Use the <Spacebar> to toggle In-band Supervision on or off. (<\*> indicates on.)

**Busy identifier sequence**

This is a key that the telephone system sends to signal that the call has encountered a busy condition.

Enter the key(s) received (0–99, A–D, \*, #).

**Busy identifier sequence timeout**

This is the time allowed, in one second intervals, for the system to send the key for busy.

Enter the time, in seconds, between 1 and 99.

**No answer identifier sequence**

This is the key that the telephone system sends to signal that the call has encountered a ring no answer state.

Enter the key(s) received (0–99, A–D, \*, #).

**Call connected identifier sequence**

This is a key that the telephone system sends to signal that the call has been answered.

Enter the key(s) received (0–99, A–D, \*, #).

**Identifier sequence timeout**

Time in seconds that the system will wait to receive the identifier.

Enter a number between 1 and 99.

**Edit Port Telephone Parameters Screen 6**

```

EDIT PORT 01 TELEPHONE PARAMETERS SCREEN 06
-----
IN-BAND SIGNALING CONTROL

      Data packet timeout: 1 seconds
      Cut over to voice delay: 0 seconds
      Data packet terminator: 00 hexadecimal
      Receive data before going off hook: < >
-----
ESC exit  F10 save  PgDn nxt scr  PgUp prv scr  F5 prv prt  F6 nt prt

```

*Figure 9-16: The In-Band Signaling Control Panel*

## In-Band Signaling Control

When In-band Signaling is used as the integration method, before a call is transferred to AVM, information about the event and the called party is given to AVM in the form of DTMF tones.

In this series of DTMF tones, each tone represents a different number. Collectively, these numbers make up a data packet.

### Data packet timeout

This is the number of seconds you want Atlas AVM to wait, after a ring, to receive data packet information before assuming that no packets are coming. If no packets are received, the call then goes to the port greeting.

Enter a number between 1 and 9. (Start at 3 recommended.)

### Cut over to voice delay

This is the number of seconds you want Atlas AVM to wait after receiving a data packet before playing the port greeting or box greeting.

Enter a number between 0 and 9.

### Data packet terminator

This is the character that signifies the end of a packet. If there is a terminating character, Atlas AVM does not have to wait for timeout.

Valid characters for this field are a,b,c,d,#,\*, and digits 0–9. *See* Figure 9-12 for an explanation of these characters.

## Edit Port Telephone Parameters Screen 7

```

EDIT PORT 01 TELEPHONE PARAMETERS SCREEN 07
INTEGRATION DATA PACKET 1
    Call type identifier bytes:00 00 00 00 00 00 hexadecimal
    Call type identifier offset:0
    Call type identifier length:0
    Packet definition:Retrieve message
    Called party box number/DTMF offset:0
    Called party box number length:0
    Calling party box number offset:0
    Calling party box number length:0
    Trunk port:1
    Go off hook:< >
ESC to exit  F10 to save  PgUp prev screen  F5 prev port  F6 next port

```

Figure 9-17: The Integration Data Packet Screens

## **Integration Data packet**

### **Call type identifier bytes**

These characters signify what type of event the data packet represents. Each character that makes up the event number must be shown as a hex character in this field. (See the hex conversion table in Figure 12-8.)

### **Call type identifier offset**

Specify where the call type identifier begins.

### **Call type identifier length**

Specify the length of the call type identifier.

### **Packet definition**

Move the cursor to this field to access the packet definition submenu, which contains the following choices:

- Retrieve message
- Forwarded
- Forwarded busy
- Forwarded no answer
- Direct to box
- Trunk
- Terminate call
- Ignore
- DTMF

### **Called party box number/DTMF offset**

Specify where the called party box number/DTMF begins.

### **Called party box number length**

Specify the length of the called party box number.

### **Calling party box number offset**

Specify where the calling party box number begins.

### **Calling party box number length**

Specify the length of the calling party box number.

### **Trunk port**

Enter the number of the trunk port.

### **Go off hook**

Use the <Spacebar> to toggle this field on or off. (<\*> indicates on.)

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## ***Fax Installation***

Fax capability for Atlas AVM is an optional add-on feature that must be purchased separately from Key System US.

The instructions provided in this chapter supplement the instructions provided in the installation manual for the GammaFax MLCP-4/AEB fax board, which is the board from GammaLink that has been certified for use with Atlas AVM. The GammaFax MLCP-4/AEB fax board can be connected to any Dialogic or Rhetorex voice board that supports an AEB bus.

Other AEB-based GammaFax boards, as well as GammaFax boards employing the MVIP bus architecture, should work with Atlas AVM; however, these boards have not been tested for compatibility.

## **Configuring the Fax Board**

There are two parts to configuring the GammaFax MLCP-4/AEB board before physically installing the board in your computer. These are:

- Setting I/O port address DIP switches
- Setting the AEB signaling mode

Both parts are discussed below.

### **Setting the I/O Port Addresses**

The GammaFax MLCP-4/AEB uses four I/O ports in the 100h to 373h range. The port addresses used are set with DIP switch settings on the board. There are sixteen different sets of four I/O port addresses that the board can be configured to use. You will need to configure the board to use I/O ports that are not already in use by other adapters in your system, such as multi- I/O boards, network adapters, and so on.

To determine which I/O port memory addresses are currently in use in your system, you can consult the documentation that came with your

system, or use a utility program that allows you to examine the 100h to 373h address range.

If you use a utility, be sure to load the any drivers that will be in use with the system when Atlas AVM is running before examining port I/O addresses. (Do not load the GammaFax drivers before examining port addresses.) When you examine port memory addresses, note that values of "FF" in a port address usually mean that the memory at that address is not being used.

## Setting the AEB Signaling

The fax board should be configured for Voice/Fax applications. Follow all instructions contained in the fax board installation manual for configuring the board and software for a voice/fax application.

There are two signaling modes for each GammaFax MLCP-4/AEB fax channel, called *SIGX* and *SIGR*, which stand for "transmit signaling" and "receive signaling" respectively. Since the voice board, rather than the fax board, drives transmit signaling under Atlas AVM, transmit signaling by the fax board must be disabled. See the fax board installation manual for the procedure to disable fax board transmit signaling.



### Caution

Failure to disable transmit signaling can damage the GammaFax MLCP-4/AEB fax board and the voice board.

## Cabling the Fax and Voice Boards

The AEB bus on the GammaFax MLCP-4/AEB board connects to the AEB bus on a Dialogic board through a 20-conductor ribbon cable, or to the ABC bus on a Rhetorex voice board through a custom 20-pin-to-9-pin AEB-to-ABC bus cable. The appropriate cable is available from the respective voice board manufacturers.

The AEB connector shell on the voice and fax boards is notched to mate with a key on the cable connectors to prevent the cable from being installed backwards. The ABC-bus end of the AEB-to-ABC cable used with Rhetorex voice boards has the first pin on the connector plugged to prevent it from being inserted backwards.

Take care to properly orient the cable connectors to the pin connectors on the board before pressing the connectors in place. If more than moderate pressure is required, verify proper cable orientation before continuing.

## Installing the GammaFax Drivers

The GammaFax drivers are found on a single Atlas AVM distribution disk labeled Fax Driver Disk which, if you purchased fax, you will receive packaged with the other Atlas AVM installation disks. The driver installation routine on this diskette correctly configures the GammaFax

initialization file for use with Atlas AVM. You should use this diskette, rather than the one shipped from GammaLink, to install the GammaFax drivers.

To install the drivers, you must be logged to the drive containing the installation diskette. For instance, if you are installing from the A drive, insert the diskette into the A drive, then type:

```
A: <ENTER>
install <ENTER>
```

Follow the instructions for driver installation that accompanied your GammaFax board. Use the following responses to the prompts displayed by the installation program:

- Install all GammaLink files
- Perform fax channel detection
- For the country code, enter a value of 1
- For the mode, select Atlas AVM

Note that if there are embedded spaces in any of the following entries, the entry must be enclosed in quotes:

- CSID
- user name or company name
- voice and fax telephone numbers

For example, to enter the name John Doe, type "John Doe".

## **For Systems with More Fax Channels than Voice Board Ports**

The driver installation program automatically creates a file named GFAX.\$DC in the target directory you chose for driver installation. This file contains commands that initialize the fax channels detected during driver installation. For the Atlas AVM fax feature to function properly, the number of initialized fax board channels must not be greater than the number of ports on the attached voice board. If the number of channels on the fax board is greater than the number of voice board ports, you will have to edit the GFAX.\$DC file to eliminate commands that initialize the extra fax channels.

For example, the following is a section from a GFAX.\$DC file created by the installation routine for a four-channel GammaFax board.

```
init 1
init 2
init 3
```

```
init 4
boardtype 1 0x12c2
boardtype 2 0x12c2
boardtype 3 0x12c2
boardtype 4 0x12c2
load 1 D:\FAX\GFXCX.BIN
load 2 D:\FAX\GFXCX.BIN
load 3 D:\FAX\GFXCX.BIN
load 4 D:\FAX\GFXCX.BIN
```

If the board is attached to a 2-port Rhetorex 232 board, the above commands will have to be edited to delete commands for channels 3 and 4. The edited commands will then appear as follows:

```
init 1
init 2
boardtype 1 0x12c2
boardtype 2 0x12c2
load 1 D:\FAX\GFXCX.BIN
load 2 D:\FAX\GFXCX.BIN
```

## Configuring the AUTOEXEC.BAT File

The following three lines must be added to your AUTOEXEC.BAT file before using the fax board.

```
SET GFAX=D:\PATH
D:\PATH\GFQRESET -P3
D:\PATH\GFDCP -B1
```

*D:* is the drive where the fax drivers are installed, and *PATH* is the path where the fax drivers are installed.

The first two lines must appear in the order shown and *before* the command that loads your voice board driver. The third line must appear after the command that loads your voice board driver.

Once the GammaFax board and drivers have been installed and the AUTOEXEC.BAT file has been configured, reboot the computer. If the GammaFax board initializes without error, the Fax option on Atlas AVM's Application menu should be enabled when you restart Atlas AVM. You may now proceed to set up the Atlas AVM fax administration screen.



## ***System Administration Considerations***

After an AVM system is installed, you need to see that the customer's desired applications and configuration are implemented. This would include such things as:

- Configuring boxes
- Recording greetings and audio menus
- Testing the system
- Training end users

If system administrators are properly trained, much of this responsibility can be turned over to them. The more hands-on experience system administrators can get, the less they will have to rely on you in the future.

## **System Administration Steps**

1. Obtain a list of users and their extension numbers. Since users' mailbox numbers generally match their extensions, you will need this information before you set up mailboxes.
2. Decide on the range of numbers to use for box numbers. By default AVM box numbers start at 1000. Plan to renumber voicemail boxes to match up with the owners' extensions. You can use any other numbers, but it is recommended that they be kept separate from the voicemail box numbers.
3. Find out if there are groups of box owners who have similar voicemail needs. Plan to set up Master boxes to copy information from so you will not have to program each box separately.
4. Use the forms in the *System Administrator's Manual* to record information about boxes to be set up. Then you will have a hard copy of the information in case you ever experience data loss or system failure.

5. Refer to the *System Administrator's Manual* for information on setting up and configuring boxes.
6. Record the audio portion for port greetings and other audiotext boxes as instructed in the *System Administrator's Manual*.
7. Have the box owners access their own mailboxes and record their personal greetings.
8. Test the configuration of the port greetings, audio menus, and voicemail boxes.

Train users on the AVM telephone interface. See the *Atlas AVM User Guide*.



## *Final Cutover*

Now that you have Atlas AVM installed, configured, tested, and the users trained, you can complete the last steps to make the system operational. This is called the ‘final cut over.’ Depending on the individual site this could entail various steps, but the most common are described below.

1. Connect single line extensions.

Some of the lines were no doubt connected during the earlier procedures. Make sure that all lines are in place.

2. Program Atlas AVM lines at the telephone switch.

You will need to program the telephone system according to what type of integration you are using.

3. Organize hunt groups.

If your telephone system supports hunt groups, you will probably want all the lines going to Atlas AVM to be in a hunt group.

4. Forward trunk calls.

If the Atlas AVM system uses an Auto Attendant, the telephone system must be programmed to send all trunk calls (calls originating from outside) to Atlas AVM.

5. Program telephone handsets.

To accommodate station to station calls, you will need to program the customer’s telephones to ‘call forward’ to Atlas AVM. This programming can probably be done at the telephone system console.

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# Glossary

## **Absolute Index Number**

An internal number assigned to a box when the system is created. This is not necessarily the same as the owner's box number. For example, when a new Atlas AVM system is installed, the first box is usually numbered 1000. The Absolute Index Number of the first box is 0000, and it never changes.

## **AccuCall Plus**

A DOS utility from Rhetoirex that allows you to create or modify a PCPM table.

## **AMIS (Audio Messaging Interchange Specification)**

An industry standard for networking voicemail systems that was designed to control how voice messaging products from various manufacturers interface with each other. Any voice messaging products designed to AMIS standards can exchange voice messages with each other.

## **Analog**

A method of telephony transmission in which the information from the source (for example, speech in a human conversation) is converted into an electrical signal that varies continuously over a range of amplitude values. Also used to refer to applications that use loop start signaling instead of digital signaling.

## **Audiotext Box**

Audiotext Boxes are used within Atlas AVM to greet callers, play Audio Menus, telephone system calls as defined by the caller's choices from the audio menu, receive and forward voice messages from other locations on a network, and implement the Auto Attendant feature. Audiotext boxes do not record voice messages.

*See also* Auto Attendant. Compare Voicemail box.

## **Auto Attendant**

A type of audiotext box that enables a caller to direct a call by pressing keys on the telephone keypad to make selections from an audio menu. The Auto Attendant is sometimes called the Company Greeting Box.

*See also* Audio Menu.

## **BIOS**

Basic Input-Output System. The set of permanently stored system service programs needed to manage the PC and consisting of drivers and other software to control peripheral units.

## **Cadence**

A repeating cycle or pattern of tones and silence intervals generated by an audio signal.

## **Call Progression Tone**

An audible tone sent from the telephone system that indicates the progress of a call. Call progression tones include dial tones, busy tones and ring tones.

## **Atlas AVM**

Atlas AVM is a DOS-based voice processing system that provides a suite of telephony features, as well as the means of integrating with a telephone switch.

## **Channel**

In DOS, a channel is a line plus a VP resource (a logical path that is used for a voice processing).

## **Closed Network**

A group of separate sites, requiring communication with each other, who have completed the programming required to allow them to exchange voicemail messages. All sites in a closed network must use Atlas AVM equipment. *Compare* Open Network.

## **CO (Central Office)**

A telephone company location that provides switching and other services for local telephone subscribers.

## **Configuration File**

A file used by the voice board drivers to obtain initialization information about specific boards in your system.

## **Configure**

A DOS utility (CONFIGUR.EXE) from Rhetorex that creates or edits a Rhetorex DOS configuration file.

*See also* the Configuration Manual for DOS.

## **Cycle**

A complete signal sequence which consists of an on period (sound) and an off period (silence).

**Default**

A preset value that is used by the software in the absence of over-riding values entered by the user.

**DID (Direct Inward Dialing)**

The ability for a caller outside a company to dial an internal extension directly without going through an attendant.

**Double Cycle**

A complete signal sequence consisting of two on periods (sound) and two off periods (silences). Both on times are usually the same.

**DTMF (Dual Tone Multi-Frequency)**

A tone (actually, a combination of a low and a high frequency tone) generated by a telephone system. Each tone has a special meaning to telephony equipment. A familiar example is the different tones made by a touchtone telephone when certain keys are pressed.

*See also* In-band Signaling Packet.

**DTMF-On Time**

The length of the DTMF tone.

**Firmware**

A set of program instructions that are resident (usually in EPROM) on an expansion board.

**Glitch**

A temporary loss of signal; temporary silence; drop out. If a silence duration is less than the PCPM Table's filter dropout parameter then the silence is not valid; in other words, the silence is a glitch.

**Hook Flash**

A station to PBX signaling method, normally activated by quickly depressing the handset cradle plunger on a telephone.

**In-band Signaling Packets**

A group of DTMF tones that have a certain meaning to telephony equipment. In voicemail applications, In-band Signaling Packets can be generated by and interpreted by both the telephone system and the voicemail software.

Because the meaning of tone groups can vary, depending on the type of telephone system being used, information specific to the type of telephone system must be entered at the time of installation. This enables Atlas AVM to correctly interpret the

tones it receives from the telephone system, and to generate tones that the telephone system can interpret. This enables calls to be passed between the telephone and Atlas AVM, and to be processed correctly according to the meaning of the specific telephone system's In-band Signaling Packet tones. *See also* DTMF.

**Interdigit Delay Time**

The length of time a caller can pause between pressing digits on the keypad when entering menu choices or a box number. If the caller pauses longer than the Interdigit Delay Time, Atlas AVM assumes that the caller is finished entering digits.

**Interrupt Request (IRQ),**

Interrupt Request. A signal sent to the central processing unit (CPU) to temporarily suspend normal processing and transfer control to an interrupt handling routine. Interrupts may be generated by conditions such as completion of an I/O process, detection of hardware failure, power failure, etc.

**ISA**

Industry Standard Architecture

**Loop Current**

The current that flows through the circuit from the telephone switch when the line is off hook.

**Loop Drop**

The transition from loop current on to loop current off.

**Loop Start**

A method of signaling the CO of an off-hook condition through the current flow within a business and between the business and the outside world.

**Off Hook**

When a telephone handset is lifted from its cradle (or equivalent condition), the telephone line state is said to be "off hook."

**Off Period**

Silence.

*See also,* Cycle and Double Cycle.

**On Hook**

When a telephone handset is returned to its cradle (or equivalent condition), the telephone line state is said to be "on hook."

A line is on hook when the line is open and no current is flowing.

**On Period**

Sound.

*See also*, Cycle and Double Cycle.

**Open Network**

An Open Network (or Open Access Network) is a group of two or more sites that use AMIS-compatible voice messaging equipment, and who have completed the programming required to allow them to exchange voicemail messages with each other. Equipment used at an Open Access Site can be, but does not have to be, Atlas AVM. *Compare* Closed Network.

*See also* AMIS.

**Passcode**

There are three types of passcodes. The first is a unique sequence of numbers, chosen by the box owner or assigned by the system administrator, that serves as a lock for the owner's box. Because only the owner or system administrator knows the passcode, unauthorized access to boxes or to programming is prevented. The passcode can be a maximum of 8 numbers long, and can be changed by the box owner at any time.

The second type of passcode is used by the system administrator to access the administration screens.

The third type of passcode is used by the system installer to access the installation-related screens.

**Pause**

The pause character is a comma <, > and can be entered as part of the dialing number. The value of the pause character is set by the system administrator.

**Personal Computer.**

In this manual, the term refers to an IBM personal computer or compatible machine.

**Peripheral**

Any equipment, apart from the central processing unit, that provides a system with outside communication or additional facilities.

**Port**

The interface point between Atlas AVM software and the telephone system. 0

**RAM**

Random Access Memory. The primary memory in a computer. Memory that can be overwritten with new information. The

contents of RAM memory are lost when the electrical power to the PC is switched off.

**RDSPexit**

A Rhetorex DOS utility that unloads the DOS RDSP Driver from PC memory.

**RDSPtest**

A Rhetorex DOS utility that tests RDSP boards.

**Ring Back**

The ring a caller hears when the telephone at the other end of the line is ringing.

**Ring Off Cycle**

The length of time between rings.

**Showjump**

A Rhetorex DOS utility that displays the jumper settings for a Rhetorex board at a given base I/O port.

*See also, the Showjump User's Guide.*

**Spike**

A temporary signal; noise. If a signal duration is less than the PCPM Table's filter spike parameter then the signal is not valid; in other words, the signal is a spike.

**Talk Off**

The false tripping of DTMF receivers caused by human speech mimicking DTMF tones.

**Telephone System**

Allows communication within a business and between the business and the outside world.

**Terminating Tone**

A call progression tone that indicates that a connection has been terminated.

**Tokens**

“w” (dial tone), “&” (default flash character) and “,” (pause)

**Trunk**

A telephone communication path between two points, with one point being the Central Office.

**Voicemail**

A system of computer hardware and special-purpose software that interfaces with a company's telephone system to provide

telephone message record, store, and forward capability.  
Voicemail is also referred to as voice messaging.

**VPP**

Volts Peak-to-Peak

**Wink**

A timed momentary off-hook pulse communications protocol

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