ATLAS 550 FXO Voice Compression and Private Frame Relay over a point-to-point T1 to an ATLAS 550 FXS Using Trunk Numbers and Dial on Offhook

Introduction

The ADTRAN ATLAS 550 can be used on a point-to-point T1 to emulate a Frame Relay circuit when a telco Frame Relay switch is not being used. An ATLAS 550 populated with a combination of FXS, FXO and VCOM modules can also support compressed voice in this private Frame Relay application. This technical support note will describe the hardware and configuration necessary to support voice compression and private frame relay over a point-to-point T1, using trunk numbers and the dial on offhook features to access a remote PBX for dialtone. **Figure 1** shows a diagram of this application.
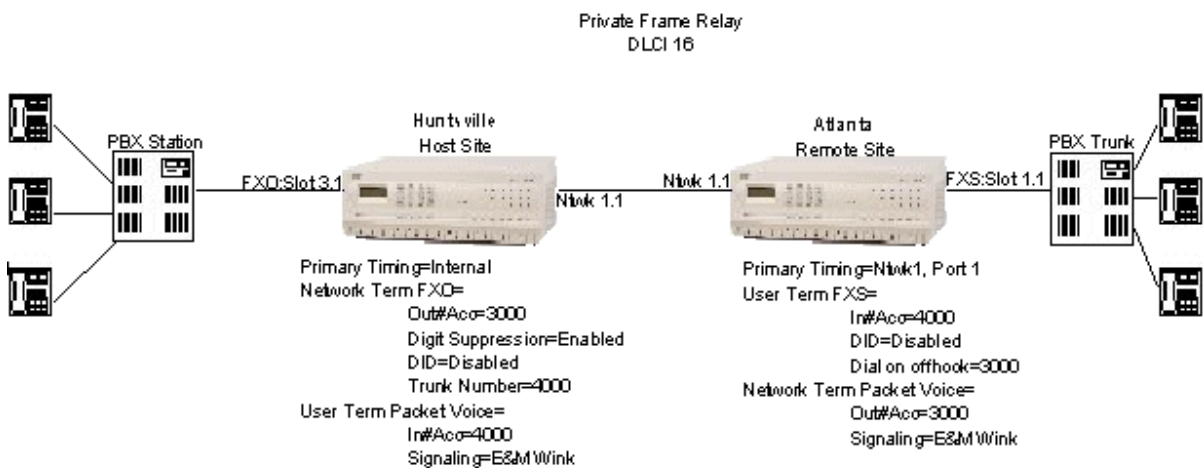

Figure 1

Before You Begin

At one end of the circuit, you will need the ATLAS 550 (Part Number: 1200305L1) populated with at least one FXO Module (Part Number: 1200310L1). The FXO Module will be used at the end that connects to the station side of the PBX. Note that the station side of a PBX provides battery and dial tone. At the other end you will need an ATLAS 550 populated with at least one FXS Module (Part Number: 1200309L1). The FXS module is used to connect to the trunk side of a PBX. Note that the trunk side of a PBX provides loop closure and does not provide dial tone. Both ends will need a Voice Compression module. This module comes in 8, 16, and 24 channel versions (Part Numbers: 1200312L1, 1200312L2, and 1200312L3, respectively). The Voice Compression (VCOM) module is a resource module only, meaning there are no physical interfaces on the board and it must be connected to any user module or the resource host module. You will also need to use the Network T1/PRI interface (or any other available T1/PRI interface) to connect the point-to-point T1.

Configuring the Host ATLAS 550

Configuring the System Timing

Since we are setting up a point-to-point T1 in this example, we will assume that there is no timing on the T1 and use Internal ATLAS timing for the clock source. To configure this, go to **System Config** from the main menu, select **Primary Timing Source**, and make sure it is set to **Internal** as shown in **Figure 2**.
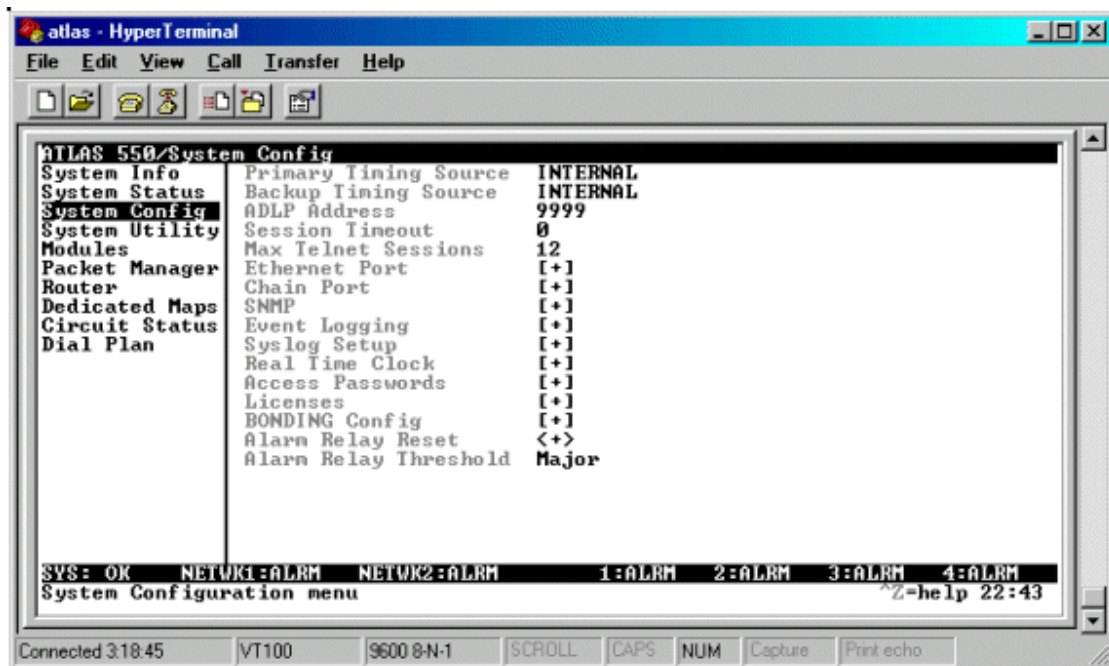

Figure 2

Configuring Packet Endpoints

The ATLAS uses Packet Endpoints to terminate Frame Relay connections. On a Private Frame Relay circuit, one ATLAS will be configured as the Frame Relay Switch, or the Network, and the other end will be the User. For our example, the Host site will be configured as the Network.

1. From the main menu, go to **Packet Manager/Packet Endpnts/Config**.
2. Press the right arrow. A new entry will automatically be created. If an entry already exists, highlight the line item number and press the **'I'** key to insert a new entry. The 'D' key may be used to delete an entry.
3. Change the **Endpnt Name** to reflect something meaningful. In Figure 3, the **Endpnt Name** is **Private FR**.
4. Change the **Protocol** to **Frame Relay**.
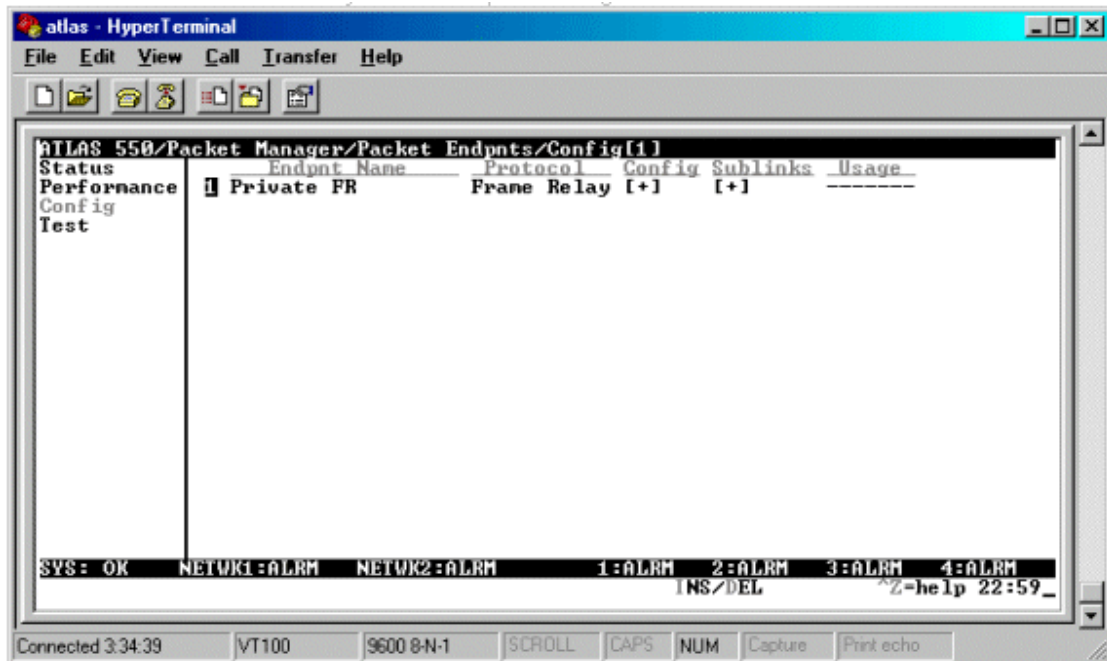5. Press <enter> on the **[+]** symbol in the **Config** column.

Figure 3

6. Change the **Signaling Role** to **Network**. In a private frame relay setup, one site must act as the **Network**, and the other as the **User**. Since we are referring to this end as the host site, we will set it **Network**.
7. Select the proper **Signaling Type**. For this example, we will use **Annex D**, but any option may be used as long as both ATLAS units are set the same. This sets the frame relay signaling type for this endpoint only and is independent of other interfaces on the ATLAS. See **Figure 4**.
8. Leave the remaining options set to the default values and use the left arrow key to go back to the previous screen.
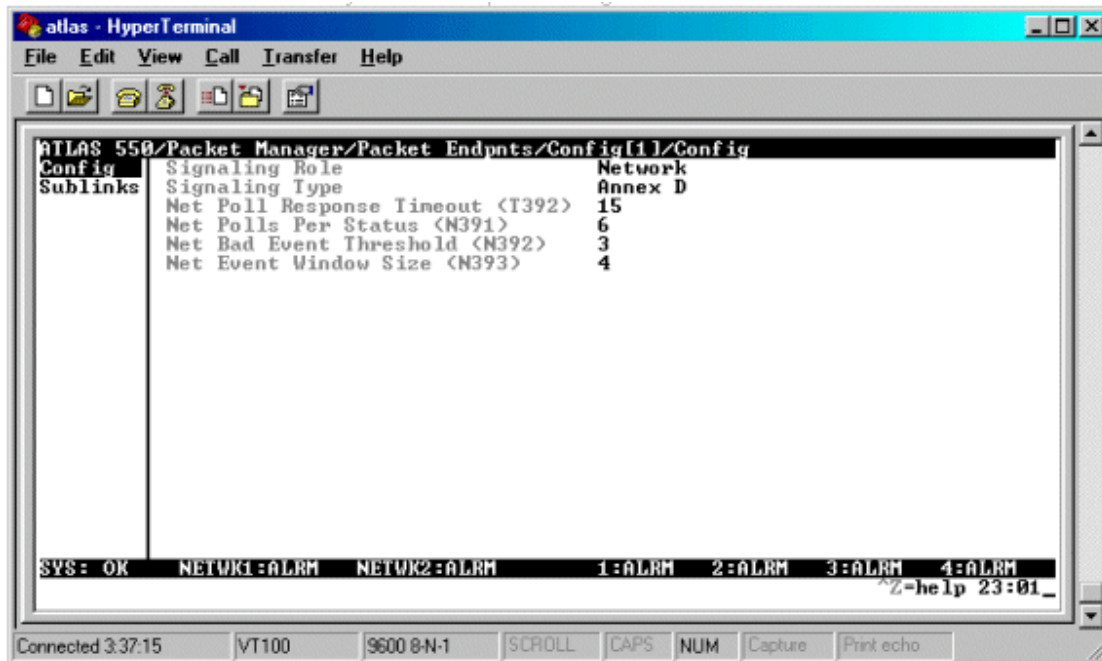
Figure 4

9.   Press <enter> on the **[+]** symbol in the **Sublinks** column.
10. Create a new entry for each Permanent Virtual Circuit (PVC) on this frame relay line.
11. Name the sublink after the remote site to which you are connecting. For our example, the remote site is *Atlanta*, so we will name the sublink **ATL**.
12. Set the **DLCI** to access that remote site. Configure the **DLCI** field to the local **DLCI** number the ATLAS should use to access the remote site. Since this is Private Frame Relay, you may choose the DLCI numbers. See **Figure 5**.
13. Set the **QOS** and **Burst** for each PVC. These fields limit the amount of bandwidth that data traffic will consume so that voice traffic will have enough bandwidth to be transmitted in a time-sensitive manner. **QOS** represents the bandwidth in kbps to be used by this PVC for data traffic. It is a guaranteed bandwidth for that PVC. A value of zero **QOS** means that data traffic on this PVC will not be limited. **Burst** represents the bandwidth in kbps that the data traffic can burst up to. A value of zero **Burst** means that the bursting of data traffic above the QOS will not be limited. For more information on **QOS** and **Burst**, highlight these fields and press **Ctrl-A**. **Figure 5** shows that the *Atlanta* sublink has been configured for a **QOS** of 52 and a **Burst** of 54. This means that the data traffic will consume a maximum of 54kbps on this PVC.
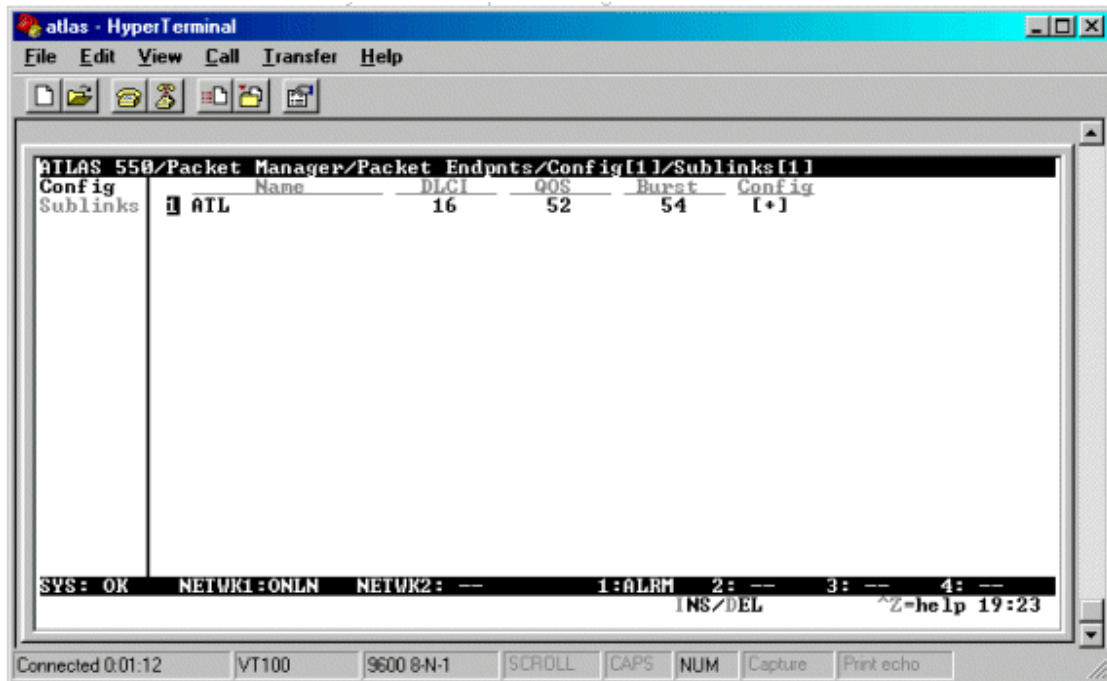
Figure 5

14. Press <enter> on **Config** and change the **Fragmentation Threshold** to 220. This field tells the ATLAS what packet size to use for each fragment. The far end ATLAS must also be set for this value. Fragmentation of the packets allows packets to be transmitted quickly instead of having the long delay associated with a large packet. Whenever an application consists of data along with compressed voice, it is necessary to turn on fragmentation. See **Figure 6**.
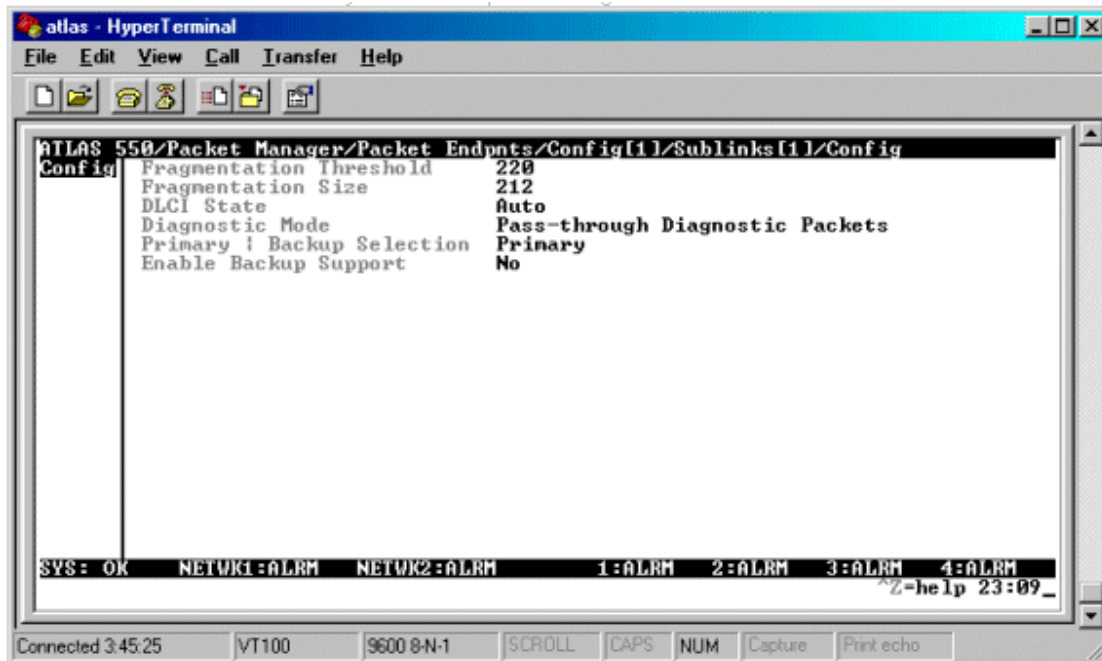15. Repeat steps 10 through 14 for the remaining sublinks.

Figure 6

Configuring Packet Connects

After the packet endpoint is created and configured with the appropriate DLCIs, the endpoint needs to be linked. In this example, we will be using the ATLAS' internal router. This configuration will require a packet connect from the endpoint, *Atlanta* sublink, to the internal router (see **Figure 7**).

1. From the main menu, go to **Packet Manager/Packet Connects**.
2. Create a new entry by using the right arrow key to move your cursor to the **FROM: PEP** field.
3. Press <enter> on **FROM: PEP** and select the packet endpoint that is terminating the frame relay line.
4. Press <enter> on **Sublink** and select the first sublink on that endpoint.
5. Press <enter> on **TO: PEP** and select **Router**. Notice that **Not Used** now appears in the corresponding **Sublink** field.
6. Press <enter> on **Protocol** and select **IP** as shown in **Figure 7**.
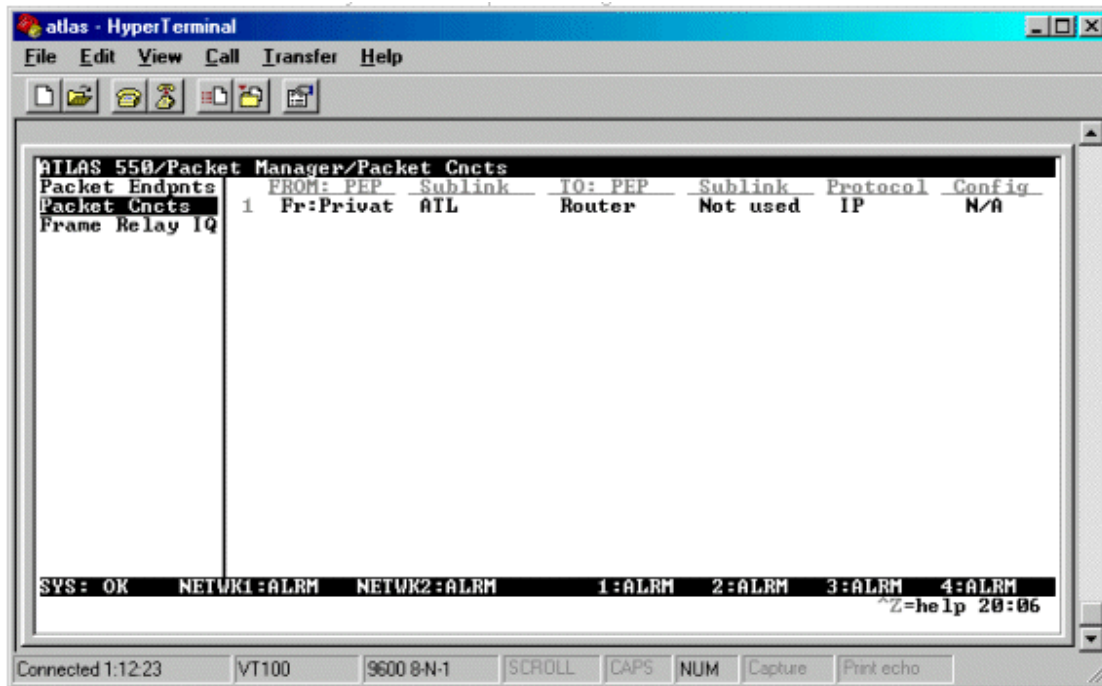7. Repeat steps 2 through 6 for each remaining sublink on the endpoint.

Figure 7

Configuring the Dedicated Maps

Now that the packet endpoint has been created and linked, the endpoint must be mapped to a physical port on the ATLAS.

1. From the main menu, go to **Dedicated Maps/Create-Edit Maps**.
2. Press <enter> on the **[Cncts=#]** option for Map 1. (# refers to the number of entries you have in the map.)
3. Insert a new entry into the map.
4. Under the **From Slt** and **Port** column, select the slot and port to which the frame relay circuit will be connected.
5. Under **To Slt**, select **PktEndpt**.
6. Under **To Prt/Pep**, select the packet endpoint created earlier which will terminate the frame relay circuit. See **Figure 8**.
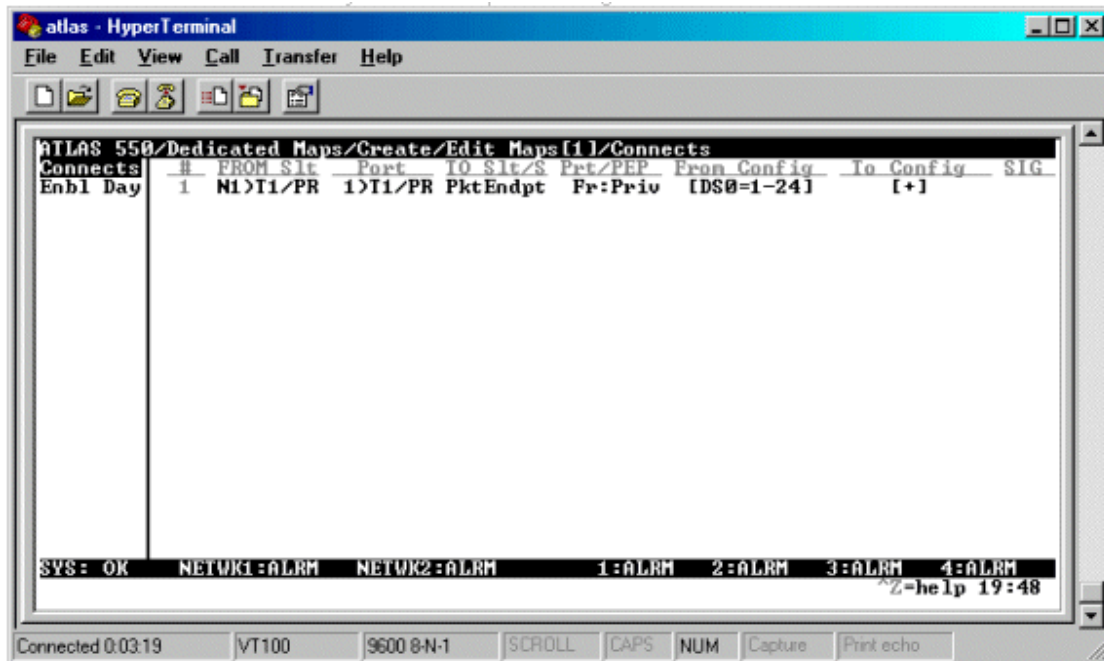
Figure 8

7. Press <enter> on the **From Config** option and the menu in **Figure 9** will be displayed.
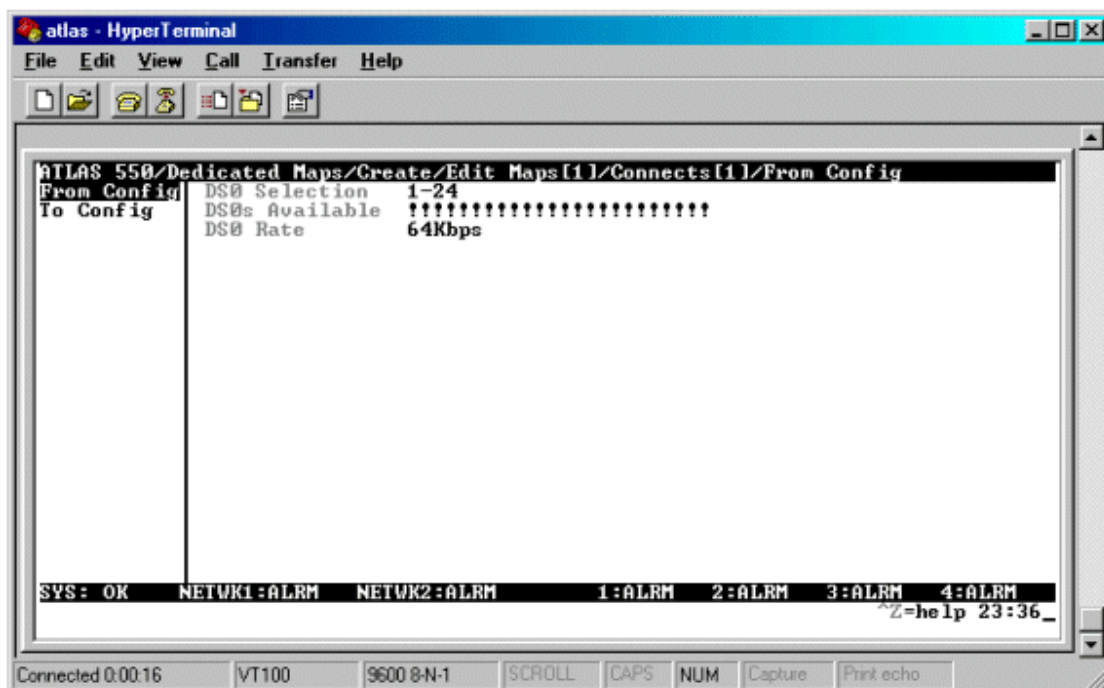

Figure 9

8. Press <enter> on **DS0 Selection** and change it to reflect the number of channels used for the frame relay connection. In the example network of Figure 1, all 24 channels will be used, so 1-24 is entered in the **DS0 Selection** menu field.

Note that the internal router is linked to the Ethernet port on the ATLAS. No user configuration is required under the **Dedicated Map** or **Packet Endpoint** menus for the internal router.

Configuring the Dial Plan

The **Dial Plan** will contain the configuration necessary to compress and uncompress the voice as well as route the voice calls based on the digits dialed. First we will configure the FXO module that will connect to the PBX. This needs to be configured in **Dial Plan/Network Term** as shown in **Figure 10**.
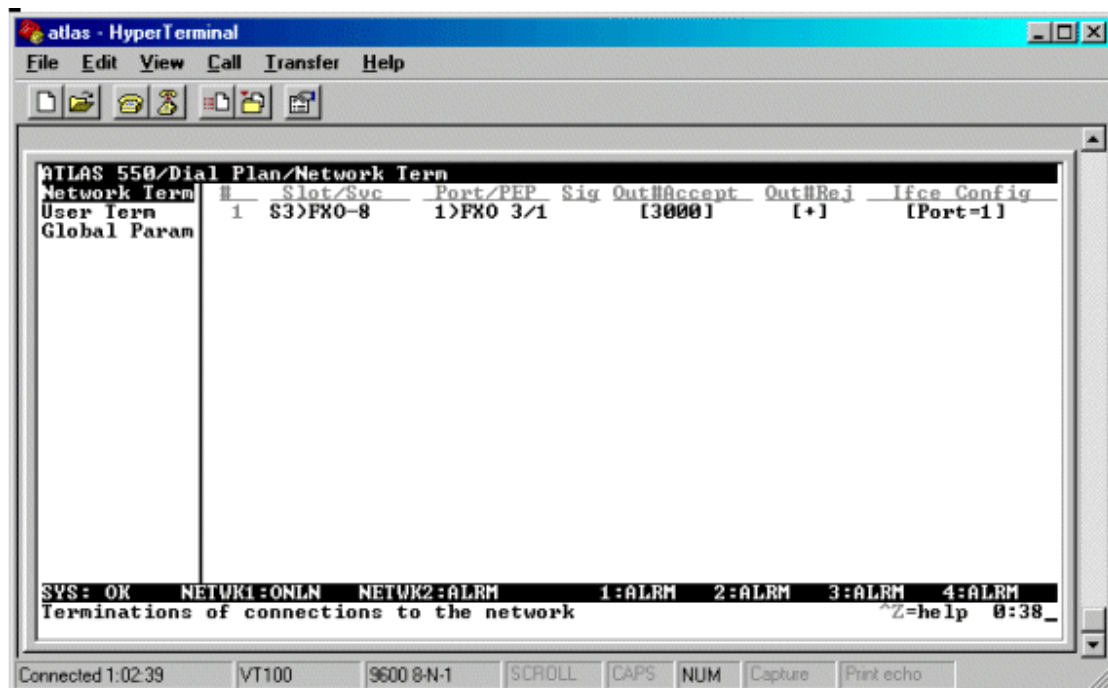

Figure 10

1. Select **Slot/Svc** and press <enter>. Select the appropriate slot for the FXO module.
2. Select **Port/PEP** and press <enter>. Select the port on the FXO module that you wish to use.
3. Select the **Out#Accept** field and press <enter>. Use the arrow keys to move under the **Accept Number** field, and then enter the number that will be coming from the remote site. The remote site will be inserting a configured digit pattern that the ATLAS will use to send the call. This is not a number sent to the PBX or dialed by a user, but simply an arbitrary number that the ATLAS will send to the host site when a user goes off-hook. This allows the ATLAS access to the host PBX without the user needing to dial any number. The **Out#Accept** for this application will be 3000.
4. Press >esc&lt: until you are back to the screen shown in **Figure 10**.
5. The **Out#Rej** field will allow you to restrict outgoing calls. Most of the time this field will be left blank.
6. Select **Ifce Config** and press <enter>. You should see the screen shown in **Figure 11**.
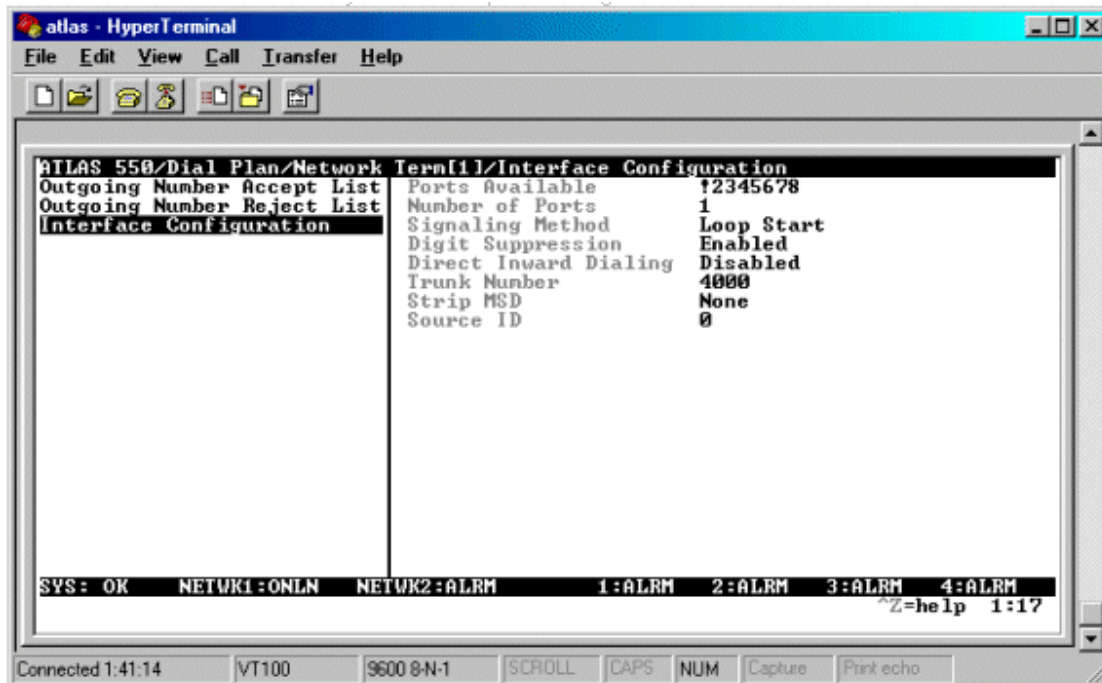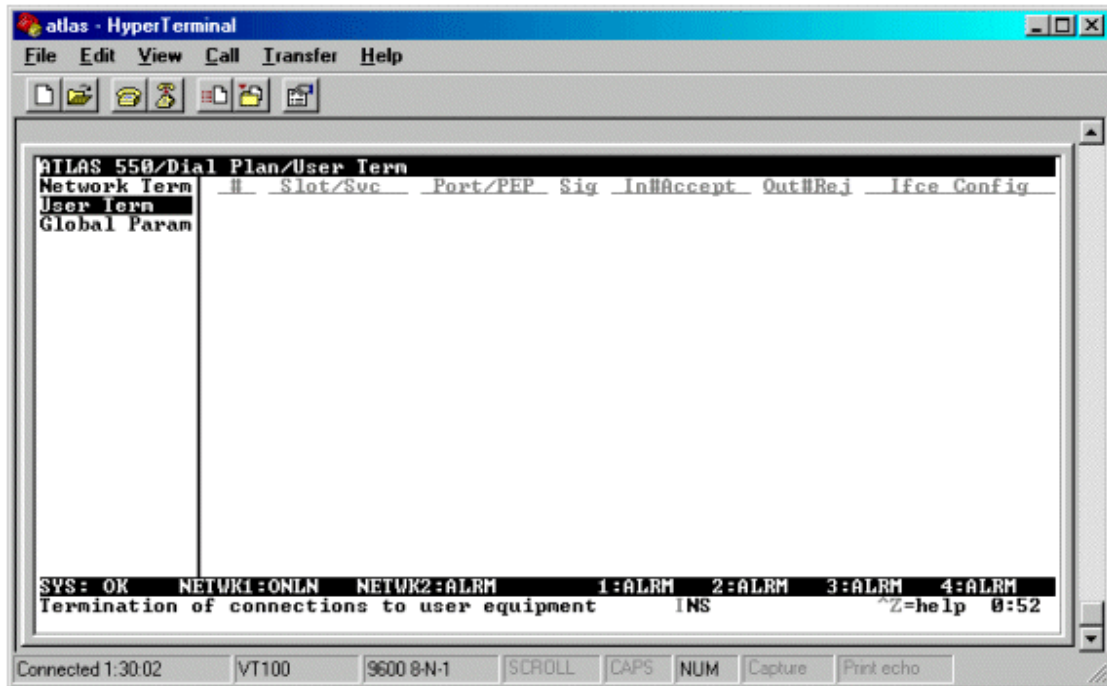
Figure 11

7. The **Ports Available** field is a read-only field and will change according to how the **Number of Ports** option is configured.

8. Select **Number of Ports** and select any number 1-8. Any number greater than 1 entered in this field means that multiple ports on this FXO module will be used to route calls with the matching **Out#Accept**. This option can be used to configure all 8 ports on the module instead of configuring 8 separate entries in the **Dial Plan**.

9. The **Signaling Method** should be set the same as the PBX or telco line that is plugging into the FXO module. If this line is coming from the telephone company, this information may be obtained by calling the line provider.

10. **Digit Suppression** determines whether or not the digits dialed by the remote end will be passed to the PBX we have connected off this port. This option applies to outbound calls on a **Network Term**. In this application, we do not want the digits passed so we will set this to **Enabled**.

11. **Direct Inward Dialing** applies to inbound calls on a **Network Term**. If this option is enabled, the ATLAS will look for incoming digits. For our application, we are using **Trunk Numbers** so we do not need to look for any incoming digits. We will set this field to **Disabled**. Note that when this field is **Enabled**, several new menu items will appear. These options are **DID Digits Transferred**, **DID Number Prefix**, **Strip MSD**, and **Source ID**.

12. When **Direct Inward Dialing** is **Disabled**, the **Trunk Number** field appears. This field is used when no DID Digits are being sent. The ATLAS uses the **Trunk Number** to route the call based on a matching number in the **In#Accept** list in the **User Term**. For example, if we set the trunk Number to **4000**, there must be a matching entry in the **User Term In#Accept** field in order for the call to be routed to that port.

13. **Strip MSD** is used to strip digits off the front of a number before it is forwarded out a port. The options are **None, 1, 2, or 3**. For most applications this will be left at **None**.

14. The **Source ID** field can be used as an alternate way of routing calls through the dial plan.

After the **Network Term** is done, the **User Term** must be configured. This is where we will configure the compressed voice.

1. Go to **Dial Plan/User Term** and press <enter>. You should see a screen similar to **Figure 12**.

Figure 12

2. Use the arrow keys to move under the **Slot/Svc** field, and then press <enter>. Select **PktVoice** and press <enter> again to save that selection.
3. Move to the **Port/PEP** field and press <enter>. Select the packet endpoint that you wish to configure the voice for.
4. Move to the **In#Accept** field and press <enter>. This will take you to a new screen. Move to the **Accept Number** field and enter the number that this port will accept. For our example, this will be the same number that we entered as the **Trunk Number** in the **Network Term** configuration on this same unit. This will allow the call to be accepted by this port, compressed, and sent to the remote site.
5. The **Out#Rej** field allows you to restrict calls. This field is usually left blank.
6. Move to the **Ifce Config** field and press <enter>. **Figure 13** shows the **Ifce Config** screen.
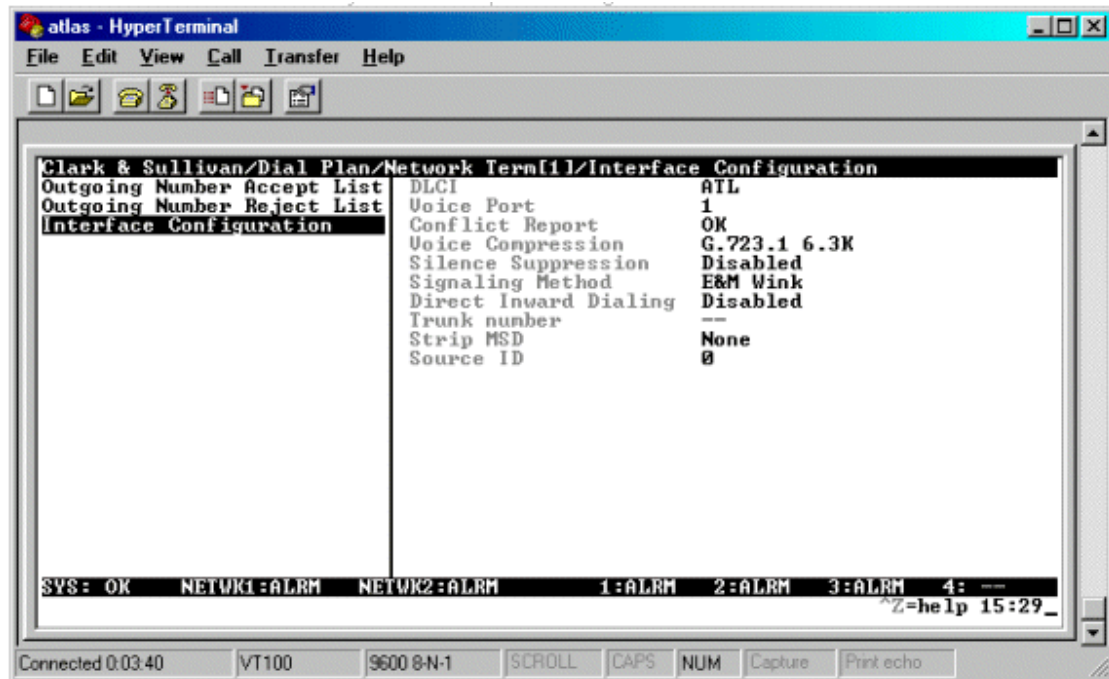
Figure 13

1. Use the arrow keys to move to the **DLCI** field, and then press <enter>. Select the **DLCI** that you are using for your voice connection and press <enter> to select it. We are using the **ATL** DLCI that we configured in the **Packet Manager**. This is the DLCI that goes to the remote site.
2. Press <enter> on **Voice Port** and enter a value of 1. This option is used to send information to the remote site regarding which physical phone port to send the call to. **Voice Port** numbers must match on both the host and remote site. A new **Voice Port** number must be used for each individual entry in the **Dial Plan**. Values can range from 1 to 255.
3. **Conflict Report** is a status field that will display **OK** once the interface is configured and not in conflict with another entry.
4. **Voice Compression** sets the compression algorithm used across the frame relay link. This must match at both the host and remote sites. Options are **G.723.1** or **Netcoder**. For our example, we will use **G.723.1**.
5. **Silence Suppression** is used to conserve bandwidth across the frame relay line when no one is talking. During periods of silence, special frames to mark the silence are sent. By enabling **Silence Suppression**, these frames will not be sent. Both ends of the circuit must be set the same in order to use this feature.
6. The **Signaling Type** determines the type of voice signaling to be used. Options are **E&M Immediate**, **E&M Wink**, **Loop Start**, and **Feature Group D**. This must be the same at both the host and remote sites. We will use **E&M Wink** for this example.
7. Enabling **Direct Inward Dialing** will allow the ATLAS to send digits out the interface. We will leave it **Disabled** for our example configuration.
8. A **Caller ID** number may be set for outgoing calls by entering a number in this field.
9. The **Source ID** field is used as an alternate method of routing calls. For our example we will leave this set to **0**.
10. Press >esc&lt: until you are back to the screen shown in **Figure 14**.
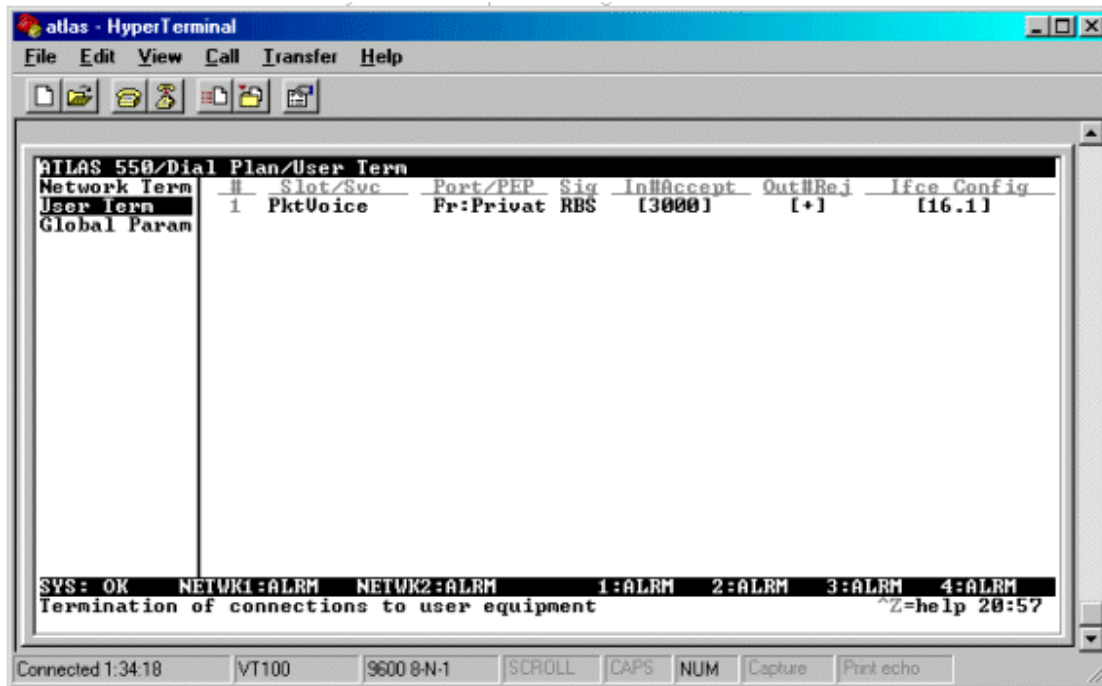
Figure 14

If you wish to add another entry to the **User Term**, you must insert another row. To do this, move to the # column and highlight the number below it. Press **'I'** for Insert and a new row will appear. Repeat the steps listed above to configure another entry.

Configuring the Remote ATLAS 550

For the example in **Figure 1**, we are referring to the remote site as the ATLAS 550 with the FXS Module installed. The configuration needed in this unit is very similar to the configuration that was just discussed. However, there are several slight differences. The first is the timing.

Configuring the System Timing

Go to **System Config/Primary Timing Source**. Since the host ATLAS is set to **Internal**, this remote ATLAS will need to take its timing from the host site. We will change the **Primary Timing** on this side to the slot and port where the T1 in between the ATLAS units is connected. For our example, this is Network 1, Port 1.

Configuring the Packet Endpoints

1. You will need to configure a packet endpoint on this remote end similar to the one we did previously at the host site. Go to **Packet Manager/Packet Endpnts/Config** and press <enter>.
2. Press the right arrow key. A new entry will automatically be created. If an entry already exists, highlight the line item number and press the **'I'** key to insert a new entry. The **'D'** key may be used to delete an entry.
3. Change the **Endpnt Name** to reflect something meaningful. At the host site we used *Private FR*. We will also use that name at this end, but you may change it if you wish.

4. Change the **Protocol** to **Frame Relay**.
5. Press <enter> on the **[+]** symbol in the **Config** column
6. At the host site, we changed the **Signaling Role** to **Network**. In a private frame relay application, one end must act as the **Network** and the other end must act as the **User**. Since we have already designated the host end as the **Network**, we will need to set the remote end for **User**. **Figure 15** shows an example of this configuration.
7. The **Signaling Type** must be set to match the ATLAS at the host site, so we will set it to **Annex D**, as we did for the host unit.
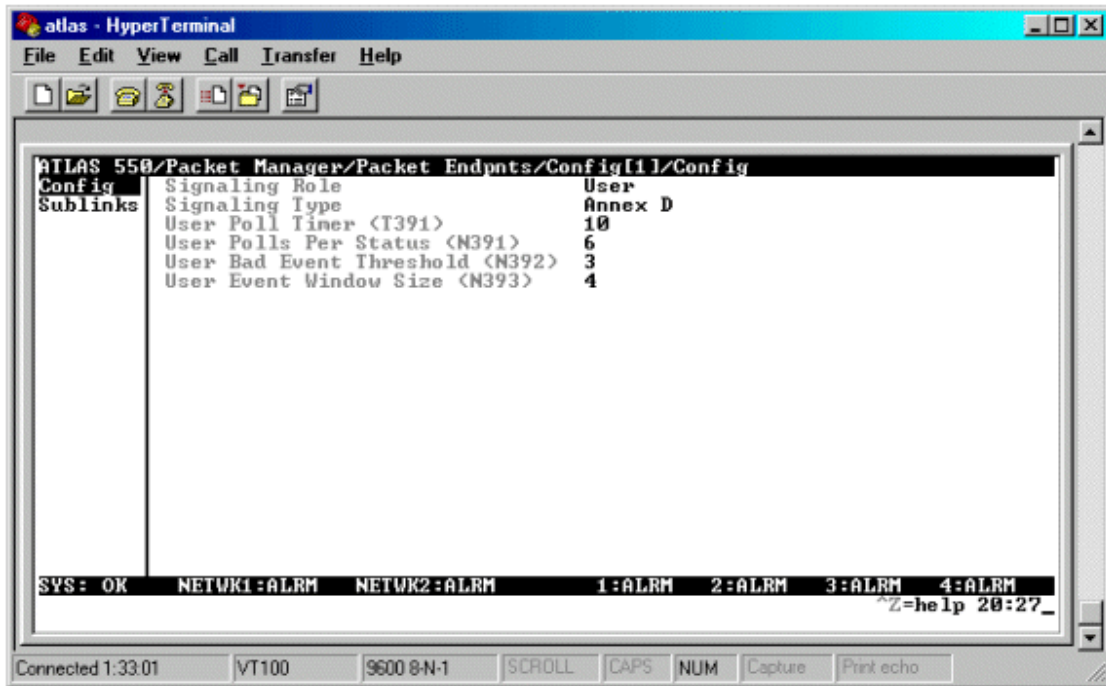

Figure 15

8. Press <enter> on the **[+]** symbol in the **Sublinks** column.
9. Create a new entry for each Permanent Virtual Circuit (PVC) on this frame relay line.
10. Name the sublink after the host site to which you are connecting. For our example, the host site is *Huntsville*, so we will name the sublink **HSV**.
11. Set the **DLCI** to access that remote site. Configure the **DLCI** field to the local **DLCI** number the ATLAS should use to access the remote site. Since this is Private Frame Relay, you may choose the DLCI numbers. The DLCI number on this end should be the same as the DLCI you selected for the host site. We used DLCI 16.
12. Set the **QOS** and **Burst** for each PVC. These fields limit the amount of bandwidth that data traffic will consume so that voice traffic will have enough bandwidth to be transmitted in a time-sensitive manner. **QOS** represents the bandwidth in kbps to be used by this PVC for data traffic. It is a guaranteed bandwidth for that PVC. A value of zero **QOS** means that data traffic on this PVC will not be limited. **Burst** represents the bandwidth in kbps that the data traffic can burst up to. A value of zero **Burst** means that the bursting of data traffic above the **QOS** will not be limited. For more information on **QOS** and **Burst**, highlight these fields and press **Ctrl-A**.

Configuring the Packet Connects

After the packet endpoint is created and configured with the appropriate DLCIs, the endpoint needs to be linked. In this example, we will be using the ATLAS' internal router. This configuration will require a packet connect from the endpoint, *Huntsville* sublink, to the internal router.

1. From the main menu, go to **Packet Manager/Packet Connects**.
2. Create a new entry.
3. Press <enter> on **FROM: PEP** and select the packet endpoint that is terminating the frame relay line.
4. Press <enter> on **Sublink** and select the first sublink on that endpoint.
5. Press <enter> on **TO: PEP** and select **Router**. Notice that **Not Used** now appears in the corresponding **Sublink** field.
6. Press <enter> on Protocol and select *IP*.
7. Repeat steps 2 through 6 for each remaining sublink on the endpoint.

Configuring the Dedicated Map

1. From the main menu, go to **Dedicated Maps/Create-Edit Maps**.
2. Press <enter> on the **[Cncts=#]** option for Map 1. (# refers to the number of entries you have in the map.)
3. Insert a new entry into the map.
4. Under the **From Slt** and **Port** column, select the slot and port to which the frame relay circuit will be connected.
5. Under **To Slt**, select **PktEndpt**.
6. Under **To Prt/Pep**, select the packet endpoint created earlier which will terminate the frame relay circuit.
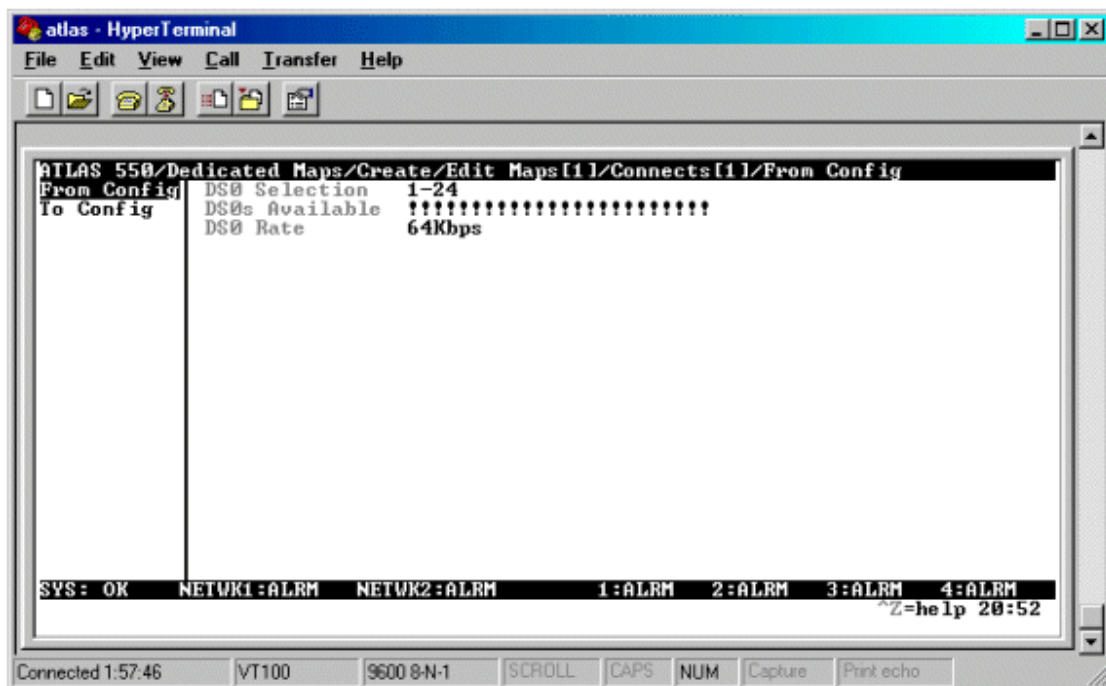7. Press <enter> on the **From Config** option and the menu in **Figure 16** will be displayed.

Figure 16

8. Highlight the **DS0 Selection** field and press<enter>. Change this to the number of channels you want to use for this connection. This should match what the host end is configured for. In our example, we are using a full T1, or 24 DS0s.

Configuring the Dial Plan

1. Go to **Dial Plan/Net Term** and press <enter>. The Packet Voice entries on this end will be configured in the Network Term. Select the **Slot/Svc** field and select **PktVoice**.
2. The **Port/PEP** will be the frame relay link that you have configured in the **Packet Endpoints**.
3. Move to **Out#Accept** and press <enter>. Under the **Accept Number** field, enter the outgoing number that will be sent to the host end. For our example, that number will be 3000.
4. Hit >esc&lt: until you are back to the **User Term** screen. Go to **Ifce Config** and press <enter>. You should now see the screen as shown in **Figure 17**.
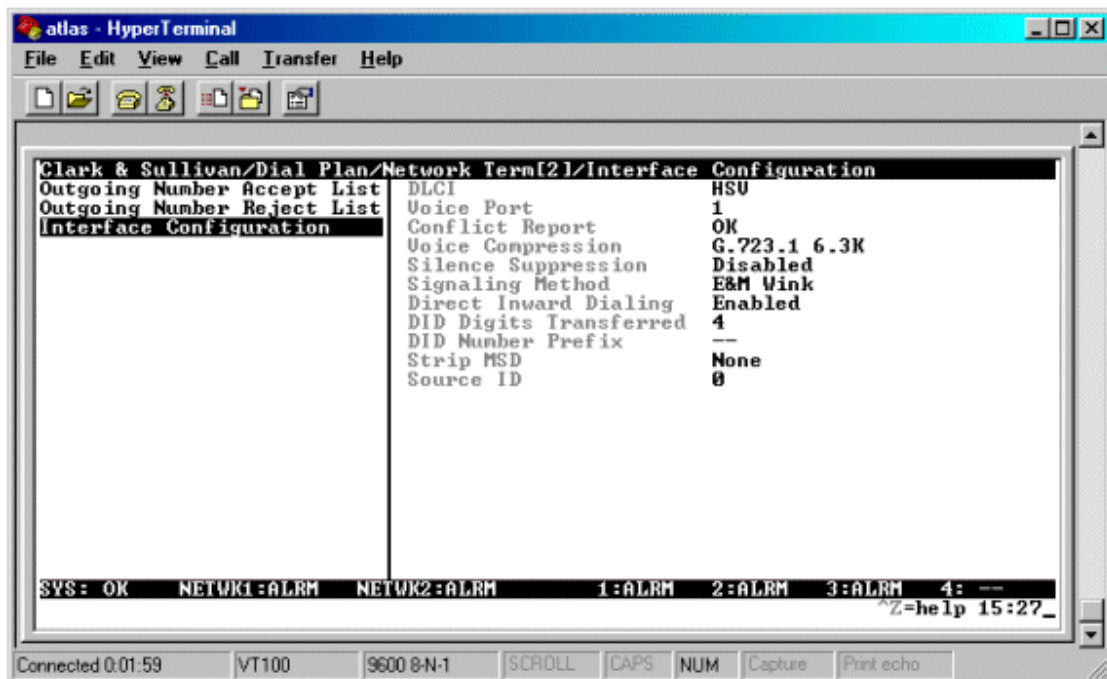

Figure 17

These options are the same ones we looked at earlier for the configuration of the host end. They will be configured the same as the host end. If you would like to go through these options again, please refer to the previous section of this technical support note that discusses these options. There is one exception when configuring the remote unit. The **Direct Inward Dialing** option must be Enabled to allow the digits sent from the host site to be used for routing the call at the remote site. We are using four digit numbers so the DID Digits Transferred option must be set to **4**.

Next we need to configure the FXS ports in the **User Term**.

1. Go to **Dial Plan/User Term** and press <enter>.
2. Under the **Slot/Svc** field, select the FXS module.

3. Go to **In#Accept** and press <enter>. Under the **Accept Number** enter the number you wish this port to accept. For the example, it will be 4000.
4. Press >esc&lt: until you are back to the **User Term** screen. Go to **Ifce Config** and press <enter>. You should now see a screen similar to **Figure 18**.

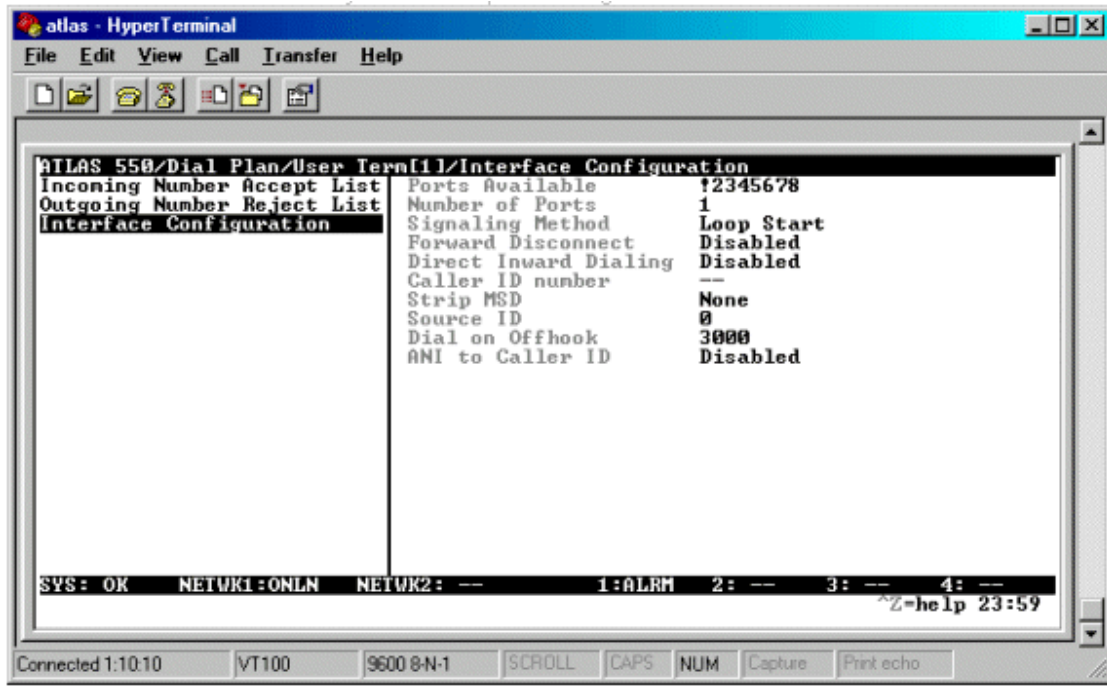ATLAS 550/Dial Plan/User Term[1]/Interface Configuration

Incoming Number Accept List	Ports Available	⬆2345678
Outgoing Number Reject List	Number of Ports	1
Interface Configuration	Signaling Method	Loop Start
	Forward Disconnect	Disabled
	Direct Inward Dialing	Disabled
	Caller ID number	--
	Strip MSD	None
	Source ID	0
	Dial on Offhook	3000
	ANI to Caller ID	Disabled

Figure 18

1. **Ports Available** is a read-only field and will change as the **Number of Ports** is adjusted.
2. **Number of Ports** can be changed to the number of ports needed for this entry (up to 8).
3. The **Signaling Method** must be configured the same as the equipment that is connected to this port. Options are **Loop Start** and **Ground Start**.
4. **Forward Disconnect** should be left at **Disable** for this application. For more information on **Forward Disconnect**, highlight the field and press **Ctrl-A**.
5. **Direct Inward Dialing** is set to **Enabled** only when we want the ATLAS to pass digits out that interface. Again, we will leave this option **Disabled**, as on the host unit.
6. The **Caller ID Number** may be added if caller id information is desired.
7. **Strip MSD** should be left at **None**. This option will strip the number of most significant digits that you indicate in this field.
8. **Source ID** should be left at **0**.
9. The **Dial on Offhook** feature allows the ATLAS to dial any number that you enter in this field as soon as you go offhook. This will be the number that is sent to the Host end. For our example, we are using 3000 to dial the host unit.

The ATLAS should now be configured properly. When we go offhook on the FXS module, configured in the User term on the remote end of the circuit, the ATLAS will automatically dial 3000. In the Network term entry we created, the packet voice entry accepts this number and compresses the voice to be sent over the private frame relay circuit to the host end. The host end accepts the call on the User term packet voice entry and decompresses the voice. Since the matching accept number at the host end is on the Network term, the call gets routed to the FXO port with that accept number, which is connected to the host PBX. On the host end, we have the digit suppression enabled so no digits will be sent to the PBX. The host PBX will then send

dialtone	back	to	the	remote	end.

When calling from the FXO side to the FXS site, the host ATLAS will send the digits in the **Trunk Number** field (4000), which is sent out the host User Term Packet Voice entry across the Private frame relay and to the remote ATLAS. At the remote site, the Packet Voice is accepted and since the **Direct Inward Dialing** is enabled on the Network term, the digits are sent to the ATLAS switchboard and the call gets routed to the FXS module with the matching **In#Accept** of 4000.

Possible Problems and Troubleshooting Tips

Physical Layer Problems:

- If you see any alarms on the ATLAS, check to see which modules are in alarm. You can determine this by looking at the status bar along the bottom of your terminal or telnet session. The numbers at the bottom indicate the slots in the ATLAS. If **ALRM** appears next to any of the slots, this indicates that at least one of the ports on the module in that slot is in alarm. To view the alarm, go to **Modules** and find the slot that is in alarm. Highlight the **Menu** column next to that slot and press <enter>. Look at the **Alarm Status** to see the active alarms.
- An active alarm is indicated by a '*' next to the text. If you have an active **LOS** and **RED** alarm, check to make sure your cables on the T1 port are plugged in. If you are connecting to a T1 Smart Jack, you will need to use a straight-through RJ-45 cable. These are the gray cables that are shipped with the ATLAS. If you are setting this up in a lab environment, you will need a T1 crossover cable in between the two ATLAS units. These are the red cables that are shipped with the ATLAS.
- If you have an active RED alarm only, check the Frame and Code options to make sure they match the frame and code type of the T1. You can look at these options by going to Modules/Menu/Configuration.
- If you have active **RED** and **BLUE** alarms, this indicates a problem in the circuit. Contact your T1 provider.
- If you have an active **YELLOW** alarm, this indicates that the other end of the circuit is in alarm. Check on that end to see what alarm is active. If the other end also has an active **YELLOW** alarm, the problem is most likely with the T1 line and you will need to contact the carrier.
- If you have an active **DS0** alarm, this means that at least one of your DS0s has a problem.

Troubleshooting the Physical Layer Problem

- Use the T1 loopback plug that is shipped with the ATLAS to determine if the port and the cable are good. The loopback plug is located on the power cord of the ATLAS. Plug one end of a straight-through RJ45 cable into the port you are trying to test on the ATLAS. Plug the other end into the loopback plug. If you alarm is still on, try a different cable. This should clear any T1 alarms you may have, and this verifies that the port and the cable are working.

Configuration Issues

- If there are no active alarms on the ATLAS, you may have a configuration problem. Go back through this technical support note and make sure the configuration is correct. If you have verified the configuration, you may need to use the event log to aid in troubleshooting.

Using the Event Log to aid in Troubleshooting

- The event log can be found in the menu structure under **System Status/Event Log**. The ATLAS can be configured to send messages of varying degrees to the event log. To set the degree of the messages that will be sent to the event log, go to **System Config/Event Logging**. The default settings are all at **Minor**. To see all information for a particular message, change it to **Info**. For this application, you might try looking at the **Switchboard** information. Please note that the event log will only hold 349 entries. You may want to use the **Syslog** feature of the ATLAS to capture all entries.