ZyXEL Prestige P128
1.50 Release Note/Manual Supplement

Date: August 17, 1998

This Release Note/Manual Supplement contains information about the new features, and bug fixes since the incremental release v1.42(B.03I).

The **New Features** section lists the new features added to this release. Please read through the **Enhancement Details** section for detail information. A new section**, Changes not Compatible to Previous Versions**, is added to this release notes. It is important to read it too.

1 New Features

SUA Enhancements

In this release, SUA has been enhanced to support the popular applications: Microsoft's tracert under Windows9x and Windows NT, CuSeeMe, IRC, RealAudio, VDOLive, Quake, and PPTP. Besides above, multiple SUA servers are supported.

Advanced ISDN Features

ISDN Supplementary Services are supported in this release. These services work properly only if the needed functions subscribed from the Telephone Company or PTT.

B-channel, CLID, and WAN IP Address Shown on Menu 24.1

B-channel, CLID, and WAN IP address are shown on Menu 24.1. Due to the space limitation in Menu 24.1, Menu 24.1 has been separated into to two screens. The original static information in Menu 24.1 has been moved to Menu 24.2.1. Now, Menu 24.1 only displays the dynamic information.

2 Changes not Compatible to Previous Versions

2.1 ISDN EPA Mechanism is Moved to PC

The CI (Command Interpreter, menu 24.8) command, "isdn ana display", is disabled in this release. A new CI command "sys epalog" is added for displaying the ISDN raw trace data. "isdn ana display" could not work with Telnet before, but "sys epalog" works fine with Telnet. So you do not have to rely on terminal emulator via serial port to capture the ISDN trace any more.

A new DOS tool **- epapc**, will be used to decode the ISDN raw trace data into meaningful Q.921 and Q.931fields. **epapc** is part of this release. If you have difficulties to run **epapc**, please send the raw ISDN trace to ZyXEL support for decoding. The proper steps to take the raw ISDN trace are:

1. type “isdn analyzer on”
2. try your ISDN call now
3. after failed ISDN connection type “isdn analyzer off”
4. Type “sys epalog”
5. Start to capture the data displayed on the screen

6. Type <CR> to show the raw ISDN trace on screen
7. Press <space bar> to keep Prestige displaying the trace until hit end
8. Stop the capturing mechanism

2.2 Busy Tone Generated after Far-end Hanging up

In the previous releases, Prestige kept silent after far-end hanging up the phone. In this release, Prestige generates busy tone after far-end hanging up. There exists a side effect for those answering machines based on silent to detect the end of the message. The work around is to issue a CI command and reboot the Prestige.

- if the answering machine is on POTS port 1, the CI command is:
  isdn set initstring s78.0=1
- if the answering machine is on POTS port 2, the CI command is:
  isdn set initstring s78.1=1
- if the answering machines are on both POTS ports, the CI command is:
  isdn set initstring s78.0=1s78.1=1

If you do not want to reboot your Prestige after issuing the above CI command, you can issue another CI command to activate it:

isdn send initstring

You can reset the function to default by issuing:

isdn set initstring

3 Supported Platforms

v1.50(B.00) supports Prestige models: P128 and P128L.

4 Enhancement Details

4.1 SUA Enhancements

SUA has been enhanced to support the popular Internet applications: MS tracert, CuSeeMe, IRC, RealAudio, VDOLive, Quake, and PPTP. There is no configuration needed for supporting these applications.

For multiple SUA server support, the ***Server IP Addr*** field in Menu 4 and 11.2 has been removed and a new Menu 15 is added.

```
Menu 15 - Multiple Server Configuration

Port #          IP  Address
---------       -----------------

1. Default      0.0.0.0
2. 0            0.0.0.0
3. 0            0.0.0.0
4. 0            0.0.0.0
5. 0            0.0.0.0
6. 0            0.0.0.0
7. 0            0.0.0.0
8. 0            0.0.0.0
```

There are up to eight SUA servers can be configured in this menu. The first one is reserved for the default server, and it can not be changed. You only need to assign the IP address to this default server. For other SUA servers, you need to assign them both of the Port number and the IP address.

If the default server has been configured, the entire unknown (port number is not configured in this Menu, nor in SUA table) WAN incoming IP packets will be routed to this IP address. Otherwise, Prestige will discard them silently if they are neither Telnet (port number 23) nor Authentication (port number 113) packets. For Telnet packets, Prestige will perform as Telnet server in this case. For Authentication packets, Prestige will reject them.

The most often used port numbers are:

Table 1. Services vs. Port number:

Service	Port number
FTP	21
telnet	23
SMTP	25
DNS (Domain Name Server)	53
www-http (Web)	80
PPTP	1723

4.2 Advanced ISDN Features

Supplemental Services are supported for North American and DSS-1 switches. The relationship among the advanced ISDN features and switch types is:

Table 2. Advanced ISDN features vs. ISDN variances:

Feature:	US	DSS-1	1TR6
Incoming Call Bumping (MP)*	y	y	y
Outgoing Call Bumping (MP)*	y	y	y
Call Waiting	y	y	n
Call Transfer	y	n	n
Call Forwarding	y	n	n
Three Way Calling (Conference Call)	y	n	n
Reminder Ring	y	n	n

Notes:

* - feature supported since v1.3
y - feature supported in this release
n - feature not supported in this release

4.2.1 Before You Begin

Advanced ISDN Features refers to Call Waiting, Call Hold, Conference Calling, Call Transfer, and Reminder Ring on the Prestige POTS ports. There are services on the serving Central Office switch that works in cooperation with the Prestige software must first be enabled. These services usually are extra charges in addition to your monthly payment.

Additional Call Offering (ACO) (in Europe the same service is better known as “Call Waiting”) is required to be subscribed on your ISDN line in order to utilizing the Call Waiting (CW) feature. Flexible Calling is required on your ISDN line in order to using the Three Way Calling (Call Conference), or Call Transfer features. You may want to check with your phone company to confirm if these services are available to you.

In some cases, your telephone company may enable the subscribed services on your first directory (phone) number only. You have to request your telephone company to enable the services on your second directory number as well if you do need the features on both of the POTS ports.

4.2.2 Advanced ISDN Features Supported by Northern American Version

4.2.2.1 Menu 2 – ISDN Setup

If the advanced ISDN features are enabled on both of your PORT ports, please follow the suggestion in this section in order to making the advanced ISDN features work properly on both of them. It is important to select **Phone 1** in ***Incoming Analog Call*** field under **1st Phone** field, and to select **Phone 2** in ***Incoming Analog Call*** field under **2nd Phone field.**

For example, if you have two telephone numbers, 5552000 and 5554000, and you want to assign 5552000 to POTS port 1, then configure it as:

```
Menu 2 - ISDN Setup

Switch Type= xyz
B Channel Usage= Switch/Switch
1st Phone #= 5552000
  SPID #= 0555200001
  Incoming Analog Call= Phone 1
2nd Phone #= 5554000
  SPID #= 0555400001
  Incoming Analog Call= Phone 2

Advanced Setup = No
```

If you want to assign 5554000 to POTS port 1, then configure it as:

```
Menu 2 - ISDN Setup

Switch Type= xyz
B Channel Usage= Switch/Switch
1st Phone #= 5554000
  SPID #= 0555400001
  Incoming Analog Call= Phone 1
2nd Phone #= 5552000
  SPID #= 0555200001
  Incoming Analog Call= Phone 2

Advanced Setup = No
```

The important point is to select **Phone 1** in ***Incoming Analog Call*** field under **1st Phone** field, and to select **Phone 2** in ***Incoming Analog Call*** field under **2nd Phone field.**

4.2.2.2 Menu 2.1 – ISDN Advanced Setup

In Menu 2, move cursor to the ***Advanced Setup*** field and select **Yes**, then press the “ENTER” key. This will pop up the ISDN Advanced Setup menu for configuring supplemental services and other option as shown below in menu 2.1.

```
Menu 2.1 -- ISDN Advanced Setup

ISDN Features Access Code:
  Conference Call= 60
  Call Transfer= 61
  Call Drop= 62
  Call Forwarding= 57

Phone 1 Call Waiting= Enable/Disable
Phone 2 Call Waiting= Enable/Disable
Preferred Phone # for 1st Out Data Call= None/1st/2nd
```

Please note that the default values entered in the feature access code fields are the values that are specified by the NI-1 standard. Please ask your Telephone Company about the right access code for your ISDN line provision. For certain RBOCs like Pacific Bell, or switches like the Northern Telecom Custom, the values could be different from above. Take the access code provided by your telephone company; otherwise, our recommended values are:

Access Code	Feature
4	Conference Call
5	Call Transfer
6	Call Drop
57	Call Forwarding

By toggling the ***Phone 1 Call Waiting*** and ***Phone 2 Call Waiting*** fields, you can enable and disable the Call Waiting feature on the POTS port phones. Default is to disable the Call Waiting feature on both phones.

The purpose for adding the ***Preferred Phone # for 1st Out Data Call field*** is to allow user to reserve a Directory (Phone) number for incoming calls. When there is no active data or voice call, the first Directory (Phone) number (DN-1) will be used to place the out data call if **None** or **1st** is selected, and DN-2 will be used if **2nd** is selected. The difference between **None** and **1st** is the B-channel requested in Setup message from Prestige to switch. There is no B-channel preference set in the Setup message if **None** is selected. The B-channel preference is **B1** if **1st** is selected, and the B-channel preference is **B2** if **2nd** is selected. Here is a brief description about the choices of this field:

None: DN-1 is used to place the out data call.
1st: DN-1 is used to place the out data call, and the preferred B-channel is B1.
2nd: DN-2 is used to place the out data call, and the preferred B-channel is B2.

We have to emphasize that the above selection is only a preference. Switch may reject the B-channel request; and different DN is used because the desired Directory (Phone) number used by other call.

4.2.2.3 Call Waiting

ISDN Call Waiting allows you to place a call on hold while you answer another incoming call on the same phone (directory) number.

4.2.2.3.1 How to Use Call Waiting

- Put your current call on hold and answer the incoming call - after hearing the call waiting indicator tone, press and immediately release the flash hook button on your telephone.
- Put your current call on hold and switch to another call - press and immediately release the flash hook button on your telephone.
- Hang up your current call before answering the incoming call – hang up the phone and wait for the phone to ring. Then answer the incoming call.
- Hang up on the current active call and switch back to the other call – hang up the phone and wait for the phone to ring. Then pick up the phone to return to the other call.

4.2.2.3.2 Why Call Waiting not Work as You Expected

1. An incoming caller will receive a busy signal if:

   - you have two calls (one active and one on hold; or both actives by using Three Way Calling) on the Directory (Phone) number the incoming caller is attempting to reach.
   - you are dialing out by using the Directory (Phone) number the incoming caller is attempting to reach, but have not yet established a connection.

2. If no action is taken (call waiting indicator tone is ignored) to pickup the call, the call waiting tones will disappear after about 20 seconds.
3. There exists a rollover call. Please take a look of **Call Rollover Support** section for detailed information.

An application note for Northern American Call Waiting examples is included in Appendix A

4.2.2.4 Call Rollover Support

In the previous releases, Prestige did not support Call Roller. Prestige supports this feature in v1.50. Unfortunately, this feature is switch dependent. Call Rollover Works only if the Switch Type is AT&T NI-1.

4.2.2.4.1 Incoming Call Rollover

If the Directory (Phone) number, DN-1, associated to POTS port 1 is used by a data call (it might be another analog call to POTS port 2 even though it is really rare), the incoming analog call intended to DN-1 will be routed to POTS port 1. It is called incoming analog call rollover. None of the advanced ISDN features works for this incoming analog rollover call.

If the Directory (Phone) number, DN-1, associated to POTS port 1 is used by an analog call, the incoming data call intended to DN-1 will be answered. It is called incoming data call rollover. The advanced ISDN features do not work on the analog call if there is an incoming data rollover call at the same time.

4.2.2.4.2 Outgoing Call Rollover

For outgoing data call, there is no problem for Prestige to pick the unused Directory (Phone) number to place the call. So, there is no outgoing data rollover call problem. It is different for the outgoing call initiated by the POTS port.

If the Directory (Phone) number, DN-1, associated to POTS port 1 is used by a data call, Prestige will use DN-2 as the caller number to place the call from POTS port 1. It is called outgoing voice call rollover. Call Waiting does not work in this case even though other advanced ISDN features work fine.

4.2.2.5 Three Way Calling (Call Conference)

The Three Way Calling feature allows you to add the third party to an existing call. The Flexible Calling service must be subscribed from the Telephone Company.

4.2.2.5.1 How to Add the Third Party to the Existing Call

- If you wish to add the third party to an existing call, the steps are:

    1. Press the flash hook button and immediately release it to put the existing call on hold and receive a dial tone.
    2. Dial the third party.
    3. Inform the third party about the conference.
    4. When you are ready to conference the call, press the flash hook button and immediately release it to establish a Three Way Conference Call.

- If you wish to cancel your attempt for some reason (the third party's line is busy, or no one answer, …), just hang up the phone and pick it back up after the phone ringing.

4.2.2.5.2 How to Remove a Party from the Three Way Calling

- If you wish to drop the last one added to the Three way calling call, just press the flash key. The last call that was added to the conference will be dropped.
- If you wish to drop yourself from the conference call, but allow the other two callers to remain connected. Just hang up your phone. If the other two remain on the line, your drop will not impact their connection.

4.2.2.6 Call Transfer

Call Transfer allows you to transfer an active call to a third party. The Flexible Calling service has to be subscribed from the Telephone Company. In fact, Call Transfer is a variance of Three Way Calling.

If you wish to transfer an active call to a third party and inform him about the transferred call, the steps are:

1. Press flash to immediately put the existing call on hold and receive a dial tone.
2. Dial the third party.
3. Inform the third party about the transfer call.
4. Press the flash hook button and immediately release it to establish a Three Way Conference Call.
5. Hang up the phone to complete the transfer.

If you wish to do a blind transfer to the third party, the steps are:

1. Press flash to immediately put the existing call on hold and receive a dial tone.
2. Dial the third party.
3. Before the third party picks up the call, you can transfer the call by pressing the flash and hanging up. The call will be automatically transferred.

4.2.2.7 Call Forwarding

The Call Forwarding feature is supported by ISDN switch directly. The Call Forwarding feature of the POTS port can be activated and deactivated by using the phone set. The Call Forwarding is a telephone feature and will not impact incoming data call. Please request your phone company for the instruction activate or deactivate the Call Forwarding feature. Here is the instruction written in Custom Calling Service section of Pacific Bell White Pages:

To turn on, press 72#. At the dial tone, enter the number you want your calls forwarded to. If the line is busy or does not answer, repeat the above process and it will be activated even if the line is still busy. To Turn off, press 73#.

4.2.2.8 Reminder Ring

Prestige will alert you every time a call has been forwarded by sending a single short ring to your telephone. This feature is just to let you know that your calls are being forwarded.

4.2.3 Call Waiting Support for DSS-1

In this release, the only supplementary service supported is Call Waiting. Please move cursor to the ***Advanced Setup*** fields in Menu 2 and select Yes, then press the “ENTER” key to pop up the ISDN Advanced Setup as below.

```
            Menu 2.1 -- ISDN Advanced Setup

           Phone 1 Call Waiting= Enable/Disable
           Phone 2 Call Waiting= Enable/Disable
```

By toggling the ***Phone 1 Call Waiting*** and ***Phone 2 Call Waiting*** fields, you can enable and disable the Call Waiting feature on the POTS port phones. Default is to disable the Call Waiting feature on both phones. Here is a brief description about the choices of this field:

Disable: disable the Call Waiting feature on the specific phone.
Enable: enable the Call Waiting feature on the specific phone.

4.3 B-channel, CLID, and WAN IP Address Shown on Menu 24.1

Menu 24.1 has been separated into to two screens. The original static information in Menu 24.1 has been moved to Menu 24.2.1 as below.

```
Menu 24.2.1 - System Maintenance - Information

     Name: 1.5FCS
     Routing: IP/IPX
     RAS S/W Version: V1.50(B.00) | 8/17/98
     ISDN F/W Version: V 07f
     Country Code: 255

     LAN
       Ethernet Address: 00:a0:c5:40:00:ee
       IP Address: 204.237.202.151
       IP Mask: 255.255.255.192
       DHCP: None
```

Menu 24.1 only displays the dynamic information now. The B-channel instead of the logical channel is shown under **CHAN** now. The CLID of the incoming data or voice call will be shown on this menu too. So is WAN IP address.

```
                Menu 24.1 - System Maintenance - Status

CHAN   Link    Type     TXPkt    RXPkt    Error  CLU   ALU    Up Time
 B1    isp     64Kbps   28923    25392      0    12%   1%     0:28:24
 B2    Voice    0Kbps   15521    13194      0     0%   0%     0:00:00

     Total Outcall Time:    11:31:06

     Ethernet                        WAN
       Status: 10M/Half Duplex        B1 IP Addr: 202.237.102.121
       TX Pkt: 59389                  B2 IP Addr:
       RX Pkt: 126924                 B1 CLID:
       Collision: 3                   B2 CLID: 12345

     LAN Packet Which Triggered Last Call: (Type: IP)

     45 00 00 38 7D 14 00 00 3F 11 C8 0B CA E7 BB 61 9D 16 01 06 00 35 00 35
     00 24 39 90 79 65 00 00 00 01 00 00 00 00 00 00 06 64 6F 6D 61 69 6E 03

                         Press Command:
   COMMANDS: 1-Drop B1  2-Drop B2  3-Reset Counters  4-Drop All  ESC-Exit
```

5 Bug fixes

1. Some Prestige users experienced system hang once a couple of weeks. When it happened, it needed human intervention to power cycle the Prestige for making it work again. In this release, Prestige will re-boot itself to eliminate the human intervention.
2. Prestige used to have a bug in its SUA mechanism. This bug caused Prestige dropping the fragmented WAN incoming IP packets. The symptom is that you have problems to certain Web sites if the Prestige is configured SUA. It is fixed in this release.

3. Prestige's behavior was unexpected if aggregation was requested in the callback case. It was because Prestige did not implement callback request and response for BAP (Bandwidth Allocation Protocol). Prestige has been enhanced to handle callback request and response for BAP in this release.
4. Prestige used to write Error or Info messages to its system log after ping and telnet finished. These messages might cause users unnecessary concerns. There will be no more this kind of messages in system log after this release.
5. Prestige did not respond to HDAP (Host Discovery and Address Assignment Protocol) requests from PWC (Prestige Web Configurator) any more after its IP address changed. It is fixed in this release.
6. The information in the call history entry is nonsense if the data in Rate column is blank in Menu 24.9.4. But Prestige should not show it anyway. Prestige does not show it any more.
7. ***My Login*** and ***My Password*** fields for Outgoing call in Menu 11.1 are not required fields anymore.
8. Add CI command "ip dhcp reset" to clean up the DHCP (Dynamic Host Configuration Protocol) table. It means that the DHCP will be empty after this CI command.
9. Prestige did not inform far-end its MAC address in IPXCP negotiation. It has been added.
10. Prestige used to route the WAN incoming IP multi-cast packets to default route. Now, it will drop these incoming multi-cast packets.
11. Prestige will exit from Menu 24.1 after unplug the RS232 cable now.
12. Prestige used to have problems to transfer big files (over 56K bytes after compression) with Ascend MAX if compression is on. It is fixed.
13. Prestige will negotiate LCP magic number with remote router now.
14. By default, Prestige used to negotiate for PPP VJ slot ID compression, it has been changed not negotiate for PPP VJ slot ID compression any more.
15. Prestige will send the ICMP redirect packet to the source of the packet if Prestige knows the gateway, which provides better route from the station to the destination than using Prestige as the gateway.
16. In Menu 24.1, the value of *<u>Total Outcall Time</u>* is the sum of the outcall time of switched channels now.
17. The system has been tuned to eliminate some of the MP queue full and Stac compression sequence error messages even though we could not eliminate all of them.
18. The range of callback timeout has been changed from "5 to 60 seconds" to "1 to 60 seconds".
19. In this release, Prestige will prevent the IP route of a newly added remote node overwriting the LAN routing entry in routing table.
20. For P128L, the default value for Idle timeout in Menu 11.1 is 0 (infinite) now.
21. In this release, Prestige will drop the link even if the far-end does not respond to Prestige's request for dropping the link in case BAP used for aggregation negotiation.
22. For eliminating new user's confusion on entering ISP's IP address. ***ISP IP Addr*** field in Menu 4 is removed.
23. For Denmark ISDN switch, the right ring frequency has now been set in the A/B ports defaults. You need no longer manually enter the CI command "isdn set initstring s121=2" as in the past.

6 Known Problem List

1. If both of the POTS port telephones are taken off hook simultaneously while an MP call is in progress, the Prestige may drop both data channels.
2. If Prestige connects to the switch that does not support in-band tone, the tone will generated by Prestige instead. In this case, Prestige will send the same tone to both POTS ports. For example, when telephone 1(telephone connects to POTS port 1) is ringing, off-hooking telephone 2 (telephone connects to POTS port 2) will cause telephone 1's sound changing from ring to dial tone. It is because Prestige generates dial tone for POTS port 2 now.

3. For European version, a global digital call will still ring and can be answered even if **MSN** is selected in Menu 2 as the ***incoming call matching*** method.
4. The POTS port (A/B adapter) dial tone may disappear if call bumping is attempted twice in rapid succession on a switch that does not support in-band tone.
5. For European version, the ISDN ***Link*** status still shows **Idle** in Menu 24.1 even if the cable is unplugged.
6. In very large IPX networks, you may have problems accessing the desired servers.
7. The TCP/IP filters configured in Menu 21 may not work properly if it is a SUA connection. It is because Prestige executes the call/output filter after SUA converting the source IP address and port number of the packets from LAN. And executes the input filter before SUA converting destination IP address and port number to local address and port number.

7 To Upgrade Prestige

Get the files from ZyXEL anonymous FTP server (ftp.zyxel.com). Upgrade your Prestige by following the instructions for your model:

Versions:
RAS S/W Version - V1.5(B.00) | 8/17/97
ISDN F/W Version - DSS1: V 07f
1TR6: V 07f
USA : V 07f

RAS and ISDN firmware files:
p128a.bin (for Northern America)
p128e.bin (for DSS1 and Prestige 128L)
p128g.bin (for 1TR6)

Commands:
ATBAx: Where x = baud rate
options available are:
1= 38.4K
2= 19.2K
3= 9.6K
4= 57.6K
5= 115.2K
ATUR: Upload Firmware file via XMODEM

Romfile: romfile.zip (romfile0)
ATUR3: Upload Romfile and clear all settings
Note: You don't need to upload this file if you are upgrading from release 1.2(b.01) unless you want to reset all configurations to factory default.

Appendix A

Application notes: Call waiting support Version: 001

Setup: User set DN1 ("1st Phone #" in Menu 2) to Phone1 and set DN2 ("2nd Phone #" in Menu 2) to Phone2. In Menu2.1, all Access Codes are set correctly according to the local Telco switch. The "`Preferred Phone # for 1st Out Data Call`" is set to use first DN. Both POTS have subscribed supplemental service and ACO turn on. We use Northern Tel NI-1 in this example, however, most of the cases, the switch type doesn’t make a difference. If the behavior of the Prestige is different for different switch type, we will also mention it in this example.


```
  Menu 2 - ISDN Setup

Switch Type= Northern Tel NI-1

B Channel Usage= Switch/Switch
1st Phone #= xxx0101
  SPID #= 408xxx0101
  Analog Call=  Phone 1
2nd Phone #= xxx0202
  SPID #= 408xxx0202
  Analog Call=  Phone 2

Advanced Setup = No
```

```
Menu 2.1 -- ISDN Advanced Setup

ISDN Features Access Code:
  Conference Call= 60
  Call Transfer= 61
  Call Drop= 62
  Call Forwarding= 57

Phone 1 Call Waiting= Enable
Phone 2 Call Waiting= Enable
Preferred Phone # for 1st Out Data Call= 1st
```

In this example, we will describe Prestige in different states, and then using a table to show some actions happened in that state and what user should hear from the phone set. The actions in the table include

Offhook : user pickup the phone
Call to DN1 : incoming voice call to DN1 (Phone 1)
Call to DN2 : incoming voice call to DN2 (Phone 2)

The responses after the action include:
Ring : user can hear phoneset ringing and able to answer the call.
Dial tone : user offhook or single flash can hear the dial tone, and able to make an outcall.
Busy tone : remote caller hears busy from switch or user picks up the phone and hear busy tone from phone set.
Call waiting tone : user hear the tone and can single flash answers it.

This document will discuss what happen when a incoming voice call to a Prestige that is in the following 3 situation:
. All B channels are idle
. Both B channels are occupied by data calls
. One channel is occupied by data call
The following sections will describe each situation in details.

1. Both B channel idle

When both B channels are idle, each phone/DN can answer up to 2 voice calls.

	Action 1	Action 2	Action 3	Action 4
	Call to DN1 / Offhook phone 1	Call to DN2 / Offhook phone2	call to DN1	call to DN2

Phone 1	Ring / Dial tone	--	Call waiting tone	--
Phone 2	--	Ring / Dial tone	--	Call waiting tone

2. Two data connections up

Two data connections could be to the same destination or to different destination. Therefore, we discuss two different cases here.

(1) Both channels are connected to user's ISP. When a voice incoming call to any DN, one data connect will be bumped, and the corresponding phone will ring. After user answer the call, if another voice incoming call to that DN, user will hear call waiting tone and able to flash to answer the 2nd incoming call.

CASE 1: Call to DN1 first.

	Action 1	Action 2	Action 3
	Call to DN1 / Offhook phone 1	Call to DN2 / Offhook phone2	call to DN1
Phone 1	Ring / Dial tone	--	Call waiting tone
Phone 2	--	Caller get busy tone / Busy tone	--

CASE2: Call to DN2 first

	Action 1	Action 2	Action 3
	Call to DN2 / Offhook phone 2	Call to DN1 / Offhook phone1	call to DN2
Phone 1	--	Caller get busy tone / Busy tone	--
Phone 2	Ring / Dial tone	--	Call waiting tone

(2) Two channels are connected to different Remote Nodes, such as one to ISP and one to user's office network. A voice incoming call to any DN will get busy tone.

	Action 1	Action 2
	Call to DN2 / Offhook phone 2	Call to DN1 / Offhook phone1
Phone 1	--	Caller get busy tone / Busy tone
Phone 2	Caller get busy tone / Busy tone	--

3. Data outcall (DN1) to ISP when both phones are onhook (not in use)
Since "`Preferred Phone # for 1st Out Data Call`" is selected as 1ST, and both phones are not in use when the outcall takes place, so the data channel will be using DN1. For incoming voice calls to DN2, user will hear ring from Phone2. For incoming voice call to DN1, the result will depend on the switch type. For AT&T NI-1, the call can be 'rolled over', so Phone1 will ring and user can answer it. For other switch type, since the switch can't do rollover, so the switch won't accept Prestige use DN2 to answer calls to DN1, the caller will hear busy tone. Below are tables for AT&T NI-1 switch type and other switch types.

(1) SWITCH TYPE: AT&T NI-1

CASE 1: An incoming voice call to the DN that's been used by a data connection can be rolled over and answered.

	Action 1	Action 2	Action 3
	Call to DN1 / Offhook phone 1	Call to DN1	Call to DN2 / Offhook phone 2
Phone 1	Ring / Dial tone	Caller get busy tone	--
Phone 2	--	--	Caller get busy tone / Busy tone

CASE 2: An incoming voice call to the DN that's not been used by a data connection can be answered, and furthermore, a second incoming voice call to the phone, user can hear call waiting tone and use single flash to answer it.

	Action 1	Action 2	Action 3
	Call to DN2 /	Call to DN1	call to DN2

	Offhook phone 2		
Phone 1	--	Caller get busy tone	--
Phone 2	Ring / Dial tone	--	Call waiting tone

(2) OTHER SWITCH TYPE:

CASE 1: An incoming voice call to the DN that's been used by a data connection can not be rolled over, and therefore, the caller will hear busy tone.

	Action 1	Action 2	Action 3
	Call to DN1	Call to DN2 / Offhook phone 2	Call to DN2
Phone 1	Busy tone	--	--
Phone 2	--	Ring / Dial tone	Call waiting tone

CASE 2: Offhook phone 1, which the DN is used by the data connection, user can dial out. At this moment, the call is rollover. Therefore, user will not get call waiting tone on phone 1 when a voice call comes into phone 1 when a rollover call occurs, and the caller will get busy tone.

	Action 1	Action 2	Action 3
	Offhook phone 1	Call to DN1	Call to DN2 / Offhook phone 2
Phone 1	Dial tone	Caller get busy tone	--
Phone 2	--	--	Caller get busy tone / Busy tone

4. Data outcall to ISP when one of the Phone is in use

In this case, no call is rollover. Therefore, any incoming voice call to the phone that's in used, user should be able to hear the call waiting tone and use single flash to answer the call.

(1) Phone 1 (DN1) is used to connect to "peer", and then a data call is triggered to ISP (DN2).

	Action 1	Action 2
	Call to DN2 / Offhook phone 2	Call to DN1 / Offhook phone1
Phone 1	--	Call waiting tone / Dial tone
Phone 2	Caller get busy tone / Busy tone	--

(2) phone 2 (DN2) is used to connect to “peer”, and then a data call is triggered to ISP (DN1).

	Action 1	Action 2
	Call to DN1 / Offhook phone 1	Call to DN2 / Offhook phone2
Phone 1	Caller get busy tone / Busy tone	--
Phone 2	--	Call waiting tone / Dial tone