



Configuration Guide for Voice/IP Gateways

Bogen PCM Zone Paging

MultiVOIP Models: MVP130-BG, MVP210-BG,
MVP410-BG, MVP810-BG



Configuration Guide

Doc # Bogen03

MultiVOIP Models MVP130-BG, MVP210-BG, MVP410-BG, MVP810-BG

This publication may not be reproduced, in whole or in part, without prior expressed written permission from Multi-Tech Systems, Inc. All rights reserved.

Copyright © 2004, by Multi-Tech Systems, Inc.

Multi-Tech Systems, Inc. makes no representations or warranties with respect to the contents hereof and specifically disclaims any implied warranties of merchantability or fitness for any particular purpose. Furthermore, Multi-Tech Systems, Inc. reserves the right to revise this publication and to make changes from time to time in the content hereof without obligation of Multi-Tech Systems, Inc. to notify any person or organization of such revisions or changes.

Record of Revisions

Revision	Description
A	Initial Release. (01/07/04)
B	Additional application changes (02/16/04)
C	Add Caution for Low Power (2/24/04)

Patents

This Product is covered by one or more of the following U.S. Patent Numbers: *6151333, 5757801, 5682386, 5.301.274; 5.309.562; 5.355.365; 5.355.653; 5.452.289; 5.453.986.* Other Patents Pending.

Trademark

Trademark of Multi-Tech Systems, Inc. is the Multi-Tech logo. Windows is a registered trademark of Microsoft.

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, Minnesota 55112
(763) 785-3500 or (800) 328-9717
U.S. Fax: 763-785-9874
Technical Support: (800) 972-2439
<http://www.multitech.com>

Bogen paging example using FXS Pass Through feature.

The example below shows a Bogen PCM/ZPM Zone Paging unit at the Corporate site connected through MultiVOIPs to amplifiers and speakers at each of two branch sites. Users at the corporate site can page users at the Branch A and/or Branch B sites.

The FXS Pass Through feature allows an “always on” audio connection to exist between the Corporate site and both Branch A and Branch B sites. The MultiVOIPs use of Silence Compression means little or no bandwidth is used when not paging.

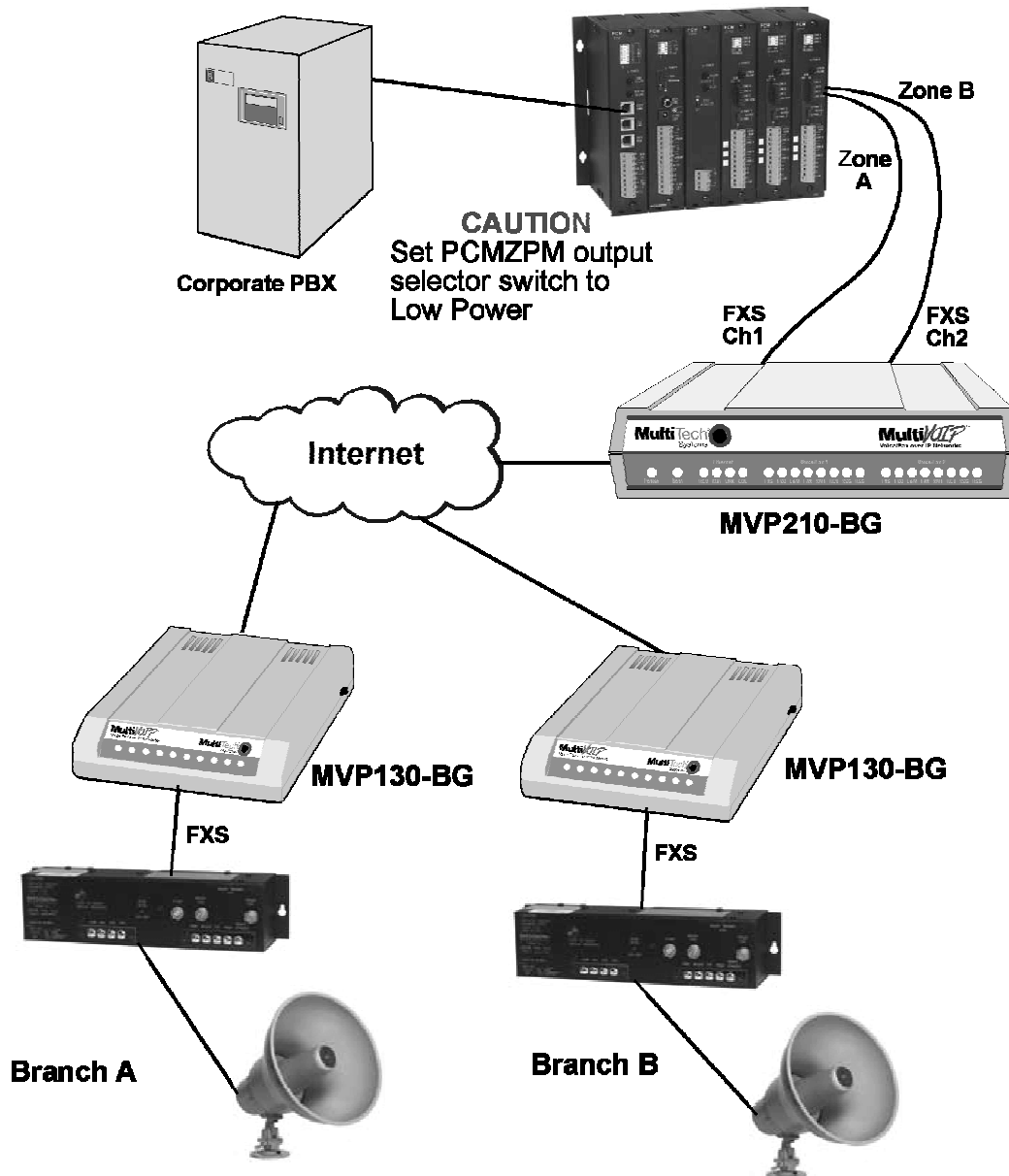


Figure 1. Bogen paging system diagram.

1. Preliminary planning: Determine the number of MultiVOIP gateways to be used for the paging network. Assign a unique IP address and identifying number to each MultiVOIP. It is helpful to write this down like below:

MultiVOIP Name	Identifying Number	IP Address	Local Voice Channel	Destination MultiVOIP Number	Destination MultiVOIP Voice Channel
Corporate	1	192.168.25.20	1	2	1
			2	3	1
Branch A	2	192.168.25.21	1	1	1
Branch B	3	192.168.25.22	1	1	2

2. In the Configuration / IP screen of the MultiVOIP Configuration software, configure each MultiVOIP with a unique **IP address**. Also configure the **mask** and **gateway** address. If all MultiVOIPs are located in the same subnet, you can leave the gateway address field blank. Click Ok when finished.

The screenshot shows the 'IP Parameters' configuration window. It is divided into several sections:

- Diff Serv Parameters:** Call Control PHB is 34, VoIP Media PHB is 46, and Frame Type is TYPE-II.
- IP Parameters:** Enable DHCP is unchecked. IP Address is 192.168.25.20, IP Mask is 255.255.255.0, and Gateway is blank.
- DNS:** Enable DNS is unchecked, and DNS Server IP Address is blank.
- FTP Server:** Enable is checked.
- TDM Routing Option:** Use TDM Routing For Intra-Gateway calls is unchecked.

Buttons for OK, Cancel, and Help are located on the right side of the dialog.

Corporate

IP Parameters

Diff Serv Parameters

Call Control PHB : 34

VoIP Media PHB : 46

Frame Type TYPE-II

IP Parameters

Enable DHCP

IP Address : 192 . 168 . 25 . 21

IP Mask : 255 . 255 . 255 . 0

Gateway : . . .

DNS

Enable DNS

DNS Server IP Address : . . .

FTP Server

Enable

TDM Routing Option

Use TDM Routing For Intra-Gateway calls

OK

Cancel

Help

Branch A

IP Parameters

Diff Serv Parameters

Call Control PHB : 34

VoIP Media PHB : 46

Frame Type TYPE-II

IP Parameters

Enable DHCP

IP Address : 192 . 168 . 25 . 22

IP Mask : 255 . 255 . 255 . 0

Gateway : . . .

DNS

Enable DNS

DNS Server IP Address : . . .

FTP Server

Enable

TDM Routing Option

Use TDM Routing For Intra-Gateway calls

OK

Cancel

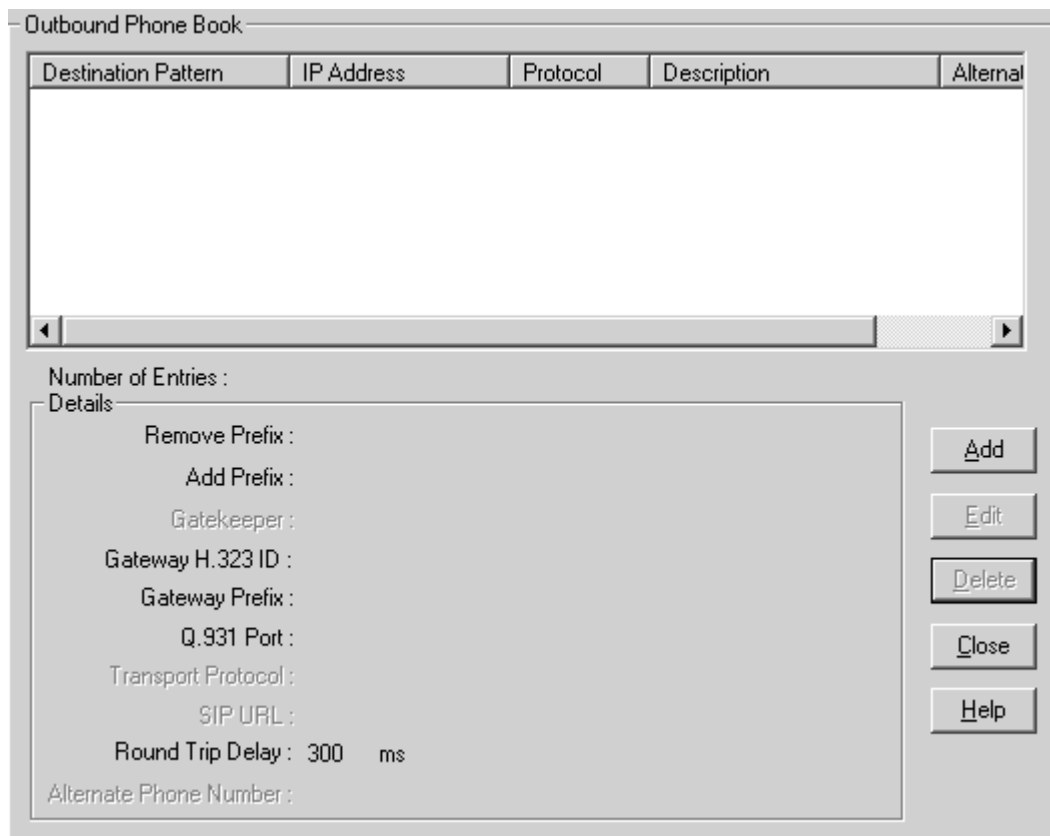
Help

Branch B

3. In the Configuration/Interface screen, Select each voice channel used for paging from the Select Channel drop-down list and configure that channel so **FXS Loop Start** and **Pass Through** are enabled. Click **Ok** when finished.

The screenshot shows the 'Interface Parameters' configuration window. The 'Select Channel' dropdown is set to 'Channel 1'. The 'Interface' section has 'FXS (Loop Start)' selected. The 'Dialing Options' section has 'Regeneration' set to 'DTMF' and 'Inter Digit Timer' set to 2 secs. The 'E&M Options' section has 'Signal' set to 'Dial Tone' and 'Wink Timer' set to 250 ms. The 'FXD Disconnect On' section has 'Current Loss' checked. The 'Ring Count' section has 'FXS' set to 8 and 'FXD' set to 2. The 'Flash Hook Options' section has 'Generation' set to 600 ms. The 'Pass Through' section has 'Enable' checked. The 'Inter Digit Regeneration Timer' is set to 100 ms. The 'Disconnect Tone Sequence' is set to 'Fax + None'. The 'Silence Detection' is set to 'None'. The 'Disconnect On Call Progress Tone' is not enabled. The 'Current Loss Detect Timer' is set to 500 ms. The 'Detection Range' has 'Min' set to 100 ms and 'Max' set to 1000 ms. The 'Mode' is set to '2Wire'. The 'Type' is set to 'TYPE II'. The 'Message Waiting Light' is not checked. The 'Current Loss' option in the 'FXS Options' section is not checked. The 'Wink' option in the 'Signal' section is not selected. The 'Tone Detection' option in the 'FXD Disconnect On' section is not checked. The 'Enable' option in the 'Disconnect On Call Progress Tone' section is not checked. The '4Wire' option in the 'Mode' section is not selected. The 'None' option in the 'Disconnect Tone Sequence' section is selected. The 'None' option in the 'Silence Detection' section is selected. The 'None' option in the 'Regeneration' section is not selected. The 'Pulse' option in the 'Regeneration' section is not selected. The 'Ground Start' option in the 'Interface' section is not selected. The 'FXD' option in the 'Interface' section is not selected. The 'E&M' option in the 'Interface' section is not selected. The 'Channel 1' option in the 'Select Channel' dropdown is selected. The 'OK', 'Cancel', 'Copy Channel', 'Default', and 'Help' buttons are visible on the right side of the window.

-
4. Go to the Phone Book / Phone Book Modify / Outbound Phone Book / List Entries screen.



Outbound Phone Book

Destination Pattern	IP Address	Protocol	Description	Alternat
---------------------	------------	----------	-------------	----------

Number of Entries :
Details

Remove Prefix :
Add Prefix :
Gatekeeper :
Gateway H.323 ID :
Gateway Prefix :
Q.931 Port :
Transport Protocol :
SIP URL :
Round Trip Delay : 300 ms
Alternate Phone Number :

Add
Edit
Delete
Close
Help

-
5. Click the Add button to add a phone book entry. In the **Destination Pattern** field, enter the identifying number for one of the remote MultiVOIP units. Enter the IP address of the remote MultiVOIP unit in the **IP Address** field. Leave the other fields set at their defaults.

Add/Edit Outbound Phone Book

Phone Number Details

Use as default entry

Destination Pattern : 2

Total Digits : 0

Remove Prefix :

Add Prefix :

IP Address : 192 . 168 . 25 . 21

Description :

Protocol Type

SIP H.323 SPP

H.323

Use GateKeeper

Gateway H.323 ID :

Gateway Prefix :

Q.931 Port Number : 1720

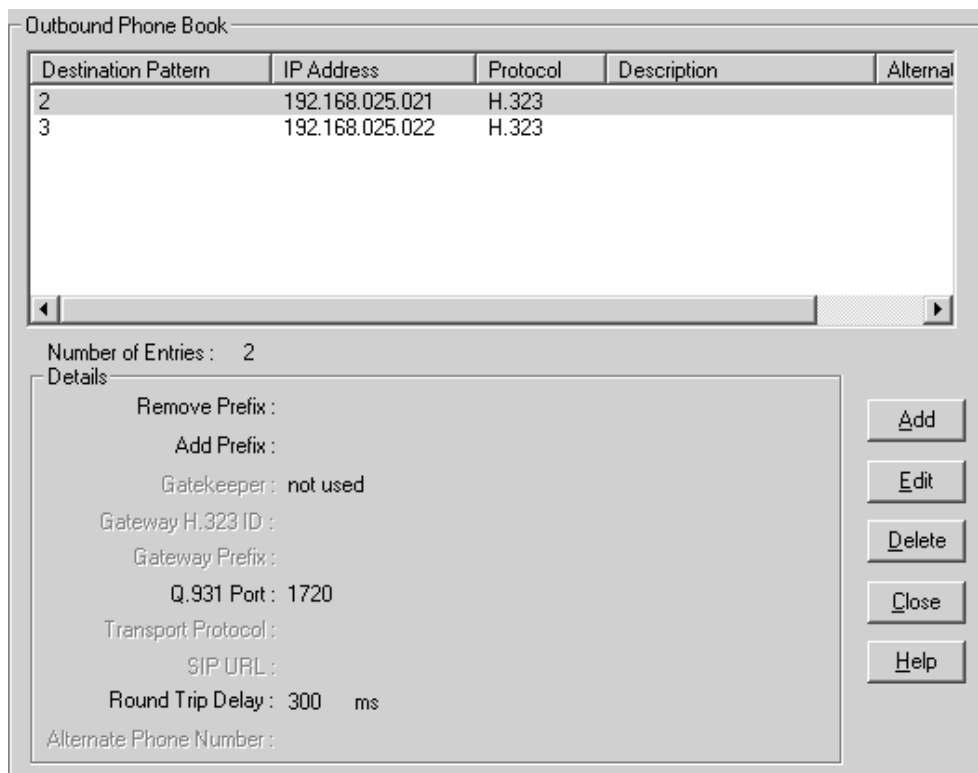
OK

Cancel

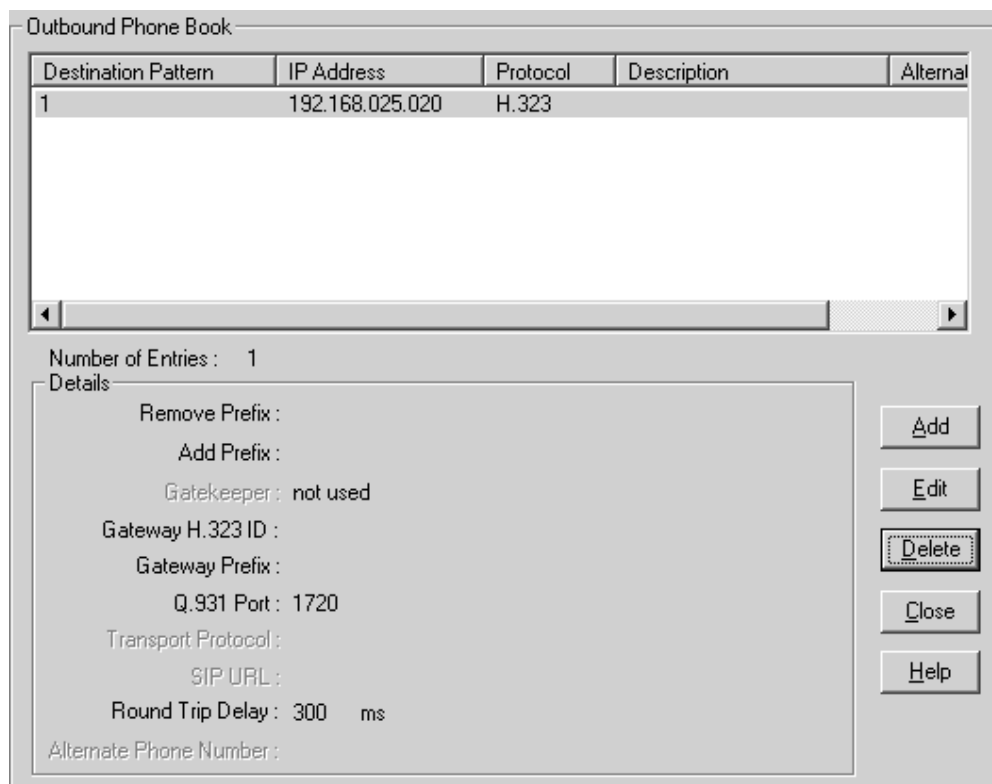
Help

Advanced

6. Click **Ok** when finished with this entry. Click the **Add** button as needed to create entries for other remote MultiVOIP units. In our example, the Corporate MultiVOIP would have two entries like below:



The Branch A and Branch B MultiVOIP entries would look like this:



- In the configuration / Voice/Fax screen, select one of the channels to be used for paging and click the **Auto Call Enable** checkbox to enable the auto call feature. In the **Phone Number** field, enter x:y where x equals the remote MultiVOIP identifying number and y equals the remote MultiVOIP Voice Channel number. For example, on Corporate MultiVOIP channel 1 you would enter **2:1** to communicate with Branch A voice channel 1. For best voice quality, configure the **Voice Coder** field to G.711 U-Law @ 64 kbps. For good voice quality requiring less bandwidth, configure the Voice Coder field to G.723.1 @ 6.3 kbps. You must use the same voice coder on channels that communicate with each other. Leave the other fields set to defaults. Configure the other channels as needed and click **Ok** when finished.

Voice/Fax Parameters

Select Channel: Channel 1

Voice Gain: Input 0 dB, Output 0 dB

Dtmf: Gain High -4 dB, Low -7 dB; Duration 100 ms; DTMF: Out Of Band - Fixed Duration

Fax: Fax Enable; Max Baud Rate 14400; Fax Volume -9.5 dB; Jitter Value 400 ms; Mode FRF 11

Coder: Manual, Automatic; Selected Coder G.711 U - law @ 64 kbps; Max bandwidth 10 kbps

Advanced Features: Silence Compression, Echo Cancellation, Forward Error Correction

Auto Call: Auto Call Enable, Generate Local Dial Tone; Phone Number 2:1

Dynamic Jitter Buffer: Minimum Jitter Value 60 ms, Maximum Jitter Value 300 ms, Optimization Factor 7

Automatic Disconnection: Jitter Value 350 ms, Call Duration 180 secs, Consecutive Packets Lost 30, Network Disconnection 300 secs

Buttons: OK, Cancel, Copy Channel, Default, Help

Corporate Channel 1

Voice/Fax Parameters

Select Channel: Channel 2

Voice Gain: Input 0 dB Output 0 dB

Dtmf Gain: High -4 dB Low -7 dB

Duration: 100 ms

DIMF: Out Of Band - Fixed Duration

Coder: Manual Automatic
Selected Coder: G.711 U - law @ 64 kbp:
Max bandwidth: 10 kbps

Advanced Features:
 Silence Compression
 Echo Cancellation
 Forward Error Correction

Fax:
 Fax Enable
 Max Baud Rate: 14400
 Fax Volume: -9.5 dB
 Jitter Value: 400 ms
 Mode: FRF 11

Auto Call:
 Auto Call Enable Generate Local Dial Tone
 Phone Number: 3:1

Dynamic Jitter Buffer:
 Minimum Jitter Value: 60 ms
 Maximum Jitter Value: 300 ms
 Optimization Factor: 7

Automatic Disconnection:
 Jitter Value: 350 ms Consecutive Packets Lost: 30
 Call Duration: 180 secs Network Disconnection: 300 secs

OK Cancel Copy Channel Default Help

Corporate Channel 2

Voice/Fax Parameters

Select Channel: Channel 1

Voice Gain: Input 0 dB Output 0 dB

Dtmf Gain: High -4 dB Low -7 dB

Duration: 100 ms

DIMF: Out Of Band - Fixed Duration

Coder: Manual Automatic
Selected Coder: G.711 U - law @ 64 kbp:
Max bandwidth: 10 kbps

Advanced Features:
 Silence Compression
 Echo Cancellation
 Forward Error Correction

Fax:
 Fax Enable
 Max Baud Rate: 14400
 Fax Volume: -9.5 dB
 Jitter Value: 400 ms
 Mode: FRF 11

Auto Call:
 Auto Call Enable Generate Local Dial Tone
 Phone Number: 1:1

Dynamic Jitter Buffer:
 Minimum Jitter Value: 60 ms
 Maximum Jitter Value: 300 ms
 Optimization Factor: 7

Automatic Disconnection:
 Jitter Value: 350 ms Consecutive Packets Lost: 30
 Call Duration: 180 secs Network Disconnection: 300 secs

OK Cancel Copy Channel Default Help

Branch A Channel 1

Voice/Fax Parameters

Select Channel: Channel 1

Voice Gain: Input 0 dB Output 0 dB

Dtmf Gain: High -4 dB Low -7 dB

Duration: 100 ms

DIMF: Out Of Band - Fixed Duration

Coder: Manual Automatic
Selected Coder: G.711 U - law @ 64 kbp:
Max bandwidth: 10 kbps

Advanced Features:
 Silence Compression
 Echo Cancellation
 Forward Error Correction

Fax:
 Fax Enable
Max Baud Rate: 14400
Fax Volume: -9.5 dB
Jitter Value: 400 ms
Mode: FRF 11

Auto Call:
 Auto Call Enable Generate Local Dial Tone
Phone Number: 1:2

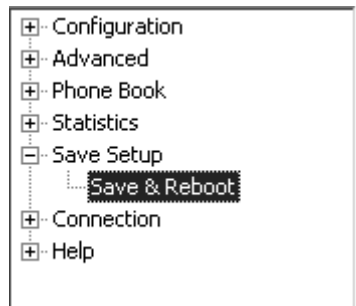
Dynamic Jitter Buffer:
Minimum Jitter Value: 60 ms
Maximum Jitter Value: 300 ms
Optimization Factor: 7

Automatic Disconnection:
 Jitter Value: 350 ms Consecutive Packets Lost: 30
 Call Duration: 180 secs Network Disconnection: 300 secs

Buttons: OK, Cancel, Copy Channel, Default, Help

Branch B Channel 1

8. Select **Save Setup / Save and Reboot** to save the configuration to the MultiVOIP.



9. Assuming a Bogen PCMZPM is located at Corporate, connect a RJ11 phone cord to channel 1 FXS/FXO jack on the Corporate MultiVOIP. Connect the two wires on the other end of the phone cord to the Zone A + and – leads on the PCMZPM. Connect another RJ11 phone cord between channel 2 FXS/FXO jack and the Zone B + and – leads on the PCMZPM. A cord with an RJ11 plug on one end and tinned leads on the other end is included for this purpose. Caution: set the PCMZPM output selector switch to Low Power
10. At the Branch sites, connect a RJ11 phone cord between the MultiVOIP channel 1 FXS/FXO jack and the TIP/RING leads of the speaker amplifier (TPU-15, for example). You cannot connect the speaker directly to the MultiVOIP.
11. When all MultiVOIPs are powered on , a permanent audio connection will exist between Corporate MultiVOIP channel 1 and Branch A MultiVOIP channel 1. Another permanent audio connection will exist between Corporate MultiVOIP channel 2 and Branch B MultiVOIP channel 1. You can now make pages From Corporate to Branch A and/or Branch B remote sites.