



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

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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.

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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.

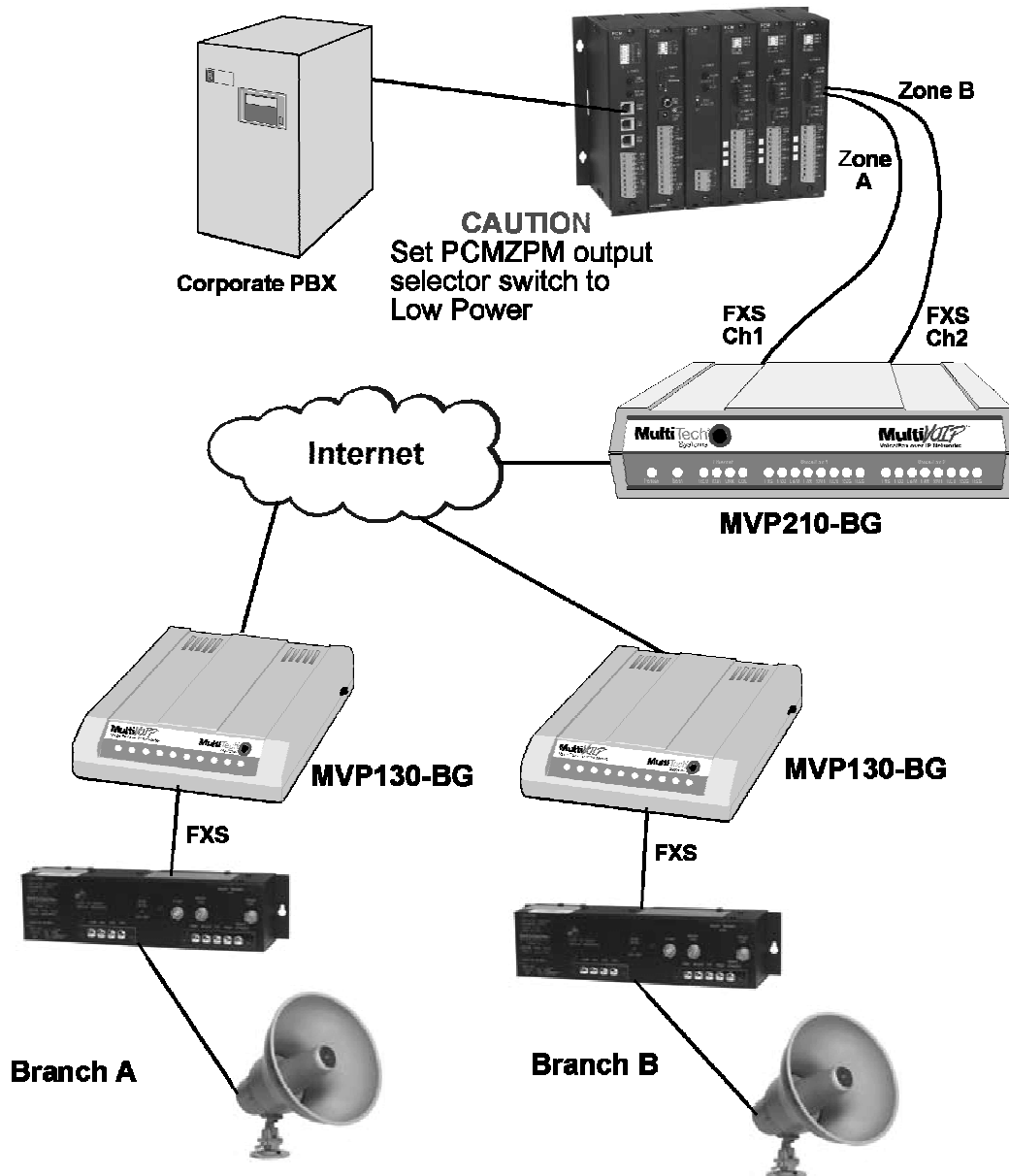


Figure 1. Bogen paging system diagram.

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.

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 . 20

IP Mask : 255 . 255 . 255 . 0

Gateway : 1 . . .

DNS

☐ Enable DNS

DNS Server IP Address : . . .

FTP Server

☒ Enable

TDM Routing Option

☐ Use TDM Routing For Intra-Gateway calls

OK

Cancel

Help

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.

Interface Parameters

Select Channel: Channel 1

Interface

- ☒ FXS (Loop Start)
- ☐ FXS (Ground Start)
- ☐ FXO
- ☐ E&M

Dialing Options

Regeneration

- ☐ Pulse
- ☒ DTMF

Inter Digit Timer: 2 secs

Inter Digit Regeneration Timer: 100 ms

Message Waiting Light: ☐

E&M Options

Signal

- ☒ Dial Tone
- ☐ Wink

Wink Timer: 250 ms

Type: TYPE II

Mode

- ☒ 2Wire
- ☐ 4Wire

Pass Through

- ☒ Enable

FXO Disconnect On

- ☒ Current Loss
- ☐ Tone Detection

Silence Detection: None

Disconnect Tone Sequence: Fax + None

Silence Timer: 15 secs

Current Loss Detect Timer: 500 ms

Ring Count

FXS: 8

FXO: 2

FXS Options

- ☐ Current Loss

Disconnect On Call Progress Tone

- ☐ Enable

Flash Hook Options

Generation: 600 ms

Detection Range

Min: 100 ms

Max: 1000 ms

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

-
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
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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

- 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 screenshot shows the 'Outbound Phone Book' window. It contains a table with the following data:

Destination Pattern	IP Address	Protocol	Description	Alternate
2	192.168.025.021	H.323		
3	192.168.025.022	H.323		

Below the table, it says 'Number of Entries : 2'. The 'Details' section is visible, showing fields for 'Remove Prefix', 'Add Prefix', 'Gatekeeper : not used', 'Gateway H.323 ID', 'Gateway Prefix', 'Q.931 Port : 1720', 'Transport Protocol', 'SIP URL', 'Round Trip Delay : 300 ms', and 'Alternate Phone Number'. On the right side, there are buttons for 'Add', 'Edit', 'Delete', 'Close', and 'Help'.

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

The screenshot shows the 'Outbound Phone Book' window. It contains a table with the following data:

Destination Pattern	IP Address	Protocol	Description	Alternate
1	192.168.025.020	H.323		

Below the table, it says 'Number of Entries : 1'. The 'Details' section is visible, showing fields for 'Remove Prefix', 'Add Prefix', 'Gatekeeper : not used', 'Gateway H.323 ID', 'Gateway Prefix', 'Q.931 Port : 1720', 'Transport Protocol', 'SIP URL', 'Round Trip Delay : 300 ms', and 'Alternate Phone Number'. On the right side, there are buttons for 'Add', 'Edit', 'Delete', 'Close', and 'Help'.

7. 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

OK Cancel Copy Channel Default Help

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 ☐ Consecutive Packets Lost: 30

☐ Call Duration: 180 secs ☐ Network Disconnection: 300 secs

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
 DTMF : **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
 DTMF : **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

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

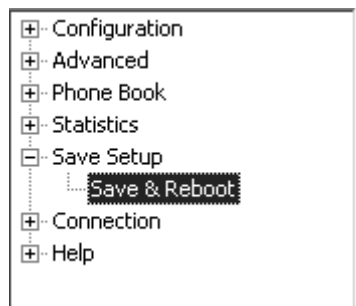
Dynamic Jitter Buffer:
Minimum Jitter Value: 60 ms
Maximum Jitter Value: 300 ms
Optimization Factor: 7

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.