

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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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.

- IP Parameters-	
Diff Serv Parameters	
Call Control PHB : 34	
VoIP Media PHB : 46	TYPE-II
- IP Parameters	
Enable DHCP	0 <u>K</u>
IP Address : 192 . 168 . 25 . 20	<u>C</u> ancel
<u>I</u> P Mask : 255 . 255 . 255 . 0	<u>H</u> elp
Gateway :	
DNS	
Enable <u>D</u> NS	
DNS <u>S</u> erver IP Address : · · · ·	
FTP Server	
✓ Ena <u>b</u> le	
TDM Routing Option	1
Use TDM <u>Bouting</u> For Intra-Gateway calls	

Corporate

□ IP Parameters □ Diff Serv Parameters	
Call Control PHB : 34	
VolP Media PHB : 46	
- IP Parameters	
Enable DHCP	<u> </u>
IP <u>A</u> ddress : 192 . 168 . 25 . 21	<u>C</u> ancel
<u>I</u> P Mask : 255 . 255 . 0	<u>H</u> elp
<u>G</u> ateway : · · ·	
DNS	1
Enable <u>D</u> NS	
DNS <u>S</u> erver IP Address :	
FTP Server	1
TDM Routing Option	1
Use TDM <u>Bouting</u> For Intra-Gateway calls	
	1

Branch A

- IP Parameters	
Diff Serv Parameters	
Call Control <u>P</u> HB : 34	
Erame Type	TYPE-II 💌
VoIP Media PHB : 46	
- IP Parameters	
	0 <u>K</u>
Enable DHCP	
IP Address; 192 . 168 . 25 . 22	
IP <u>A</u> ddress : 192 . 168 . 25 . 22	<u>C</u> ancel
<u>I</u> P Mask : 255 . 255 . 255 . 0	Usia
	<u>H</u> elp
Gateway:	
DNS	
Enable <u>D</u> NS	
DNS Server IP Address :	
FTP Server	
🔽 Enable	
TDM Routing Option	
Use TDM <u>R</u> outing For Intra-Gateway calls	

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	Dialing Options	ок
(FXS (Loop Start))	Regeneration Inter Digit Tim	
	C Eulse	Cancel
C FXS (<u>G</u> round Start)	🖸 🖸 DTMF 🗖 Message	e Waiting Light
O <u>F</u> X0		Copy Channel
	Inter Digit Regeneration Timer	100 ms
○ <u>E</u> &M		Default
Ethi Online	- FXO Disconnect On	Ring Count Help
E&M Options	Current Loss	F <u>X</u> S 8
Signal Contract		
© Dial <u>⊺</u> one © <u>W</u> ink	Tone Detection	F <u>x</u> 0 2
	Silence Detection	FXS Options
Wi <u>n</u> k Timer 250 ms	None 💌	Current Loss
,	Disconnect Tone Sequence	
		Disconnect On Call Progress Tone
Type TYPE II	** 💌 + None 💌	Enable
- Mode		Flash Hook Options
C Zwire C 4wire	Silence Timer 15 secs	Generation : 600 ms
		Detection Range
Pass Through	Current Loss	Min: 100 ms
En <u>a</u> ble	Detect Timer 500 ms	Max: 1000 ms

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

Outbound Phone Book				
Destination Pattern	IP Address	Protocol	Description	Alternal
•				
Number of Entries :				
Details Remove Prefix	,.			
Add Prefix				Add
Gatekeepe				Edit
Gateway H.323 ID				===
Gateway Prefix				<u>D</u> elete
Q.931 Port				Close
Transport Protoco	I:			
SIP URL	.:			<u>H</u> elp
Round Trip Delay	: 300 ms			
Alternate Phone Number				

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	1
Use as default entry	
Destination Pattern : 2	<u> </u>
<u>T</u> otal Digits : 0	<u>C</u> ancel
<u>R</u> emove Prefix :	
	<u>H</u> elp
Add Prefix :	
[P Address:] 192 . 168 . 25 . 21	Advanced
Description :	
Protocol Type	
© <u>S</u> IP	
H.323	1
☐ Use <u>G</u> ateKeeper	
Gateway H. <u>3</u> 23 ID :	
Grateway Prefi <u>x</u> :	
<u>Q</u> .931 Port Number : 1720	

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

Destination Pattern	IP Address	Protocol	Description	Alterna
	192.168.025.021	H.323		
1	192.168.025.022	H.323		
1				
Number of Entries : ; Details	2			
Remove Pre	fix :			Add
Add Pre	fix :			
Gatekeep	er : not used			<u>E</u> dit
Gateway H.323	D :			
alatornay intoco i	0			<u>D</u> elete
Gateway Pre	(18.1			
Gateway Pre	mx : ort : 1720			Close
Gateway Pre	ort : 1720			<u>C</u> lose
Grateway Pre Q.931 Pr	ort : 1720 col :			<u>C</u> lose <u>H</u> elp
Gateway Pre Q.931 Pr Transport Protoc	ort : 1720 col : 31. :			

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



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		
	Fax	0 <u>K</u>
Input 0 💌 dB Output 0 💌 dB	Fax Enable	<u>C</u> ancel
Dtmf	Max Baud Rate 14400	
	Fax Volu <u>m</u> e -9.5 💌 dB	Copy Channel
High -4 💌 dB Low -7 💌 dB	Jitter Value 400 ms	De <u>f</u> ault
Duration 100 ms	Mode FRF 11	<u>H</u> elp
DIMF: Out Of Band - Fixed Duration		
	Advanced Features	
Coder Manual O Automatic	Silence Compression	
	_	
Selected Coder G.711 U - law @ 64 kbp:	Echo Cancellation	
Max bandwidth 10 kbps	Forward Error Correction	
Auto Call		
🔽 Auto Call Enable 🔲 Generate Local Dial T	one	
Phone Number 2:1		
- Dynamic Jitter Buffer		
Minimum Jitter Value 60 ms		
Maximum Jitter ⊻alue 300 ms		
Optimization Factor 7		
Automatic Disconnection		
	ecutive Packe <u>t</u> s Lost ³⁰	
	ork Disconnection 300 secs	
	J	

Corporate Channel 1

-Voice/Fax Parameters		
Select Channel Channel 2		OK
Voice Gain	Fax	
Input 0 💌 dB Output 0 💌 dB	▼ Fa <u>x</u> Enable	Cancel
	Max Baud Rate 14400	
Gain	Fax Volume -9.5 💌 dB	Copy Channel
High -4 💌 dB Low -7 💌 dB	Jitter Value 400 ms	Default
		Dejauk
Duration 100 ms	Mode FRF 11 💌	Help
DIMF: Out Of Band - Fixed Duration		
Coder	Advanced Features	
	Silence Compression	
Selected Coder G.711 U - Iaw @ 64 kbp: 💌	🔽 Echo Cancellation	
Max bandwi <u>d</u> th 10 kbps		
	Forward Error Correction	
Auto Call		
🔽 Auto Call Enable 🔲 Generate Local Dial T	one	
Phone Number 3:1		
Description D. West		
Dynamic Jitter Buffer Minimum Jitter Value 60 ms		
Maximum Jitter Value 300 ms		
Optimization Factor 7		
Automatic Disconnection		
	· • • • • • • • • • • • • • • • • • • •	
	ecutive Packe <u>t</u> s Lost ³⁰	
Call Duration 180 secs Netwo	ork Disconnection 300 secs	3
· · · · · · · · · · · · · · · · · · ·	$1 \cdot C$	
Corpora	te Channel 2	

-Voice/Fax Parameters		
Select Channel Channel 1		0K [
-Voice Gain	Fax	0 <u>K</u>
Input 0 💌 dB Output 0 💌 dB	Fax Enable	<u>C</u> ancel
Dtmf	Max Baud Rate 14400	
Gain	Fax Volu <u>m</u> e •9.5 💌 dB	Copy Channel
High -4 💌 dB Low -7 💌 dB	Jitter Value 400 ms	De <u>f</u> ault
Dur <u>a</u> tion 100 ms	Mode FRF 11 💌	<u>H</u> elp
DIMF: Out Of Band - Fixed Duration		
	vanced Features	
	Silence compression	
Selected Coder G.711 U - law @ 64 kbp:	Echo Ca <u>n</u> cellation	
Max bandwidth 10 kbps	Forward Error Correction	
Auto Call		
🔽 Auto Call Enable 🔲 Generate Local Dial Tone		
Phone Number 1:1		
Dynamic Jitter Buffer		
Minimum Jitter Value 60 ms		
Maximum Jitter ⊻alue 300 ms		
Optimization Factor 7		
	ive Packe <u>t</u> s Lost 30	
	Disconnection 300 secs	3

Branch A Channel 1

-Voice/Fax Parameters	
Select Channel Channel 1	
Voice Gain	0 <u>K</u>
Input 0 dB Output 0 dB Max Baud Rate 14400	<u>C</u> ancel
Dtmf Fax Volume 9.5 dB	Copy Channel
High -4 💌 dB Low -7 💌 dB Jitter Value 400 ms	De <u>f</u> ault
Duration 100 ms	<u>H</u> elp
DIMF: Out Of Band - Fixed Duration	
Coder Advanced Features	
Selected Coder G.711 U - law @ 64 kbp: 💌 🛛 🔽 Echo Cancellation	
Max bandwigth 10 kbps Forward Error Correction	
- Auto Call	
🔽 Auto Call Enable 🔲 Generate Local Dial Tone	
Phone Number 1:2	
Dynamic Jitter Buffer	
Minimum Jitter Value 60 ms	
Optimization Factor 7	
Automatic Disconnection	
☐ Jitter Value 350 ms ☐ Consecutive Packe <u>t</u> s Lost 30	
Call Duration 180 secs C Network Disconnection 300 sec	s

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.