Key Product:	**8180 SIP Audio Alerter**
Application:	**[Multicasting] – Voice Paging in Multiple Locations Simultaneously**
Summary:	**An 8180 can auto-answer and broadcast page audio through its internal speaker AND simultaneously multicast to other 8180 speakers on the network. This allows voice paging in multiple locations or to otherwise cover large areas. Only one auto-answering 8180 need be registered with the SIP server and setup as the Master unit while all other 8180 devices can be setup as Slaves.**

[Overview]

With Version 2.1 each 8180 may be configured to broadcast or receive multicast audio. This feature will allow for Zoned Multicast paging and is now also compatible with Polycom Group paging. In order for this to work multicasting must be enabled using the 8180 web interface ("Control Panel").

If set to Master Mode, the 8180 will send its page audio over the network to a multicast IP address instead of, or in addition to, the speaker. If set to Slave Mode, the 8180 will watch for audio sent to the Multicast IP address and play the audio over its speaker.

One use for multicasting is where SIP licenses are required for each endpoint. By having one 8180 multicast to others, only a single license is required.

To set up multicasting, first determine which 8180 you want to auto-answer, play, and transmit page audio. This will then be configured as the Master unit. Then set up one or more other 8180s to receive and play the page audio (called "Slaves").

With the new features in Version 2.1 and expanding on the 8180 multicast capability, 9 zones are now supported in a priority hierarchy from "Emergency" to "Music". Each 8180 may belong to one or more zones. To support zoned paging an 8180 configured as a master may be optioned to require zone selection prior to paging. Separate zones may be paged simultaneously if multiple masters exist in the network.

If you are currently using Polycom group paging the Version 2.1 8180 is now compatible. The 8180 may receive or broadcast multicast paging compatible with Polycom phone on-premise group paging protocol. This enables the 8180 to augment telephone speaker paging for common areas or noisy environments. In addition the 8180 may be called to page and simultaneously multicast to Polycom phones.

[Requirements/Prerequisites]

1. At least one 8180 that is registered with the SIP server and has a page extension set up and available to use.
2. One or more additional 8180s that are attached to the network (but not necessarily registered with the SIP server).
3. For setup, you'll need to know your SIP Server Name (or IP address) as well as information related to the page extension referred to above: Extension name, Auth ID, and password. You should be able to get this information from whoever is administering your SIP Server.

[Using the 8180 Control Panel]

When an 8180 is connected to the PoE network it will automatically obtain an IP address during initialization. The IP address can be determined by pressing the Menu key on the back of the 8180 until you hear the words "Device Information". Then press the Select key to have the 8180 speak the IP address.

1. Alternatively, you can search the network with the Network Device Locator tool which will list all Algo SIP devices it sees on the network, complete with IP addresses, Product Names, and Device Names.
2. Enter the 8180's IP address in your internet browser to bring up the Control Panel. Note that the default password is "algo" (all lowercase and without quotation marks).

It's a good idea to provide descriptive Device Names for each 8180 that you have on a network. That will make them easy to locate and test later, particularly if you have a number of 8180s spread throughout a facility. You can do this in the Control Panel by using the Admin>Device Name field.

[Configuring the Master 8180]

1. Using the IP address of the 8180 that you want to use as your Master unit, open the Control Panel.
2. In the SIP tab, make sure that you have entered your information on the Ring and Page Extension, as well as the address for the SIP Server.

3. Under Basic Settings -> Ring, set the Ring Tone you wish to use. Once configured click "Save"

4. Under Basic Settings -> Page, set the Page Tone you wish to use. Once configured click "Save"

ALGO
Communication Solutions

5. Test the ring and page extension setup by clicking "Save Settings", then dialing the extension above from any other phone on the system (note the calling phone does not need to be SIP). The 8180 should auto-answer, play a pre-announce ring or page tone, and allow you to talk over the speaker.
6. Under Basic Settings -> Multicast, set the Multicast Mode to Master/Sender and leave Multicast Zone in the default setting (224.0.2.60:50000). Note that the 8180 will still auto-answer and page over its speaker as before. ***Enabling the master multicast feature does not in any way alter the other functionality of the 8180***.

Multicast Master/Sender Settings

By selecting DTMF Selectable Zone you will be able to ring/page multiple zones at the same time, whereas by selecting Single Zone only one zone will receive the ring/page.

Master Ring Mode

By selecting the Send Audio Stream all Slave Mode units will sound the ring/page tone configured from within the Master unit, whereas by selecting Send Control Signal the ring/page tone will sound based on the configuration of the Slave Mode units.

Multicast Settings

| Status | **Basic Settings** | Advanced Settings | System | Logout |

SIP   Ring   Page   **Multicast**

Multicast Settings

Here you can configure multicast settings.

Multicast Mode

Multicast Mode   ○None ◉Master/Sender ○Slave/Receiver

Multicast Master/Sender Settings

Zone Selection Mode   ○DTMF Selectable Zone ◉Single Zone

Master Single Zone   [Music ▾]

Master Ring Mode   ○Send Control Signal ◉Send Audio Stream

Zone Definition

Priority Call (DTMF:9)	224.0.2.60:50000
All Call (DTMF:0)	224.0.2.60:50001
Zone 1 (DTMF:1)	224.0.2.68:50002
Zone 2 (DTMF:2)	224.0.1.116:5001
Zone 3 (DTMF:3)	224.0.2.60:50004
Zone 4 (DTMF:4)	224.0.2.60:50005
Zone 5 (DTMF:5)	224.0.2.60:50006
Zone 6 (DTMF:6)	224.0.2.60:50007
Music (DTMF:N/A)	239.168.3.9:20000

Polycom Group Paging/Push-to-Talk

Paging/PTT Mode   ○Disabled ◉Group Paging ○Push-to-Talk

Default Channel   [Group 1 ▾]

7. Click "Save Settings" and close the web interface.

Note: When using a Polycom phone as the paging source, the Polycom phone must be set to use the G.711 or G.722 audio codec. Note that the Polycom phone must be configured with the "Compatibility" setting ("ptt.compatibilityMode") disabled in order for this codec setting to be applied.

[Configuring the Slave 8180]

1. Under Basic Settings -> Multicast, set the Multicast Mode to Slave/Receiver and leave Multicast Zone in the default setting (224.0.2.60:50000). Enabling the Slave/Receiver multicast feature does not in any way alter the other functionality of the 8180.

    Each 8180 has its own IP address, and shares a common multicast IP and port number (multicast zone) for multicast packets.

    The master unit transmits to a configurable multicast zone, and all the slave units listen to the multicast zones assigned to them. When a master unit receives a valid SIP ring event, it will ring locally and also send out a special packet to the multicast zone. The network switches and router see the packet and deliver it to all the members of the group. When a page audio call occurs, the master unit will transmit all the RTP audio packets to the multicast zone and all the slave units will receive them.

2. Ensure that the multicast IP and port number are the same on all the master and slave units of one group. The user may define multiple zones by picking different multicast IP addresses and/or port numbers.

    1. Multicast IP addresses range: 224.0.0.0/4 (from 224.0.0.0 to 239.255.255.255)
    2. Port numbers range: 1 to 65535
    3. By default, the 8180 is set to use the multicast IP address 224.0.2.60 and the port numbers 50000-50008

    Multicast IP addresses and ports are configured by default. These typically do not need to be changed unless they conflict with other multicast addresses on the network. If changed they must be changed on all 8180s in the multicast group.

    Make sure that the multicast IP address and port number do not conflict with other services and devices on the same network.

3. Select one or more multicast zones for the 8180 to listen to. Note that multicast zone priority is based on the zone definition list order (top to bottom).

4. Once configured click "Save Settings" and close the web interface.

| Status | **Basic Settings** | Advanced Settings | System | Logout |

SIP   Ring   Page   **Multicast**

Multicast Settings

Here you can configure multicast settings.

Multicast Mode

Multicast Mode	○None ○Master/Sender ●Slave/Receiver

Multicast Slave/Receiver Settings

Slave Zones	☑Priority Call ☑All Call ☐Music
	☑Zone 1 ☐Zone 2 ☐Zone 3
	☐Zone 4 ☐Zone 5 ☐Zone 6

Zone Definition

Priority Call (DTMF:9)	224.0.2.60:50000
All Call (DTMF:0)	224.0.2.60:50001
Zone 1 (DTMF:1)	224.0.2.60:50002
Zone 2 (DTMF:2)	224.0.2.60:50003
Zone 3 (DTMF:3)	224.0.2.60:50004
Zone 4 (DTMF:4)	224.0.2.60:50005
Zone 5 (DTMF:5)	224.0.2.60:50006
Zone 6 (DTMF:6)	224.0.2.60:50007
Music (DTMF:N/A)	224.0.2.60:50008

Polycom Group Paging/Push-to-Talk

Paging/PTT Mode	○Disabled ●Group Paging ○Push-to-Talk

Default Channel	Group 1 ▼
Priority Channel	Group 24 ▼
Emergency Channel	Group 25 ▼

Note: When using a Polycom phone as the paging source, the Polycom phone must be set to use the G.711 or G.722 audio codec. Note that the Polycom phone must be configured with the "Compatibility" setting ("ptt.compatibilityMode") disabled in order for this codec setting to be applied.

For More Information:

Sales   1.604.454.3790   sales@algosolutions.com
Support   1.604.454.3792   support@algosolutions.com
Algo Communication Products Ltd., 4500 Beedie Street, Burnaby, BC, Canada V5J 5L2