Application Note

Using a Dialogic® Media Gateway Series as a PSTN Gateway with an Asterisk IP-PBX Server

Executive Summary

This application note describes a method to configure the Dialogic® 1000 Media Gateway Series or Dialogic® 2000 Media Gateway Series and the Asterisk IP-PBX Server software to interoperate without connecting phone lines into your server running Asterisk.

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Introduction

Customers interested in using an Asterisk IP-PBX solution frequently require Public Switched Telephone Network (PSTN) connection to a telephony network. However, they may not want to use Digium telephony blades if they are implementing a full IP solution on their internal network. This application note discusses a method for configuring the Dialogic® 1000 Media Gateway Series or Dialogic® 2000 Media Gateway Series and the Asterisk software to interoperate without connecting phone lines into your server running Asterisk.

A note about terminology: the term "Dialogic® Media Gateway Series" is used in this application note to refer to the Dialogic 1000 Media Gateway Series and the Dialogic 2000 Media Gateway Series collectively, as well as to refer to a single gateway from either such Series.

Approach

For the approach discussed in this application note, the Asterisk IP-PBX is implemented using the Asterisk "DEMO" dial plan. Calls arrive from an ISDN line, and are terminated to the Dialogic Media Gateway Series. The Dialogic Media Gateway Series translates the PSTN ISDN calls from ISDN to SIP and is configured to deliver the SIP calls to the Asterisk IP-PBX.

Once it has answered the calls, the Asterisk IP-PBX plays the Asterisk demo prompt, but any Asterisk functionality can be used at this point. A typical configuration would play a prompt and allow the inbound caller to dial an extension to reach an internal party or call center agent.

The Asterisk IP-PBX does not implement any transcoding, but handles the RTP stream for the duration of the call.

Environment

A PSTN phone line is required to terminate into the Dialogic Media Gateway Series. Any ISDN protocol supported by the Dialogic Media Gateway Series could be used, but it is implemented in this application note using NI2 protocol. SIP is configured, and SIP calls are sent over the Ethernet network to the Asterisk IP-PBX. This is often used in a call center environment, but could also be used in a small- to medium-sized business, for example.

Features and Benefits of Dialogic® Media Gateway Series as PSTN Front-End to Asterisk

Many small businesses and call centers moving away from traditional phone systems for in-house use can find Asterisk to be a low-cost VoIP switching platform. However, even businesses who find the costs of an all-IP system attractive still generally want to connect calls to and from the PSTN. The Asterisk IP-PBX provides a dial plan and a switching point for the internal calls between stations. The Asterisk IP-PBX also can route outbound calls to the Dialogic Media Gateway Series, which in turn sends calls out to the PSTN. Potential benefits of this configuration are that it can allow for a 100 percent IP switching and wiring solution inside the company, while also enabling external calls to be carried by the public network. Another potential benefit is that because TDM termination points are separate from the server running the switching solution, you can locate them in separate physical locations.

Using the Dialogic Media Gateway Series as a front-end to an IP small business or call center thus can provide a "best of both worlds" approach to a network build-out. The internal network can sometimes be shared with the existing data network, though, for all but the smallest environments, greater reliability generally can be obtained by running parallel networks with similar hardware for voice and data. This can significantly reduce the cost of telephony wiring and also allows for use of standard network equipment. By keeping the IP-PBX in the data center with other servers providing data-related functionality (for example, accounting software, customer-interaction tracking, etc.), both the equipment and personnel are consolidated into one group, rather than having separate data and telecom staff and hardware. And retaining connectivity to the PSTN via a separate phone device on the desk satisfies a user's comfort with existing telephony equipment. VoIP can provide additional benefits by creating the opportunity for new applications, and additional productivity from new network-based tools.

Topology of Solution

Low Density Topology

In a low density environment, such as less than twelve-seat offices, a Dialogic® 1000 Media Gateway Series (DMG1000 Gateways) provides enough ports to enable connectivity to the PSTN, allowing you to increase the number of internal stations later.

A small call center could implement an eight- to twelve-seat phone system with the following components:

- One Asterisk server
- One DMG1000 Gateways
- One to two 16-port Ethernet switches
- Eight to twelve SIP phones

The actual ratio of outside lines to internal callers will be application- and environment-specific, and this particular ratio is just an example. Figure 1 depicts an example of what a possible end solution could look like.

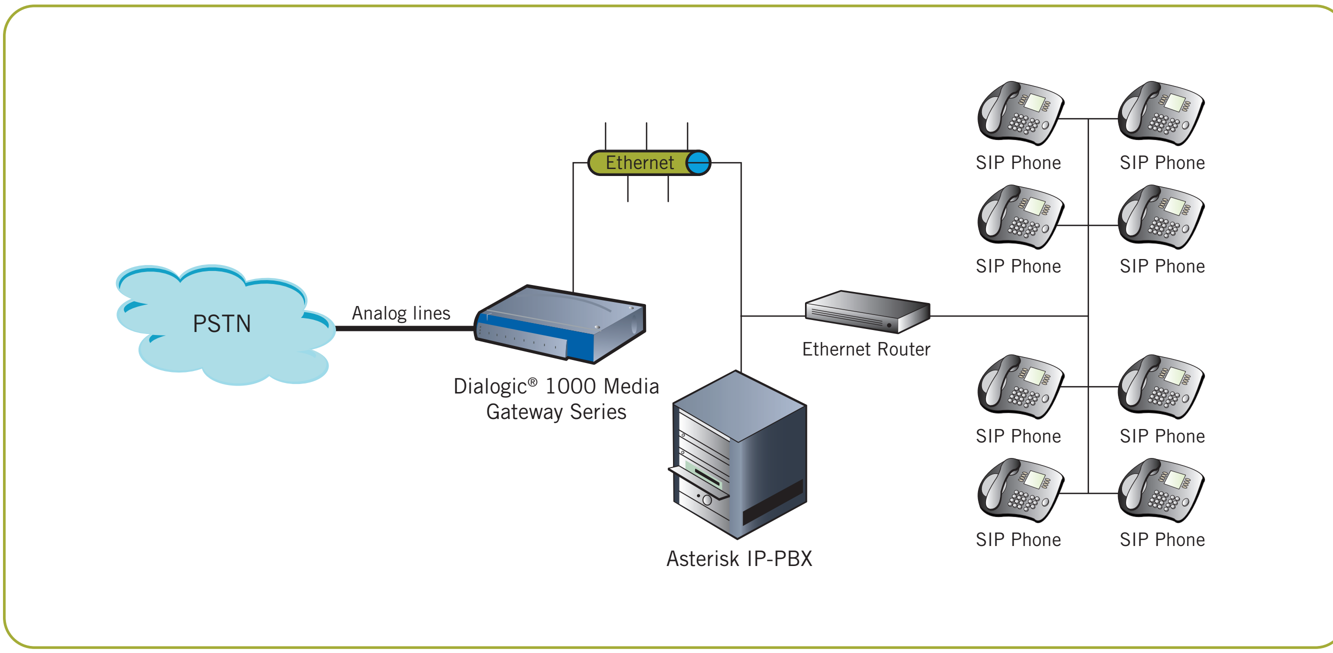

Figure 1. An Example of a Low Density Topology including a Dialogic® 1000 Media Gateway Series

High Density Topology

In a higher density environment, such as 24- to 60-seat offices, a Dialogic® 2000 Media Gateway Series (DMG2000 Gateways) provides enough ports to permit 80 percent of stations to connect to outside lines at a given time.

A medium-sized business could implement a 24- to 60-seat phone system with the following components:

- One to two Asterisk servers
- One DMG2000 Gateways
- One to two 48-port Ethernet switches
- Twenty-four to sixty SIP phones

Again, this particular ratio of outside lines to internal parties will be application- or use case-specific and can vary depending on end user's requirements. Figure 2 depicts what one possible end solution could look like.

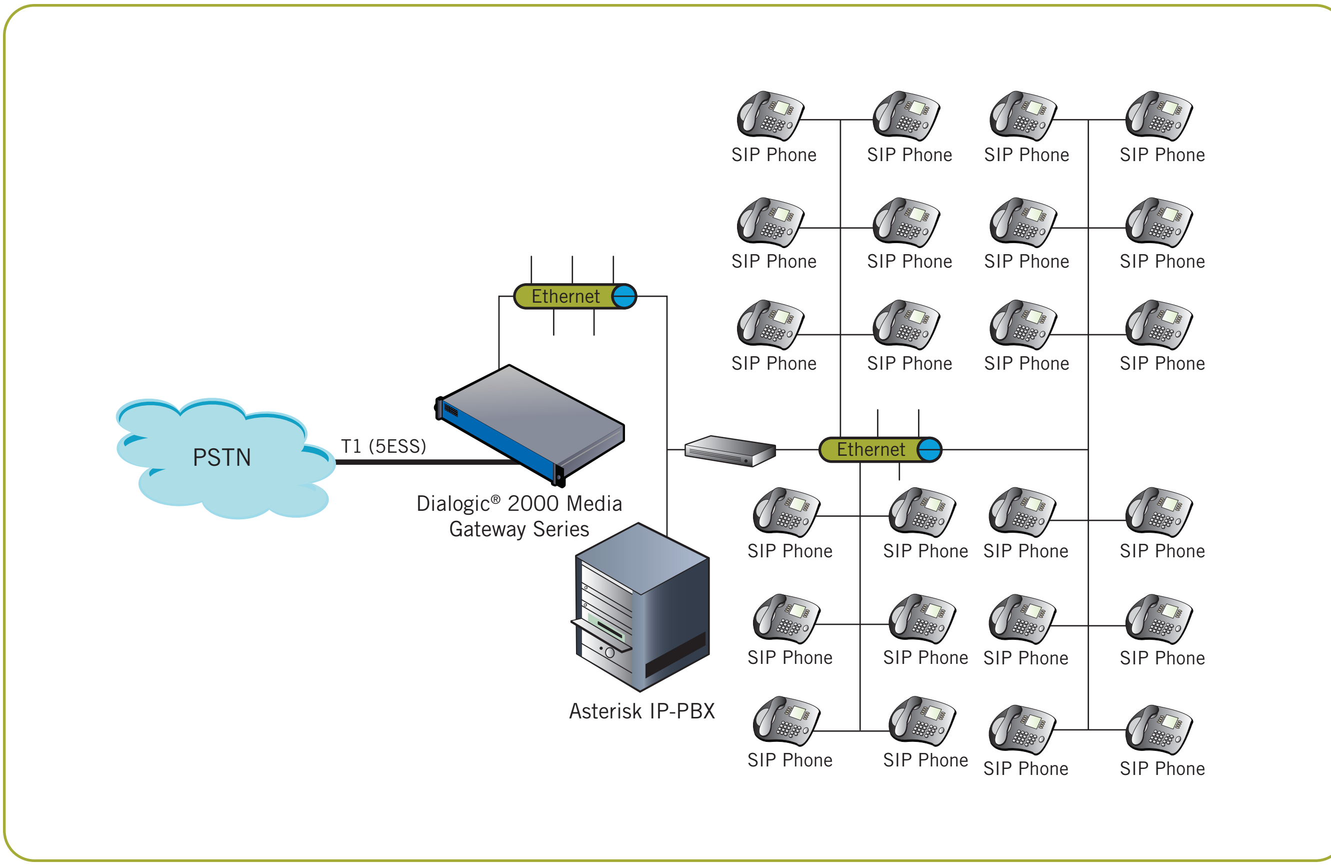

Figure 2. An Example of High Density Topology including a Dialogic® 2000 Media Gateway Series

Configuring Asterisk and Dialogic® Media Gateway Series

Only two Asterisk configuration files require modification to implement the connection between the Asterisk server and the Dialogic Media Gateway Series. The files requiring modification are extensions.conf and sip.conf (see the *For More Information* section for the location to download these files).

Extensions.conf has been modified so that when you arrive at the "demo" welcome prompt on the Asterisk server, you can dial 501, and it will place an outbound SIP call to the Dialogic Media Gateway Series. The Dialogic Media Gateway Series will translate your call from SIP to ISDN, and place an outbound call on the T1/E1 line. The sip.conf file has been modified to allow the SIP phone to register as user 7003.

The Dialogic Media Gateway Series low density (DMG1000 Gateways) and high density (DMG2000 Gateways) configurations have been changed to set the T1 line for 5ESS, user-side.

Summary

This application note described how it is possible to connect your Asterisk IP-PBX to the world of the PSTN without connecting phone lines into your server running Asterisk.

For More Information

A Zip file containing the two configuration files can be downloaded at http://www.dialogic.com/goto/?10856

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