Deploying SIP Phones into Unified Communications Solutions with the Dialogic® Media Gateway Series

Technology Brief

Communication is an important part of business enterprises, and many are adopting Unified Communications (UC) solutions to improve business processes and enhance communications. These UC solutions typically use Voice over IP (VoIP) to deliver phone calls and are connected to PSTN-based telephone networks either by direct VoIP to the enterprise's existing IP-PBX or by using VoIP gateways, such as those from the Dialogic® Media Gateway Series (DMG Gateways) to provide connectivity to the enterprise's TDM-based PBX or direct to the PSTN.

One of the notable elements of these UC solutions is the end user terminal devices. These terminal devices can include wired and wireless IP phones, including proprietary digital feature phones integrated into a UC solution as part of a legacy TDM PBX, as well as legacy analog phones. In most cases, the IP phones use the Session Initiation Protocol (SIP) and therefore will be referred to as SIP phones in this technology brief. Also addressed is the deployment of SIP phones into a UC solution.

SIP Phone Functionality

When SIP phones were first introduced more than a decade ago, they were quite expensive and lacked the full feature set provided by traditional digital feature phones. This situation has changed over time so that current full featured SIP phones often have video capabilities and are available for between $100 and $150USD. In order to realize the full feature set of SIP phones, they must be designed for the UC solution into which they are being deployed. For example, Cisco and Avaya offer SIP phones and UC solutions, and each company's SIP phones provide a full feature set when deployed in its company's respective solution. Similarly, Microsoft has introduced its Microsoft® Office Communications Server UC solution and has certified a set of SIP phones with a full set of functionality for use in the solution. It is notable that when a SIP phone is deployed into a UC solution for which it is was not designed or certified, the feature set can be degraded.

Degradation of Functionality for Non-Certified SIP Phones

Presence indication, a critical piece in enhancing UC productivity, is an example of a feature that, in many cases, is not available when deploying a non-certified SIP phone into a UC solution. Table 1 lists a typical SIP phone feature set and the features that can be expected to be available when certified and non-certified SIP phones are deployed into UC solutions.

Feature	Available in Certified SIP Phones	Available in Non-Certified SIP Phones
Make call	Yes	Yes
Receive call	Yes	Yes
Call hold	Yes	Yes
Consultative hold	Yes	Yes
Ad hoc conferencing	Yes, up to six parties	Yes, up to three parties
Last party drop	Yes	Yes
Forward all my calls/SAC	Yes	Yes
Forward my calls when busy or no answer	Yes	Yes
Attended call transfer	Yes	Yes
Unattended call transfer	Yes	Yes
Hunt groups	Yes	Depends on local proxy capabilities
Inbound call management	Yes (CM COR)	Depends on local proxy capabilities
Outbound call management	Yes (CM COR)	Depends on local proxy capabilities
Calling party block	Yes	No
Calling party unblock	Yes	No
Call park	Yes	No
Call unpark	Yes	No
Call pickup	Yes	No
Directed call pickup	Yes	No
Extended call pickup	Yes	No
Priority call	Yes	No
Auto callback	Yes	No
Malicious call trace	Yes	No
Malicious call trace cancel	Yes	No
EC500 on/off	Yes	No
Transfer to voicemail	Yes	No
Whisper page	Yes	No
Recording voice call to messaging	Yes	No
Bridge line and call appearances	Yes	No
Extend call	Yes	No
Hold recall	Yes	No
Transfer recall	Yes	No
Busy indicator	Yes	One button dial, yes; busy indicator, no
Message waiting indicator	Yes	No

Table 1. Features Available in Certified and Non-Certified SIP Phones

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Connection of Non-Certified SIP Phones

As noted in Table 1, many SIP phone features are not available when a SIP phone is deployed into a UC solution for which the SIP phone was not designed. Because of the accompanying severe degradation in feature functionality, it is not advantageous from a feature standpoint to deploy non-certified SIP phones into a UC solution. If however, it becomes necessary to deploy non-certified SIP phones into a UC solution, DMG Gateways can provide a bridge between the SIP phones and the other UC solution components. Figure 1 shows an example of a typical UC solution deployment that includes DMG Gateways in such a role. The DMG Gateways can route calls between the UC solution environment and the non-certified SIP phones, and can also route calls between the UC solution and the PSTN.

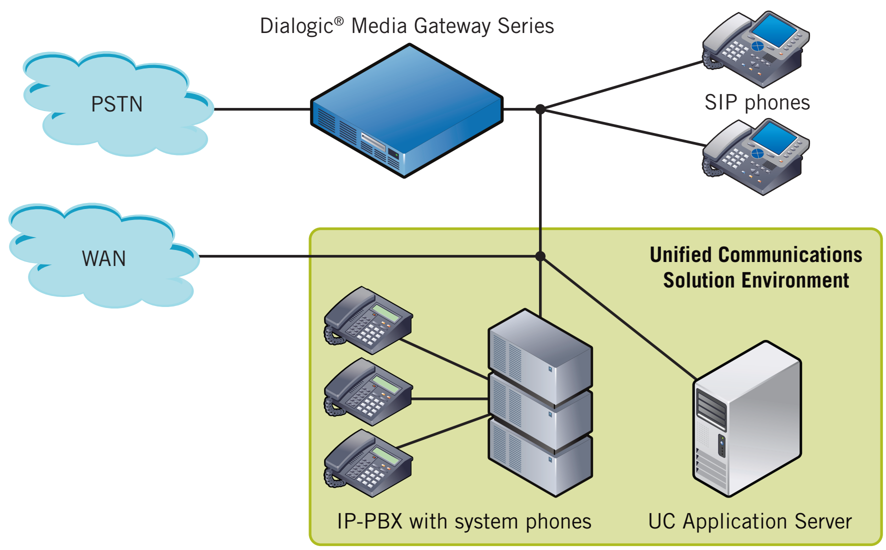

Figure 1. A UC Solution Deployment Example using Dialogic® Media Gateway Series (DMG Gateways)

Another common deployment scenario is to have the primary UC solution components deployed in a central (or headquarters) location, and other portions of the UC solutions, including the non-certified SIP phones and potentially legacy analog devices, deployed at remote sites (see Figure 2 for an example). Here, similar to the first scenario, the DMG Gateways can route calls between the SIP phones and the legacy analog devices, and between the SIP phones and the other UC solution components.

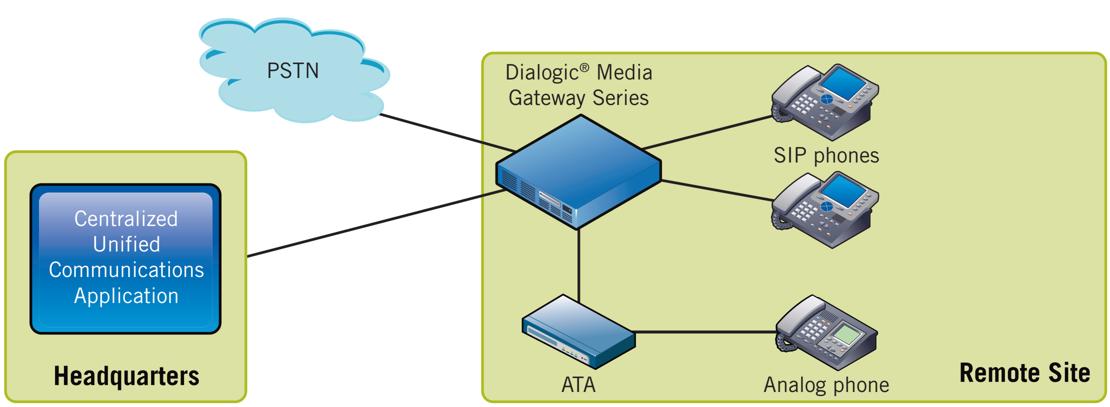

Figure 2. A UC Solution using Dialogic® Media Gateways Series (DMG Gateways) Located at a Remote Site

Survivability of Non-Certified SIP Phones

As noted previously, DMG Gateways can route calls between the remote SIP phones and the remote legacy analog devices and between the SIP phones and the other UC solution components. If the connection to the headquarters, or to the primary UC solution components, is broken, DMG Gateways can perform in one of the following two failure scenarios:

- **No proxy server deployed at the remote location** — The SIP phones and the legacy analog devices can send and receive calls to and from external locations via the connection to the PSTN through the DMG Gateways; however, they would not be able to send or receive calls to or from one another.

- **Proxy server deployed at the remote location** — The SIP phones and legacy analog devices can send and receive calls to and from one another, or to and from external locations via the DMG Gateways.

For More Information

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