Application Note

Using VoIP Interface Adapters with ClearOne Teleconferencing Equipment

Introduction
This document introduces a solution designed to enable VoIP communications using ClearOne's installed (XAP 400 and XAP TH2) and portable (MAX Wireless and MAX EX) teleconferencing equipment.

The application focuses on third party handset-to-Ethernet interface equipment — specifically Sipura's SPA-2000 and Cisco's ATA 186 VoIP gateway devices. These devices provide a VoIP interface through the conferencing equipment's telephone interface port.

Network Setup Considerations
This document only provides an overview of network configuration and programming. Any modifications to a corporate network including the addition of equipment, should always be overseen by the company's Network/Telecom Administrator. Adding uncertified or improperly configured equipment to a network can severely hamper or disable the network.

While specific network setup instruction is outside of this document's scope, there are several questions you should consider: Do I have a DHCP (Dynamic Host Configuration Protocol) server to gain an IP address automatically? Do I need to register with any SIP or H.323 supporting hardware in the configuration of my VoIP unit? Is my network connection (internal and/or external) capable of supporting the VoIP connection (typically 90Kbps)? Both the Sipura and the Cisco models provide excellent network configuration sections in their product user manuals. You will find links to their web sites at the end of this document.

Hardware Connections
Attaching the VoIP equipment requires absolutely no modifications to the ClearOne equipment. Both the Cisco and Sipura units simply emulate a standard phone line on the equipment side. The network port on the VoIP interface is an RJ-45 which attaches to the corporate LAN. The connection to the conferencing system is accomplished through a standard telephone cord connected between the VoIP interface and the Line jack on the conferencing system.

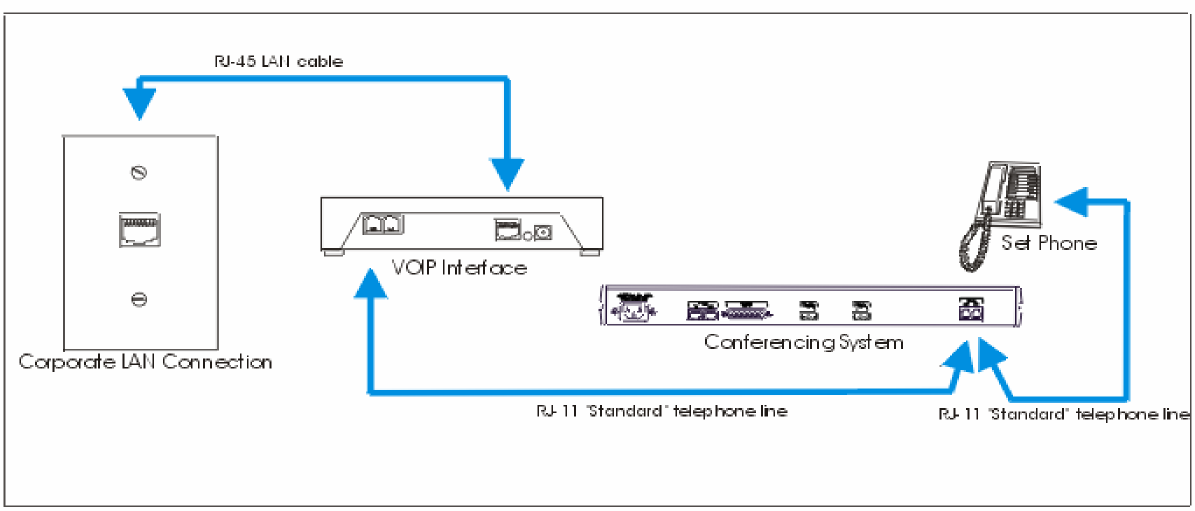

Fig. 1

ClearOne Communications
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Telephone Interface Setup

Setup of the telco properties is similar with both the Sipura and Cisco units. Listed below are the recommended settings for optimal performance when interfacing to ClearOne's conferencing products. Additional settings that are not listed will need to be set up according to your particular network. It is recommended that you consult your Network/Telecom Administrator and the Cisco and Sipura product manuals prior to system installation/configuration.

Cisco ATA 186		Sipura SPA-2000	
Parameter	**Recommended Setting**	**Parameter**	**Recommended Setting**
TX Codec	(1)*	Preferred Codec	(G711u)
RX Codec	(1)	DTMF TX Method	(Auto)
Polarity	(0x00000000)	Silence Supp Enable	No
Ring On/Off Time	(2,4,25)		
Dial Tone	Use default string		
Busy Tone	Use default string		
Ring Back Tone	Use default string		
Alert Tone	Use default string		
FX Input Level	(-1)		
FXS output Level	(-4)		

*Parameters in parentheses ( ) are the factory default settings.

***Note**: The "Preferred Codec" option dictates the compression format for the audio passing over the network. With high bandwidth, either G711a or G711u is preferable. If your connection is slower, under 90 Kbps, it is best to choose a higher compression format such as 729a.*

Making a Call

The dialing function is performed using DTMF tones generated by the conferencing system, which are then interpreted by the VoIP adapter. The format of the dialed strings usually includes the IP address of the party to be connected to (refer to the VoIP adapter's user manual for specific dialing format information). With more advanced VoIP networks, you can simply dial an extension number if a SIP (Session Initiated Protocol) server is used to associate the extension with the actual IP of the called party. Most adapter units will only recognize touch-tone dialing and have no pulse dialing option.

Conclusion

Both the Cisco and Sipura VoIP gateways provide a simple and viable solution to interface ClearOne's teleconferencing products to a VoIP conferencing application. As in any conferencing application, results are dependent on a multitude of conferencing conditions such as room noise levels, room acoustics, conference levels, network bandwidth and talker to mic distances.

Although implementation of this technology into a conferencing system may not require expertise on network integration, it is recommended that your Network/Telecom Administrator be consulted prior to installation.

Links to VoIP manufacturers

Cisco VoIP 180 Series: http://www.cisco.com/en/US/products/hw/gatecont/ps514/index.html
Sipura Products SPA Series: http://www.sipura.com/products/index.htm

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